# PES INSTITUTE OF TECHNOLOGY

## DEPARTMENT OF ELECTRONICS AND COMMUNICATION

## FINAL YEAR PROJECT SYNOPSIS

## VIII SEMESTER

## SESSION: JAN – JUN 2017


SUBMITTED BY:

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**Title: Design of a Real-time speech recognition system.**

## Introduction:

Speech is the principal source of communication among humans to show their ideas, feelings and thoughts to each other. Speech recognition technology has made it possible for computer to listen human voice commands and interpret human languages. Speech recognition is the process of converting a given input signal into sequence of words, by means an algorithm implemented as a computer program. Speech Recognition system allows a computer to identify the words that a person speaks into readable text. Speaker, or voice, recognition is a biometric modality that uses an individual's voice for recognition purposes. (It is a different technology than "speech recognition", which recognizes words as they are articulated, which is not a biometric.) The speaker recognition process relies on features influenced by both the physical structure of an individual's vocal tract and the behavioral characteristics of the individual.

## Problem statement:

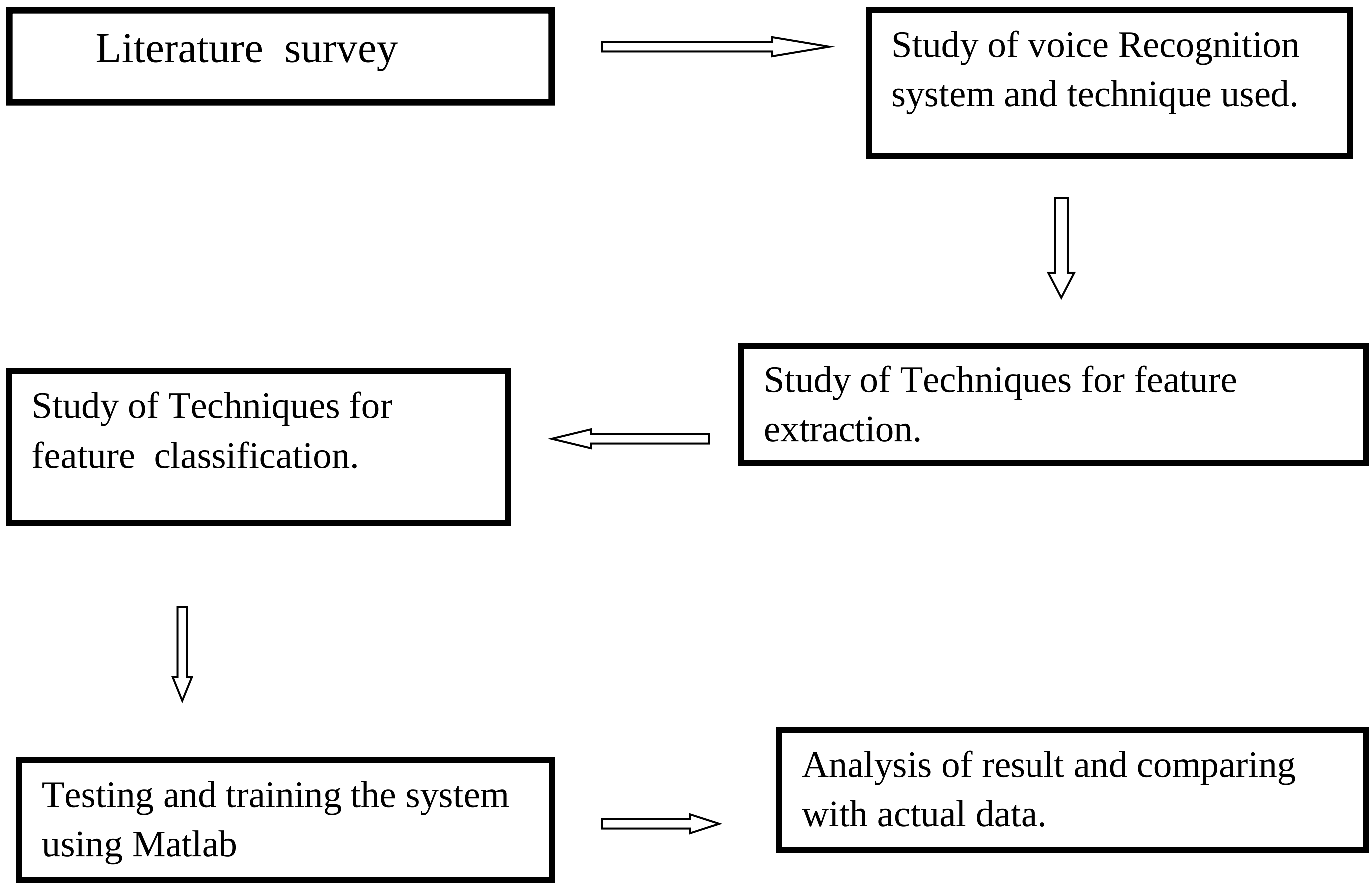
The problem of recognizing speech with the help of a computer is a difficult problem, and the reason for this is the complexity of the human language. Problems includes such as to the problem of natural language understanding , background noise, changes in pronunciation according to whether words are spoken in isolation or in sentence and in variations of accent between individuals, grammatical mistakes , Speaker variability, Amount of data and search space.

## Objectives:

The objective of the project is

1. To develop an automatic speech recognizer for English language.
2. To critically review literature related to Speech recognition. To identify speech corpus elements exhibited in English language.
3. To implement an isolated word speech recognizer that is capable of recognizing and responding to speech.
4. To train the above developed system in order to make it speaker independent.
5. To develop real time speaker recognition system using features extraction algorithms.

## Block diagram:




## Methodology:

1. <u>End-point detection of speech signal:</u> The implementation for the speaker verification system first addresses the issue of finding the endpoints of speech in a waveform. The algorithm finds the start and end of speech in a given waveform, allowing the speech to be removed and analyzed. The endpoint detection algorithm is used here. This algorithm gives the entire region where speech exists in an input signal.

2. <u>Features Extraction:</u> The task of the acoustic front-end is to extract characteristic features out of the spoken utterance. The regular frontend includes among others, the following algorithmic blocks: Fast Fourier Transformation (FFT), calculation of logarithm (LOG), the Discrete Cosine Transformation (DCT) and sometimes Linear Discriminate Analysis (LDA). Widely used speech features for auditory modeling are cepstral coefficients obtained through Linear Predictive Coding (LPC). Another well-known speech extraction is based on Mel-frequency Cepstral Coefficients (MFCC).

3. Features classification: It provides simulation of information processing analogues to human nervous system. Multilayer feed forward network with back propagation algorithm is the common choice in classification and pattern recognition. Hidden Markov Model, Gaussian Mixture Model, Vector Quantization are the some of the techniques for acoustic features to visual speech movement.

## Software and Hardware Requirement:

Software: MATLAB 2016

## Literature Survey:

1)Deep learning approaches to problems in speech recognition, computational chemistry, and natural language text processing By George Edward Dahl A thesis submitted in conformity with the requirements for the degree of Doctor of Philosophy Graduate Department of Computer Science University of Toronto
2)Feature Extraction and Classification Techniques for Speech Recognition: A Review Nidhi Desai1, Prof.Kinnal Dhameliya2, Prof.Vijayendra Desai3 1M.Tech. [Electronics and Communication] Student, Department Of Electronics and Communication Engineering, C.G.P.I.T, Bardoli, Gujarat 2Asst.Professor, Department Of Electronics and Communication Engineering, C.G.P.I.T, Bardoli, Gujarat 3Asst.Professor, Department Of Electronics and Communication Engineering, CKPCET, Surat, Gujarat
3)Real Time Speaker Recognition System using MFCC and Vector Quantization Technique
Roma Bharti Mtech, Manav rachna international university Faridabad
Priyanka Bansal Mtech Manav rachna international university Faridabad
4)Machine Learning Paradigms for SpeechRecognition: An Overview Li Deng, Fellow, IEEE, and Xiao Li, Member, IEEE
5)Speech Recognition Using Deep Learning Algorithms Yan Zhang, SUNet ID: yzhang5 Instructor: AndrewNg
6)Speech-To-Text Conversion (STT) System Using Hidden Markov Model (HMM) Su Myat Mon, Hla Myo Tun
7)Speech-to-Text and Speech-to-Speech Summarization of Spontaneous Speech Sadaoki Furui, Fellow, IEEE, Tomonori Kikuchi, Yousuke Shinnaka, and Chiori Hori, Member, IEEE
8) A New Silence Removal and Endpoint Detection Algorithm for Speech and Speaker Recognition Applications *G. Saha, Sandipan Chakroborty, Suman Senapati* Department of Electronics and Electrical Communication Engineering Indian Institute of Technology, Khragpur, Kharagpur-721 302, India
9)Speaker Recognition Using MFCC and Vector Quantisation Geeta Nijhawan 1, Dr. M.K Soni 2
10)D.A. Reynolds, "Experimental evaluation of features for robust speaker identification," IEEE Trans. Speech Audio Process., vol. 2(4), pp. 639-43, Oct. 1994.