3rd & 4th Year
Dr. Bhaskar

NATIONAL INSTITUTE OF TECHNOLOGY, TIRUCHIRAPPALLI

COURSE OUTLINE TEMPLATE			
Course Title	Digital speech processing (Elective course) (3rd and 4th years)		
Course Code	ECPE20	No. of Credits	3
Department	B.Tech, ECE	Faculty	Dr. E. S. Gopi
Pre-requisites Course Code	ECPC15		
Course Coordinator(s) (if, applicable)	Nil		
E-mail	esgopi@nitt.edu	Telephone No.	9500423313
Course Type	☐ Core course	☑ Elective course	

COURSE OVERVIEW

Speech production model-1D sound waves-functional block of the Vocal tract model –Linear predictive co-efficients (LPC) -Auto-correlation method-Levinson-durbin algorithm-Auto-co-variance method-Lattice structure-Computation of Lattice co-efficient from LPC-Phonetic Representation of speech-Perception of Loudness - Critical bands – Pitch perception – Auditory masking.

Feature extraction of the speech signal: Endpoint detection-Dynamic time warping- Pitch frequency estimation: Autocorrelation approach- Homomorphic approach-Formant frequency estimation using vocal tract model and Homomorphic approach-Linear predictive co-efficient -Poles of the vocal tract-Reflection co-efficient-Log Area ratio.

Cepstrum- Line spectral frequencies- Functional blocks of the ear- Mel frequency cepstral co-efficients- Spectrogram-Time resolution versus frequency resolution-Discrete wavelet transformation.

Pattern recognition for speech detection: Back-propagation Neural Network-Support Vector Machine- Hidden Markov Model (HMM)-Gaussian Mixture Model(GMM) -Unsupervised Learning system: K-Means and Fuzzy K-means clustering - Kohonen self-organizing map-Dimensionality reduction techniques: Principle component analysis (PCA), Linear discriminant analysis (LDA), Kernel-LDA (KLDA), Independent component analysis (ICA).

Non-uniform quantization for Gaussian distributed data- Adaptive quantization-Differential pulse code modulation- Code Exited Linear prediction (CELP)-Quality assessment of the compressed speech signal Text to Speech (TTS) analysis –Evolution of speech synthesis systems-Unit selection methods - TTS Applications.

COURSE OBJECTIVES

The purpose of this course is to explain how DSP techniques could be used for solving problems in speech communication.

COURSE OUTCOMES (CO)

Course Outcomes	Aligned Programme Outcomes (PO)
At the end of the course student will be able to, CO1: illustrate how the speech production is modeled	PO1

CO2:summarize the various techniques involved in collecting the features from the speech signal in both time and frequency domain.	PO1, PO11
CO3:summarize the functional blocks of the ear	PO1
CO4:compare the various pattern recognition techniques involved in speech and speaker detection	PO1,PO11
CO5: summarize the various speech compression techniques	PO1
PO1: Graduates of Electronics and Communication Engineering Programme will have the ability to apply the knowledge on Mathematics, Science and Engineering concepts in Complex engineering problems.	
PO11:To apply engineering & management principles in their own / team projects in Multidisciplinary environment.	

COURSE TEACHING AND LEARNING ACTIVITIES,

Week	Topic	Mode of Delivery
1	Introduction to speech production model, 1D sound waves	Slide presentation and chalk and talk method
2	Vocal tract model, Computation of LPC, Autocorrelation model	Slide presentation and chalk and talk method
3	Levinson- durbin algorithm, Auto co-variance method, Lattice structure, Computation of Lattice co-efficient from LPC	Slide presentation and chalk and talk method
4	Phonetic representation of speech, perception of loudness, Critical bands, pitch perception, Auditory masking	Slide presentation and chalk and talk method
4	Feature extraction of the speech signal :Endpoint detection, Dynamic time warping, Homomorphic filtering	Slide presentation and chalk and talk method
5	LPC, poles of the vocal tract, Pitch frequency, Formant frequencies, Line spectral frequency	Slide presentation and chalk and talk method
6	Cestrum, Spectrogram, Discrete wavelet transformation	Slide presentation and chalk and talk method
7	Pattern recognition for speech detection: Backpropagation Neural network, Support Vector Machine	Slide presentation and chalk and talk method
8	Hidden Markov model, Gaussian Mixture model	Slide presentation and chalk and talk method
8	K-means algorithm and Fuzzy k-means algorithm, K-SOM	Slide presentation and chalk and talk method
9	Dimensionality reduction techniques, PCA, LDA, KLDA, ICA	Slide presentation and chalk and talk method
10	Speech compression techniques: DPCM,CELP	Slide presentation and chalk and talk method
10	Speech compression techniques: TTS, TTS applications	Slide presentation and chalk and talk method

COURSE ASSESSMENT METHODS				
S. No.	Mode of Assessment	Week/Date	Duration	
1	Quiz 1	August last week	1 hour	15%
2.	Quiz 2	October first week	1 hour	15%
3.	Matlab based Assignments	Continous assessment	-	10%
3	Audio slide preparation	First week of November	-	10%
4	End semester exam	Second week of Novemeber	3 hours	50%

ESSENTIAL READINGS : Textbooks, reference books Website addresses, journals, etc

1. *L.R.Rabiner and R.W.Schafer," Introduction to Digital speech processing",now publishers USA,2007*
2. *E.S.Gopi,"Digital speech processing using matlab", Springer,2014.*
3. *Recent literature in Digital speech processing*

COURSE EXIT SURVEY

1. **Self-assessment feedback by the students**
2. **Overall performance of the students in the assessment.**

COURSE POLICY (including plagiarism, academic honesty, attendance, etc.)

[1] Copying is strictly not allowed for submitting the project audio slide. However discussion with the peers is allowed.
[2] Minimum attendance requirement is 75% to write the end semester exam.

[3] Other policy is as the institute norms.

ADDITIONAL COURSE INFORMATION

Interaction through piazza (www.piazza.com) is mostly encouraged. Or "Acadly"

FOR SENATE'S CONSIDERATION

Course Faculty ______ **CC-Chairperson** ______ **HOD** ______

(Dr.E.S.GOPI)