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 ≤1.25dB, Passband edge =200Hz
Stopband attenuation≥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.

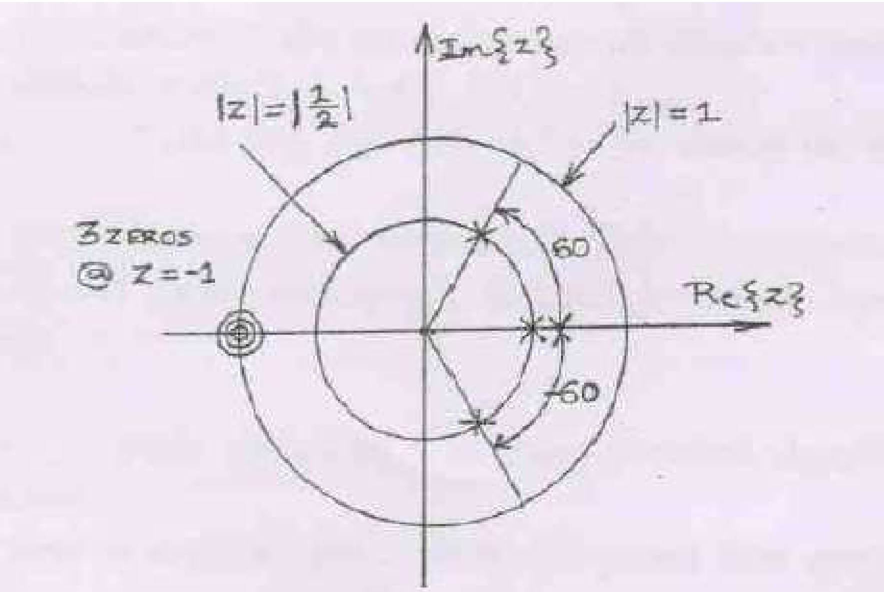

Fig. Q8 (b)

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≤ |ω| ≤ π/4

0 ; π/4 ≤ ω ≤ π

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

w(n) = {1 ; 0≤ n ≤ 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 ≤ |ω| ≤ π/2

{ 0 ; π/2 ≤ ω ≤ π

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

$H_d(e^{j\omega}) = \{e^{-j3w}$; -3π/4 ≤ ω ≤ 3π/4

{ 0 ; 3π/4 ≤ |ω| ≤ π

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

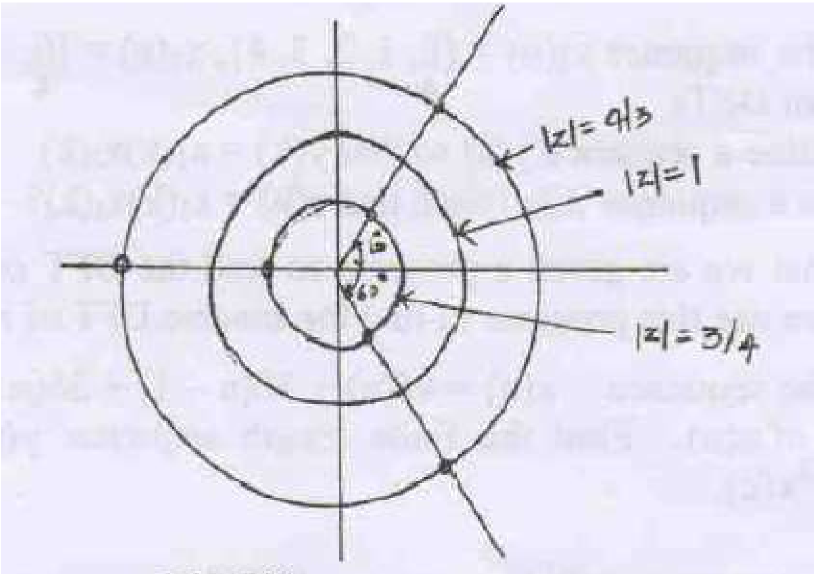

Fig.7(a)

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?

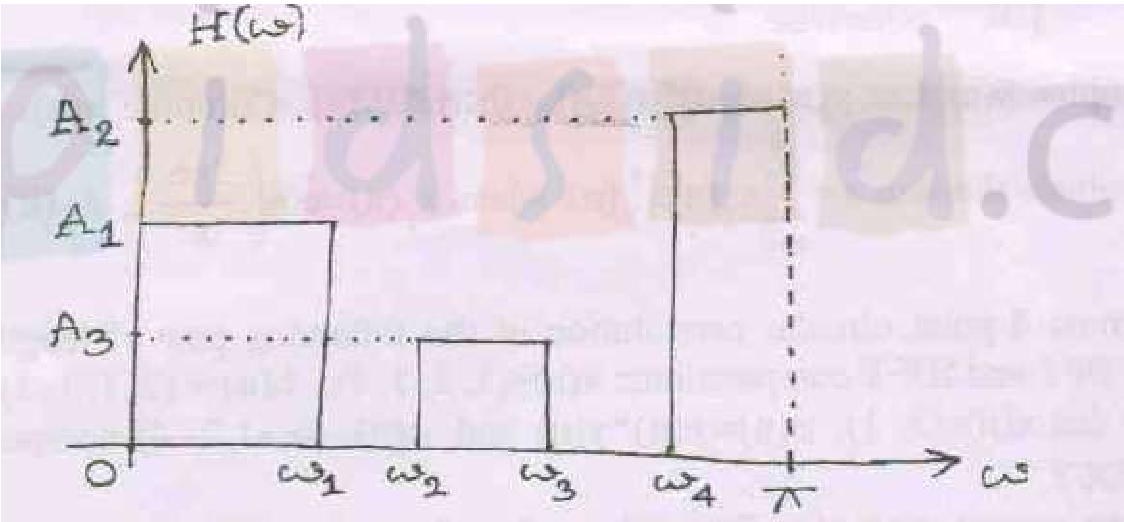

Fig. Q5 (b)

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.

$$\mathrm{Hd}(\omega)=\begin{cases} e^{-j3\omega} & \frac{\pi}{6} \le |\omega| \le \pi \\ 0 & |\omega| < \frac{\pi}{6} \end{cases} \quad \omega(n)=0.54-0.46\cos\left(\frac{2\pi n}{N-1}\right),$$ 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:

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

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

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

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