EC6502 – PRINCIPLES OF DIGITAL SIGNAL PROCESSING

TWO MARKS

UNIT I

DISCRETE FOURIER TRANSFORM

1. State the properties of DFT? (University)

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

2. Define circular convolution?

Let $x_1(n)$ and $x_2(n)$ are finite duration sequences both of length N with DFTs $X_1(K)$ and $X_2(k)$

If $X_3(k)=X_1(k)X_2(k)$ then the sequence $x_3(n)$ can be obtained by circular convolution defined as

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

3. 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** durationL+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.

4. 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.

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

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.

6. 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

7. 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.

8. 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

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

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

9. What are the 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.

10. What is overlap-add method?

In this method the size of the input data block $x_i(n)$ is L. To each data block we append M-1 zeros and perform N point circular convolution of $x_i(n)$ 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.

11. What is overlap-save method?

In this method the data sequence is divided into N point sections $x_i(n)$.Eachsection 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 $x_i(n)$ with h(n) the first M-1 points will not agree with the linear convolution of $x_i(n)$ 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 $x_i(n)$ N h(n). This process is repeated for, all sections and the filtered sections are abutted together.

12. Why FFT is needed?

The direct evaluation DFT requires N_2 complex multiplications and N_2 –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.

13. What is FFT? (University)

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 DFTcomputation time. It is based on the fundamental principle of decomposing the computation of DFT of a sequence of length N into successively smaller DFTs.

14. How many multiplications and additions are required to compute N point DFT using radix-2 FFT? (University)

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 respectively.

15. 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 an integer, then this algorithm is known as radix-2 algorithm.

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 (DIF).

18. What are the applications of FFT algorithm?

The applications of FFT algorithm includes

1) Linear filtering
2) Correlation
3) Spectrum analysis

19. What is the relationship between z-transform and DTFT?
The z-transform of x(n) is given by

$$x(z) = \sum_{n=-\infty}^{\infty} x(n)\, Z^{-n} \quad ; \text{where } z = re^{j\omega} \qquad \ldots\ldots\ldots\ldots\ldots\ldots (1)$$

Substituting z in x(z) we get,

$$x(z) = \sum_{n=-\infty}^{\infty} x(n)\, r^{-n} e^{-j\omega n} \qquad \ldots\ldots\ldots\ldots\ldots\ldots (2)$$

The Fourier transform of x(n) is given by

$$X(e^{j\omega}) = \sum_{n=-\infty}^{\infty} x(n)\, e^{-j\omega n} \qquad \ldots\ldots\ldots\ldots\ldots\ldots (3)$$

Equation (2) and (3) are identical, when r = 1.
In the z-plane this corresponds to the locus of points on the unit circle |Z|=1.
Hence $X(e^{jw})$ is equal to H(z) evaluated along the unit circle, or $X(e^{jw}) = x(z)|_{z=e^{jw}}$
For $X(e^{jw})$ to exist, the ROC of x(z) must include the unit circle.

20. Define DFT of a discrete time sequence?

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

$$x(k) = \sum_{n=0}^{N-1} x(n) e^{-j2\pi nk/N} \qquad K=0,1,2,3,\ldots N-1$$

21. Define IDFT?

The IDTFT of the sequence of length N is defined as

$$X(n) = (1/N) \sum_{k=0}^{N-1} x(k) e^{j2\pi nk/N} \qquad n=0,1,2,3,\ldots N-1$$

22. Define DTFT and IDTFT of a sequence? (University)

The DTFT (Discrete Time Fourier Transform) of a sequence x(n) is defined as

$$X(w) = \sum_{n=-\infty}^{\infty} x(n) e^{-jwn}$$

The IDTFT is defined as

$$x(n) = 1/2\pi \int_{-\pi}^{\pi} X(w)\, e^{jwn}\, dw$$

23. What is the drawback in DTFT?

The drawback in discrete time fourier transform is that it is continuous function of w and cannot be processed by digital systems.

24. State periodicity property with respect to DFT?

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

x(n+N) = x(n) for all n.
x(k+N) = x(k) for all k.

25. State linearity property with respect to DFT?

If $x_1(k)$ and $x_2(k)$ are N-point DFTs of finite duration sequences $x_1(n)$ and $x_2(n)$, then DFT $[a\, x_1(n) + b\, x_2(n)] = a\, x_1(k) + b\, x_2(k)$, a, b are constants.

26. State time reversal property with respect to DFT?

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

$DFT[x((-n))_N] = DFT[x(N-n)] = x((-k))_N = x(N-k)$

27. State circular time shifting property with respect to DFT?

If DFT[x(n)] = x(k), then DFT $[x((n-l))] = x(k)\, e^{-j2\prod k1/N}$

28. What is the basic operation of DIF algorithm? (University)

The basic operation DIF algorithm is called butterfly in which two inputs G(n) and H(n)are combined to give $x_1(k)$ and $x_2(k)$

$$x_1(k)= G(n)+ H(n)$$

$$x_2(k)=\{G(n)- H(n)\}\ W_N^k$$

Where, W_N^k is the **twiddle factor**

UNIT II

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

1. IIR filter
2. 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

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

3. State the structure of IIR filter?

IIR filters are of recursive type whereby the present o/p sample depends on present i/p, past i/p samples and o/p samples. The design of IIR filter is realizable and stable. The impulse response h(n) for a realizable filter is

h(n)=0 for n<=0

4. State the advantage of direct form _structure over direct form _structure.

In direct form _structure, the number of memory locations required is less than that of direct form structure.

5. 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.

6. 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.

7. What do you understand by backward difference?

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

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

The above equation is called backward difference equation.

8. 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 Ω-axis maps onto the perimeter of the circle of radius ½ in the Z-plane.

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

In this method of digitizing an analog filter, the impulse response of resulting digital filter is a sampled version of the impulse response of the analog filter.

10. Give the bilinear transform equation between S-plane & Z-plane.

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

11. 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

$$\mathbf{S=2/T(1-Z^{-1}/1+Z^{-1})}$$

12. 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 Ω-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.

13. What is Warping Effect? (University)

In Bilinear Transformation,

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

For small values of ω,

$\Omega = 2/T\ \omega/2 = \omega/T$

For low frequency, the relationship between Ω and ω are linear. Therefore digital filters have the same amplitude as the analog filter.

But for the high frequency, the relationship is not linear and the distortion is reduced in the digital filter. This is known as warping effect.

14. 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.

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

$\Omega p = 2/T \tan \omega_p/2$

$\Omega s = 2/T \tan \omega_s/2$

15. What are the advantages & disadvantages of bilinear transformation? (University)

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.

16. What is the advantage of cascade realization?

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

17. Define signal flow graph? (University)

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

18. What is transposition theorem & transposed structure?

The transpose of a structure is defined by the following operations.

- Reverse the directions of all branches in the signal flow graph
- Interchange the input and outputs.
- Reverse the roles of all nodes in the flow graph.
- Summing points become branching points.
- Branching points become summing points.

According to transposition theorem if we reverse the directions of all branch transmittance and interchange the input and output in the flow graph, the system function remains unchanged.

19. Define IIR filter?

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

20. 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 Frequency Transformation F

21.Write the expression for order of Butterworth filter? (University)

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

22. Write the steps in designing chebyshev filter?

1. Find the order of the filter.
2. Find the value of major and minor axis λ
3. Calculate the poles.
4. Find the denominator function using the above poles.
5. The numerator polynomial value depends on the value of n.
   If n is odd: put s=0 in the denominator polynomial.
   If n is even put s=0 and divide it by $(1+e^2)^{1/2}$

23. Write the expression for the order of chebyshev filter?

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

UNIT III
FIR FILTER DESIGN

1. What are FIR filters?

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?

1. Choose the desired frequency response $H_d(w)$
2. Take the inverse fourier transform and obtain $H_d(n)$
3. Convert the infinite duration sequence $H_d(n)$ to h(n)
4. Take Z transform of h(n) to get H(Z)

3. What are advantages of FIR filter? (University)

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

4. What are the disadvantages of FIR Filter? (University)

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 linear phase characteristic in FIR
filter? (University)**

The necessary and sufficient condition for linear phase characteristic 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.

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

1. Fourier series method and window method
2. Frequency sampling method. 3. Optimal filter design method.

7. Distinguish between FIR filters and IIR filters.

FIR filter	IIR filter
1. These filters can be easily designed to	These filters do not have linear phase.
2. FIR filters can be realized recursively and non-recursively.	IIR filters are easily realized recursively.
3. Greater flexibility to control the shape of their magnitude response.	Less flexibility, usually limited to specific kind of filters.
4. Errors due to round off noise are less severe in FIR filters, mainly because feedback is not used.	The round off noise in IIR filters is more.

8. What is Gibb's phenomenon? (University)

One possible way of finding an FIR filter that approximates $H(e^{jw})$ would be to truncate the infinite Fourier series 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.

9. List the steps involved in the design of FIR filters using windows.

1. For the desired frequency response $H_d(w)$, find the impulse response $h_d(n)$ using Equation

$$h_d(n)=1/2\pi \int_{-\pi}^{\pi} H_d(w)e^{jwn}dw$$

2. Multiply the infinite impulse response with a chosen window sequence w(n) of length N to obtain filter coefficients h(n),i.e.,

$$h(n)= \begin{cases} h_d(n)w(n) & \text{for } |n|\leq(N-1)/2 \\ 0 & \text{otherwise} \end{cases}$$

3. Find the transfer function of the realizable filter (N-1)/2

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

10. What are the desirable characteristics of the window function?

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 as ω tends to π.

11. Give the equations specifying the following windows.

a. Rectangular window
b. Hamming window
c. Hanning window
d. Bartlett window
e. Kaiser window

a. Rectangular window:

The equation for Rectangular window is given by

$$W(n) = \begin{cases} 1 & 0 \le n \le M-1 \\ 0 & \text{otherwise} \end{cases}$$

b. Hamming window:

The equation for Hamming window is given by

$$W_H(n) = \begin{cases} 0.54-0.46 \cos 2\pi n/M-1 & 0 \le n \le M-1 \\ 0 & \text{otherwise} \end{cases}$$

c. Hanning window:

The equation for Hanning window is given by

$$W_{Hn}(n) = \begin{cases} 0.5[1- \cos 2\pi n/M-1\ ] & 0 \le n \le M-1 \\ 0 & \text{otherwise} \end{cases}$$

d. Bartlett window:

The equation for Bartlett window is given by

$$W_T(n) = \begin{cases} 1-\dfrac{2|n-(M-1)/2|}{M-1} & 0 \le n \le M-1 \\ 0 & \text{otherwise} \end{cases}$$

e. Kaiser window:

The equation for Kaiser window is given by

$$W_k(n) = \begin{cases} \dfrac{I_o[\alpha\sqrt{1-(2n/N-1)^2}]}{I_o(\alpha)} & \text{for } |n| \le \dfrac{N-1}{2} \\ 0 & \text{otherwise} \end{cases}$$

where α is an independent parameter.

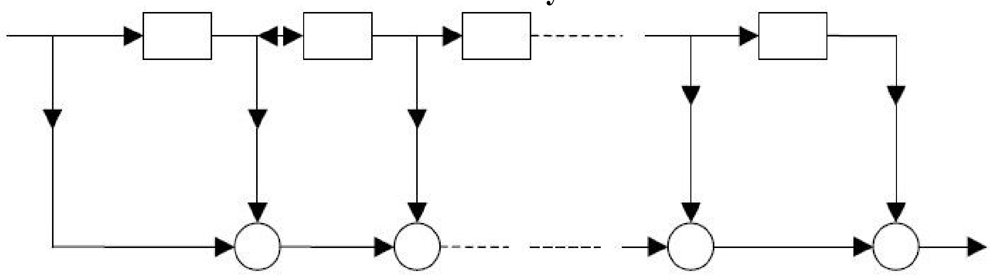
12. What are the advantages of Kaiser window?

- 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 rectangular window.

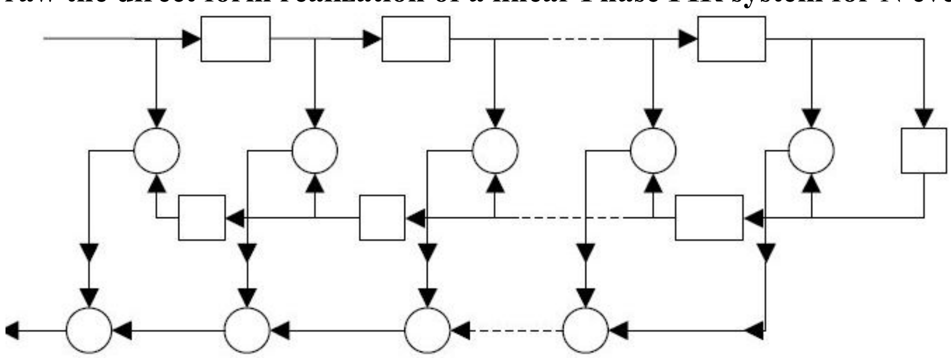
13. What is the principle of designing FIR filter using frequency sampling method?

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 the IDFT of this set of samples.

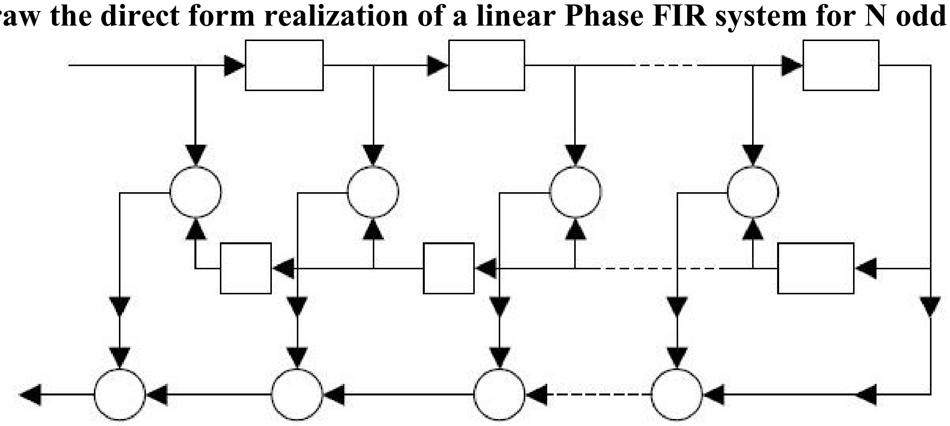
14. Draw the direct form realization of FIR system.



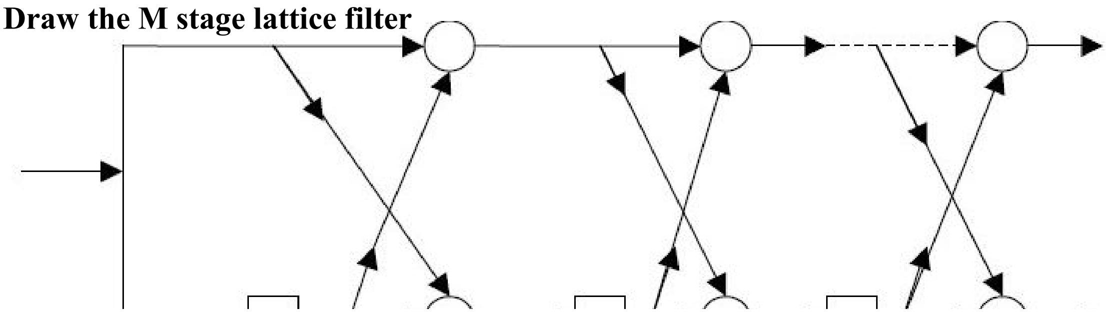
15. Draw the direct form realization of a linear Phase FIR system for N even



16. Draw the direct form realization of a linear Phase FIR system for N odd



17. Draw the M stage lattice filter



18. State the equations used to convert the lattice filter coefficients to direct form FIR Filter coefficient.

$$\alpha_m(0) = 1$$
$$\alpha_m(m) = k_m$$
$$\alpha_m(k) = \alpha_{m-1}(k) + \alpha_m(m) \bullet \alpha_{m-1}(m-k)$$

19. State the equations used to convert the FIR filter coefficients to the lattice filter Coefficient.

For an M_stage filter , $\alpha_{m-1}(0) = 1$ and $k_m = \alpha_m(m)$

$$\alpha_{m-1}(k) = \frac{\alpha_m(k) - \alpha_m(m) \bullet \alpha_m(m-k)}{1-\alpha_m^2(m)} , \quad 1 \le k \le m-1$$

20. Compare Hamming window with Kaiser window.

Hamming window	Kaiser window
1.The main lobe width is equal to $8\pi/N$ 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 α and N. The side lobe peak can be varied by varying the parameter α.

21. List the Butterworth polynomial for various orders.

N	Denominator polynomial
1	S+1
2	S^2+.707s+1
3	(s+1)(s^2+s+1)
4	(s^2+.7653s+1)(s^2+1.84s+1)
5	(s+1)(s^2+.6183s+1)(s^2+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)

22. What are the advantages and disadvantages of FIR filters? (University)

Advantages:

1. FIR filters have exact linear phase.
2. FIR filters are always stable.
3. FIR filters can be realized in both recursive and non recursive structure.
4. Filters with any arbitrary magnitude response can be tackled using FIR sequence.

Disadvantages:

1. 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.
2. Large storage requirement is requirement
3. Powerful computational facilities required for the implementation.

UNIT 4 FINITE WORD LENGTH EFFECTS

1. What are 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.

2. What is meant by fixed point number? (University)

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.

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

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

4. 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.

5. What is meant by 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.

6. 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.

7. What is meant by floating point representation? (University)

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.

8. What are the advantages of floating point representation?

1. Large dynamic range
2. Overflow is unlikely.

9. What are the quantization errors due to finite word length registers in digital filters?(University)

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

10. What is input quantization error?

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.

11. What is 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 a bit which produces the error.

12. What are the different quantization methods? (University)

1. Truncation
2. Rounding

13. What is truncation? (University)

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

14. What is rounding? (University)

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

15. What are the two types of limit cycle behavior of DSP?

1. Zero limit cycle behavior
2. Over flow limit cycle behavior

16. What are the methods to prevent overflow?

1. Saturation arithmetic and
2. Scaling

UNIT V
MULTIRATE SIGNAL PROCESSING

1. State some applications of DSP?

1. Speech processing
2. Image processing
3. Radar signal processing.

2. Define sampling rate conversion.

Sampling rate conversion is the process of converting the sequence x(n) which is got from sampling the continuous time signal x(t) with a period T, to another sequence y(k) obtained from sampling x(t) with another period T'.

3. State the methods to convert the sampling rate.

There are two methods:

- Resampling after reconstruction
- Conversion in digital domain

4. What is multirate signal processing?

Multirate signal processing is the technique of processing the signal with multiple sampling rates.

Advantages:

- Computational complexity is less
- Finite arithmetic effects are less
- Filter order required are low
- Sensitivity to filter coefficient lengths is less

5. State the applications of multirate signal processing.

- Sub-band coding
- Voice privacy using analog phone lines
- Signal compression by subsampling
- A/D and D/A convertors

The above applications come under the areas given below:

- Communication Systems

- Speech and audio processing systems
- Antenna systems
- Radar Systems

6. What is decimation?

Decimation is the process of reducing the sampling rate of the signal. It is otherwise called down- sampling or sampling rate compression.

7. What is interpolation?

Interpolation is the process of increasing the sampling rate of the signal. It is otherwise called down-sampling or sampling rate expansion.

8. Give short note on sub-band coding?

signals which occupy contiguous frequency bands analysis filter bank. These signals are down-sampled, yielding sub- band signals, which are then compressed using encoders. The compressed signals are multiplexed and transmitted. On the receiving side, reverse operations are carried out. This process yields better compression ratio, because each sub-band signal can be represented using a different number of bits.

9. Give brief not on Speech Processing.

Speech processing includes processing like encoding, synthesis and recognition.
Encoding is performed to remove the redundant signal in a speech signal.
Compression/coding is performed in transmitter side and thus **synthesis** is required in receiver side.
Recognition is used to recognize both the speech and the speaker.

10. What are the different techniques of voice compression and coding?

- Waveform coding
- Transform Coding
- Frequency band encoding
- Parametric methods

11. Give short notes on image enhancement.

Image enhancement focuses mainly on the features of an image. The various feature enhancements are sharpening the image, edge enhancement, filtering, contrast enhancement, etc.

12. Give short notes on adaptive filters.

Adaptive filters are linear filters used in various areas where the statistical knowledge of the signals to be filtered/analyzed are not known a priori or the signals may be slowly time variant. Both IIR and FIR filters can be used in adaptive filtering, but FIR filters are mostly used due to its simplicity and adjustable zeros.

13. State the applications of adaptive filtering.

- Adaptive noise cancelling

- Line Enhancing
- Frequency Tracking
- Channel Equalization
- Echo cancellation