17892

BRAINWARE UNIVERSITY

Term End Examination 2025-2026
Programme – B.Tech.(RA)-2023
Course Name – MOOC-Real-Time Digital Signal Processing
Course Code - PEC-ECR501C
(Semester V)

Full Marks : 70 **Time : 3:0 Hours**

[The figure in the margin indicates full marks. Candidates are required to give their answers in their own words as far as practicable.]

Group-A

(Multiple Choice Type Question) 1 x 15=15

1. *Choose the correct alternative from the following :*

(i) Define the concept of real-time signal processing.
a) Processing signals with human interaction only
b) Processing signals without delay as data arrives
c) Processing stored signals in batch mode
d) Processing signals using analog techniques

(ii) State the purpose of DSP hardware.
a) To create music
b) To slow down signals
c) To execute digital signal processing tasks efficiently
d) To reduce power supply requirements

(iii) Classify signal types in real-time DSP systems.
a) Static and transient signals
b) Continuous, Discrete, Deterministic, and Random
c) Electrical and magnetic signals
d) Binary and analog signals only

(iv) Compare fixed point and floating point in DSP.
a) Both store unlimited data
b) Floating point offers better precision, fixed point offers speed and less memory use
c) Fixed point is more expensive than floating point
d) Both are same in performance

(v) Define the concept of filter stability in IIR filters.
a) All poles lie outside the unit circle
b) All poles lie inside the unit circle
c) All zeros lie inside the unit circle
d) Filters have equal coefficients

(vi) Classify IIR filters based on their pole-zero configuration.
a) Gaussian and triangular
b) All-pass, band-pass, low-pass, high-pass
c) Butterworth, Chebyshev, Elliptic
d) Binary and analog

(vii) State the importance of filter length in FIR filter design.
a) Determines output gain
b) Controls tap size
c) Affects frequency response sharpness
d) Reduces sampling time

(viii) Describe the implementation of cascaded IIR filters in hardware.

a) Using one large filter
b) Direct form II only
c) Connecting multiple lower-order filters in series
d) Using windowing method

(ix) Define a random process and its characteristics in signal processing.

a) A predictable signal with known values
b) Set of random variables indexed by time
c) Constant-valued signal
d) Deterministic signal only

(x) Explain the concept of the LMS algorithm.

a) Algorithm for compression
b) Statistical filter with fixed weights
c) Adaptive filter minimizing mean square error
d) Fourier-based algorithm

(xi) Identify the main steps involved in the LMS algorithm implementation.

a) Sampling, filtering, modulating
b) Initialization, error computation, weight update
c) ADC, DAC, computation
d) Feedback, amplification, scaling

(xii) Define the concept of convergence in the LMS algorithm.

a) When error keeps increasing
b) Algorithm crashes
c) Filter output reaches the desired signal closely
d) When output is noise

(xiii) Classify different types of adaptive filters used in signal processing.

a) Analog and digital filters
b) LMS, RLS, Kalman filters
c) FIR and IIR only
d) Linear and logarithmic

(xiv) Describe the significance of digital signal generation in telecommunications.

a) Only improves volume
b) Enhances signal timing
c) Removes filters
d) Enables efficient, reliable, and noise-resistant data transmission

(xv) State the impact of quantization on digital signal generation.

a) Improves accuracy
b) Adds continuous values
c) Increases delay
d) Introduces quantization noise and limits precision

Group-B

(Short Answer Type Questions) 3 x 10=30

2. Define real-time signal processing. (3)
3. Summarize the key components of DSP hardware. (3)
4. Explain the impact of real-time processing in DSP system design. (3)
5. Describe the advantages of FIR filters over IIR filters. (3)
6. Identify the concept of the Discrete Fourier Transform (DFT). (3)
7. Explain the impact of quantization effects in signal processing. (3)
8. Analyze the effect of quantization noise on digital filters. (3)
9. List the components of a graphic equalizer. (3)
10. Analyze the effect of sampling rate on audio signal quality. (3)
11. Analyze the relationship between the mean square error (MSE) and the LMS algorithm's performance. (3)

OR

Analyze the effect of different initial weight settings on the LMS algorithm. (3)

Group-C

(Long Answer Type Questions) 5 x 5=25

12. Explain the design differences in DSP systems for real-time applications, covering hardware and software considerations. (5)

13. Examine the effects of quantization on signal accuracy and precision, and provide examples. (5)
14. Explain the challenges and solutions in using FFT for real-time signal processing in embedded systems. (5)
15. Discuss the importance of windowing in FFT for improving frequency resolution in spectral analysis. (5)
16. Explain the impact of quantization effects on IIR filter design. How can these effects be minimized? (5)

OR

Analyze the stability issues associated with IIR filters and the methods used to ensure stability in their design. (5)

****