KINGS

COLLEGE OF ENGINEERING

DEPARTMENT OF ELECTRICAL AND ELECTRONICS ENGINEERING

QUESTION BANK

Subject Code/Name: IT 1252/Digital signal Processing Year/Sem:IV/ VII

UNIT – I SIGNALS & SYSTEMS

PART A (2marks)

1. Determine the energy of the discrete time sequence (2)

$$x(n) = (½)^n,\ n\geq0$$
$$=3^n,\quad n<0$$

2. Define multi channel and multi dimensional signals. (2)
3. Define symmetric and anti symmetric signals. (2)
4. Differentiate recursive and non recursive difference equations. (2)
5. What is meant by impulse response? (2)
6. What is meant by LTI system? (2)
7. What are the basic steps involved in convolution? (2)
8. Define the Auto correlation and Cross correlation? (2)
9. What is the causality condition for an LTI system? (2)
10. What are the different methods of evaluating inverse z transform? (2)
11. What is meant by ROC? (2)
12. What are the properties of ROC? (2)
13. What is zero padding? What are it uses? (2)
14. What is an anti imaging and anti aliasing filter? (2)
15. State the Sampling Theorem. (2)
16. Determine the signals are periodic and find the fundamental period (2)
    i) $\sin\sqrt{2}\,\pi t$
    ii) $\sin 20\pi t+ \sin 5\pi t$
17. Give the mathematical and graphical representations of a unit sample, unit step

sequence. (2)

18. Sketch the discrete time signal $x(n) = 4\,\delta(n+4) + \delta(n) + 2\,\delta(n-1) + \delta(n-2) - 5\,\delta(n-3)$ (2)
19. Find the periodicity of $x(n) = \cos t(2\pi n / 7)$ (2)
20. What is inverse system? (2)
21. Write the relationship between system function and the frequency response. (2)
22. Define commutative and associative law of convolutions. (2)
23. What is meant by Nyquist rate and Nyquist interval? (2)
24. What is an aliasing? How to overcome this effect? (2)
25. What are the disadvantages of DSP? (2)
26. Compare linear and circular convolution.(2)
27. What is meant by section convolution? (2)
28. Compare over lap add and save method. (2)
29. Define system function. (2)
30. State Parseval's relation in z - transform. (2)

PART B

CLASSIFICATION OF SYSTEMS:

1. Determine whether the following system are linear, time-invariant (16)

   (a) $y(n) = Ax(n) + B$. (4)

   i(a) $y(n) = x(2n)$. (4)

   ii(a) $y(n) = n\,x^2(n)$. (4)

   iv) $y(n) = a^{x(n)}$ (4)

2. Check for following systems are linear, causal, time in variant, stable, static (16)

   i) $y(n) = x(2n)$. (4)

   ii) $y(n) = \cos(x(n))$. (4)

   iii) $y(n) = x(n)\cos(x(n)$ (4)

   iv) $y(n) = x(-n+2)$ (4)

3. (a)For each impulse response determine the system is (a) stable i(a) causal

   i) $h(n) = \sin(\pi n / 2)$. (4)

ii) $h(n) = \delta(n) + \sin \pi n$ (4)

(b)Find the periodicity of the signal $x(n) = \sin(2\pi n / 3) + \cos(\pi n / 2)$ (8)

4. (a)Find the periodicity of the signal

i) $x(n) = \cos(\pi /4) \cos(\pi n /4)$. (4)

ii) $x(n) = \cos(\pi n^2 / 8)$ (4)

(b) State and proof of sampling theorem. (8)

5. Explain in detail about A to D conversion with suitable block diagram and to reconstruct the signal. (16)

6. What are the advantages of DSP over analog signal processing? (16)

CONVOLUTION:

8. Find the output of an LTI system if the input is $x(n) = (n+2)$ for $0 \le n \le 3$

and $h(n) = a^n u(n)$ for all n (16)

9. Find the convolution sum of $x(n) = 1$ n = -2,0,1

$= 2$ n= -1

$= 0$ elsewhere

and $h(n) = \delta(n) - \delta(n-1) + \delta(n-2) - \delta(n-3)$. (16)

10. (a)Find the convolution of the following sequence $x(n) = u(n)$; $h(n) = u(n-3)$. (8)

(b)Find the convolution of the following sequence $x(n) = (1,2,-1,1)$, $h(n) = (1, 0 ,1,1)$. (8)

12.Find the output sequence y(n) if $h(n) = (1,1,1)$ and $x(n) = (1,2,3,1)$ using a circular Convolution. (16)

13. Find the convolution y(n) of the signals (16)

$x(n) = \{ \alpha^n, -3 \le n \le 5$ and $h(n) = \{ 1, 0 \le n \le 4$

0, elsewhere } 0, elsewhere }

Z TRANSFORM:

14. State and proof the properties of Z transform.(16)

15.(a) Find the Z transform of

i) $x(n) = [(1/2)^n - (1/4)^n] u(n)$ (2)

ii) $x(n) = n(-1)^n u(n)$ (2)

iii) $x(n) (-1)^n \cos(\pi n/3) u(n)$ (2)

iv) $x(n) = (½)^{n-5} u(n-2) + 8(n-5)$ (2)

(b) Find the Z transform of the following sequence and ROC and sketch the pole zero diagram.

i) $x(n) = a^n u(n) + b^n u(n) + c^n u(-n-1)$, $|a| < |b| < |c|$ (4)

ii) $x(n) = n^2 a^n u(n)$ (4)

17. Find the convolution of using z transform (16)

$x_1(n) = \{ (1/3)^{n,}$ $n>=0$

$(1/2)^{)-n}$ $n<0$ }

$x_2(n) = (1/2)^n$

INVERSE Z TRANSFORM:

18. Find the inverse z transform (8x2=16)

$X(z) = \log(1-2z)$ $z < |1/2|$

$X(z) = \log(1+az^{-1})$ $|z| > |a|$

$X(z) = 1/1+az^{-1}$ where a is a constant

$X(z)=z^2/(z-1)(z-2)$

$X(z) = 1/(1-z^{-1})(1-z^{-1})^2$

$X(z)= Z+0.2/(Z+0.5)(Z-1)$ $Z>1$ using long division method.

$X(z) = 1- 11/4\, z^{-1} / 1-1/9\, z^{-2}$ using residue method.

$X(z) = 1- 11/4\, z^{-1} / 1-1/9\, z^{-2}$ using convolution method.

19. A causal LTI system has impulse response h(n) for which Z transform is given by H(z)

$1+ z^{-1} / (1-1/2\, z^{-1})(1+1/4\, z^{-1})$ (16)

i) What is the ROC of H (z)? Is the system stable?

ii) Find THE Z transform X(z) of an input x(n) that will produce the output $y(n) = -1/3\,(-1/4)^n u(n) - 4/3\,(2)^n u(-n-1)$

iii) Find the impulse response h(n) of the system.

ANALYSIS OF LTI SYSTEM:

20. The impulse response of LTI system is h(n)=(1,2,1,-1).Find the response of the system to the input x(n)=(2,1,0,2) (16)

21. Determine the magnitude and phase response of the given equation

$y(n) = x(n)+x(n-2)$ (16)

22. Determine the response of the causal system $y(n) - y(n-1) = x(n) + x(n-1)$ to inputs $x(n)=u(n)$ and $x(n) = 2^{-n} u(n)$.Test its stability (16)

23. Determine the frequency response for the system given by

$$y(n)-y3/4y(n-1)+1/8\ y(n-2) = x(n)- x(n-1) \qquad (16)$$

24. Determine the pole and zero plot for the system described difference equations

$$y(n)=x(n)+2x(n-1)-4x(n-2)+x(n-3) \qquad (16)$$

25. A system has unit sample response $h(n) =-1/4\ \delta(n+1)+1/2\ \delta(n)-1/4\ \delta(n-1)$. Is the system BIBO stable? Is the filter is Causal? Find the frequency response? (16)

26. Find the output of the system whose input- output is related by the difference equation

$y(n) -5/6\ y(n-1) +1/6\ y(n-2) = x(n) -1/2\ x(n-1)$ for the step input. (16)

27. Find the output of the system whose input- output is related by the difference equation

$y(n) -5/6\ y(n-1) +1/6\ y(n-2) = x(n) -1/2\ x(n-1)$ for the $x(n) =4^{n} u(n)$. (16)

UNIT – II

FAST FOURIER TRANSFORM

PART A (2marks)

1. How many multiplication and additions are required to compute N point DFT using radix 2 FFT? (2)
2. Define DTFT pair. (2)
3. What are Twiddle factors of the DFT? (2)
4. State Periodicity Property of DFT. (2)
5. What is the difference between DFT and DTFT? (2)
6. Why need of FFT? (2)
7. Find the IDFT of Y (k) = (1, 0, 1, 0) (2)
8. Compute the Fourier transform of the signal x(n) = u(n) – u(n-1). (2)
9. Compare DIT and DIF? (2)
10. What is meant by in place in DIT and DIF algorithm? (2)
11. Is the DFT of a finite length sequence is periodic? If so, state the reason. (2)
12. Draw the butterfly operation in DIT and DIF algorithm? (2)
13. What is meant by radix 2 FFT? (2)
14. State the properties of $W_N^{\ k}$? (2)
15. What is bit reversal in FFT? (2)
16. Determine the no of bits required in computing the DFT of a 1024 point sequence

with SNR of 30dB. (2)

17. What is the use of Fourier transform? (2)

18. What are the advantages FFT over DFT? (2)

19. What is meant by section convolution? (2)

20. Differentiate overlap adds and save method? (2)

21.Distinguish between Fourier series and Fourier transform. (2)

22.What is the relation between fourier transform and z transform. (2)

23. Distinguish between DFT and DTFT. (2)

PART B

1.(a) Determine the Fourier transform of x (n) =$a^{|n|;}$ -1<a<1 (8)

(b) Determine the Inverse Fourier transform H (w) = $(1\text{-}ae^{-jw})^{-1}$ (8)

2. State and proof the properties of Fourier transform (16)

FFT:

3. Determine the Discrete Fourier transform x (n) = (1, 1, 1, 1) (16)

4. Derive and draw the 8 point FFT-DIT butterfly structure. (16)

5. Derive and draw the 8 point FFT-DIF butterfly structure. (16)

6.Compute the DFT for the sequence.(0.5,0.5,0.5,0.5,0,0,0,0) (16)

7.Compute the DFT for the sequence.(1,1,1,1,1,1,0,0) (16)

8.Find the DFT of a sequence x(n)=(1,1,0,0) and find IDFT of Y(k) =(1,0,1,0) (16)

9. If x (n) = sin (nΠ/2), n=0, 1, 2, 3

h (n) = 2^n , n=0,1,2,3.Find IDFT and sketch it. (16)

10.(a) Find 4 point DFT using DIF of x(n) =(0,1,2,3) (8)

(b).Proof x(n)*h(n) =X(z) H(z) (8)

11.Discuss the properties of DFT. (16)

12.Discuss the use of FFT algorithm in linear filtering. (16)

13.Explain the application of DFT in linear filtering and spectral analysis? (16)

UNIT - III IIR FILTER DESIGN

PART A (2 marks)

1. Define canonic and non canonic form realizations**.** (2)

2. Draw the direct form realizations of FIR systems? (2)

3. Mention advantages of direct form II and cascade structure? (2)

4. Define Bilinear Transformation. (2)

5. What is prewar ping? Why is it needed? (2)

6. Write the expression for location of poles of normalized Butterworth filter. (2)

7. Distinguish between FIR and IIR Filters. (2)

8. What is linear phase filter? (2)

9. What are the design techniques available for IIR filter? (2)

10. What is the main drawback of impulse invariant mapping? (2)

11. Compare impulse invariant and bilinear transformation. (2)

12. Why IIR filters do not have linear phase? (2)

13. Mention the properties of Butterworth filter? (2)

14. Mention the properties of Chebyshev filter? (2)

15. Why impulse invariant method is not preferred in the design of high pass IIR filter? (2)

16. Give the transform relation for converting LPF to BPF in digital domain. (2)

PART - B

Structures of IIR systems:

1. Obtain the cascade and parallel form realizations for the following systems (16)

$$Y(n) = -0.1(n-1) + 0.2\, y(n-2) + 3x(n) + 3.6\, x(n-1) + 0.6\, x(n-2)$$

2. Obtain the Direct form I and II

$$y(n) = -0.1(n-1) + 0.72\, y(n-2) + 0.7x(n) - 0.252\, x(n-2) \qquad (16)$$

3. Obtain the (a) Direct forms i(a) cascade ii(a) 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)

4. Find the direct form I and II

$$H(z) = 8z^{-2}+5z^{-1}+1 / 7z^{-3}+8z^{-2}+1 \qquad (16)$$

5.Find the direct form –I, cascade and parallel form for (16)

$$H(Z) = z^{-1} - 1 / 1 - 0.5\, z^{-1} + 0.06\, z^{-2}$$

IIR FILTER DESIGN:

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

7. (a)Discuss the limitations of designing an IIR filter using impulse invariant method. (8)

(b)Derive bilinear transformation for an analog filter with system function $H(s) = b/ s + a$ (8)

8 (a)For the analog transfer function $H(s) = 2 / (s+1)(s+3)$ .Determine $H(z)$ using bilinear transformation. With T=0.1 sec (8)

(b)Convert the analog filter $H(s) = 0.5\ (s+4)\ /\ (s+1)(s+2)$ using impulse invariant transformation T=0.31416s (8)

9. The normalized transfer function of an analog filter is given by

$H_a(s^n) = 1/\ s_n^2 + 1.414\ s_n + 1$. Convert analog filter to digital filter with cut off frequency of 0.4 π using bilinear transformation. (16)

10. Design a single pole low pass digital IIR filter with -3db bandwidth of 0.2Π by using bilinear transformation. (16)

11. For the constraints

$$0.8 \le |H(e^{jw})| \le 1,\ 0 \le \omega \le 0.2\pi$$

$|H(e^{jw})| \le 0.2,\ 0.6\pi \le \omega \le \pi$ with T= 1 sec .Determine system function H(z) for a Butterworth filter using Bilinear transformation. (16)

12.Design a digital Butterworth filter satisfying the following specifications

$$0.7 \le |H(e^{jw})| \le 1,\ 0 \le \omega \le 0.2\pi$$

$|H(e^{jw})| \le 0.2,\ 0.6\pi \le \omega \le \pi$ with T= 1 sec .Determine system function H(z) for a Butterworth filter using impulse invariant transformation. (16)

13. Design a digital Chebyshev low pass filter satisfying the following specifications

$$0.707 \le |H(e^{jw})| \le 1,\ 0 \le \omega \le 0.2\pi$$

$|H(e^{jw})| \le 0.1\ \ 0.5 \le \omega \le \pi$ with T= 1 sec using for bilinear transformation. (16)

14.Design a digital Butterworth High pass filter satisfying the following specifications

$$0.9 \le |H(e^{jw})| \le 1,\ 0 \le \omega \le \pi/2$$

$|H(e^{jw})| \le 0.2,\ 3\pi/4 \le \omega \le \pi$ with T= 1 sec. using impulse invariant transformation. (16)

15. Design a realize 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 ω = 0.75 π rad. (16)

UNIT - IV FIR FILTER DESIGN

PART A (2marks)

1. What are Gibbs oscillations?(2)
2. Explain briefly Hamming window(2).
3. If the impulse response of the symmetric linear phase FIR filter of length 5 is h(n) = {2, 3, 0, x, y), then find the values of x and y.(2)
4. What are the desirable properties of windowing technique?(2)
5. Write the equation of Bartlett window.(2)

6.Why IIR filters do not have linear phase?(2)

7.Why FIR filters are always stable?(2)

8.Why rectangular window are not used in FIR filter design using window method?(2)

9.What are the advantages of FIR filter? (2)

10.What are the advantages and disadvantages of window? (2)

11.What is the necessary condition and sufficient condition for the linear phase characteristic of a FIR filter? (2)

12.Compare Hamming and Hanning window? (2)

13.Why triangular window is not a good choice for designing FIR Filter? (2)

14.Why Kaiser window is most used for designing FIR Filter? (2)

15.What is the advantages in linear phase realization of FIR systems? (2)

PART B

1. 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. (16)
2. Explain the need for the use of window sequence in the design of FIR filter. Describe the window sequence generally used and compare the properties. (16)
3. Explain the type 1 design of FIR filter using Frequency sampling technique. (16)
4. A LPF has the desired response given below (16)

$$H(e^{j\omega}) = \begin{cases} e^{-3j\omega}, & 0 \le \omega \le \pi/2 \\ 0. & \pi/2 \le \omega \le \pi \end{cases}$$

.Determine the filter coefficients h(n) for M=7 using frequency sampling technique.

5. Design a HPF of length 7 with cut off frequency of 2 rad/sec using Hamming window. Plot the magnitude and phase response. (16)

6. Explain the principle and procedure for designing FIR filter using rectangular window (16)

7. Design a filter with

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

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

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

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

$0. \quad |\omega| \le 3\pi/4$.Calculate the value of h(n) for N=11 and hence find H(z).

9.Design an ideal differentiator with frequency response $H(e^{j\omega}) = jw \quad -\pi \le \omega \le \pi$ using hamming window for N=8 and find the frequency response. (16)

10.Design an ideal Hilbert transformer having frequency response

$H(e^{j\omega}) = j \quad -\pi \le \omega \le 0$

$-j \quad 0 \le \omega \le \pi$ for N=11 using Blackman window. (16)

FIR structures:

12.(a) Determine the direct form of following system (8)

$H(z) = 1+2z^{-1} - 3z^{-2} + 4z^{-3} - 5z^{-4}$

(b) Obtain the cascade form realizations of FIR systems (8)

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

UNIT - V FINITE WORDLENGTH EFFECTS

PART A (2marks)

1. What are the three quantization errors due to finite world length registers in digital filters?(2)
2. What do you mean by limit cycle oscillations? (2)
3. Explain briefly quantization noise. (2)
4. Represent 15.75 in fixed point and in floating point representations. (2)
5. What is the need for scaling in digital filters? (2)
6. List the well known techniques for linear phase FIR filter? (2)

7.What is quantization step size? (2)

8.State the advantages of floating point over fixed point representations? (2)

9.Why rounding preferred over truncation in realizing digital filter? (2)

10.What is meant by dead band? (2).

11.What is over flow limit cycle? How overflow can be eliminated? (2)

12.Sketch the noise probability density functions for rounding? (2)

13.Sketch the noise probability density functions for truncation?. (2)

14.What is meant by finite word length effect in digital filter? (2)

15.Explain the fraction 7/8 and -7/8 in sign magnitude ,1's, 2's complement. (2)

16.Convert in decimal to binary 20.675 (2)

17. Convert in binary to decimal 1110.01 (2)

18. What is product quantization error? (2)

19. What is input quantization error? (2)

20.What is coefficient quantization error? (2)

PART B

1. Explain in details about quantization in floating point realizations of IIR filter? (16)

2. Describe the effects of quantization in IIR filter. Consider a first order filter with difference equation y(n) = x(n)+0.5 y(n-1) assume that the data register length is three bits plus a sign bit. The input x(n) = 0.875 δ(n).Explain the limit cycle oscillations in the above filter, if quantization is performed by means of rounding and signed magnitude representation is used.(16)

3. Explain briefly

    (1) Effects of coefficient quantization in filter design. (6)

    (2) Effects of product round off error in filter design. (6)

    (3) Speech recognition (4)

4. Explain briefly

    (A)Define limit cycle oscillation. Explain. (8)

    (B) Explain the different representation of fixed and floating point representation. (8)

5.Two first order LPF whose system function are given below connected in cascade. Determine the over all output noise power (16)

$H_1(z) = 1/1-0.9z^{-1}$ and $H_2(z) =1/1-0.8z^{-1}$

6. (a) Describe the quantization error occur in rounding and truncation in twos complement.(8)

(b) Draw a sample and hold circuit and explains its operation? (8)

7. (a)Explain dead band in limit cycles? (8)

(b)Draw the stastical model of fixed point product quantization and explain (8)

8. (a)What is dead band of a filter? Derive the dead band of second order linear filter? (12)

(b)Consider all pole second order IIR filter described by equation $y(n) = -0.5\ y(n-1) - 0.75\ y(n-2) + x(n)$.Assuming 8 bits to represent pole, determine the dead band region governing the limit cycle. (4)

9. Determine the variance of the round off noise at the output of two cascaded of the filter with system function $H(z) = H_1(z) . H_2(z)$ where $H_1(z) = 1 / 1-0.5\ z^{-1}$ $H_2(z) = 1 / 1-0.25z^{-1}$ (16)

10. Explain with suitable examples the truncation and rounding off errors (16)

11 .a) Explain the application of DSP in Speech processing? (8)

b) What is a vocoder? Explain with a block diagram? (8)

12. Determine the dead band of the filter of $y(n) = 0.95\ y(n-1) + x(n)$ (16)