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APJ ABDUL KALAM TECHNOLOGICAL UNIVERSITY

FIRST SEMESTER M.TECH DEGREE EXAMINATION, DECEMBER 2017

Electronics & Communication Engineering

1. Signal Processing

2. VLSI and Embedded Systems

01EC6311 Speech Signal Processing

Answer *any two full* questions from *each* part

Limit answers to the required points.

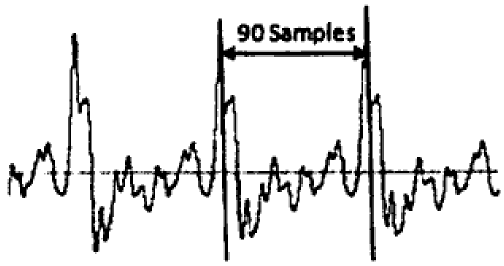
Max. Marks: 60 Duration: 3 hours

PART A

1. a. Analyze the acoustic modeling of the vocal tract in speech processing. (5)

   b. Describe the state of the Glottis during the pronunciation of the following phonemes: i) /k/, (ii) /b/, (iii) /a/, (iv) /ʃ/ (2)

   c. Determine F_0 of the following signal if the signal is sampled at 22050 Hz. (2)



2. a. Explain frequency domain analysis of speech signals using filter banks. (2)

   b. Explain how cepstral analysis is used in speech signal processing, with relevant block diagram. (5)

   c. Suppose uniform filter bank analysis is used to extract the parameters of a speech segment. If the bandwidth of each filter is 100 Hz and speech signal is recorded with sampling frequency 12 KHz, determine the required number of filters to cover the entire spectrum of the speech segment. (2)

3. a. Describe the use of zero crossing rate and autocorrelation function determination in time-domain analysis of speech signals. (5)

   b. Differentiate between narrowband and wideband spectrograms. (4)

PART B

4. a. Derive the expression for autocorrelation matrix in linear prediction analysis of speech signals. (5)

   b. State the property of the Toeplitz matrix that makes it suitable for using recursive algorithms like the Levinson Durbin algorithm. (1)

   c. Write the Levinson Durbin equations used in solving autocorrelation matrix. (3)

5. a. Explain the application of Hidden Markov model in speech analysis. (6)

   b. Describe the sinusoidal model of speech. (3)

6. a. Derive the Covariance matrix of LPC analysis. (5)

   b. Describe Gaussian Mixture Modeling of speech. (4)

PART C

7. a. Explain how sub-band coding is performed in speech coding. (6)

   b. Describe any one vocoder based on the CELP algorithm. (6)

8. a. With the help of a block diagram, explain how a Text-to-speech system is implemented. (6)

   b. Describe how segmentation is achieved in speech recognition systems. (4)

   c. An audio signal is recorded using sampling frequency Fs = 8 kHz, encoded with 16 bit and recorded in mono. How much memory is required to store 100ms signal in PCM WAV format? (2)

9. a. Distinguish between adaptive transform coders and harmonic coders. (6)

   b. Explain the issues faced in language identification and voice transmission over the internet. (6)