KINGS

COLLEGE OF ENGINEERING

DEPARTMENT OF INFORMATION TECHNOLOGY

QUESTION BANK

SUB.CODE : EC1358 **BRANCH / YEAR / SEM : IT / III /VI**

SUB.NAME : DIGITAL SIGNAL PROCESSING

UNIT- I

SIGNAL AND SYSTEMS

PART A (2 MARKS)

1. What is a continuous and discrete time signal?
2. Give the classification of signals?
3. What are even and odd signals?
4. What are deterministic and random signals?
5. What are energy and power signal?
6. What are the operations performed on a signal?
7. What are the properties of a system?
8. What is memory system and memory less system?
9. What is an invertible system?
10. What are time invariant systems?
11. What is SISO system and MIMO system?
12. What is the output of the system with system function H1 and H2 when connected in cascade and parallel?
13. What do you mean by periodic and non-periodic signals?
14. Determine the convolution sum of two sequences x(n) = {3, 2, 1, 2} and h (n) = {1, 2, 1, 2}
15. Find the convolution of the signals?
16. What are the properties of convolution?
17 .What is alasing effect?
18. State sampling theorem.
19. what is an LTI system?
20..What are the classification of discrete time system?
21.What are the types of systems?

PART –B (16 MARKS)

1. i) Find the convolution of X(n) and h(n)

X (n) = $(1/2)^n$ u(n)
h (n) = $(1/3)^n$ u(n) (08)

ii) Find the z-transform of x (n), x (n) = $(1/2)^{n-5}$ u(n-2.) + 8(n-5). (08)

2. i) Find the inverse Z- transform of X(z),

$x(z)=\frac{1}{1+az-1}$, Where a constant. (08)

(ii) Find the z-transform of autocorrelation function. (08)

3. A discrete time system can be static or dynamic, linear or non-linear, Time variant or time invariant, causal or non causal, stable or unstable. Examine the following system with respect to the properties also
   - y(n)=cos(x(n))
   - y(n)=x(-n+2)
   - y(n)=x(2n)
   - y(n)=x(n) cos ωn (16)
4. Determine whether each of the following systems defined below is (i) casual (ii) linear (iii) dynamic (iv) time invariant (v) stable
   - y(n) = log10[{x(n)}]
   - y(n) = x(-n-2)
   - y(n) = cosh[nx(n) + x(n-1) (16)
5. Find the convolution of the two signals
   x(n) = 3nu(-n); h(n) = (1/3)n u(n-2)
   x(n) = (1/3) –n u(-n-1); h(n) = u(n-1)
   x(n) = u(n) –u(n-5); h(n) = 2[u(n) – u(n-3)] (16)

6. Compute the convolution y(n) of the signals
   x(n)= a^n -3≤n≤5 0 elsewhere and
   h(n)= 1 0≤n≤4 0 elsewhere (16)

UNIT- II

FAST FOURIER TRANSFORMS

PART A (2 MARKS)

1.Differentiate DTFT and DFT.
2.Differentiate between DIT and DIF algorithm.
3. DIF-Frequency is decimated and input is natural order output is bit reversed format.
4. How many stages are there for 8 point DFT?
5. How many multiplication terms are required for doing DFT by expressional method and FFT method?
6. Write the steps in designing chebyshev filter?
7. What is the different method of evaluating inverse z-transform?
8. State the initial value theorem and the final value theorem.
9. State parsevals relation in z-transform.
10 What is zero padding? What are its uses?

11. Define discrete Fourier series?
12. When the DFT x(k) of a sequence x(n) is imaginary?
13. When the DFT X(k) of a sequence x(n)is real?
14 If X (K) is DFT of a sequence x(n), then DFT of imaginary part of x(n) is__________.
15. Explain circular frequency shifting property of DFT?
16. What is FFT?
17 What is meant by radix -2 FFT
18. What are the methods of used for the sectional convolution?
19. What is meant by sectional convolution?
20. Differentiate between DIT and DIF algorithm.

PART –B (16 MARKS)

1.Determine the DFT of the sequence

x(n) = 1/4, for 0<=n <=2

= 0, otherwise

2. Derive the DFT of the sample data sequence x(n) = {1,1,2,2,3,3}and compute the corresponding amplitude and phase spectrum.
3.Given x(n) = {0,1,2,3,4,5,6,7} find X(k) using DIT FFT algorithm.
4.Find the inverse DFT of X(k) = {1,2,3,4}

UNIT- III

IIR FILTER DESIGN

PART A (2 MARKS)

1.What is warping effect?
2.Write a note on pre warping.
3.Give the bilinear transform equation between s plane and z plane.
4.Why impulse invariant method is not preferred in the design of IIR filters other than lowpass filter?
5.What is meant by impulse invariant method?
6.What do you understand by backward difference?
7.What are the properties of chebyshev filter?
8.Give the magnitude function of Butterworth filter?
9.Give the expression for poles and zeroes of a chebyshev type 2 filters.
10.How can you design a digital filter from analog filter?
11.Write down bilinear transformation.
12.Differentiate Butterworth and Chebyshev filter.
13.What is filter?
14.What are the types of digital filter according to their impulse response?
15.How phase distortion and delay distortion are introduced?

PART –B (16 MARKS)

1.Derive the condition of FIR filter to be linear in phase.
Conditions are
- Group delay and Phase delay should be constant
- show the condition is satisfied. (16)

2. i) Describe impulse invariant mapping technique for designing IIR filter. (08)
ii) Develop cascade and parallel realization of the system described by the difference equation $y(n) + (3/8)\, y(n-1) - (3/32)\, y(n-2) - (1/64)\, y(n-3) = x(n) + 3x(n-1) + 2x(n-2)$ (08)

3. Design a Butter worth digital filter to meet the following constraints.
$0.9 \leq |H(\omega)| \leq 1,\ 0 \leq \omega \leq \Pi/2$
$|H(\omega)| \leq 0.2,\ 3\Pi/4 \leq \omega \leq \Pi$
Use bilinear transformation mapping technique. Assume T=1 Sec (16)

4. Consider the system described by
$$y(n)-(3/4)y(n-1)+(1/8)y(n-2)=x(n)+(1/3)x(n-1)$$
Determine and draw all possible realization structures. (16)

5. Explain the following terms briefly.
i. Frequency sampling structure. (04)
ii. Lattice structure for IIR filter (04)
iii.Perturbation error. (04)
iv. Limit cycles. (04)

UNIT- IV

FIR FILTER DESIGN

PART A (2 MARKS)

1. Write the steps involved in FIR filter design.
2. What are advantages of FIR filter?
3. What are the disadvantages of FIR FILTER?
4. What is the necessary and sufficient condition for the linear phase characteristic of a FIR filter?
5. List the well known design technique for linear phase FIR filter design?
6. Define IIR filter?
7. For what kind of application, the antisymmetrical impulse response can be used?
8. For what kind of application, the symmetrical impulse response can be used?
9. What is the reason that FIR filter is always stable?
10. What condition on the FIR sequence h(n) are to be imposed n order that this filter can be called a liner phase filter?
11. Under what conditions a finite duration sequence h(n) will yield constant group delay in its frequency response characteristics and not the phase delay?
12. State the condition for a digital filter to be causal and stable?

13. What are the properties of FIR filter?
14. When cascade from realization is preferred in FIR filters?
15. What are the disadvantages of Fourier series method?
17. What are the desirable characteristics of the window?
18. What are the advantages of Kaiser window?
19. For what type of filter frequency sampling method is suitable?
20. What are the necessary and sufficient condition for linear phase characteristic in FIR filter?

PART –B (16 MARKS)

1. Derive the condition of FIR filter to be linear in phase. (16)
2. Describe briefly the different methods of power spectral estimation?
   i. Bartlett method (06)
   ii. Welch method (06)
   iii. Blackman-Turkey method and its derivation. (04)
3. Design a digital low pass filter FIR filter of length 11,with cut off frequency of 0.5 kHz and sampling rate 2 kHz using hamming window. (16)
4. Explain the design of FIR filter using frequency sampling technique. (16)
5. Use Fourier series method in conjunction with a hamming window to design an approximation to an ideal LPF with magnitude response

$$H(e^{j\omega}) = 1 \quad \omega \le \frac{\pi}{3}$$
$$= 0 \quad \text{otherwise}$$

Compare the response with that obtained from an unwindowed design for N=11. (16)

UNIT- V

FINITE WORD LENGTH EFFECTS

PART-A (2 MARKS)

1. Define white noise?
2. What do you understand by a fixed-point number?
3. What is the objective of spectrum estimation?
4. List out the addressing modes supported by C5X processors?
5. What is meant by block floating point representation? What are its advantages?
6. What are the advantages of floating point arithmetic?
7. How the multiplication & addition are carried out in floating point arithmetic?
8. What do you understand by input quantization error?
9. List the on-chip peripherals in 5X.
10. What is the relationship between truncation error e and the bits b for representing a decimal into binary?
11. What is meant rounding? Discuss its effect on all types of number representation?
12. What is meant by A/D conversion noise?
13. How would you relate the steady-state noise power due to quantization and the b bits representing the binary sequence?
14. What is overflow oscillation?
15. What are the methods used to prevent overflow?
16. What are the two kinds of limit cycle behavior in DSP?

a. Zero input limit cycle oscillations.
b. Overflow limit cycle oscillations.

PART –B (16 MARKS)

1. Derive the expression for steady state I/P Noise Power and Steady state O/P Noise power. (16)
2. Draw the product quantatization model for first order and second order filter and write the difference equation and draw the noise model. (16)
3. For the second order filter draw the direct form II realization and find the scaling factor S0 to avoid over flow. Find the scaling factor from the formula

$$I = \frac{1+r2}{(1-r2)(1-2r2\cos 2\o =r4)} \qquad (16)$$

4. Explain briefly about various number representation in digital computer. (16)

5. Consider the transfer function H (Z) =H1 (Z) H2 (Z) where H1 (Z) =1/1-a1Z-1
H2(z) = 1/ 1-a2Z-1
Find the o/p Round of noise power Assume a1=0.5 and a2= 0.6 and
Find o.p roundoff noise power. (16)
6. What is meant by A/D conversion noise? Explain in detail? (16)