IV B.Tech – I Semester

(20EC7724) DIGITAL SPEECH AND AUDIO SIGNAL PROCESSING
(Minors)

Int. Marks	Ext. Marks	Total Marks	L	T	P	C
30	70	100	3	1	0	4

Pre-Requisites: Digital Signal Processing

Course Objectives:

- To familiarize the basic mechanism of speech production and the basic concepts of of speech signals
- To study the speech processing methods in time and frequency domain
- To study the analysis of speech signals
- To give an overall picture about various applications of speech processing
- To learn various transform coders for audio coding

UNIT-I: Speech Basic Concepts

Fundamentals of Digital Speech Processing: Anatomy & Physiology of Speech Organs, The process of Speech Production, Acoustic Phonetics, Articulatory Phonetics, The Acoustic Theory of Speech Production- Uniform lossless tube model, effect of losses in vocal tract, effect of radiation at lips, Digital models for speech signals.

UNIT-II: Time and Frequency Domain Methods for Speech Processing

Time domain parameters of Speech signal – Methods for extracting the parameters: Energy, Average Magnitude – Zero crossing Rate – Silence Discrimination using ZCR and energy Short Time Fourier analysis – Formant extraction – Pitch Extraction using time and frequency domain methods.

Homomorphic Speech Analysis:

Cepstral analysis of Speech – Formant and Pitch Estimation – Homomorphic Vocoders.

UNIT-III: Linear Predictive Analysis of Speech

Formulation of Linear Prediction problem in Time Domain – Basic Principle – Auto correlation method – Covariance method – Solution of LPC equations – Cholesky method – Durbin's Recursive algorithm – lattice formation and solutions – Comparison of different methods – Application of LPC parameters – Pitch detection using LPC parameters – Formant analysis – VELP – CELP.

UNIT-IV: Application of Speech Signal Processing

Automatic Speech & Speaker Recognition: Basic pattern recognition approaches, Parametric representation of speech, Evaluating the similarity of speech patterns, Isolated digit Recognition System, Continuous digit Recognition System. Hidden Markov Model (HMM) for Speech: Hidden Markov Model (HMM) for speech recognition, Viterbi algorithm, Training and testing using HMMS, Speaker Recognition: Recognition techniques, Features that distinguish speakers, Speaker Recognition Systems: Speaker Verification System, Speaker Identification System.

UNIT-V: Audio Coding and Transform Coders

Lossless Audio Coding-Lossy Audio Coding- ISO-MPEG-1A,2A,2A Advanced, 4Audio Coding - Optimum Coding in the Frequency Domain - Perceptual Transform Coder -Brandenburg-Johnston Hybrid Coder - CNET Coders - Adaptive Spectral Entropy Coding -Differential Perceptual Audio Coder - DFT Noise Substitution -DCT with Vector Quantization -MDCT with Vector Quantization.

Course Outcomes:

After successful completion of the course, the students can be able to

S.No	Course Outcome	BTL
1	Know basic concepts of speech production	L1
2	Understand the speech analysis, speech coding and parametric representation of speech and apply it in practical applications	L2
3	Extract the LPC coefficients that can be used to synthesize or compress the speech	L3
4	Develop systems for various applications of speech processing	L3
5	To understand the basic audio coding methods	L2

Correlation of Cos with Pos & PSOs:

CO	PO1	PO2	PO3	PO4	PO5	PO6	PO7	PO8	PO9	PO10	PO11	PO12	PSO1	PSO2
CO 1	3	3	1	-	-	-	-	-	-	-	-	-	3	-
CO 2	2	3	1	-	-	-	-	-	-	-	-	-	2	-
CO 3	2	1	1	-	-	-	-	-	-	-	-	-	3	-
CO 4	3	2	2	-	-	-	-	-	-	-	-	-	2	-
CO 5	3	2	1	-	-	-	-	-	-	-	-	-	3	-

Text Books:

1. Digital Processing of Speech Signals – L. R. Rabiner and R. W. Schaffer, PHl, 2004.
2. Digital Processing of Speech Signals. L.R Rabinar and R W Jhaung, 1978, Pearson Education.
3. Digital Audio Signal Processing – Udo Zolzer, 2nd Edition, Wiley.

Reference Books:

1. Speech Processing and Synthesis Toolboxes – Donald G. Childers, John Wiley & Sons, 1999.
2. Fundamentals of Speech Recognition – L.R. Rabiner and B. H. Juang, PHI, 1999.
3. Speech Communications: Human & Machine – Douglas O'Shaughnessy, IEEE Press, 2^{nd} Edition, 1999.
4. Discrete-Time Speech Signal Processing: Principles and Practice - Thomas F. Quatieri, 1^{st} Edition, PHI.