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DEPARTMENT OF ELECTRONICS & COMMUNICATION ENGG

EC 6502/PRINCIPLES OF DIGITAL SIGNAL PROCESSING (REGULATION 2013)

III YEAR/ V SEMESTER

UNIT WISE EXPECTED UNIVERSITY EXAMINATION QUESTIONS

QUESTION BANK

UNIT-1

Discrete Fourier Transform

Part-A

1. State sampling theorem?
2. **Define DFT pair equation? (NOV/DEC 11)**
3. State and prove the parseval s relation for DFT?
4. **What do you meant by the term 'Bit reversal' applied to FFT?**
5. Draw and explain the basic butterfly diagram of DIF & DIT FFT algorithms?
6. **List out four properties of DFT? (NOV/DEC 10)**
7. State complex conjugate DFT property?
8. What is correlation?
9. What is overlap add method & save method? **(MAY/JUN 13)**
10. How many addition & multiplication are needed to compute 'N'-Point FFT?
11. What is meant by twiddle factor?
12. **Relation between DFT & DTFT? (NOV/DEC 14)**
13. Why the result of linear & circular convolution is not same?
14. **What are the advantages of direct computation of DFT? (NOV/DEC 11)**
15. State the properties of twiddle factor that are used in FFT
16. Find the circular convolution of x(n()={1,2,-1} and h(n)={1,2,2}
17. State there advantages of FFT over DFTs. **(NOV/DEC 10)**
18. Compute the IDFT for the sequence X(k)={1,4,1,4}
19. Compute the 4 point DFT using DIT FFT algorithm for the sequence x(n)={1,2,3}
20. **State the properties of Twiddle factor that are used in FFT.**
21. Draw the butterfly diagram for decimation in frequency FFT algorithm.
22. **How many stages of decimation are required in the case of a 64 point radix 2 DIT FFT algorithm**?

PART-B

1. (a) Obtain an 8-point DIF FFT flow graph from the first principle (8)
   (b) **Using the above flow graph compute DFT of x (n) = {cosnΠ/4, where 0<_n<_7** (8)
2. (a) Determine the 8-point DFT of the sequence x(n)={0,0,1,1,1,0,0,0} (16)
   (b) 1. State and prove the convolution theorem of DFT (8)
   2. **Explain DIT FFT algorithm & flowchart (NOV/DEC 10,11)** (8)
3. (a) Find the DFT of the sequence x(n)={1,1,0,0} & find the IDFT of the Sequence y (k) = {1, 0, 1,0} (8)
   (b) Draw the butterfly diagram using DIT FFT algorithm for the following sequence x(n)= {1,0,0,0,0,0,0,0,} (16)

4. **(a) 1. Draw the butterfly flow diagram of the radix-2 DIT FFT algorithm** **(8)**
   2. Compute the DFT of the sequence x (n) = {0, 1, 2, 3} using DIF FFT Algorithm (NOV/DEC 11) (8)

   (b) 1. Determine the 8-point DFT of the sequence x(n)={1,1,1,1,1,1,0,0} (8)
   2. What are the steps to be involved for the radix DIT FFT algorithm? Explain. **(NOV/DEC 10)** (8)

5. (a) draw the 8-point DIF FFT flow graph of the sequence x(n)={2,2,2,2,1,1,1,1} (8)

   (b) Obtain the 8-point DIT FFT flow graph of the radix-2 algorithm. (8)
6. find the 8-point DFT of the sequence by using x(n)={0,1,2,3,4,5,6,7} (16)
7. **find the DFT of the sequence x(n)={1/2,1/2,1/2,1/2,0,0,0,0}(MAY/JUNE 12)** (8)
8. Find the IDFT for the sequence by using DIF FFT algorithm (16)
   x(n)={28,-4+9.656j, -4+4j,-4+j1.656,-4,-4-1.656j,-4-4j,-4-j9.656}

UNIT-II
IIR Filter Design

PART A

1) Compare Butterworth and chebyshev filters?
2) **What is prewarping? Why it is needed? (NOV/DEC 13,11)**
3) **What are the limitations of impulse invariant technique? (MAY/JUN 11)**
4) Give the transform relation for converting the low pass to band pass in analog domain?
5) **Find the digital filter equivalent for H(s)=1/(s+8). (NOV/DEC 12)**
6) Determine the order of the analog Butterworth filter that has 2db pass band attenuation at a frequency of 20rad/sec and at least -10db stop band attenuation at 30rad/sec.
7) Find the digital transfer function h(z) by using impulse invariant method for the analog transfer function H(S)=1/S+2 .assume that T=0.1 sec.
8) **State the relationship between the analog and digital frequencies when converting an analog filter to digital filter using bilinear transformation**.
9) Sketch the mapping of S-plane and Z-plane in bilinear transformation.
10) **What is warping effect? (NOV/DEC 10)**
11) **State the properties of Butterworth filter**. **(NOV/DEC 11)**
12) Give the expression for location of poles of normalized Butterworth filter.
13) Distinguish between FIR and IIR system.
14) List the various method of calculating the coefficients of IIR filter.
15) What are the salient features of low pass Butterworth filter?
16) What are the different structures used for realizations of an IIR system?
17) State two advantage of bilinear transformation.
18) **Mention the properties of chebyshev filter**. **(NOV/DEC 13)**
19) Draw the response curve of Butterworth and chebyshev.
20) What are the advantages of direct form-II realization when compared to direct form-I.
21) What is the relationship between the Z-transform and Laplace transform?
22) **What is frequency warping? (NOV/DEC 11,12)**
23) What are the characteristics of IIR?
24) How a digital filter is designed using impulse invariance method?
25) **Why IIR filter do not have linear phase?**
26) What are the requirements for an analog filter to be stable and causal?
27) What is importance of poles in filters design?
28) What are the properties that are maintained same in the transfer of analog filter into a digital filter?
29) Define ripple in a filter?

30) Find the digital transfer function h (z) by using impulse invariant method for the analog transfer function H(s) =1/(s+2). Assume T=1sec.

31) Write a note on frequency transformation.

PART B

1. Derive bilinear transformation for an analog filter with system function
   H(s) = b / (s + a) (8)
2. Obtain the(i) Direct forms ii) cascade iii) parallel form realizations for the following systems
   y (n) = 3/4(n-1) – 1/8 y(n-2) + x(n) +1/3 x(n-1) (16)

3.Derive the mapping relation in approximation of derivatives using backward difference method. (6)

4. Design a digital low pass Butterworth filter using bilinear transformation method to meet the following specifications. Pass band ripple ≤ 1.25 dB, pass band edge = 200Hz, stop band attenuation ≥ 15 dB, stop band edge = 300 Hz. Sampling frequency = 2 kHz. (16)

5. i)Find the transfer function of a low pass analog Chebyshev filter to meet the following requirements.
Pass band edge 1 rad /sec, pass band ripple 0.1 dB, stop band attenuation is at least 40 dB for 2 rad /sec. (NOV/DEC 11) (12)

ii) Compare IIR and FIR filters. (4)

6. Obtain the direct form I, direct form II, cascade and parallel structures for the system given by the difference equation y(n) = y(n-1) -1/2 y(n-2) + x(n) – x(n-1) + x(n-2) (16)

7. **Convert into Digital filter using bilinear transform using cutoff frequency=1 R/S and T=1 sec.**
H(s) = 2 / {(s+4)(s+2)}

8. Obtain the direct form-I,II, cascade and parallel form realizations for the following systems
Y (n) = -0.1(n-1) + 0.2 y (n-2) + 3x (n) +3.6 x (n-1) +0.6 x (n-2) (16)

9. Realize the following system using parallel and cascade forms. Use second order sub-systems. (16)

$$H(z)=\frac{10\left(1-\frac{1}{2}z^{-1}\right)\left(1-\frac{2}{3z^{-1}}\right)\left(1+2z^{-1}\right)}{\left(1-\frac{3}{4}z^{-1}\right)\left(1-\frac{1}{8z^{-1}}\right)\left(1-\left(\frac{1}{2}+\frac{j1}{2}\right)z^{-1}\right)\left(1-\left(\frac{1}{2}-\frac{j1}{2}\right)z^{-1}\right)}$$

10. **Convert the analog filter with system function $H_a(s)=\frac{s+0.1}{(s+0.1)^2+9}$ into a digital filter by means of the impulse invariance method.(NOV/DEC 11)** (8)

11. Convert the analog band pass filter $H(s)=\frac{(\Omega u-\Omega l)s}{s^2+(\Omega_u-\Omega l)s+\Omega u\Omega l}$ with upper and lower band edge frequencies Ωu and Ωl respectively into a digital filter by means of a bilinear transformation. Assume $w_u=3\pi/5$ and $w_l=2\pi/5$. (16)

12 .Find the direct form II .H (z) $=8z^{-2}+5z^{-1}+1 / 7z^{-3}+8z^{-2}+1$ (8)

13. Find the direct form –I,II, cascade and parallel form for (16)
   $H(Z) = z^{-1} -1 / 1 – 0.5 z^{-1}+0.06 z^{-2}$

14. Explain the method of design of IIR filters using bilinear transform method. (16)

15. **For the analog transfer function H(s) = 2 / (s+1) (s+3) .Determine H (z) using bilinear transformation. With T=0.1 sec (NOV/DEC 10)** (8)

16. a)Convert the analog filter H(s) = 0.5 (s+4) / (s+1)(s+2) using impulse invariant transformation T=0.31416s (8)

17. For the constraints 0.8 ≤ |H (e^{jw})| ≤1, 0 ≤ ω ≤ 0.2π ; |H (e^{jw})| ≤0.2, 0.6π ≤ ω ≤π with T= 1 sec .Determine system function H(z) for a Butterworth filter using Bilinear transformation. (16)

18. Design a digital Butterworth filter satisfying the following specifications 0.7 ≤ |H (e^{jw})| ≤1, 0 ≤ ω ≤ 0.2 π ;
|H (e^{jw})| ≤0.2, 0.6π ≤ ω ≤π with T= 1 sec .Determine system function H(z) for a Butterworth filter using impulse invariant transformation. (16)

19. Design a digital Chebyshev low pass filter satisfying the following specifications 0.707 ≤ |H (e^{jw})| ≤1, 0 ≤ ω ≤ 0.2π ; |H (e^{jw})| ≤0.1 0.5 ≤ ω ≤π with T= 1 sec using for bilinear transformation. (16)

20.**Design a digital Butterworth High pass filter satisfying the following specifications**
0.9 ≤ |H (e^{jw})| ≤1, 0 ≤ ω ≤ π/2
|H (e^{jw})| ≤0.2, 3π/4 ≤ ω ≤π with T= 1 sec. using impulse invariant transformation
(16)

21. Design a digital filter using bilinear transformation for the following specifications

i) Monotonic pass band and stop band
ii) -3.01 db cut off at 0.5 π rad
iii) Magnitude down at least 15 db at $\omega = 0.75\ \pi$ rad. (16)

22. **Obtain an analog Chebyshev filter transfer function that satisfies the constraints. (16)**
$1/\sqrt{2} \leq |H(j\Omega)| \leq 1; \quad 0 \leq \Omega \leq 2$ (NOV/DEC 11, 12)
$|H(j\Omega)| < 0.1; \quad \Omega \geq 4$

23. Obtain H (Z) from Ha(S) when T=1sec and Ha(S) $=2S/S^2+0.2S+1$ using bilinear transformation.
24. Obtain H (Z) from Hs(S) when T=1sec and Ha(S) $=S^3/(S+1)(S^2+S+1)$ using bilinear transformation.
25. Obtain H (Z) from Hs(S) when T=1sec and Ha(S) $=(S+0.3)/(S+0.3)^2+16$ using bilinear transformation.
26. Design the Butterworth filter and transferred into digital filter by impulse invariant method. The specification of desired low pass filter is, (16)

$0.8 \leq |H(w)| \leq 1.0; \quad 0 \leq w \leq 0.2Z$
$|H(w)| \leq 0.2; \quad 0.32Z \leq W \leq Z.$

27. **Design the chebyshev digital filter using bilinear transformation is (16)**
$0.8 \leq |H(w)| \leq 0.1; 0 \leq w \leq 0.2Z$ (NOV/DEC 132)
$|H(w)| \leq 0.2; \quad 0.32Z \leq w \leq Z$

28. Design the Butterworth filter using bilinear transformation and the specification of the desired low pass filter is,

$0.8 \leq |H(w)| \leq 1.0; \quad 0 \leq w \leq 0.2Z$
$|H(w)| \leq 0.2; \quad 0.32Z \leq W \leq Z.$ (16)

29. Design the Chebyshev filter using by impulse invariant and the specification of the desired low pass filter is,

$0.9 \leq |H(w)| \leq 1.0; \quad 0 \leq w \leq 0.25Z$
$|H(w)| \leq 0.24; 0.5 \leq w \leq Z$ (16)

UNIT-III
FIR Filtre Design

Part-A

1. Write procedure for designing FIR filter using windows?
2. **What do you meant by linear phase FIR filter? (NOV/DEC 10)**
3. What is the condition for linear phase characteristics of an FIR filter?
4. Why is window function used in FIR design?
5. What are the desirable characteristics of the frequency response of the window function?
6. **Define Hamming and Blackman window functions. (NOV/DEC 11)**
7. What are the merits and demerits of FIR filters?
8. **Distinguish between recursive and no recursive realization. (APR/MAY 13)**
9. What are the different types of filters based on impulse response?
10. What are the different types of filters based on frequency response?
11. What are the desirable and undesirable features of FIR filters?
12. What are the design techniques of designing FIR filters?
13. **What is the condition for the impulse response of FIR filter to satisfy for constant group and phase delay and for only constant group delay?**
14. What condition on the FIR sequence h(n) are to be imposed in order that this filter can be called a linear phase filter?
15. State the condition for a digital filter to be causal and stable.
16. What are properties of FIR filter?
17. **How the zeros in FIR filter is located? Explain briefly? (NOV/DEC 12)**
18. What is Gibbs phenomenon or oscillations?
19. What are the desirable characteristics of the window?
20. What is the principle of designing FIR filter using windows?
21. **What is the necessary and sufficient condition for linear phase characteristics in FIR filter?**

22. What is the principle of designing FIR filter using frequency sampling method?
23. For what type of filters frequency sampling method is suitable?
24. Mention the merits and demerits of FIR filter.
25. **Show that the filter with h (n) = {-1, 0, 1} is a linear phase filter. (MAY/ JUN 12)**

Part- B

1. Describe the design of FIR filters using frequency sampling technique. (8)
2. Explain the principle and procedure for designing FIR filter using hamming window. (8)
3. **The desired response of a LPF is (NOV/DEC 10) (16)**

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

Determine H(ω) for M=7 using Hamming window and rectangular window. (16)

4.The desired response of a LPF is **(APR/MAY 13)**

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

Determine H(w) for M=7 using Hamming window and rectangular window. (16)

5. **Determine direct form realization for $H(z) = 1 + 2z^{-1} + 3z^{-2} + 4z^{-3} - 5z^{-4}$ (NOV/DEC 11)**
(8)

6. Obtain the cascade form realization of FIR system
$H(z) = 1 + 5/2\, z^{-1} + 2z^{-2} + 2z^{-3}$ (8)

7. Design a band pass filter which approximates the ideal filter with cutoff frequencies at 0.2 rad/sec. The filter order is N=7. Use hamming window. (16)

8. Design a linear phase FIR filter approximating the ideal response (16)

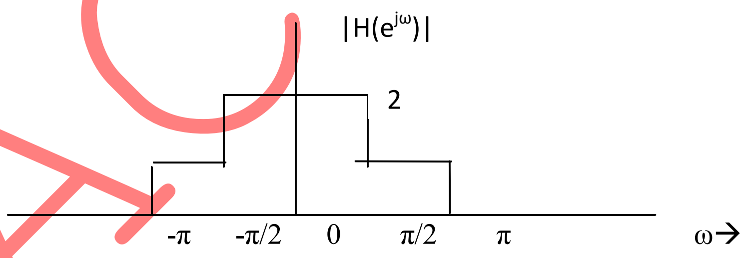
$$H(\omega) = \begin{cases} 1 & \text{for } |\omega| \le \pi/6 \\ 0 & \text{for } \pi/6 \le |\omega| \le \pi \end{cases}$$

9.i) **List the various steps in designing a FIR filter (APR/MAY 12)** (8)

ii) Derive the frequency response of linear phase FIR filter when impulse response is symmetrical and N is odd. (8)

10.i) Write the magnitude and phase function of FIR filter when impulse response is symmetric and N is even. (8)

ii) Design an FIR filter using rectangular window. (first 10 coefficients only). The magnitude specification is given below. (8)



11. i) Determine the coefficient of a linear phase FIR filter of length M=15 which has a symmetric unit sample response and a frequency response that satisfies the conditions (16)

$$H_r(2\pi k/15) = \begin{cases} 1, & k = 0,1,2,3 \\ 0.4 & k = 4 \\ 0, & k = 5,6,7 \end{cases}$$

12. i) Prove that an FIR filter has linear phase if the unit sample response satisfies the condition h(n) = ± h(M-1-n), n =0,1,….. M-1. Also discuss symmetric and anti symmetric cases of FIR filter. (8)

ii) Explain the need for the use of window sequence in the design of FIR filter. Describe the window sequences generally used and compare the properties. (8)

13. i) Design a low pass FIR with 11 coefficients for the following specifications.
Pass band frequency edge is 0.25 kHz and sampling frequency is 1 kHz. (Use rectangular window). Realize the above filter by direct form structures. (16)

14. i) Using a Kaiser window, design a low pass FIR filter for the following specifications.
Fs = 20 kHz, F stop = 5 kHz, A stop = 80 dB, F pass = 4 kHz, A pass = 0.1 dB. (10)
ii) Explain the characteristics of any four window functions used in FIR filter design. (6)

15. Obtain direct form and cascade form realization for the transfer function of an FIR system given by
$H(z) = (1- \frac{1}{4} z^{-1} + 3/8\, z^{-2})\ (1 - 1/8z^{-1} - \frac{1}{2} z^{-2})$

16. i) Explain the principle and procedure for designing FIR filter using rectangular window. (8)
ii) Obtain the cascade form realization of FIR system (8)

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

17. **Design a filter with**

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

0, $\pi/4 \le \omega \le \pi$ using a Hamming window with N=7. (16)

18. Design a FIR filter whose frequency response (16)

$H(e^{j\omega}) = 1,\ \pi/4 \le \omega \le 3\pi/4$

0, $|\omega| \le 3\pi/4$. Using Blackman Window N=7. (16)

UNIT IV

FINITE WORD LENGTH EFFECTS

Part -A

1. **What is limit cycle oscillation? How is it overcome?**
2. Give the IEEE 754 standard for the representation of floating point numbers.
3. What is Zero input limit cycle oscillation?
4. What is the steady state noise power at the output of an LTI system due to the quantization at the input to b bits?
5. **Express the fraction (-7/32) in signed magnitude and two's complement notation using 6 bits. (APR/MAY 13)**
6. What do you meant by limit cycle oscillation in digital filters?
7. **What are the effects of finite word length in digital filters? (APR/MAY 11)**
8. List the errors which arise due to quantization process.
9. Discuss the truncation error in quantization process. **(APR/MAY 13)**
10. Write expression for variance of round-off quantization noise.
11. **What is sampling?**
12. **Define limit cycle Oscillations, and list out the types. (APR/MAY 10)**
13. When zero limit cycle oscillation and Over flow limit cycle oscillation has occur?
14. Why? Scaling is important in Finite word length effect.
15. What are the differences between Fixed and Binary floating point number representation?
16. **What is the error range for Truncation and round-off process? (APR/MAY 13)**

5. Express the fraction (-9/32) in signed magnitude and two's complement notations using 6 bits.

6. What are the various factors which degrade the performance of digital filter implementation when finite word length is used?

7. Define asymptotically unbiased estimate and consistent estimate.

Part-B

1. Explain the fixed point and floating point representation of numbers. (8)
2. **Write notes on quantization noise. Derive the formula for noise power. (APR/MAY 13)** (8)
3. The output of an A/D is fed through a digital system whose system function is H(z)=(1-β)z/(z- β), 0< β<1.Find the output noise power of the digital system. (8)
4. The output of an A/D is fed through a digital system whose system function is H(z)=0.6z/z-0.6. Find the output noise power of the digital system=8bits. (8)
5. Discuss in detail about quantization effect in ADC of signals. Derive the expression for Pe(n) and SNR. (16)
6. a) **Write short notes on limit cycle oscillation (APR/MAY 13)** (8)
   b) Explain in detail about signal scaling (8)
7. A digital system is characterized by the difference equation Y(n)=0.95y(n-1)+x(n). Determine the dead band of the system when x(n)=0 and y(-1)=13. (16)
8. **Two first order filters are connected in cascaded whose system functions of individual sections are $H_1(z)=1/(1-0.8z^{-1})$ and $H_2(z)=1/(1-0.9z^{-1})$ .Determine the overall output noise power.** (16)

UNIT V
MULTIRATE SIGNAL PROCESSING

Part -A

1. **What is Multi rate digital signal processing? (APR/MAY 10)**
2. Write the different applications of Multi rate DSP?
3. What are the different areas in which Multi rate DSP is used?
4. Give the advantages of Multi rate DSP.
5. **Define Decimation. (APR/MAY 12)**
6. **Define Interpolation. (APR/MAY 13)**
7. What is the need for sampling rate conversion?
8. Obtain the decimated signal y(n) by a factor three from the input signal y(n) = x(Mn) where M=3.
9. Obtain the two fold expanded signal y(n) of the input signal x(n) x(n) = { n ,n>0
   0, otherwise

   y(n) = { x(n/M) ,n=multiples of M
   0 otherwise where M=2
10. What is the need for Multi rate signal processing?
11. **Give some examples of Multi rate Digital system**.
12. Write the input output relationship for a decimation processing a factor of 5.
13. **With an example explain the sampling process. (APR/MAY 13)**
14. What is meant by aliasing? How aliasing can be avoided?
15. The signal x(n)is defined by x(n) ={ an , n>0
    0 ,otherwise

    obtain the decimated signal with a factor of 3.
16. For the signal x(n) given in the above question(Q.no15),obtain the interpolated signal with a factor of 3.
17. **Draw the block diagram of Multi rate stage decimator and interpolator. (MAY/ JUN 11)**
18. Draw the block diagram of a general poly phase frame work for decimator and interpolator.
19. Draw the signal flow graph of a system represented by the input, output relationship y(n) = 3x(n) +5x(n-1) – 4x(n-2) – 6y(n-2).
20. What is the need for multistage filter implementation?
21. What are the drawbacks in Multistage filter implementation.
22. Obtain the poly phase structure of the filter with the transfer function H(Z) = 1-3 Z-1 / 1+4Z-1.
23. What are the advantages of poly phase decomposition?
24. What is QMF filter?

25. What are the errors in QMF filter bank?
26. What are digital filter banks? Give some application where these filter banks are used?
27. What is meant by sub coding? **(MAY/ JUN 11)**
28. List the different methods of speech coding techniques.
29. What is meant by signal compression?
30. What is the function of channel Vo- coder?
31. List the different types of channel Vo- coder? **(MAY/ JUN 10)**

PART-B

1. Explain sampling rate reduction by an integer factor 'D'. Derive input and output frequency spectra relation. (8)
2. (a) Explain the need for multistage implementation of sampling rate conversion. (8)
(b) Draw the block diagram of sub-band coding of speech signal and explain.
3. Explain in detail about Multi rate Digital signal processing. **(APR/MAY 13)** (8)
4. Explain with the block diagram, the general poly phase framework for interpolator and decimator. (8)
5. With neat block diagram explain about sub coding.
6. **Explain QMF filter bank.(APR/MAY 13,12)** **(8)**
7. **Explain about different applications of Multi rate digital signal processing.**
8. **Explain about sampling rate conversion by a factor I/D. (APR/MAY 13)** (8)
9. Explain the poly phase decomposition for (16)
(i) FIR filter structure
(ii) IIR filter structure.

UNIT-II(two mark questions with answer)

1. **Compare Butterworth and chebyshev filters?**

Ans.:

Sl.no	Butterworth	Chebyshev
1.	This filters are all pole design.	This filters are all pole design.
2.	At Ω_c, the magnitude of normalized butterworth filter is $1/\sqrt{2}$.	At Ω_c, the magnitude of normalized butterworth filter is $1/\sqrt{(1+\varepsilon^2)}$.
3.	It has maximally flat passband and stopband.	The magnitude response is equiribble in passband and monotonic in the stopband.

2. **What is prewarping? Why it is needed?**

✓ Because of the non-linear mapping; the amplitude response of the digital IIR filter is expanded at lower frequencies and compressed at higher frequencies in comparison to the analog filter .this is called as frequency wrapping .

To compensate this effect, prewarping or prescaling procedure is used.this procedure is as follows,

✓ The ' Ω' scale is changed to prewarpped scale denoted by Ω* and
Ω*=2/Ts[tan(wTs/2)].

✓ Then analog filter transfer function H(s) is obtained by using the value of Ω*.

✓ By applying the bilinear transformation, the desired digital frequency response H(z) is obtained.

3.What are the limitations of impulse invariant technique?

✓ The 'Ω' is analog frequency and it is changed from π/Ts to –π/Ts, while the digital frequency 'w' varies from –π to π. That means from π/Ts to –π/Ts 'w' maps from –π to π. Let k be any integer.
Then , we can write the general range of Ω as (k-1)π/Ts to (k+1)π/Ts; but for this range also 'w' maps from –π to π.
Thus mapping from the analog frequency 'Ω' to digital frequency 'w' is many to one. This mapping is not one to one mapping.

✓ Analog filters are not band limited so there will be aliasing due to the sampling process. Because of this aliasing, the frequency response of resulting digital filter will not be identical to the original frequency of the analog filter.

✓ The change in the value of sampling time (Ts) has no effect on the amount of aliasing.

4.Give the transform relation for converting the low pass to band pass in analog domain?

✓ The transformation for converting a normalized low pass filter to the band pass filter with the cut off frequency Ωl to Ωu
S->(s2 +ΩlΩu)/(s(Ωu-Ωl))

5. **Find the digital filter equivalent for H(s)=1/(s+8).**

✓ Apply bilinear transformation,

$H(z)=H(s)|s=2(1-z^{-1})/T(1+z^{-1})$

T=1sec

$H(z)=1/\{2(1-z^{-1})/1(1+z^{-1})\}+8$

$=1/\{(2-2z^{-1}+8+8z^{-1})/(1+z^{-1})$

$=(1+z^{-1})/(6+6z^{-1})$

$=(1+z^{-1})/6(1+z^{-1})$

H(z)=1/6

6. **Determine the order of the analog butterworth filter that has a -2db pass band attenuation at a frequency of 20rad/sec and atleast -10db stopband attenuation at 30rad/sec.**
   - ✓ Pass band attenuation = -2db
   - ✓ Pass band frequency=20rad/sec
   - ✓ Stop band attenuation=-10db
   - ✓ Stop band frequency=30rad/sec
   - ✓ N=log A/log(1/k)
   - ✓ N=4

7. **Find the digital transfer function h(z) by using impulse invariant method for the analog transfer function H(S)=1/S+2 .assume that T=0.1 sec.**

Ans.:

By impulse invariant method,

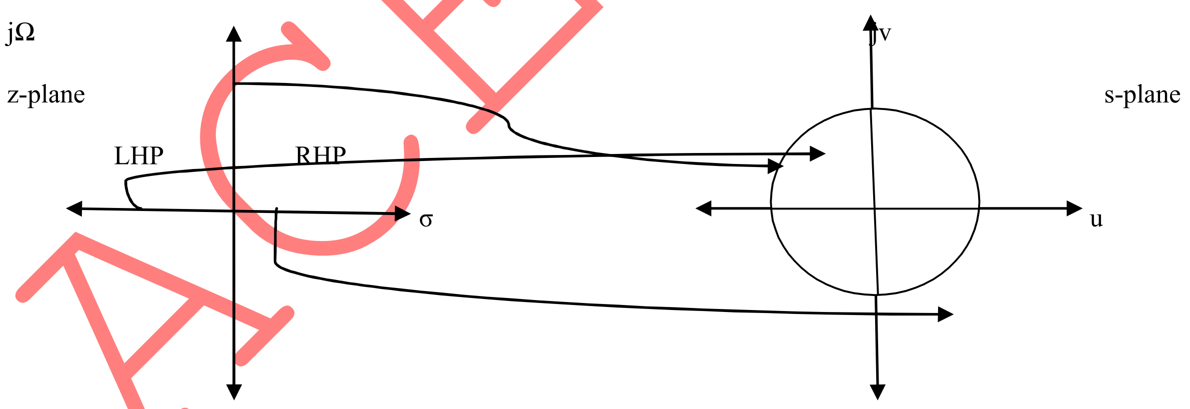
$H(z)= \sum_{k=1}^{N} \frac{Ck}{1-ePkTZ-1}$

$H(z)=1/1-0.3678z^{-1}$

8. **State the relationship between the analog and digital frequencies when converting an analog filter to digital filter using bilinear transformation.**

$$H(z)= H(s)|_{s\text{-->}\frac{2}{T}\left[\frac{1-z^{-1}}{1+z^{-1}}\right]}$$

9. **Sketch the mapping of S-plane and Z-plane in bilinear transformation.**



10. **What is warping effect?**

Refer q no:2

11. **State the properties of butterworth filter.**
   - The butterworth filters are all pole designs
   - At Ω_c (cut off frequency), the magnitude of normalized butterworth filter is 1/√2.
   - The filter order N,completely species the filter and if N increases , the magnitude response approaches the ideal response.

12. **Give the expression for location of poles of normalized butterworth filter.**

$S_k = e^{j\varphi_k}$, **k=1,2 3 ……N, where** $\varphi_k = \frac{\pi}{2} + \frac{(2k-1)\pi}{2N}$

13. Distinguish between FIR and IIR system.

Sl.No.	FIR	IIR
1.	It can be designed with exact linear phase	IIR not having linear phase.
2.	FIR can be realized as recursive and non-recursive filters.	IIR recursive filters.
3.	Magnitude response can be controlled easily.	It is restricted
4.	Feedback is not used. So, round off is not effective.	Round off affect the response.

14. List the various method of calculating IIR filter.

- ✓ Impulse invariant method
- ✓ Bilinear transformation
- ✓ Approximation derivative method

15. What are the salient features of low pass butterworth filter?

- ✓ The magnitude response is nearly constant (equal to 1) at lower frequencies.
- ✓ That means pass band is maximally flat.
- ✓ There are no ripples in pass bend and stop band.
- ✓ The maximum gain occurs at $\Omega=0$ and its magnitude of $H(0)=1$.The magnitude response is monotonically decreasing.

16. What are the different structures used for realizationan IIR system.

- ✓ Direct form-I
- ✓ Direct form-II
- ✓ Cascade form
- ✓ Parallel form
- ✓ Transposed form
- ✓ Lattice-ladder form

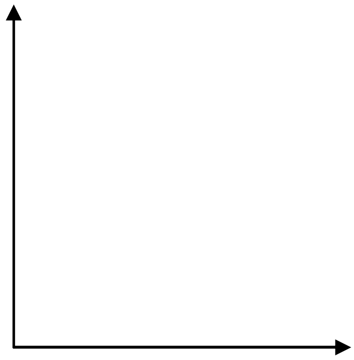
17. State two advantage of bilinear transformation.

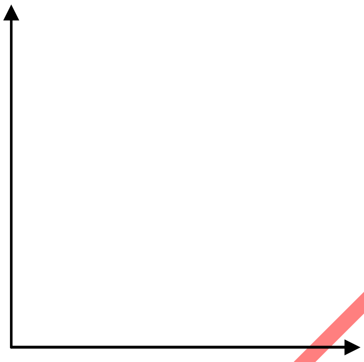
- ✓ The bilinear transformation is one to one mapping
- ✓ Bilinear transformation can be used to design digital filters with prescribed magnitude response with piecewise constant values.

18. Mention the properties of chebyshev filter.

- ✓ The magnitude response is equiripple in the pass band and monotonic in stop band
- ✓ These all are pole designs.
- ✓ The normalized magnitude function has a value of $1/\sqrt{1+\epsilon^2}$ at Ω_c (cut off frequency).

19. Draw the response curve of butterworth and chebyshev.





20. What are the advantages of direct form-II realization when compared to direct form-I.

- ✓ It requires minimum storage space.
- ✓ Good round off noise property.

21. What is the relationship between the Z-transform and laplace transform.

By Bilinear Transformation:$H(z)=H(s)|_{s\rightarrow\frac{2}{T}\left[\frac{1-Z^{-1}}{1+Z^{-1}}\right]}$

By Impulse invariant:$H(z)=\sum_{k=1}^{N}\frac{Ck}{1-e^{pkT}Z^{-1}}$

22. What is frequency warping? Refer q no. 2

23. what are the characteristics of IIR?

- ✓ The physically realizable iir filters does not have linear phase.
- ✓ The IIR filter specifications includes desired characteristics for magnitude response only.

24. How a digital filter is designed using impulse invariance method.

- ✓ Step 1: Analog frequency transfer function H(s) will be given. If it is not given then, obtain expression for H(s) from the given specification
- ✓ If required expand H(s) by using partial fraction expansion (PFE).
- ✓ Obtain 'Z' transform of each (PFE) term using impulse invariance transformation equation

  $H(Z)=\sum_{k=0}^{N} Ck/(1-e^{PkT}Z^{-k})$
- ✓ The required digital IIR filter is designed.

25. Why IIR filter do not have linear phase?

- ✓ Linear phase filter must have a system function that satisfies the condition $H(Z)=\pm Z^{+N}H(Z^{-1})$

  Where Z^{-N} represent a delay of N units of time.
- ✓ But if this is the case, for every pole inside the unit circle there must be a pole outside the unit circle.
- ✓ Hence, the filter , would be unstable. Consequently aausal and stable IIR filter cannot have linear phase.

26. What are the requirements for an analog filter to be stable and causal?

- ✓ The analog filter transfer function $H_a(s)$ should be a rational functions of s and coefficient of s should be real.
- ✓ The poles should lie on the left half of s-plane.
- ✓ The numbers of zeros should be less than or equal to number of poles.

27. What is importance of poles in filters design?

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

28. What are the properties that are maintained same in the transfer of analog filter into a digital filter?

- ✓ The jω of the s-plane should be mapped into the unit circle of the z-plane. Then there will be direct relationship between the 2 frequencies.
- ✓ The left half of the s-plane should be mapped into the unit circle of the z-plane. Then the stable analog filter is converted into stable digital filter.

29. Define ripple 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 δp and that stopband is denoted as δ.

Unit-III

Part-A

1.Write procedure for designing FIR filter using windows?

1. For the desired frequency response $H_d(e^{jw})$, find the impulse response $h_d(n)$ using the equation

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

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

$$h(n)=h_d(n)w(n) \text{ for } |n|\leq\frac{N-1}{2}$$

$$0 \quad \text{Otherwise}$$

3. Find the transfer function of the realizable filter.

$$H'(z)=Z^{-(N-1)/2}\ H(z)$$

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

2.What do you meant by linear phase FIR filter?

The transfer function of a FIR causal filter is given by $H(z)=\sum_{n=0}^{(N-1)} h(n)z^{-n})$, where h(n) is the impulse response of the filter.

The fourier transform of h(n) is $H(e^{j\omega})$,which is periodic in frequency with period 2π,$H(e^{j\omega})=\pm|H(e^{j\omega})|e^{j\Theta(\omega)}$,where $|H(e^{j\omega})|$ is magnitude response and Θ(ω) is phase response.

3.What is the condition for linear phase characteristics of an FIR filter?

For the FIR filter with linear phase we can define Θ(ω)=-α(ω),-π≤ω≤π,where α is a constant phase delay in the samples,

$$\sum_{n=0}^{N-1} h(n)\sin\omega(n-\alpha)=0.$$

4.Why is window function used in FIR design?

One possible way of finding an FIR filter that approximates $H(e^{j\omega})$ would be to truncate the infinite fourier series at n=± $\left(\frac{N-1}{2}\right)$. Abrupt truncation of the series will lead to oscillations in the passbandandstopband .these oscillations can be reduced through the use of lass abrupt truncation of the fourier series. This can be achieved by multiplying the infinteimpulse response with a finite weighing sequence w(n) , called a window.

5.What are the desirable characteristics of the frequency response of the window function?

The desirable characteristics are,

i)the central lobe of the frequency response of the window should contain most of the energy and should be narrow.

ii)the highest side lobe level of the frequency response should be small.

iii)the side lobes of the frequency response should decrease in energy rapidly as ω tends to π.

6.Define Hamming and Blackman window functions?

The frequency response of blackman window the peak side lobe level is down about 57db from the main lobe peak,an improvement of 16db relative to the hamming window.
The side lobe attenuation of a low pass filter using blackman window.

7.What are the merits and demerits of FIR filters?

- It's always stable
- With exactly linear phase the filter can be designed easily
- The filters structures realized in both recursive and nonrecrussive
- Various kind of filter designing are available.

8. Distinguish between recursive and nonrecursive realization.

The FIR filters are of non-recursive type, whereby the present output sample is depend on the present input sample and previous input samples, whereas the IIR filters are of recursive type , whereby the present output sample depends on the present input ,past input samples and output samples.

9. 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 filter 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 nonrecursive type whereby the present output sample is depend on the present input sample and previous input samples.

10. what are the different types of filters based on frequency response?

The filters can be classified based on frequency response.They are(i) lowpass filter (ii)Highpass filter(iii) bandpass filter (iv)bandreject filter.

11. what are the desirable or advantages and undesirable or disadvantages features of FIR filters?

(i)FIR filters have exact linear phase.

(ii)FIR filters are always stable.

(iii)FIR filters can be realized in both recursive and nonrecursive structure.

(iv)Filters with any arbitrary magnitude response can be tackled using FIR sequence.

12.what are the design techniques of designing FIR filters?

There are three well-known methods for designing FIR filters with linear phase.These are (1)windows method(2)frequency sampling method(3)optimal or minimax design.

13.what is the condition for the impulse response of FIR filter to satisfy for constant group and phase delay and for only constant group delay?

For linear phase FIR filter to have both constant group delay and constant phase delay

$\Theta(\omega)= -\alpha\omega \quad -\pi\leq\omega\leq\pi$

For satisfying above condition $h(n)=h(N-1-n)$

i.e..The impulse response must be symmetrical about $n=(N-1)/2$

if only constant group delay is desired then

$\Theta(\omega)=\beta-\alpha\omega$

i.e...The impulse response is antisymmetrical about $n=(N-1)/2$

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

The conditions are (i)symmetric condition $h(n)=h(N-1-n)$

(ii)anti-symmetric condition $h(n)=-h(N-1-n)$

15.state the condition for a digital filter to be causal and stable.

A digital filter is causal if its impulse h(n)=0,for n<0.

A digital filter is stable if its impulse response is absolutely summable,i.e.,$\sum_{n=-\infty}^{\infty}|h(n)| < \infty$

16. What are properties of FIR filter?

(i)FIR filter is always stable.

(ii) A realizable filter can always be obtained.

(iii)FIR filter has a linear phase response.

17. How the zeros in FIR filter is located? Explain briefly ?

18.What is Gibbs phenomenon or oscillations?

One possible way of finding an FIR filter that approximates $H(e^{j\omega})$ would be to truncate the infinite Fourier series at n=±[(N-1)/2].Direct truncation of the series will lead to fixed percentage overshoots and undershoots before and after an approximated discontinuity in the frequency response.

19.what are the desirable characteristics of the window?

The desirable characteristics of the window are

(i)The central lobe of the frequency response of the window should contain most of the energy and should be narrow.

(ii)The highest side lobe level of the frequency response should be small.

(iii)The side lobes of the frequency response should decrease in energy rapidly as ω tends to π.

20. What is the principle of designing FIR filter using windows?

One possible way of obtaining FIR filter is to truncate the infinite Fourier series at n=±[(N-1)/2 where N is the length of the desired sequence.But abrupt truncation of the Fourier series results in oscillation in the passband and stopband.These oscillations are due to slow convergence of the Fourier series.To reduce these oscillations the Fourier coefficients of the filter are modified by multiplying the infinite impulse response with a finite weighing sequence ω(n) called a window where

$\omega(n)=\omega(-n)\neq 0$,for $|n|\leq(N-1)/2$

$=0$,for $|n|>(N-1)/2$

After multiplying window sequence ω(n) with $h_d(n)$ we get a finite duration sequence h(n) that satisfies the desired magnitude response.

$h(n)=h_d(n)\omega(n)$,for all $|n|<(N-1)/2$

$=0$,for $|n|>(N-1)/2$

21.what is the necessary and sufficient condition for linear phase characteristics in FIR filter?

The necessary and sufficient condition for linear phase characteristics in FIR filter is,the impulse response h(n) of the system should have the symmetry property,i.e.,h(n)=h(N-1-n)

Where N is the duration of the sequence.

22.what is the principle of designing FIR filter using frequency sampling method?

In the 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 DFT of this set of samples.

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

Frequency sampling method is attractive for narrow band frequency selective filters where only a few of samples of the frequency response are non zero.

24.CompareHanning and Hamming window technique.

Sl.no	Hanning	Hamming
1.	In window spectrum sidelobe magnitude decreases with increasing ω	In window spectrum sidelobe magnitude remain s constant
2.	The maximum sidelobe magnitude in	The maximum sidelobe magnitude in

	window spectrum is -31db	window spectrum is -41db
3.	For design this window minimum stopband attenuation is 44db	For design this window minimum stopband attenuation is 51db
4.	The width of the mainlobe window spectrum is 8π/N	The width of the mainlobe window spectrum is 8π/N

25.What are the possible types of impulse response for linear phase FIR filter?

1) N=odd, symmetric.
2) N=odd, Antisymmetric.
3) N=even, symmetric.
4) N=even, Antisymmetric.