

Research Article

# Development of Marathi Speech based System for Human Robot Interaction

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## Abstract

Now a days various smart applications and devices are present having English voice as input but English is not a local language in Maharashtra. The English illiterates are not efficiently communicates with a system because of language barrier. Our system overcomes this problem and found solution for Marathi speaking people all over world. Proposed system is composed of hardware components like HM2007 and Raspberry Pi which are essential for voice recognition and synthesis of Marathi language. With the help of the motors and Ultrasonic sensor the system will detect obstacle and communicate with the user. We have also used the opensource software like Festival and Festvox for generating the voice from Marathi text. We will also analyse the voice sample efficiency of user given input. So this system will help the Marathi peoples for executing their task in Marathi language.

**Keywords:** Raspberry pi, hm2007, motor, l239d, voice recognition, text-to-speech, marathi speech synthesis, python

## Introduction

Since ten decades, research in Automatic Speech Recognition System has made significant development such as Microsoft Translator, Google voice search, Apple's Siri etc. This system is now used with great interest in both industry and academia. [1]. In voice recognition and communication system, speech recognition is actively considered, and it has tremendous progress due to digital signal processing. Any speech recognition system consists of three fundamental steps- pre-processing, feature extraction and classification. Surrounding and noise makes speech recognition system complicated as it affects a lot on performance and accuracy of the Automatic Speech Recognition System. Speech Recognition accuracy is used for performance measurement of speech recognition system [2]. Human uses a speech as one of the effective medium for communication. Speech is a digital signal made up of various components like time, amplitude and frequency. Because of this, in different frequency bands, transitions occur at different time. In simple word, we can define Automatic Speech Recognition (ASR) as a system which extracts, recognize and translate properties of speech using computer device. The purpose of the developing ASR system is to enable human real time interaction with a machine in natural language regardless of speech accent, speech noise, environment and size of vocabulary. ASR systems can be developed for recognizing isolated words, connected word and

continuous speech. ASR systems has various industrial and academia applications such as telecommunication, for impaired hearing people, learning language for children's etc. [3]. Speech is a multi-component signal with varying time, frequency and amplitude. Due to this variability, transitions may occur at different times in different frequency bands. The technology developed for extraction, recognition, and translation of the speech characteristics by using the smart computerized device is called an Automatic Speech Recognition (ASR). The main purpose of ASR is to develop a technology which helps human to interact with the machine in our natural language in real time environment regardless of vocabulary size, noise, speech characteristics or accent [51, 52]. ASR can be implemented for isolated word recognition, connected word recognition, and continuous speech recognition. This paper is divided into five sections. First section discusses introduction to Speech Recognition System, second section discusses overview of Automatic Speech Recognition System, in third section, literature review, fourth section discusses about ASR systems for Indian languages and paper is concluded in fifth section. Marathi is one of the most widely spoken language of the world (It is ranked between four and seven based on the number of speakers), with nearly 100 million native speakers. However, this is one of the most resource language which lack speech application to solve above problem we have proposed model which will be beneficial for the English illiterates blind people in Maharashtra.

Sr. No.	Title	Authors name	Year	Review
1	Minimum Prediction Residuals Applied to Speech Recognition	Itakura et. al.	1975	Proposed the idea of using LPC which is a data compressing method to speech recognition system.
2	Speech recognition using LPC analysis	Ostrander et. al.	1982	Developed English digit speech recognition system using MFCC as extraction method and HMM model.
3	Speech recognition using the probabilistic neural network	Low et. at.	1998	Describes about implementation of speech recognition system on a mobile robot for controlling movement of the robot. The methods used for speech recognition system are Linear Predictive Coding (LPC) and Artificial Neural Network (ANN). Back propagation method is used to train the ANN.
4	Voice recognition algorithms using mel-frequency cepstral coefficient (MFCC) and dynamic time warping (DTW) techniques	Muda et al.	2010	Has talked about two speech acknowledgment model which are vital in speech recognition system they are MFCC and DTW. The non-parametric strategy for demonstrating the human sound-related observation framework, Mel Frequency Cepstral Coefficients (MFCCs) is use as extraction strategies. The nonlinear grouping arrangement known as Dynamic Time Warping (DTW) presented by Sakoe Chiba has been utilized as components coordinating procedures.
5	Techniques for feature extraction in speech recognition system: A comparative study	Shrawankar	2013	Has discussed some features extraction techniques and their advantages and disadvantages. Some hybrid feature extraction methods discussed in the paper are RASTA-PLP, MFCC-LPC and HFCC-E. MFCC feature requires less computational time and it is robust for noise free environment.

The proposed system can be modified as per the user. Many researchers have been done experimentation in the research for getting expressive speech synthesis. Proposed model will sort out the problems of English illiterates and also for the blind Marathi speaking people in all over the world. The main idea behind the project is to make use of available

## Proposed Methodology

### A. Architecture:

#### 1. Machine Learning

Machine learning deals with the development of algorithms, which helps machine to learn by inductive inference which is based on the observations data which represent incomplete information of the statistical phenomenon. Classification, also known as pattern recognition, is a very important task in Machine Learning, with this machine "learn" to

recognize complex pattern automatically, it distinguishes between exemplars which are based on their different patterns, and makes intelligent decision. A pattern classification task has three modules namely data representation (feature extraction) module, feature selection or reduction module, and classification module. The first module aims to find invariant features which describes the difference in the classes. These second module of feature selection and feature reduction is used to decrease down the dimensionality of the feature vectors for classification. The classification module finds the actual mapping between the pattern and label based on features. The objective of the present work is to investigate the machine learning method in the application of automatic recognition of emotional state from human speech. Different Machine Learning Algorithms are based on the input available at the time of training: (a) Supervised learning algorithms are trained on labelled examples, i.e., input where the desired output is known.

The supervised algorithm generalizes functions or mappings from input to outputs, thus it can be used to hypothetically generate an output for previous unseen inputs. (b) Unsupervised learning algorithms operate on unlabelled examples, i.e., inputs where the desired outputs are unknown. Here the objective is to discover structure in the data (e.g. through a cluster analysis), not to generalize a mapping from input to output. (c) Semi-Supervised learning combines labeled as well as unlabeled examples to generate a suitable function or classifier.

**B. Algorithm**

There are two basic steps in any supervised learning system, training and testing. For any speech recognition system basic steps in training and testing are almost common and step in this are preprocessing, feature extraction and classification.

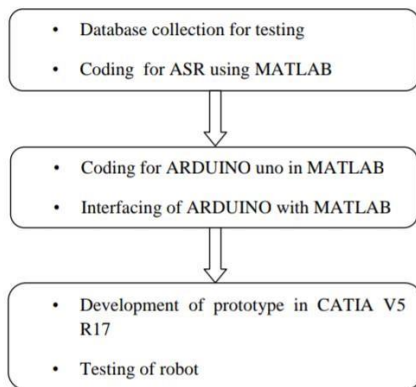


Figure 3.1: General outline of the work.

The general overview of the development of speech controlled wheel chair robot is described in the figure 3.1. At first, speech samples of English commands (right turn, left turn, start and stop) were collected using android mobile phone with 8 kHz sampling frequency. Virtual testing and training speech recognition environment was developed in MATLAB 13b version. In this proposed model there are two phases training and testing, in the training phase, training speech samples collected from the volunteers are processed and stored in the database. And in testing phase unrecognized voice input is processed and recognized using KNN and PNN classifier. Training and testing phases are shown in the figure

3.2. First the speech signal is pre-processed using first order FIR low pass filter and then framing and windowing is applied on speech signal, detail process is explained in latter part of the chapter. Voiced and unvoiced part is separated using prosodic features (Zero crossing rate, short term energy), then derived features (MFCC, LPC) are extracted from the signal. Features are quantized and the values are stored in database. In testing all the above mentioned steps are followed and the quantized data is compared with the database using KNN/PNN classifier for recognition. This information of recognized voice is given to

Arduino board using MATLAB Arduino interface. Arduino gives signal to L293D motor driver controller according to the programming done for different directions

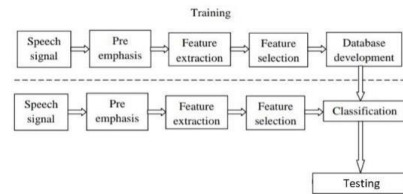


Figure 3.2: Speech Recognition System.

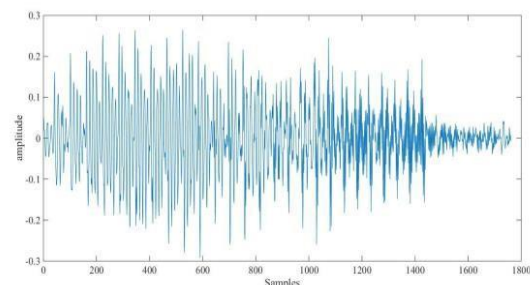
In the proposed work, we have used four possible combinations of isolated word recognition algorithm. Table 1 contains the details of the four combinations. In combination 1 speech input from the sample file is pre-processed after that LPC feature are extracted then features are classified using KNN classifier. In combination 2 all steps are same as in combination 1, only in step 3 12 MFCC features are extracted. In combination 3 all steps are same as combination 1 except step 5, in this PNN classifier is used instead of KNN. In combination 4 all steps are same as combination 3 except step 3, in this MFCC is used instead of LPC. These four combinations are tested for speech samples from database.

Table 3.1 various combination of feature for isolated word recognition system.

Combination	Step 1 Input speech signal	Step 2 Pre-processing	Step 3 Feature extraction	Step 4 Feature selection	Step 5 Classification
1	Audio sample file	Low pass filter (FIR)	Linear prediction coding(LPC)	Split vector quantization	KNN classifier
2	Audio sample file	Low pass filter (FIR)	Mel-frequency cepstrum coefficient(MFCC)	Split vector quantization	KNN classifier
3	Audio sample file	Low pass filter (FIR)	Linear prediction coding(LPC)	Split vector quantization	PNN classifier
4	Audio sample file	Low pass filter (FIR)	Mel-frequency cepstrum coefficient(MFCC)	Split vector quantization	PNN classifier

**Result and Discussion**

Automatic Speech Recognition (ASR) system is becoming the most important interface technique in the area of Robotics for the Socially Assistive Robots (SAR). Incorporation of ASR in wheel chair will increase comfort and simplify the controls. So, in the system speech signal is given through microphone to the MATLAB based program, it is executed and following outputs are obtained. For the command "left" output of digital FIR low pass filter is as shown in figure 4.1.





The signal is then segmented using zero crossing rate and short-term energy the figure 4.1 shows the segmented and original signals. 14 MFCC and 12 LPC feature vector as collected from the speech samples for feature classification. These features are quantized using split vector quantization to compress the data and efficiency are increased.

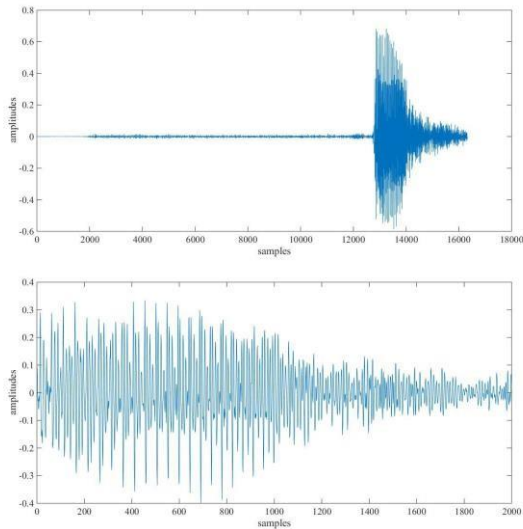


Figure 4.2: (a) Original signal and (b) voiced signal.

Table 4.1: Percentage accuracy of different combinations of algorithms. Sl. No. Combinations Efficiency (%)

1	Combination 1	72.5
2	Combination 2	78.75
3	Combination 3	76.25
4	Combination 4	83.75

In the above-mentioned table 4.1 accuracies of different proposed combinations are listed, the readings show that the combination 4 with MFCC feature extraction method and PNN classifier gives the maximum accuracy amongst the all combinations with 83.75 %.

LPC features which are extracted for speech recognition contains 12 coefficients for any speech sample. These 12 coefficients for different commands like left, right, start and stop are shown in figures 4.3, 4.4, 4.5 and 4.6 respectively.

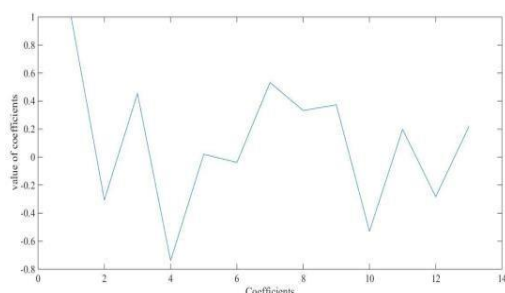


Figure 4.3: LPC for command left

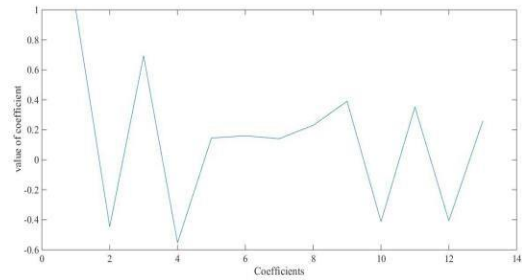


Figure 4.4: LPC for command right

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