The following questions should be addressed in Chapter one

1.Signals and Systems Basics Question:

Explain the basic elements of a DSP system with a neat block diagram. Discuss at least three practical applications of digital signal processing.

2. Classification and Properties of Signals

Question: Define and classify signals into analog and digital, deterministic and random, periodic and aperiodic, even and odd, and energy and power signals. Provide examples for each type.

3. Signal Transformation

Question: Explain the following transformations of the independent variable in signals with examples:(a) Time shifting(b) Time scaling(c) Amplitude scaling(d) Time inversion(e) Combined operations

4. Discrete-Time System Classification

Question: Classify discrete-time systems into static and dynamic, time-invariant and time-variant, linear and nonlinear, causal and noncausal, and stable and unstable systems. Provide examples for each classification.

5. Linear Time-Invariant (LTI) Systems

Question: Derive the convolution summation formula for discrete-time systems. Discuss the properties of convolution (commutative, associative, distributive) and explain how causality and stability of an LTI system are determined.

The following questions should be addressed in Chapter Two

1.Definition, Region of Convergence, and Relationship to Causality and Stability

a) Define the Z-transform and explain its significance in signal processing.

b) b) Explain the Region of Convergence (ROC) for the Z-transform. How does the ROC relate to the causality and stability of a system? Illustrate with examples.

2.Properties of Z-transform

a) Derive the linearity and time-shifting properties of the Z-transform. Provide relevant examples to illustrate these properties.

b) Explain the convolution property of the Z-transform and describe its application in analyzing discrete-time systems.

Inverse Z-transform

a) Describe the steps to determine the inverse Z-transform of a function using partial fraction expansion. Solve an example to demonstrate the process.

b) Using the power series expansion method, find the inverse Z-transform of $X(z) = \frac{1}{1-0.5z^{-1}}$.

4. Applications and Theoretical Understanding

a) State and prove Parseval's theorem for Z-transform. Explain its physical significance.

b) Use long division to find the inverse Z-transform of $X(z) = \frac{z}{z-2}$. Comment on the stability of the system based on the ROC.

The following questions should be addressed in Chapter Three

1. Frequency Response of LTI Systems

Question: Explain the frequency response of an LTI system. Derive the response of an LTI system to a complex exponential input and discuss its significance in analyzing system behavior.

2. System Function and Pole-Zero Analysis

Question: Derive the system function for a linear constant-coefficient difference equation. Explain how the frequency response of the system is related to its pole-zero locations in the z-plane, with examples.

3. Linear Phase and Causality

Question: Discuss the concept of linear phase in an LTI system. Explain the relationship between linear phase and causality, and illustrate with an example how phase characteristics affect signal processing.

The following questions should be addressed in Chapter Four

1. Structures for FIR Filters

a) Describe the direct form structure for FIR filters. Derive the output equation and explain its implementation.

b) Compare the cascade and lattice structures for FIR filters. Highlight their advantages and use cases.

2. Structures for IIR Filters

a) Explain the difference between Direct Form I and Direct Form II structures for IIR filters. Which form is more efficient and why?

b) Describe the lattice and lattice-ladder structures for IIR filters. Discuss their applications and benefits in filter design.

3. Quantization Effects

a) Explain the effects of truncation and rounding in digital filter implementation. How do these affect the stability and accuracy of the system ?

b) Discuss the strategies to minimize quantization effects in FIR and IIR filter structures.

The following questions should be addressed in Chapter Five

1. Classical Filter Design

a) Explain the procedure for designing Butterworth filters using polynomial approximations. Illustrate with an example.

b) Compare Butterworth and Chebyshev filters in terms of their magnitude response, cutoff sharpness, and order requirements.

2. Impulse Invariance and Bilinear Transformation

a) Describe the impulse invariance method for designing IIR filters. Derive the equations and provide an example.

b) Explain the bilinear transformation method for filter design. How does it handle frequency warping?

3. Properties of Chebyshev and Elliptic Filters

a) Discuss the key properties of Chebyshev filters. Compare Type I and Type II Chebyshev filters.

b) Explain the characteristics of Elliptic filters and how they differ from Butterworth and Chebyshev filters.

The following questions should be addressed in Chapter Six

1. Gibbs Phenomena in FIR Filter Design

a) What is the Gibbs phenomenon in FIR filter design? Explain its causes and impact on the filter's frequency response.

b) Discuss methods to minimize the Gibbs phenomenon in FIR filter design.

2. Applications of Window Functions

a) Compare the characteristics and applications of different window functions: Rectangular, Triangular, Hanning, Hamming, and Kaiser.

b) Explain how the choice of a window function affects the design and performance of an FIR filter.

3. FIR Filter Design Using Window Functions

a) Design an FIR low-pass filter using the Hamming window method. Provide the steps and calculations involved.

b) Discuss the trade-offs between main-lobe width and side-lobe level for different window functions in FIR filter design.

4. FIR Filter Design by Frequency Sampling Method

a) Describe the frequency sampling method for FIR filter design. Explain its advantages and limitations.

b) Design an FIR filter using the frequency sampling method for a given set of frequency specifications.

The following questions should be addressed in Chapter Seven

1. Definition and Representation of DFT

Question: Define the Discrete Fourier Transform (DFT) and its inverse. Derive its mathematical representat and explain its significance in signal processing.

2. Properties of DFT

Question: Discuss the key properties of the DFT, including:(a) Linearity(b) Time shift(c) Frequency shift Conjugation and conjugate symmetry(e) Convolution and multiplication. Illustrate each property with exampl

3. Circular Convolution

Question: Explain the concept of circular convolution and its relationship to linear convolution. Derive mathematical formulation of circular convolution and provide a step-by-step example.

4. Fast Fourier Transform (FFT) Algorithms

Question: Explain the Fast Fourier Transform (FFT) algorithm. Compare the decimation-in-time (DIT) a decimation-in-frequency (DIF) algorithms, and outline their computational steps with examples.

5. Computational Complexity of FFT

Question: Discuss the computational complexity of the FFT algorithm compared to the direct computation of DFT. Explain how the FFT reduces the number of computations and its impact on practical applications.

Marks Grid

Attempt all the questions			
1 a) b)	Ch#1 Ch#1	7+8	15
2 a) b)	Ch#2 Ch#3	7+8	15
3 a) b)	Ch#4 Ch#4	7+8	15
4 a) b)	Ch#5 Ch#5	7+8	15
5 a) b)	Ch#6 Ch#6	7+8	15
6 a) b)	Ch#7 Ch#7	7+8	15
7. Write short notes on (any two) a) Ch#1 b) Ch#2 c) Ch#3		5+5	10
There should be exactly two OR questions, and no more, from the same chapter in the respective questions, except for Question No. 7.			

Marks Distribution Table

S. No	Chapter	Hrs.	Total Marks (Approx.)
1.	1.Discrete Signals and systems	8	18
2.	2. Z-transform	6	13
3	3. Analysis of LTI system in frequency domain	5	12
4.	4. Discrete filter structures	6	13
5.	5. IIR Filter Design	7	16
6.	6. FIR Filter Design	7	16
7.	7.DiscreteFourier Transform	6	13
The questions are prepared based on the above marks distribution			