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| **ECx539** | **Speech Processing** |  **L-T-P: 3-0-0; Total 42 Lectures** |

***Pre-requisite:*** Basics of Signal Processing and Random Processes

***Course Objectives:***

1. To provide a comprehensive introduction to speech signal processing and statistical modeling methods.
2. To provide comprehensive knowledge on the development of human-machine interactive systems using speech signals for different applications.

***Course Contents:***

**Unit-I:** Speech production and perception, information sources in speech signal, linguistic aspect of speech, acoustic and articulatory phonetics, nature of speech, models for speech analysis and perception, short-term processing of speech, time, frequency and time-frequency analysis, development of short-term Fourier transform (STFT), transform and filter-bank views of STFT. [7 L]

**Unit-I1:** Basis and development cesptrum analysis of speech, real and complex cepstrum, pitch detection, formant estimation, Mel-frequency cepstral coefficient (MFCC), delta and delta-delta MFCC, Linear Prediction (LP) analysis, LP analysis of speech, solution of LP equation using Levinson-Durbin’s method, normalized error, LP spectrum, LP cepstrum, LP residual. [12L]

**Unit-III:** Speech enhancement: objective, issues, enhancement of noisy speech, reverberant speech and multi-speaker speech using time, frequency and time-frequency approaches. [5L]

**Unit-IV:** Basic concepts of pattern recognition: feature extraction, modeling, testing, Objective, issues, block diagram description of automatic speech recognition (ASR) system, development of ASR system using vector quantization (VQ), dynamic time warping (DTW), Hidden Markov Model (HMM) and Neural networks (NN). [10L]

**Unit-V:** Objective, issues, block diagram description of speaker recognition system, classification of speaker recognition systems, development of speaker recognition system using VQ, Gaussian mixture model (GMM), Adapted-GMM and I-vetor [8L]

***Text Books:***

1. L.R. Rabiner and R.W. Schafer, Digital Processing of Speech Signals Pearson Education, Delhi, India, 2004
2. L. R. Rabiner, B. H. Jhuang and B. Yegnanarayana, “Fundamentals of speech recognition”, Pearson Education, 2009.

 ***Reference Books***

1. J. R. Deller, Jr., J. H. L. Hansen and J. G. Proakis Discrete-Time Processing of Speech Signals, Wiley-IEEE Press, NY, USA, 1999
2. T. F. Quatieri, “Discrete time processing of speech signals”, Pearson Education, 2005.

***Course Outcomes (CO):***

At the end of this course, the students should be able to:

1. Understand the basic concepts of speech production and perception, information source in the speech signal, applications of speech signal processing for the modern world.
2. Understand the concepts of analysis and processing of speech signals for extracting relevant information and enhancement of speech signals in the presence of different background noises.
3. Understand the concepts pattern recognition system and different statistical modeling approaches.
4. Understand how to develop human-machine interactive systems using speech signals.