1. Prove Symmetry and periodicity property of twiddle factor 6m
2. Establish relation between DFT and Fourier Series 4m
3. Determine the circular convolution of the sequence **x(n)={1,2,3,1} and h(n)={4,3,2,2}** using DFT and IDFT equation 10m
4. Determine the linear convolution of the sequence **x(n)={2,2,3} and h(n)={2,3}** using DFT and IDFT equation 10m
5. Find the IDFT) of **X(k) = {4, 1 -j, -2, 1+j}** 4m
6. The five samples of the 8 point DFT X(k) are given **as X(0) = 0.25, X(1)= 1.25 - j0.3018 , X(6)=X(4)=0, X(5)= 0.125-j0.0518.** Determine the remaining sample if the sequence **x(n) is real valued?** 3m
7. Compute of the sequence **x(n)=sin(3n** $\pi/8$) for N=4 . Plot magnitude and phase of X(k) ((any x(n) can be given)) 6m
8. What is the need of FFT? Determine the following for a 128 point FFT computation number of 1) Stages 2)Butterflies in each stage 3) Butterflies needed for entire computation 4) Total number of complex multiplication 5) Total number of complex Addition 6)speed improvement factor compared with direct computation 8m
9. Determine the circular convolution of the sequence **x(n)={1,2,3,1} and h(n)={0,3,1,2}** using DIT FFT 10m (any sequence can be given)
10. Find the circular convolution of **x(n)={4 ,3,2,1} and h(n)= {1, 0, 1, 0}** using DIF-FFT algorithm (any sequence can be given). 10m
11. Find the IDFT **X(k)={36, -4+i9.7, -4+i4, -4+i1.7, -4, -4-i1.7, -4-i4, -4-j9.7}** using DIF-FFT algorithm show clearly all intermediate steps 10m((any sequence can be given))
12. Determine the 8-point DFT of the sequence **x(n)={8, 8, 8, 8, 1, 1, 0, 0},** sketch the magnitude and phase spectra . (any sequence can be given) 10m
13. What is in-place computation? What is total number of complex addition and multiplication required for N=256, if DFT is computed directly and if FFT is used. 4m

14. Explain bit reversal property used in FFT algorithm for N=16 4m

15. State and prove the circular (i) Time-shift and (ii) Frequency shift properties of an N-point sequence. **6M**

16. Define DFT and IDFT of a signal. Establish relation between DFT and Z-transform. **6M**

17. Let X(k) be a 14-point DFT of length - 14 real sequence x(n). The first K-samples of X(k) are given by **X(0)=12, X(1)=-1+3j, X(2)=3+4j, X(3)=1-5j, X(5)=6+3j, X(6)=-2-3j), X(7)= 10.** Find the remaining samples of X(k).**Also** evaluate the following **i) X(0) ii) X(7) iii)**$\sum_{n=0}^{13} x(n)$ **iv)** $\sum_{n=0}^{13} |x(n)|$ **10M**

18. In the direct computation of N-point DFT of x(n), how many (i) Complex additions (ii) Complex multiplications (iii) Real multiplication (iv)Real additions and (v) Trigonometric functions, evaluations are required? **10M**

19. For **x(n)={ 1, -2, 3-4, 5, -6}** without computing its DFT find the following

1) X(0) 2) $\sum_{k=0}^{5} X(k)$ **3) X(3) 4)** $\sum_{k=0}^{5} |X(k)|^2$ **5)** $\sum_{k=0}^{5} (-1)^k X(k)$ **10M**

20. Derive DIT-FFT algorithm for N=4. Draw the complete signal flow graph. **8M**

21.Find the 8-point DFT of the sequence **x(n)=1 for n even**
0 for n odd, **8M**

22. Consider a signal of length 4 defined by **x(n)={1 2 3 1}.** Compute 4-point DFT by solving explicitly 4X4systemoflinearequationsdefinedbyDFTformula. **6M**

23. A 4-point sequence **x(n)={1,2,3,4}** has DFT X(k) for 0≤K≤3. Find the signal values which has DFT **X((K-1))**$_{\mathbf{N}}$ without performing DFT and IDFT **4M**

24.Compute 4 point circular convolution given by x(n)={1,8,1,8} and h(n)={2,9,2,9} using DFT and IDFT method. **6M**

25. What is the need of FFT? Determine the following for a 128 point FFT computation number of

1) Stages 2)Butterflies in each stage 3) Butterflies needed for entire computation 4) Total number Of complex multiplication 5) Total number of complex Addition 6) speed improvement factor compared with direct computation 8M

26. Explain the shuffling of data and bit reversal as applied to DIT-FFT algorithm for N=8 4M

27. State and prove the following property
1) symmetry property 2)parseval's theorem 8M

28. Explain bit reversal property used in FFT algorithm for N=16 3M

29. Find the IDFT **X(k)={36, -4+i9.7, -4+i4, -4+i1.7, -4, -4-i1.7, -4-i4, -4-j9.7}** using DIF-FFT algorithm show clearly all intermediate steps. 10M

30. An analog signal is sampled at 10 kHz and the DFT of 512 samples is computed. Determine the Frequency spacing between the spectral samples of DFT 5M

31.consider the finite length sequence length **x(n)=δ (n)-2 δ(n-5)** Find i)The 10 point DFT of x(n)
ii) sequence y(n) that has a DFT Y(k)=$e^{-j4\pi k/10}$ X(k)W(k) where X(k) is 10 point DFT of x(n) and W(k) is10 point DFT of W(n)=u(n)-u(n-6) 12M

32. Given the 8 point sequence **x(n)= 1 ; 0≤n≤3**

= 0 ; 4≤n≤7.

Compute DFT of the sequence **x_1(n)= 1 ; n=0**

= 0; 1≤n≤4

= 1; 5≤n≤7. Using the Properties of DFT.

33. Given the sequences **x(n)= cos(n $\pi/2$)** and h(n)=2^n.Compute 4-point circular convolution. 5M

34.Given that **x(n)={2,1}, w(n)=x(n)*y(n)** and **w(n)={6,-1,7,-4 }.** Compute the sequence **y (n)** using DFT. 7M

35. Explain how the DFT can be used to compute N equi spaced samples of Z-Transform of an N point sequence on a circle of radius r. 4M

37. Let x(n) be a given sequence with N points with X(k) the corresponding DFT. Denote the operation of finding the DFT a s follows: **X(k)=F(x(n)).**What is the resulting sequence x(n) operated upon 4 times i.e determine**(k)=F(F(F(F(x(n))))).** **6M**

38. What is linear filtering? Explain how DFT is used in Linear filtering? 6M

39. Find the 4 point DFT of the sequence x(n)={ 1,-1,1,-1},also using time shift property find the DFT of the sequence y(n)=x((n-2))$_4$

40.Compute the 4 point DFT of the sequence x(n)={1,0,1,0}also find y(n), if Y(k)=X((k-2))$_4$

41. The 5 point DFT of a complex sequence x(n) is given as X(k)+{j,1+j,1_j2,2+j2,4+j}.Compute Y(k),if y(n)=x^*(n)

1	**Derive the expression for order and cut-off frequency of Butterworth filter starting from magnitude response.**	
2	Derive an expression for transfer function of normalised analog low pass Butterworth filter of order 3 starting from Magnitude squared response.	
3	**Derive the expression for poles from the squared magnitude response of Butterworth LPF .**	
4	Explain analog to analog frequency transformation.	
5	Design an analog Butterworth LPF to meet the following specification. $0.9 \le \lvert H(j\Omega) \rvert \le 1$ for $0 \le \Omega \; 0.2\pi$ $\lvert H(j\Omega) \rvert \le 0.2$ for $0.4\pi \le \Omega \le \pi$ Note: pass band and stop band frequencies may be given in Hz also.	
6	(1) Design an analog Butterworth LPF to meet the following specifications. Assume T = 1 sec. $0.9 \le \lvert H(j\omega) \rvert \le 1$ $0 \le \omega \le 0.35\pi$ $\lvert H(j\omega) \rvert \le 0.275$ $0.7\pi \le \omega \le \pi$.	
7	(3) Qn: Design a Butterworth analog highpass filter to meet the following specifications. (i) Maximum passband attenuation of 2dB. (ii) Passband edge frequency 200 rad/sec. (iii) minimum stopband attenuation = 20dB. (iv) stopband edge frequency = 100 rad/sec. Note: pass band and stop band frequencies may be given in Hz also.	

8	Design a Butterworth analog bandpass filter that will meet the following specifications. (i) Maximum attenuation of -3.0103 dB at lower and upper cut off frequency of 50 Hz & 20 kHz. (ii) Minimum attenuation of 20 dB at 20 Hz & 45 kHz. (iii) Monotonic frequency response.	
9	**Transform the third order Butterworth normalised LPF to high pass filter with passband edge at 2 rad/s .(Transfer function can be directly written).**	
10	**A prototype low pass filter has the system response H(s)= 1/(s^2 +2s + 1). Obtain a bandpass filter with Ω_0 = 2 rad/s and B= 10 rad/s.**	
11	Design an analog BWLPF with (i) A_p in dB = -2 dB at Ω_p = 100 r/s (ii) A_s in dB = -20 dB at Ω_s = 200 r/s	
12	A Butterworth lowpass filter has to meet the following Specifications a. Passband gain, A_p = -1 dB at Ω_p = 4 rad/sec b. Stopband attenuation greater than or equal to 20 dB at Ω_s = 8 rad/sec Determine the transfer function $H_a(s)$ of the lowest - order Butterworth filter to meet the above specifications.	
13	(20) Design a High pass filter to satisfy the following specifications. (i) Monotonic response (ii) $A_p^{\text{in dB}}$ = -2 dB at 400 rad/sec (iii) $A_s^{\text{in dB}}$ = -20 dB at 100 rad/sec $\lvert H(j\Omega)\rvert$	

14	Design a Butterworth HPF that will meet the following Specification A_p in dB = -2dB at PBEF = 200 r/s A_s in dB = -20dB at SBEF = 100 r/s	
15	Design an analog filter with maximally flat response in the passband and acceptable attenuation of -2dB at 20 rad/s. The attenuation in the stopband should be more than -10dB beyond 30 rad/s.	
16	Design an analog filter with maximally flat response (BW) in the passband and an acceptable attenuation of -2 dB at 20 radians/second. The attenuation in the stopband should be more than 10 dB beyond 30 radians/second.	
17	Design Butterworth filter for following specification $0.8 \leq \|H_a(s)\| \leq 1$ for $0 \leq F \leq 1000\,Hz$ $\|H_a(s)\| \leq 0.2$ for $F \geq 5000\,Hz$	
18	Find the order "N" of a lowpass Butterworth filter to meet the following specifications. $\delta_p = 0.001$, $\delta_s = 0.001$ $\Omega_p = 1$ rad/sec , $\Omega_s = 2$ rad/sec	
19	Differentiate between Butterworth and Chebyshev filter	
20	Differentiate between analog and digital filter	

21	12 $H(s) = \frac{1}{s^2+\sqrt{2}s+1}$ represents the transfer function of a low pass filter with ~~pa~~ cut-off frequency of 1 rad/s. Using frequency transformation, find the transfer function of the following analog filters. a) a low-pass filter with cutoff frequency 10 rad/s. b) a highpass filter with cut-off frequency of 100 ~~Hz~~ rad/s c) a bandpass filter with centre frequency $\Omega_0 = 100$ rad/s and bandwidth $B_0 = 10$ rad/s	
22	Given that $\lvert H(j\Omega)\rvert^2 = \frac{1}{1+16\Omega^4}$, determine (10) the analog filter transfer function H(S).	
23	Design a Butterworth filter for the following specifications: $0.8 \le \lvert H_a(s)\rvert \le 1$ for $0 \le F \le 1\ kHz$ and $\lvert H_a(s)\rvert \le 0.2$ for $F \ge 5kHz$	
24	Design an analog Butterworth low pass filter that has –2 dB or better (i.e., lesser than 2 dB) at frequency of 20 rad/sec and atleast –10 dB of attenuation at 30 rad/sec. Use Butterworth polynomials infactored form are $(s^2+0.7654s+1)$ $(s^2+1.8478s+1)$ **(10 Marks)**	
25	Let $H(s) = \frac{1}{s^2+s+1}$ represent the transfer function of LPF with a passband of 1 rad/sec. Use frequency transformation (Analog to Analog) to find the transfer function of a band pass fitter with passband 10 rad/sec and a centre frequency of 100 rad/sec. **(03 Marks)**	

26	Determine the order and cutoff frequency of the analog butterworth filter such that it has a -2dB pass band attenuation at a frequency of 20 rad/sec and at least 15 dB stop band attenuation at 30 rad/sec. **(9 Marks)**	
27	a. Design an analog bandpass filter to meet the following frequency domain specifications: i) −3.0103 dB upper and lower cut-off frequency of 50 Hz and 20 KHz. ii) A stopband attenuation of atleast 20 dB at 20 Hz and 45 KHz. iii) Monotonic frequency response. **(10 Marks)** b. Find the poles of the polynomial of order 5. Find $H_5(S)$ and gain at $\Omega = 1$ rad/sec in dB, for a Butterworth filter. **(10 Marks)**	
28	A prototype low pass filter has the system response $H(s) = \frac{1}{s^2+2s+1}$. Obtain a band pass filter with $\Omega_0 = 2$ rad/sec and $B_0 = 10$ rad/sec. **(05 Marks)**	
29	Transform $H(s) = \frac{1}{(s^2+s+1)(s+1)}$ normalized Butterworth LPF to HPF with passband edge at 2 rad/sec. **(04 Marks)**	
30	Let $H(s) = \frac{1}{s^2+\sqrt{2}s+1}$ represent the transfer function of a lowpass filter with a passband of 1 rad/sec. Use frequency transformation to find the transfer functions the following analog filters: i) A lowpass filter with passband of 10 rad/sec. ii) A highpass filter with cut off frequency of 10 rad/sec. **(08 Marks)**	
31	a. A system function $H_5(s)$ represents a 1 rad/sec fifth-order normalized butter worth filter, i) Give H5(s) in both the polynomial and factored forms. ii) What is the gain $\lvert H_5(j\Omega)\rvert$ at $\Omega = 1$ rad/sec? What is the gain in decibels? **(10 Marks)** b. Determine the transfer function of a normalized butterworth filter of order N = 6. Show the pole locations in the s-plane. **(10 Marks)**	
32	Transform $H(s) = \frac{1}{(s^2+s+1)(s+1)}$ normalized Butterworth LPF to HPF with passband edge at 2 rad/sec. **(04 Marks)**	
33	Let $H(s) = \frac{1}{(s+1)(s^2+s+1)}$ represent a LPF with passband of 1 rad/sec. Find H(s) for (i) LPF with passband 2 rad/sec. (ii) HPF with cutoff frequency 2 rad/sec. (iii) BPF with passband 10 rad/sec and center frequency of 100 rad/sec. (iv) BSF with stopband of 2 rad/sec and center frequency of 10 rad/sec. **(08 Marks)**	

Digital IIR Filter Questions

1.Derive an expression for Bilinear transformation used for transforming an analog filter to digital filter 10M

2. Design a digital low pass filter to satisfy the following pass band ripple
$1 \leq |H(\Omega)| \leq 0$ for $0 \leq \Omega \leq 1404\,\pi$ rad/sec and stop band attenuation
$|H(j\Omega)| > 60$dB for $\Omega \geq 8268\,\pi$ rad/sec.
Sampling interval $T_s = \frac{1}{10^{-4}}$ sec. Use BLT for designing. 10M

3. Design a digital filter H(z) that when used in an A/D - H(z) – D/A structures given an equivalent analog filter with the following specifications :
Passbandripple : ≤ 3.01 dB, Passband edge: 500Hz
Stopband attenuation: ≥15dB, Stopband edge: 750 Hz
Sample Rate: 2 KHz. Use Bilinear transformation to design the filter on an analog system equation. Use Butterworth filter prototype. Also, obtain the difference equation. 12M

4.Transform the analog filter. $H_a(S) = \frac{s+1}{s^2+5s+6}$ into H(z) using impulse invariant transformation Take T = 0.1 sec. 10M

5. A second-order analog notch-filter has the transfer function $H(s) = \frac{s^2 + \Omega_0^2}{s^2 + Ks + \Omega_0^2}$ using bilinear transformation show that the transfer function H(z) of the digital notch filter is

$$H(z) = \frac{1}{2}\left[\frac{(1+\alpha) - 2\beta(1+\alpha)z^{-1} + (1+\alpha)z^{-2}}{1 - \beta(1+\alpha)z^{-1} + \alpha z^{-2}}\right] \text{where } \alpha = \frac{1+\Omega_0^2 - K}{1+\Omega_0^2 + K} \quad and \quad \beta = \frac{1-\Omega_0^2}{1+\Omega_0^2}$$ 10M

6. A second-order Butterworth lowpass analog filter with a half-power frequency of 1rad/second is converted to a digital filter H(z), using the bilinear transformation at a sampling rate, $\frac{1}{T} = 1Hz$

a. What is the transfer function H(s) of the analog filter?
b. What is the transfer function H(z) of the digital filter?
c. Are the dc gains of H(z) and H(s) identical? Explain.
d. Are the gains H(z) and H(s) at their respective half-power frequencies identical? Explain 10M

7. Design a digital Butterworth filter H(z) given an equivalent analog filter with the following specifications :
passband ripple ≤ 3db , stopband edge frequency of 750Hz , stopband attenuation of 15db , passband edge frequency = 500Hz and sampling rate is 2KHz . Design using bilinear transformation. 9M

8. Convert the analog filter into a digital filter whose system function is$H(s) = 2/(s + 1)(s + 3)$ using bilinear transformation with T = 0.1 sec. 6M

9.Distinguish between IIR and FIR filters. 4M

10. The transfer function of analog low pass filter $H(s)=\frac{(s-1)}{(s^2-1)(s^2+s+1)}$ Find H(z) using impulse invariance method. Take T=1sec. 6M

11. Let $H(s)=\frac{1}{s^2+\sqrt{2s}+1}$, a second-order low pass Butterworth filter prototype having the half-power point at $\Omega=1$. Determine the system function for the digital bandpass filter using bilinear transformation. The cutoff frequencies for the digital filter should lie at $\omega L =\frac{5\pi}{12}$ and $\omega u= \frac{7\pi}{12}$. Take T=2. 6M

12**.** Design the digital lowpass Butterworth filter using Bilinear transformation method to meet the following specifications. Take T=2 sec.

Passband ripple $\leq$1.25dB, Passband edge =200Hz
Stopband attenuation$\geq$15dB, Stopband edge = 400Hz
Sampling frequency= 2kHz 12M

13.A z-plane pole-zero plot for a certain digital filter is shown in figure Q8 (b). The filter has unity gain at DC. Determine the system function in the form,

$H(z)=A\left[\frac{(1+a1z^{-1})(1+b1z^{-2}+b2z^{-2})}{(1+c1z^{-1})(1+d1z^{-1}+d2z^{-2})}\right]$ giving the numerical values for parameters A,a1,b1,b2,c1,d1 and d2.

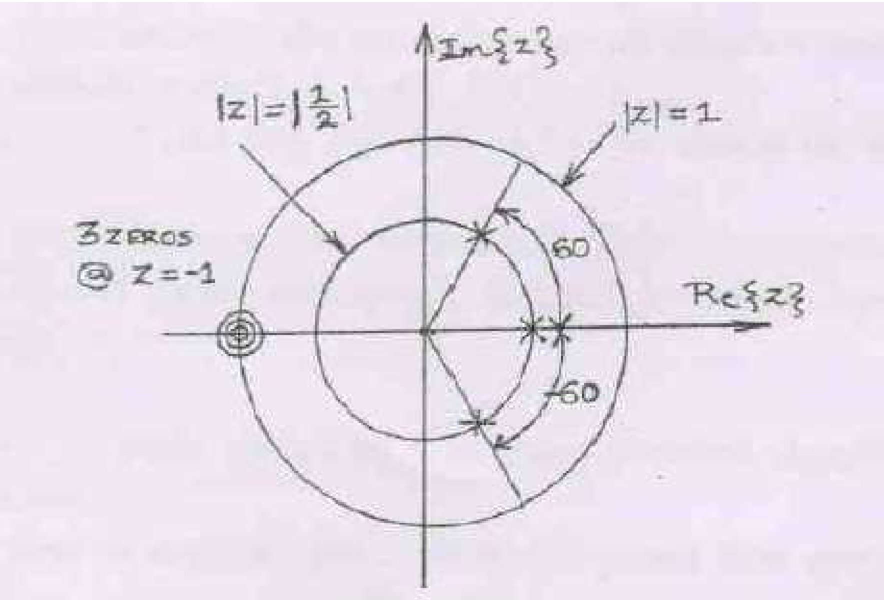

Fig. Q8 (b)

14. Convert the analog with system function Ha(s)=$\frac{(s+0.1)}{(s+0.1)^2+9}$ into a digital filter (IIR) by means of impulse invariance method. 8M

15. A digital lowpass filter is required to meet the following specifications

$$20\log|H(\omega)|_{\omega=0.2\Pi} \geq -1.9328dB$$

$$20\log|H(\omega)|_{\omega=0.6\Pi} \leq -13.9794dB$$

The filter must have a maximally flat frequency response. Find H(z) to meet the above specifications using impulse invariant transformation.

FIR Filter Questions

1. A filter is designed with the following desired frequency response

$$Hd(w) = \begin{cases} 0, & -\frac{\pi}{4} \leq w \leq \frac{\pi}{4} \\ e^{-j2w}, & \frac{\pi}{4} \leq |w| \leq \pi \end{cases}$$

Find the frequency response of the FIR filter designed using a rectangular window defined as

$$\mathrm{WR(n)} = \begin{cases} 1, & 0 \leq \mathrm{n} \leq 4 \\ 0, & \text{otherwise} \end{cases}$$ 10M

2. A LPF is to be designed with the following desired frequency response:

$H_d(e^{j\omega}) = H_d(\omega) = e^{-j3w}$;$0\leq |\omega| \leq \pi/4$

0 ; $\pi/4 \leq \omega \leq \pi$

Determine the filter coefficients h(n) , given rectangular window w(n) defined

w(n) = {1 ; $0\leq n \leq 4$

{0 ; otherwise 7M

3. Determine the filter coefficients h(n) obtained by sampling $H_d(\omega)$ given by

$H_d(e^{j\omega}) = \{e^{-j3w}$; $0 \leq |\omega| \leq \pi/2$

{ 0 ; $\pi/2 \leq \omega \leq \pi$

Also obtain the frequency response H(ω). Take N = 7. 10M

4. Realise FIR linear phase filter for N to be even. 8M

5. For the desired frequency response

$H_d(e^{j\omega}) = \{e^{-j3w}$; $-3\pi/4 \leq \omega \leq 3\pi/4$

{ 0 ; $3\pi/4 \leq |\omega| \leq \pi$

Find H(ω) for N = 7 using hamming window. 10M

6. Mention few advantages and disadvantages of windowing technique. 5M

7. Consider the pole zero plot, as shown in below fig

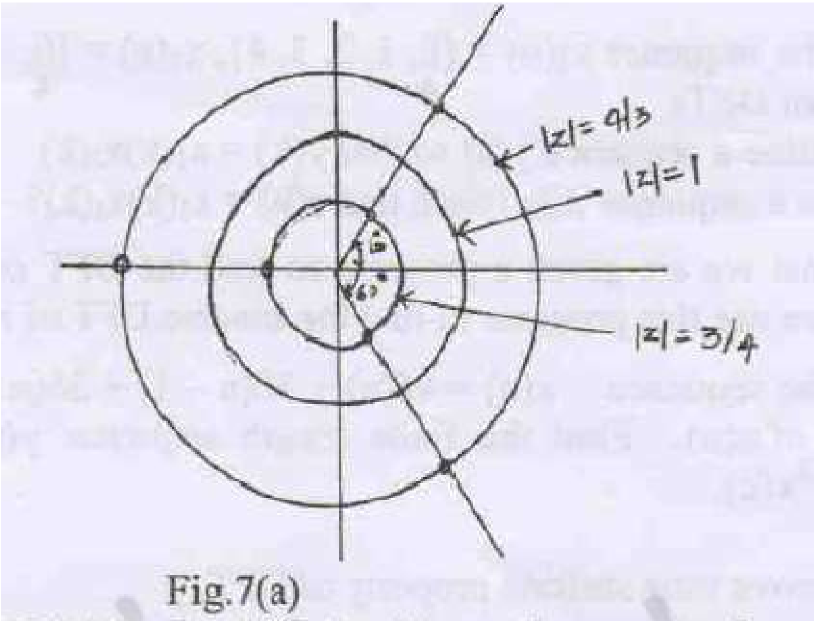

Fig.7(a)

i) Does it represent an FIR filter?

ii) Is it a linear phase system? 4M

8. Figure below shows the frequency response of an infinite-length ideal multi-band real filter. Find h(n), impulse response of this filter. Present the sketch of implementation of ω(n)h(n)(Truncated impulse response of this filter) via block diagram. Where ω(n) is a finite length window sequence?

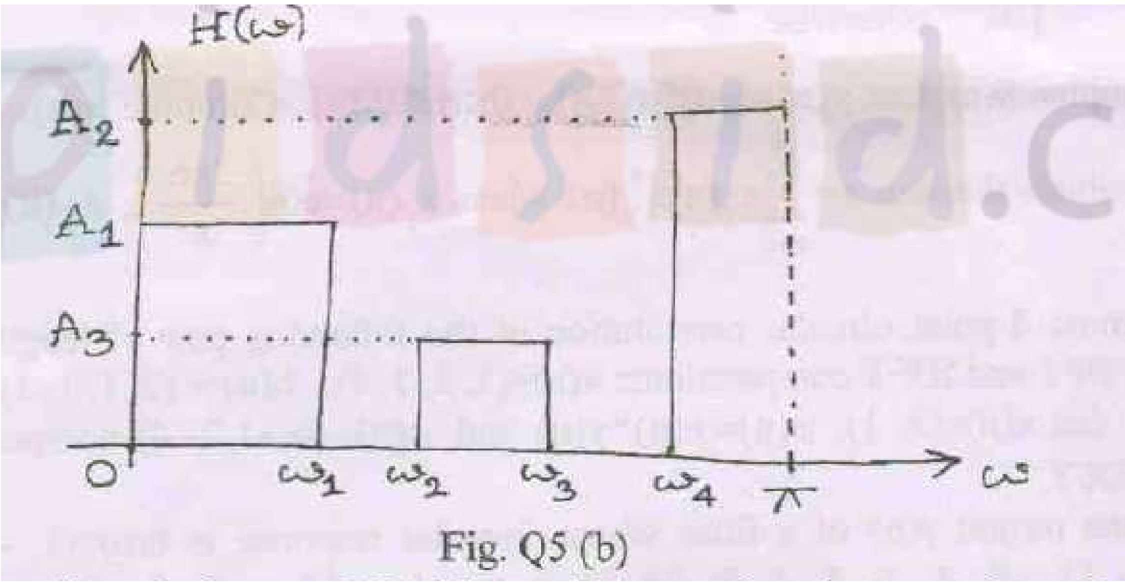

Fig. Q5 (b)

9. We are interested to design an FIR filter with a stopband attenuation of 54db and $\Delta\ \omega=0.05\pi$ using windows. Provide the means to achieve precisely this attenuation using suitable window function. 3M

10.Design a linear phase highpass filter using the Hamming window for the following desired frequency response.

$$\mathrm{Hd}(\omega)=\begin{cases} e^{-j3\omega} & \frac{\pi}{6} \le |\omega| \le \pi \\ 0 & |\omega| < \frac{\pi}{6} \end{cases}$$

$\omega(n)=0.54-0.46\cos\left(\frac{2\pi n}{N-1}\right)$, where N is the length of the Hamming window. 8M

11. Design a linear phase lowpass FIR filter with 7taps and a cut off frequency of $\omega c= 0.3\pi$ using the frequency sampling method. 6M

12 **.** Derive an expression for frequency response of a symmetric impulse response for N-odd. 8M

13. A lowpass filter is to be designed with the following desired frequency response:

$$\mathrm{Hd}(e^{j\omega}) = \begin{cases} e^{-ij\omega}, & |\omega| \le \frac{\pi}{4} \\ 0, & \frac{\pi}{4} \le |\omega| \le \pi \end{cases}$$

Determine the filter coefficients hd(n) h(n) if $\omega(n)$ is a rectangular window defined as follows

$$\mathrm{WR(n)}=\begin{cases} 1, & 0 \le n \le 4 \\ 0, & otherwise \end{cases}$$

Also, find the frequency response, H(ω) of the resulting FIR filter. 12M

14. List the steps in the design procedure of a FIR filter using window functions. 6M

15**.** List the advantages and disadvantages of FIR filter. 4M

16. Obtain the coefficients of an FIR filter to meet the specifications given below using the window method.

Passband edge frequency: 1.5KHz
Stopband edge frequency: 2KHz
Minimum stopband attenuation: 50dB
Sampling frequency: 8KHz (Obtain minimum 10 coefficients) 12M

17. An analog signal contains frequencies upto10KHz. This signal is sampled at 50KHz. Design an FIR filter having a linear-phase characteristic and a transition band of 5KHz. The filter should

provide minimum 50dB attenuation at the end of transition band(Obtain minimum of 10 coefficients) 12M

18. Derive the expression and Realize the FIR filter based on frequency sampling design 10M

19. A LPF is to be designed with the following desired frequency response:

$$H_d(e^{j\omega}) = H_d(\omega) = e^{-j3w} \quad ;0\leq |\omega| \leq \pi/2$$

$$0 \quad ; \pi/2 \leq \omega \leq \pi$$

Determine h(n) based on frequency-sampling technique. Take N=7.

20. Determine the filter coefficients h(n) obtained by sampling Hd(ω) given by

$$H_d(\omega) = H_d(\omega) = e^{-j3w} \quad ;0\leq |\omega| \leq \pi/2$$

$$0 \quad ; \pi/2 \leq \omega \leq \pi$$

Also obtain the frequency response H(ω). Take N=7.

1. Consider a IIR filter with system function

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

Sketch the direct form-I and direct form-II realizations of the filter

2. Starting from the basic equation for filters show how to realize IIR filter in Direct form-II.
3. Derive lattice structure for realization of FIR system.
4. A LTI system is described by the following input- output relation 2y(n)-y(n-2)-4y(n-3)=3x(n-2). Realize the system in DF-I and DF-II
5. Consider a FIR filter with system function H(z) = 1 + 2.88z-1 + 3.4048z-2 + 1.74z-3 + 0.4z-4 Sketch the direct form and lattice realizations of the filter and determine in detail the corresponding input - output equations.(Any equation defining filter can be given)
6. Realize the FIR filter given by $h(n)=(\frac{1}{2})^n[u(n)-u(n-4)]$ using direct form I 6m
7. Obtain the Direct form I, II and cascade realization of the following system

Y(n)= 0.75y(n-1)-0.125y(n-2)+6x(n)+7x(n-1)+x(n-2) .(Any equation defining filter can be given)

8. Obtain the direct form realization of the linear phase FIR filter given by $1+2/3Z^{-1}+15/8Z^{-2}$
9. Let the coefficients of a 3 stage FIR filter lattice structure be $k_1=0.1$, $k_2=0.2$, $k_3=0.3$. Find the coefficients of the direct form I FIR filter and draw its block diagram. .(K1,K2,K3 can be of any value)
10. Determine the coefficients K_m of the lattice filter corresponding to FIR filter described by the system function $H(Z)= 1+2Z^{-1}+1/3Z^{-2}$. Also draw the corresponding second order lattice structure.(Can be of any system H(Z))
11. Given the FIR filter with the following difference equation y(n)=x(n)+3.1x(n-1)+5.5x(n-2)+4.2x(n-3)+2.3x(n-4) . Sketch the lattice realization of the filter. (Any difference equation is possible)
12. A FIR filter is given by the following difference equation determine its lattice form. y(n)=2x(n)+4/5x(n-1)+3/2x(n-2)+2/3x(n-3)

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Question Bank for DSP Processor

Module4

1. Compare Von Neumann and Harvard Architecture of processors.
   Or
   List and briefly explain the feature that show that Harvard Architecture of processors is efficient compared to Von Neumann architecture.
2. Illustrate Von Neumann and Harvard Architecture of processors and explain in detail.
3. What is significance of Multiplier and Accumulator (MAC) Unit in DSp? Illustrate and explain the operation of MAC unit.(Li Tan).
4. What is shifter? Explain how shifter helps in handling overflow in DSp?(Li Tan)
5. Explain the operation of circular buffer.
6. Explain Fixed point number representation and Floating point number representation used in DSp s
7. Explain IEEE Floating-Point Formats
8. Convert the Q-15 signed numbers to the decimal.
   a. i)0110100100110100 ii) 1.101101110100010
   b. iii)1010111100110100 ii) 0.101101110100010
9. Find the Q-15 representation for the decimal number

   i) -0.2567 ii) 0.45632 iii) 245 iv) -253

10. Add the following two-Q-15 numbers and convert sum to decimal.

    i)1.101010111000001 + 0.010001111011010

    ii) 1. 1 1 0 1 0 1 1 1 0 0 0 0 0 1 0 + 0.1 0 0 0 1 1 1 1 0 1 1 0 0 1 0
11. Add the following floating-point numbers and determine the sum in decimal. The two numbers are in the format → **4bit (MSB) for exponent and lower 12 bits for mantissa**
    1101 011100011011 + 0100 101111100101

12. Multiply the numbers by converting them to floating format and verify the result. (Use the format specified in above problem)
    0.640136718 x 2^-2 & -0.638183593 x 2^5
13. Add the following two floating-point numbers and check for the overflow (Use same format in above problem)
    0111 011000000000 & 0111 010000000000
14. Convert the following IEEE single precision format to the decimal format
    101000000.010…0
15. Convert the following number in IEEE double precision format to the decimal format:
    001000 . . . 0:110 . . . 0000

Module5

16. Draw the functional block diagram of TMS320C6713 architecture and explain in detail.

17. List the salient features of TMS320C6713 Digital Signal Processor.
18. Explain Functional units with respect to TMS320C6X processor.
19. Explain the memory configuration of C6x processors.
20. Illustrate and briefly explain internal memory configuration of L2 memory in C6x processor.
21. Illustrate the mode of operation of indirect addressing mode in TMS320C6X processor with examples.
22. Explain in brief the Pipelining operation of TMS320C6X processor with an example. **Or** What is pipelining? Illustrate and explain different stages and phases of pipelining.
23. Explain Fetch and execute packets in TMS320C6X processor.
    Or
    Illustrate how DSp takes care of parallel instructions.
24. Illustrate the working of add, subtract and multiply instruction with respect to TMS320C6X processor showing example.
25. What is AMR register? Explain how it is used in circular addressing mode with an example.
26. What is the bit pattern to be loaded in AMR for following requirement in circular buffer addressing? Write the code segment for the same.
    i) Register B6 has to be used as pointer for circular addressing using Bk1
    ii) Buffer size of 128 bytes to be set.

27. Explain the assembly code format used in DSp
28. Explain with examples ADD/SUB/MPY instructions
29. Explain with examples i)Branch/Move instructions ii)Load/Store instructions
30. Explain the operation done by following instructions
    i. ADD .L1 A3,A7,A7
    ii. LDH .D2 *B2++,B7
    iii. MPY .M2 A7,B7,B6
    iv. NOP 5
    v. LDH .D2 *B2++, B7
    Or
    Identify the addressing modes in the following and explain the operation
    i) ADD ++A7, A8 ii) ADD A7, A8++
    iii) MVKL .S1 x,A4 iv) STW .D2 A1, *+A4[20]

31. What is assembler directive? Identify the operation done by assembler directive **.**float, **.**double, **.***sect.*
32. Illustrate indirect addressing mode with examples.
    Or Explain pre increment, post decrement addressing with examples.
33. What is the hierarchy (priority) of interrupts in C6x processor? Write the interrupt service table.
34. Explain the various interrupt control registers
35. Explain how DSp processes the maskable and nonmaskable interrupts.
36. Explain Interrupt Acknowledgment process in detail.