VOICE BASED MESSAGE SCHEDULING ANDROID APPLICATION

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Abstract – Ever wish you could just speak your thoughts into a document instead of writing or typing them? Doing so may be easier than you think. Message scheduling technology has been around for decades in one form or another. It was made popular by technology companies such as IBM, the Department of Defense, and medical offices. Over the last few years it has become increasingly more popular with the general public and much more accurate at deciphering what we are saying. This has positively impacted everyone from commuters wanting to dial a number without looking at the keypad to individuals with disabilities needing to send an SMS.

Keywords – Scheduling, Short Message Service, Voice Recognition.

I. INTRODUCTION

Voice is the basic, common and efficient form of communication method for people to interact with each other.Today speech technologies are commonly available for a limited but interesting range of task. This technologyenables machines to respond correctly and reliably to human voices and provide useful and valuable services.As communicating with computer is faster using voice rather than using keyboard, so people will prefer suchsystem. Communication among the human being is dominated by spoken language, therefore it is natural forpeople to expect voice interfaces with computer.

This can be accomplished by developing voice recognition system:speech-to-text which allows computer totranslate voice request and dictation into text. Voice recognition system:speech-to-text is the process ofconverting an acoustic signal which is captured using a microphone to a set of words. The recorded data can beused for document preparation.

A number of voice recognition systems are available on the market. The most powerful can recognize thousands of words. However, they generally require an extended training session during which the computer system becomes accustomed to a particular voice and accent. Such systems are said to be speaker dependent. A speaker dependent system is developed to operate for a single speaker. These systems are usually easier to develop, cheaper to buy and more accurate, then but not as flexible as speaker adaptive or speaker independent systems. Speaker –dependent software works by learning the unique characteristics of a single person's voice, in a way similar to voice recognition. New users must first "train" the software by speaking to it, so the computer can analyze how the person talks. This often means users have to read a few pages of text to the computer before they can use the speech recognition software.

The main disadvantages of this system are as follows:

- Speaker Dependent
- Slow Recognition

This paper presents techniques for message scheduling and speech-to-speech automatic summarization based on speech unit extraction and concatenation. For the former case, a two-stage summarization method consisting of important sentence extraction and word-based sentence compaction is investigated. Sentence and word units which maximize the weighted sum of linguistic likelihood, amount of information, confidence measure, and grammatical likelihood of concatenated units are extracted from the speech recognition results and concatenated for producing summaries. For the latter case, sentences, words, and between-filler units are investigated as units to be extracted from original speech. These methods are applied to the summarization of unrestricted-domain spontaneous presentations and evaluated by objective and subjective measures. It was confirmed that proposed methods are effective in spontaneous speech summarization.

II. BACKGROUND WORK

Kuldip K. Paliwal and et al in the year 2004 had discussed that without being affected by their popularity forfront end parameter in speech recognition, the cepstral coefficients which had been obtained from linearprediction analysis is sensitive to noise. Here, the use of spectral subband centroids had been discussed by themfor robust speech recognition. They discussed that performance of recognition can be achieved if the centroidsare selected properly as in comparison with MFCC. to construct a dynamic centroid feature vector a procedurehad been proposed which essentially includes the information of transitional spectral information [1].

EsfandierZavarehei and et al in the year 2005, studied that a time-frequency estimator for enhancement of noisyspeech signal in DFT domain is introduced. It is based on low order auto regressive process which is used formodelling. The time-varying trajectory of DFT component in speech which has been formed in Kalman filterstate equation. For restarting Kalman filter, a method has been formed to make alteration on the onsets ofspeech. The performance of this method was compared with parametric spectral subtraction and MMSEestimator for the increment of noisy speech. The resultant of the proposed method is that residual noise isreduced and quality of speech in improved using Kalman filters [2].

Ibrahim Patel and et al in the year 2010, had discussed that frequency spectral information with mel frequency isused to present as an approach in the recognition of speech for improvement of speech, based on recognitionapproach which is represented in HMM. A combination of frequency spectral information in the conventionalMel spectrum which is based on the approach of speech recognition. The approach of Mel frequency utilizes thefrequency observation in speech within a given resolution resulting in the overlapping of resolution featurewhich results in the limit of recognition. In speech recognition system which is based on HMM, resolutiondecomposition is used with a mapping approach in a separating frequency. The result of the study is that there isan improvement in quality metrics of speech recognition with respect to the computational time and learningaccuracy in speech recognition system [6].

Puneet Kaur, Bhupender Singh and Neha Kapur in the year 2012 had discussed how to use Hidden MarkovModel in the process of recognition of speech. To develop an ASR (Automatic Speech Recognition) system theessential three steps necessary are pre-processing, feature Extraction and recognition and finally hidden markovmodel is used to get the desired result. Research persons are continuously trying to develop a perfect ASRsystem as there are already huge advancements in the field of digital signal processing but at the same timeperformance of the computer are not so high in this field in terms of speed of response and matching accuracy.The three different technique used by research fellows are acoustic phonetic approach, pattern recognitionapproach and knowledge-basedapproach [4].

ChadawanIttichaichareon and PatiyuthPramkeaw in the year 2012 had discussed that signal processing toolboxhas been used in order to implement the low pass filter with finite impulse response. Computationalimplementation and analytical design of finite impulse response filter has been successfully accomplished byperforming the performance evaluation at signal to noise ratio level. The results are improved in terms ofrecognition when low pass filters is used as compared to those process which involves speech signal withoutfiltering [3].

Geeta Nijhawan, Poonam Pandit and Shivanker Dev Dhingra in the year 2013 had discussed the techniques ofdynamic time warping and Mel scale frequency cepstral coefficient in the isolated speech recognition. Differentfeatures of the spoken word had been extracted from the input speech. A sample of 5 speakers has beencollected and each had spoken 10 digits. A database is made on this basis. Then feature has been extracted usingMFCC.DTW is used for effectively dealing with various speaking speed. It is used for similarity measurementbetween two sequence which varies in speed and time [5].

III.PROPOSED WORK

System Model

The system can be shown in the Fig-1 in which mainly has one entityi.e. Mobile User and these are considering main module of the system.

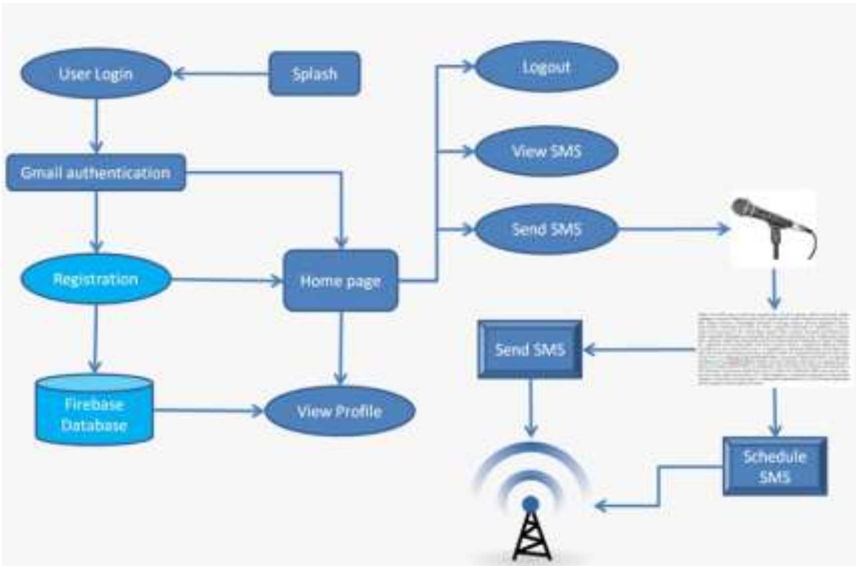

Fig. 1: System Overview

The system has following implementation modules:

Voice Recognition:

Speech Recognition (SR) is the translation of spoken words into text. It is also known as "automatic speech recognition", "ASR". Our system analyze the person's specific voice and use it to fine tune the recognition of that person's speech, resulting in more accurate transcription. The best part of the project is user no need to enter the message he want to send, you simply speak the message content then the voice will be converted into the text, In this module we getting voice input from the user and record that voice, after recording user voice, we will process the and recognize the text then show it to user.

Pick Suggestion

Even though User speak the English fluently, system can't recognize the voice 100% because each and every human having different speech speed and pitch, so system cant perfectly translate their voice into text, but it will match the user voice into many different texts and show that to the user to pick user desired output. If the desired text is not displayed in the suggestion, then user can close the suggestion and again speak.

Pick Contact

To send SMS we need the message content and the recipient who is going to get the message. So user must give the contact number of his choice to send SMS. He knows the number then he can directly enter the number into recipient textbox. If he can't remember the number and he saved it in the contacts. Then he can simply choose to pick contact option. If user chose to pick number from contact, all the available contact with their phone number will be displayed in list to the user. Then he can pick any one of the number to send SMS, picked number will be inserted in the recipient field.

Send SMS

In this module, the system will send the SMS to the recipient using SMS Manager in the Telephony Service. In this we have the recipient number and the message to be send. We will first create an object for SMS Manager in the Telephony service, using the method *sendTextMessage* system will sent the message. After sent SMS to desired recipient we need to update the message content to view and retry the sent SMS in the user SMS database.

The proposed system can be implemented as follows: In this proposed method, we implemented an android application to convert speech to text effectively and send SMS efficiently.In this application we schedule the SMS using user friendly Date and Time pickers.This allows users to schedule one SMS at a time.In this system we use firebase DB to Manage the SMS Sent information and user information.

IV. RESULTS AND DISCUSSION

In this system we developed mobile based Message Scheduling to reduce the food wastage in parties. The following screens show that our system is more users friendly and efficient.

Figure-2 Showing the menu page with some options are displayed when opens the application.

Fig. 2: Menu Page

Figure-3:Shows that the user can send the messages to their contacts clicking on mic.It converts the message speech to text.

Fig. 3: Speech Recognition

Figure-4:Shows that the user can schedule the messageswill deliveredto their contacts in a particular time and date.

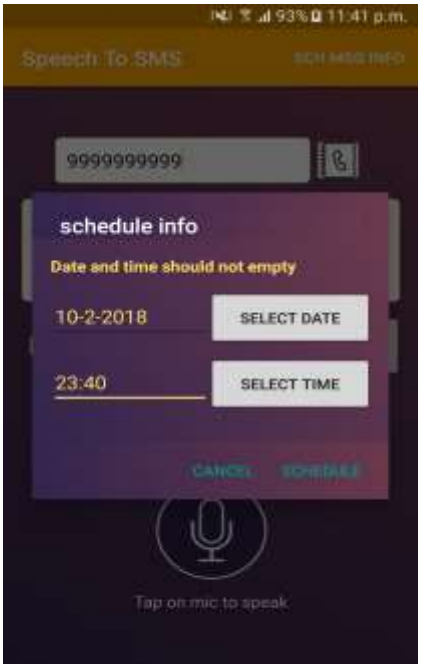

Fig. 4:Schedule Message

Figure-5: Showing the list of messages that are sent by the end user.

Fig. 5:

And the user can check the scheduled info of each message shown in Figure-6.

Fig. 6: Schedule Message Info

V. CONCLUSION

In this project, we implemented an android application "voice based sms system" in which the user can give the speech and the system convert the voice speech into text and then the user send the sms by picking the contact information manually. This effectively useful to the many user to send SMS. This system allows users to scheduling the SMS using user friendly Date picker and time picker.In the current system, we schedule one SMS at a time. In future we use the server system to many SMS scheduling at a time.

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