Hall Ticket No | | | | | | | | | | | Question Paper Code: ACAC05

INSTITUTE OF AERONAUTICAL ENGINEERING

(Autonomous)
Dundigal-500043, Hyderabad

B.Tech V SEMESTER END EXAMINATIONS (REGULAR/ SUPPLEMENTARY) - FEBRUARY 2024
Regulation: UG20
IMAGE AND SPEECH PROCESSING

Time: 3 Hours CSE (ARTIFICIAL INTELLIGENCE AND MACHINE LEARNING) **Max Marks: 70**

Answer ALL questions in Module I and II
Answer ONE out of two questions in Modules III, IV and V
All Questions Carry Equal Marks
All parts of the question must be answered in one place only

MODULE – I

1. (a) List the fundamental steps in image processing. Distinguish between histogram equalization and histogram matching. [BL: Understand| CO: 1|Marks: 7]

   (b) Implement a non-linear intensity transformation algorithm for enhancing specific details in an image. [BL: Apply| CO: 1|Marks: 7]

MODULE – II

2. (a) Discuss the significance of entropy in the context of information theory and redundancy. How is entropy related to compression efficiency? [BL: Understand| CO: 2|Marks: 7]

   (b) Implement a lossy predictive coding algorithm for compressing a color image. Evaluate the impact of prediction errors on the visual quality of the compressed image. [BL: Apply| CO: 2|Marks: 7]

MODULE – III

3. (a) Differentiate between acoustic phonetics and articulatory phonetics for speech processing. [BL: Understand| CO: 3|Marks: 7]

   (b) Design a diagram illustrating the key stages of speech production, from the initiation of speech through articulation and sound emission. Explain each stage in detail. [BL: Apply| CO: 3|Marks: 7]

4. (a) Elaborate the concept of quantization in the context of digital speech signals. How does quantization impact the fidelity of the digitized speech signal?[BL: Understand| CO: 4|Marks: 7]

   (b) Given a speech signal waveform, compute and plot its short-time Fourier transform (STFT). Highlight the regions in the time-frequency domain that correspond to different speech sounds. [BL: Apply| CO: 4|Marks: 7]

MODULE – IV

5. (a) Explain the concept of time-dependent processing in speech analysis. Why is it necessary to analyze speech signals in short-time intervals rather than considering the entire signal at once? [BL: Understand| CO: 5|Marks: 7]

(b) Outline the impact of windowing on the accuracy of pitch period estimation using the autocorrelation function. Propose strategies to minimize estimation errors.
[BL: Understand| CO: 5|Marks: 7].

6. (a) Develop a speech processing algorithm that utilizes time-dependent features to enhance speech analysis. [BL: Understand| CO: 5|Marks: 7]

(b) Examine the applications of pitch period estimation beyond speech processing. In what other domains or fields can accurate pitch estimation be valuable? [BL: Understand| CO: 5|Marks: 7]

MODULE – V

7. (a) Write about filter banks in the context of signal processing. What is their role, and properties should an ideal filter bank possess? [BL: Understand| CO: 6|Marks: 7]

(b) Implement a filter bank summation method using fast fourier transform (FFT) for a given input signal. Evaluate the efficiency of the implementation in terms of computational complexity.
[BL: Apply| CO: 6|Marks: 7]

8. (a) Summarize the concept of analysis by synthesis in speech processing. How does this approach utilize a model to analyze and synthesize speech signals? [BL: Understand| CO: 6|Marks: 7]

(b) What is linear predictive analysis? With linear predictive analysis, explain the lossless tube model. [BL: Understand| CO: 6|Marks: 7]

– ○ ○ ◯ ○ ○ –