NIRMA UNIVERSITY
SCHOOL OF TECHNOLOGY, INSTITUTE OF TECHNOLOGY
B.Tech. Electronics & Communication Engineering
Semester - VI
Department Elective II

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Course Code	2ECDE01
Course Title	Speech and Audio Signal Processing

Course Outcomes (COs):
At the end of the course, the students will be able to
1. Comprehend the speech production and hearing models.
2. Design and apply models for speech and audio signal processing.
3. Apply speech coding, speech enhancement and speaker recognition algorithms for speech and audio processing.
4. Implement the methods for speech enhancement and speech coding for speech signals.

Syllabus: **Teaching Hours:45**

UNIT 1: Introduction **06**
Introduction, Anatomy and physiology of speech production, categorization of speech sounds, Prosody, Parameters of Speech: Pitch and Formants.

UNIT II: Analysis and Synthesis of Speech and Audio signals **12**
Spectral Analysis Models, Linear Predictive Coding Model for Speech Recognition, The autocorrelation method, The covariance method, Short-Time Fourier Transform Analysis and Synthesis, Short-Time Fourier Transform Magnitude, Filter Bank Summation method, Overlap-Add method.

UNIT III: Frequency Domain Pitch Estimation **08**
A correlation-based Pitch Estimator, Pitch Estimation based on Comb Filter, Pitch Estimation based on a Harmonic Sine wave Model.

UNIT IV: Speech Coding **06**
Vector Quantization, Frequency-Domain Coding, Model-based Coding.

UNIT V: Enhancement of Speech and Audio Signals **07**
Spectral subtraction, Cepstral Mean Subtraction, Wiener Filtering.

UNIT VI: Speaker Recognition **06**
Spectral Features required for Speaker Recognition, Minimum Distance classifier, Gaussian Mixture Model.

Self-Study:
The self-study content will be declared at the commencement of the semester. Around 10% of the questions will be asked from self-study content.

Assignments:
The students will be given 8- 10 programming/simulation/ projects assignments based on the above syllabus as follows:

i. Analysis and Synthesis of Speech and Audio signals
ii. LPC Model for Speech Signal
iii. Pitch Estimation Algorithm
iv. Speaker Recognition Algorithm
v. STFT Analysis of Speech and Audio Signals
vi. Speech and Audio Compression Algorithm
vii. Enhancement of Audio and Speech signal
viii. Speech Coding Algorithm

ix. Speech Recognition Algorithm
x. Adaptive filtering for Speech and Audio Signal

Suggested Readings:

1. T.F. Quartieri, Discrete-Time Speech Signal Processing: Principles and Practice, Prentice Hall
2. L.R.Rabiner, R.W.Schafer, Theory and Applications of Digital Speech Processing, Prentice Hall
3. B. Gold, N. Morgan, D. Ellis, Speech and Audio Signal Processing: Processing and Perception of Speech and Music, Wiley-Blackwell
4. T. Dutoit, F. Marqués, L.R. Rabiner, Applied signal processing: a MATLAB-based Proof of Concept, Springer
5. Ian Vince Mcloughlin. Speech and Audio Processing: A MATLAB-based Approach, Cambridge University Press

L = Lecture, T = Tutorial, P = Practical, C = Credit