A Smart Texting System For Android Mobile Users

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Abstract— Texting that is SMS is the important function of any Mobile phone and we know that the mobile phone usage in World is spreading rapidly and has gone through great changes due to new developments and innovations in mobile phone technology. This paper based on creating a smart texting system for mobile user because it convert voice into text.. In other words, messages can be voice/speech typed. In this paper we will make use of a dictating-machine prototype for the English language, which recognizes in real time natural-language sentences built.. A speech to text converter is developed to send SMS .It is found that large-vocabulary speech recognition can offer a very competitive alternative to traditional text entry.

Keywords: Short Message Service (SMS); speech acquisition; Hidden Markov Model (HMM); HMM-based recognition.

I. INTRODUCTION

The space and time complexity is very much important because of the short memory provided by the mobile system. There is no doubt that more and more Mobile phone users are using short message service (SMS) instead of making voice calls. In order to satisfy the needs and demands of users, mobile phone manufacturers are constantly adapting and innovating to ensure that they can survive in this competitive market. An important innovation in SMS technology lately is the speech recognition technology that can convert voice messages into text messages. In other words, messages can be voice/speech typed. Currently, voice messages can only be converted into text messages in the form of normal/standard text using fully spelled words.

Now let's limit our focus towards short massage system it is text messaging service component of phone, using standardized communications protocols that allow the exchange of short text messages between mobile phone devices. SMS text messaging is the most widely used data application in the world, with 2.4 billion active users, or 74% of all mobile phone subscribers.

The cell phones are very important part of modern life. Many of us need to make a call or massage at anytime from anywhere. Many of them needs their cell phones when they can't do so e.g. At the time of driving, cooking accidents may occur because of this activity an speech to text converter for mobile design for this purpose so to avoid accidents. The study of speech to text conversion is from 1970s where the first experiment of phoneme- to-grapheme conversion, this conversion consists of segmentation of phoneme string into word. This work is again extended to stenotype-to-grapheme conversion. Voice massaging is slowly and gradually reducing the importance of text massaging because it is safer to massage at the time of cooking and driving. This paper introduces an idea about the speech-to-text conversion for SMS application. This software enable user to send the SMS without using keypad with fully spelled word.

II. WHY ANDROID

Operating system have developed a lot in last 15 years. Starting from black and white phones to recent smartphones or mini computers, mobile OS has come far away. One of the most widely used mobile OS these days is **ANDROID**. **Android** is a software bunch comprising not only operating system but also middleware and key applications. After original release there have been number of updates in the original version of Android. It is the software stack of mobile devices. Android SDK provides the API's that is necessary to begin developing applications on the Android platform using the Java programming language. Android includes an embeddable browser built upon WebKit, the same open source browser engine powering the iPhone's Mobile Safari browser.
An Android application consists of one or more of the following classifications: 1)Activities: Is the application that has a visible UI is implemented with an activity. When a we selects an application from the home screen or application launcher, an activity is started.2)Services: A service should be used for any application that needs to persist for a long time, such as a network monitor or update-checking application. 3)Content providers: We A content provider's job is to manage access to persisted data, such as a SQLite database. Suppose for the bigger system like Speech to text conversion system or one that makes data available to multiple activities or applications, a content provider is the means of accessing your data.4)Broadcast receivers: This is use to launched to process a element of data or respond to an event, such as the receipt of a text message. Following figure showing the Architecture of Android platform. The key application of the speech to text conversion system is recognition technology. Speech input adds another dimension to the mobile phones. The Google's Voice Search application, the is required is many times pre-installed on many Android devices and available in Android Market, and it provides powerful features. Now we can dictate our message instead of typing it. Just tap the new microphone button on the keyboard, and you can speak in just about any context in which you would normally type.

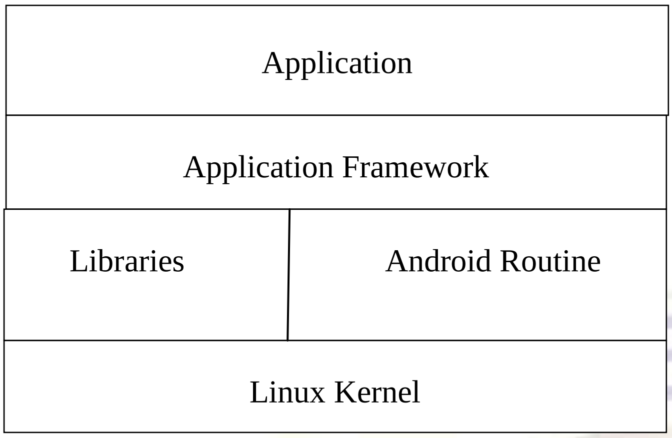

fig1: android architecture

III. DESIGN MODULE

This project converts speech into text. At the run time speech data is acquire from microphone and converted into text speech frames the speech frames are then pass for preprocessing and after preprocessing of the sample frames HMM-based training is applied on speech frame. Functionally the project divided into three modules

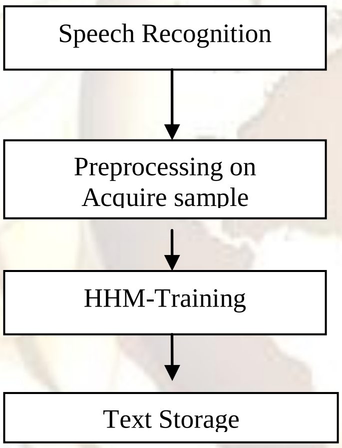

Fig:-1 Speech to text conversion system it is divided into three modules.

IV. SPEECH RECOGNITION

Speech recognition is very important part of this system in this phase speech samples are obtained from speaker at real time and stored for preprocessing. For speech recognition we require microphone to receive voice speech signals,

Speech acquisition can be easily done by the microphone present in the mobile phone, In the acquisition phase the different M/C is depends upon the its own configuration, hence there is need to store the sample of different users to make system more compatible to any type of voice.

To recognize the speech HMM-based automatic recognition was conducted. For continuous phoneme recognition, an 86% phoneme correct was achieved for the normal-hearing. To achieve speech preprocessing sphinx frame work is used this is the best tool found to acquiesce speech signals. Sphinx is design with high flexibility modularity. Recognition or pattern classification is the process of comparing the unknown test pattern with each sound class reference pattern and computing a measure of similarity (distance) between the test pattern and each reference pattern.. The digit is recognized using a maximum likelihood estimate, such as the Viterbi decoding algorithm, which implies that the digit whose model has the maximum probability is the spoken digit.

Preprocessing, feature vector extraction, and codebook generation are same as in HMM training. The input speech sample is preprocessed and the feature vector is extracted. Then, the index of the nearest codebook vector for each frame is sent to all digit models. The model with the maximum probability is chosen as the recognized digit.

Result:

Sr. No.	Voice	text
1	Speech	Nick/now is
2	Cute	Spite

Table 1:- showing output generated by acquisition module

V. SPEECH PREPROCESSING

The voice which is taken at the real time will require noise free speech signals now to reduce noise we need to consist of background noise that need to be removed. The preprocessing reduces the amount of efforts in next stages. Input to the speech preprocessing is speech signals which then converted into speech frames and gives unique sample

Steps:

1. The system must identify useful or significant samples from the speech signal. To accomplish this goal, the system divides the speech samples into overlapped frames.
2. The system performs checks for the voice activity using endpoint detection and energy threshold calculations.
3. The speech samples are then passed through a pre-emphasis filter.
4. The frames with voice activity are passed through a Hamming window. The system performs autocorrelation analysis on each frame.
6. The system finds linear predictive coding (LPC) coefficients using the Levinson and Durbin algorithm.

We apply a Hamming window to each frame to minimize signal discontinuities at the beginning and end of the frame.

VI. HMM TRAINING

An important part of speech-to-text conversion using pattern recognition is training. Training involves creating a pattern representative of the features of a class using one or more test patterns that correspond to speech sounds of the same class. A model commonly used for speech recognition is the HMM, which is a statistical model used for modeling an unknown system using an observed output sequence. The system trains the HMM for each digit in the vocabulary using the Baum-Welch algorithm. The codebook index created during preprocessing is the observation vector for the HMM model.

VII. CONCLUSION

This project demonstrate us the idea of smart texting system for mobile users. But speech recognition but the recognition fails because it requires preprocessing after acquiring of speech. The system requires training of voice because of the purpose of it should recognize the correct voice of that user. Suppose userA having this system in his mobile his voice should be train so as to get correct recognition of text.

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