Hall Ticket No | | | | | | | | | | | Course Code: ACAC05

INSTITUTE OF AERONAUTICAL ENGINEERING

(Autonomous)

Dundigal, Hyderabad - 500 043

B TECH V SEMESTER CIE–II EXAMINATIONS, FEBRUARY– 2024

Regulation: UG20

IMAGE AND SPEECH PROCESSING

Time: 2 Hours **CSE (AI&ML)** **Max Marks: 20**

Answer any FOUR questions

All parts of the question must be answered in one place only

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1. (a) List out the different losses existing in vocal tract. Comment on the relation between the transfer function of vocal tract and digital filters [BL: Understand| CO: 4|Marks: 2]

   (b) Demonstrate the wave propagation of sound in concatenated lossless tube using signal flow graph and derive the reflection coefficient. [BL: Apply | CO: 4|Marks: 3]

2. (a) Illustrate the term autocorelation function with relevant diagram. Draw the block diagram representation of short-time zero-crossings. [BL: Understand| CO: 5|Marks: 2]

   (b) What is AMDF? How will you use AMDF for pitch measurement? Differentiate between computation of pitch using AMDF and autocorrelation method.

   [BL: Understand | CO: 5|Marks: 3]

3. (a) Interpret the role of window function in speech signal processing? Why a window size of 20-30 is normally preferred for short-time speech signal processing

   [BL: Understand| CO: 5|Marks: 2]

   (b) Explain the linear filter operation of a short-time spectrum analysis with related equations and · block diagram. Also discuss about magnitude of the short time spectrum using both low pass filters and band pass filter [BL: Understand | CO: 5|Marks: 3]

4. (a) Summarize the linear predictive coding analysis and its uses in speech signal processing applications. [BL: Understand| CO: 6|Marks: 2]

   (b) Determine the second-order linear prediction inverse filter, A(z), for which the two LSF are 666.67 Hz and 2000 Hz, when Fs = 8000 samples/sec [BL: Apply | CO: 6|Marks: 3]

5. (a) Comment on the model of frequency-domain processing of speech via STFA and STFS methods.

   [BL: Understand| CO: 6|Marks: 2]

   (b) A speech signal is sampled at a rate of 10,000 samples/sec (i.e., Fs = 10,000). A Hamming window of length L samples is used to compute the STFT of the speech signal. The STFT is sampled in time with period R, and in frequency at N = 1024 frequencies.

   i) It can be shown that the main lobe of the Hamming window has a symmetric full width of approximately 8 /L. How should L be chosen if we want the full width of the main lobe to correspond to approximately 200 Hz analog frequency?

   ii) What is the spacing (in Hz) between sample points in the frequency dimension?

   [BL: Apply | CO: 6|Marks: 3]

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