<u>Course Name:</u> Audio and Speech Signal Processing **<u>Semester -VIII</u>**

Question Bank

Unit 1

1. Describe the structures involved in the production of speech, including the major components of the respiratory, phonatory, and articulatory systems.
2. Define Phonemes and explain the key steps involved in voice production
3. Discuss the function of the vocal folds and their role in regulating pitch and intensity during speech.
4. Explain different category of speech sound i.e. phonemes

Unit 2

5. Explain Linear Predictive Coding (LPC) model
6. Explain the source filter model of Human Speech production system.
7. A phoneme whose pitch is 100 Hz, is sampled at 6 kHz. It has two formants: a weak one at 500 Hz and a stronger one at 2 kHz. Determine the approximate pole locations of H(z).
8. Describe the process and significance of applying the Short Time Fourier Transform (STFT) to analyze signals in the time-frequency domain.

Unit 3

9. What is the principle behind a correlation-based pitch estimator? Explain how it works and mention its advantages and limitations.

10. Describe the functioning of pitch estimation based on a comb filter. How does the comb filter method estimate pitch.
11. Discuss the concept of pitch estimation using a harmonic sine wave model. How does this model represent pitch, and what are the key steps involved in estimating pitch based on it?

12. Explain how the accuracy of pitch estimation methods can be evaluated. What are some common metrics used to measure the performance of these methods.

Unit 4

13. Explain the concept of vector quantization in speech coding. How does it work, and what are its advantages and limitations compared to other coding techniques?

14. Describe the process of frequency-domain coding in speech compression. What are the key steps involved, and how does frequency-domain coding exploit the properties of speech signals to achieve compression?

15. Discuss the transform-based approach to speech coding. What transforms are commonly used, and how are they applied to speech signals for compression purposes.

16. Describe common audio compression standards, such as MP3, AAC, and FLAC. What are the key features of each standard, and how do they differ in terms of compression algorithms, supported features, and compatibility with playback devices?

Unit 5

17. Describe the role of various spectral features in speaker recognition systems, including the extraction process and their significance.

18. Discuss the principles behind Wiener filtering for speech and audio signal enhancement. How does Wiener filtering estimate the clean signal from its noisy version, and what are its applications in real-world scenarios?

19 What spectral features are commonly used for speaker recognition tasks? Explain how features like Mel-Frequency Cepstral Coefficients (MFCCs), Linear Predictive Coding (LPC), and Perceptual Linear Prediction (PLP) are extracted and utilized in speaker recognition systems.

Course Teacher
Prof. Pradeep Barde