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. Explain anatomy of speech production system with neat diagram
3. Explain the key steps involved in Speech production. Also compare the Vocal Tract and Nasal Tract in Speech Production Systems
4. Discuss the function of the vocal folds and their role in regulating pitch and intensity during speech.
5. Explain in detail following terms related to Speech production
   a) Voiced Speech b) Unvoiced Speech c) Pitch and d) Formants
6. Explain classification of speech sounds in detail with diagrams.

Unit 2

7. Explain Linear Predictive Coding (LPC) model of Speech production
8. Explain LPC in speech compression.
9. Explain LPC analysis-synthesis system with block diagram.
10. Explain the autocorrelation method for pitch detection and LPC analysis
11. Explain the source filter model of Human Speech production system.
12. 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).
13. Describe the process and significance of applying the Short Time Fourier Transform (STFT) to analyze signals in the time-frequency domain.
14. Explain the Short-Time Fourier Transform (STFT) for speech analysis and synthesis

Unit 3

15. Explain the correlation-based pitch estimation method in detail.
16. What is the principle behind autocorrelation-based pitch estimator? Explain how it works and mention its advantages and limitations.

17. Explain the functioning of pitch estimation based on a comb filter. How does the comb filter method estimate pitch.

18. Explain the concept of pitch estimation based on harmonic sine wave model. How does this model represent pitch, and what are the key steps involved in estimating pitch based on it?

19. Explain how the accuracy of pitch estimation methods can be evaluated. What are some common metrics used to measure the performance of these methods.
20. Compare the following pitch estimation methods:

Correlation-based method and Comb filter method

Unit 4

21. 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?
22. Explain the Vector Quantization (VQ) technique for speech coding

23. 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?

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

26. 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?

27. Explain various audio compression techniques used in speech processing.

Unit 5

28. Explain the spectral subtraction and Cepstral Mean Subtraction method for speech enhancement
29. Explain Cepstral Mean Subtraction (CMS) and Wiener filtering techniques for speech enhancement.
30. Explain the spectral features used in speaker recognition systems
31. Explain Minimum Distance classifier and Gaussian Mixture Model (GMM) for speaker recognition.
32. Describe the role of various spectral features in speaker recognition systems, including the extraction process and their significance.
33. Discuss the principles behind Wiener filtering for speech and audio signal enhancement.
34. How does Wiener filtering estimate the clean signal from its noisy version, and what are Its applications in real-world scenarios?
35. 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.