# Online Speech Recognition System

Nishit Chaturvedi, Jijo Eype, Raveeraj Dhage, Saurabh Bathe, Prof.Amol Dumbare
Student, Student , Student, Student, Assistant Professor
Department of Computer Engineering,
Pimpri Chinchwad college of Engineering and Research,Ravet, Pune, India

***Abstract:***
Online speech recognition system is an important advancement in the field of automatic speech recognition systems. Online Speech Recognition System is one of the applications of Natural Language Processing. The study on various technologies have been there in the research for more than five decades like the Hidden Markov Model (HMM),Gaussian Mixture Models and feature extraction models like Linear Predictive Cepstral Coefficient(LPCC).This System accepts a speech input from the microphone and the speech captured through microphone is compared with database containing phrases, words and phonemes and the matched speech is converted to text and displayed on the text field using an online extension system.

***IndexTerms:*** HMM, LPCC, NLP, Automatic Speech Recognition

## I. Introduction

The ultimate dream of speech recognition is to enable people to effectively interact with the modern computer systems and to provide a basic human-computer interaction with simple techniques and tools. Some of these typical applications include voice dialling, call routing, data entry and dictation, command and email writing with speech aid. With the help of an online speech recognition system the speech will be automatically converted to text with the Hmm and Lpcc algorithms and provide the users with hectic free solution to the speech processing.

## II. Layout of Paper

The paper is divided into following parts viz, Introduction, Layout of paper, Existing system, Proposed system Architecture Diagram, Mathematical Model, Technologies and Algorithms used, Acknowledgements ,Future Work, Conclusion and References

## III. Existing System

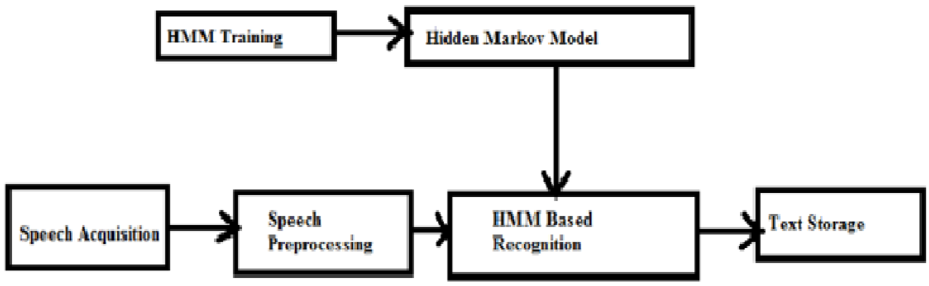
The Existing system provides solutions of speech conversion capabilities which can’t be used in an extension system and some of the algorithms used are neural networks and deep neural networks which is efficient but computation cost can be costly and even the algorithms are complex to design on current conditions.

## IV. Proposed System

The Proposed system is an online web-based Chrome extension system which uses the most popular Hmm method for speech processing along with Lpcc for database functionalities and with the help of a microphone the speech from the user is accepted and the taken speech is converted to text using above algorithms.

## 1. Architecture Diagram:

The proposed system in preventing the accidents provides the following functionalities




1. A standard microphone for capturing the speech of the user through online extension.
2. A Javascript and Ajax based front-end implementation for displaying the functionalities to start and stop speech functions through extension system with Hmm states.
3. A Php and MySql based back-end to store compare and retrieve words from the database using Lpcc capabilities.
4. After comparing the words from the database the most appropriate words based on Hmm states are displayed on the text field specified.

### 1.1 Mathematical Model

System Description:

**Input:**
Speech from user through microphone.

**Process :**
Process1 : User Sends the Speech to Online Extension System
Process2 : The Accepted Speech is processed using Hmm observations and states.
Process3 : The Outputted words through Hmm probability is compared with the database.
Process4 : Using Lpcc process final words to be displayed are selected.

**Output:**
O=The Speech recognized is converted to text

## V. Technologies and Algorithms Used

### 1. Hidden Markov Model Algorithm:

Hidden Markov Model algorithm is a statistical Markov model in which the system being modelled is assumed to be a Markov process with hidden states. In Markov Model the state is directly visible to the observer, and therefore the state transition probabilities are the only parameters, while in the Hidden Markov model, the state is not directly visible, but the output dependent on the state is visible. The task is to compute, given the model's parameters and a sequence of observations, the distribution over hidden states of the last latent variable at the end of the sequence.

### 2. LPCC

It is a variant of linear system and is abbreviated as Linear Predictive Cepstral Coefficients.
Speech systems developed based on these features have achieved a very high level of accuracy for speech recorded in a clean environment without the interruption of noise and outliers. The features extracted from database using the energy values of linearly arranged words or phrases, equally emphasize the contribution of all frequency components of a speech signal.

## VI. Acknowledgements

I would like to take this opportunity to thank my internal guide Prof.Amol Dumbare for giving me all the help and guidance I needed. I am really grateful to them for their kind support. Their valuable suggestions were very helpful. I am also grateful to Prof. Mahendra Salunke, Head of Computer Engineering Department, PCCOER for his indispensable support, suggestions. In the end our special thanks to Principal Dr. H.U. Tiwari for providing various resources such as laboratory with all needed software platforms, continuous Internet
connection, for Our Project.

## VII. Future Work

The Proposed system is an online web-based extension system for converting the speech to text and in future we will be extending this system to work for multiple browsers as currently it supports only Google Chrome browser and we may replace the current Hmm algorithm with more generalized algorithm based on deep neural networks.

## VIII. Conclusion

The proposed system is developed to convert the user given speech to text using mainly two important algorithms namely Hmm and Lpcc and as it is an online extension system it is compatible to be used in almost all websites and it will be useful in many applications including voice enabled email, voice enabled searching etc.

## References


1. Prachi Khilari, Bhope V. P, " A Review On Speech To Text Conversion Methods," International Journal of Advanced Research in Computer Engineering & Technology (IJARCET) Volume 4 Issue 7, July 2015.
2. M.A. Anusuya, S.K. Katti, "Speech Recognition by Machine: A Review," International Journal of Computer Science and Information Security, vol. 6, no. 3, pp. 181-205, 2009.
3. Nidhi Desai, Kinnal Dhameliya, Vijayendra Desai, "Feature Extraction and Classification Techniques for Speech Recognition: A Review, " International Journal of Emerging Technology and Advanced Engineering, vol. 3, no. 12, December 2013, ISSN 2250-2459.
4. M. A. Anusuya, S. K. Katti, "Front end Analysis of Speech Recognition: A review," International Journal of Speech Technology, Springer, vol. 14, pp. 99-145, 2011.
5. S. Yadav Kavita, M.M. Mukhedkar, "Review on Speech Recognition," International Journal of Science and Engineering, vol. 1, no. 2, pp. 61-70, 2013.