

Voice Input based Attendance System

Jeetvan Shah, Vikas Salunkhe, Jitendra Saturwar, Omkar Parab.



Abstract: System that marks attendance is software for students studying in different institutes which can also be used in companies the knowledge is stored in Admin device who takes attendance, the scholar user can give attendance in their respective classes before entering the category. This system will also be beneficial in calculating eligibility criteria of attendance of a student. The aim of developing such a system is to computerize the tradition way of taking attendance. Another purpose for developing this software is to use voice bio-metrics which give authentication of attendance using very small data size and generate the report at the top of the session with the help of automated system. The scope of the project is that the device on which the software is installed and therefore the voice authentication is completed, that is the system is developed as a Python application with firebase as a database, and it will work for a any teacher or faculty. Afterwards the program will be able to be embedded in a device.

Keywords: Authentication, Python, Eligibility, Voice bio-metrics.

I. INTRODUCTION

We as humans speak and hear one another through a verbal conversation or through a device. At present several attempts are made to create a technology during which humans and machine can interact in a manual way. Obviously such an interface would yield great benefits. Handwriting recognition was developed but has not fulfilled the main goal. But now attempts are made to develop vocally interactive computers to understand this dream. i.e. Computer which will give speech as output given textual input (speech synthesizer), and recognize speech which is given as input (speech recognizer). Most of the time speech recognizer recognizes speech, but this system cares who the speaker is and not what the speaker is saying.

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II. PROJECT IDEA

To get a perspective of how speaker recognition works it is desirable to possess knowledge of speech waves and what features of it is utilized in the evaluation process.

In humans, thoughts are being formed into sentences and specific words and therefore the various nerves control the texture of the vocal tract that is the tongue, mouth, vocal cords etc. To supply the specified sound. Each element resonates at a fundamental harmonics of it and thus have high energy at those frequencies. The harmonics at the first three levels are very high and are said to be formant frequencies. Each element has a singular fundamental and hence unique formant frequencies and it's this feature that permits the identification of every phoneme at the popularity stage.

III. EXISTING SYSTEM AND RESEARCH

Currently there is no such integrated system that can convert and which uses the idea of speaker recognition to mark attendance of a student. There are softwares available that are used to unlock and get access to a specific place. As compared to any automated systems, Automatic Speech Recognition (ASR) are going to be handy when its speed and efficiency is above the present method in order that savings are often made. But as mentioned above ASR systems haven't quite reached that elite position. On the other hand ASR systems is now more convenient than ever before. For example: Mac power secretary for the power mac, Dragon Dictate for windows (New electronics). And when speech-independent continuous speaker recognition systems are developed speaker recognition are going to be one among the most used methods of input and can cause the event of vocally protected computers. After creating such a software people will be able to use it as efficiently as any other software whether it be finger print based or facial recognition based. The teachers will have very less trouble in analyzing the data and it will be very helpful since students won't be able to mark any proxy attendance. There are few softwares that are based on voice recognition which is basically text to speech and then that specific text in use to do further process. Text to speech has gone on a new level and speaker recognition is coming into picture. Some technologies are available but it is on a prototype phase but as new techniques evolve, we should be able to see embedded system based on this method. People can adapt to it very quickly and it will be very understandable.

IV. METHODOLOGY (PROPOSED SYSTEM)

The requirement analysis to setup the Attendance System by using speech, must answer to the subsequent set of questions: what's output expectancy this system:



Output expected may be a secure attendance marking system which detects the Speech from users then send the attendance to teacher . The teacher user of the system has got to go online the system via a Log in then access to the report is given. In identification phase most of the calculations takes place in between the data of voice stored in the database with that of the present input of the individual.

The time of recognition depends on the limit of feature vectors, their respective dimensionality, the complexity of the data model of the individual and also the number of speakers. During this project, we specialize in achieving Real-Time talker identification supported GMM modeling method and compute the log-likelihood function to make a decision the one that is now speaking, and that we partition our project into two parts. The primary is training part and therefore the second is test part.

When executing the python file on instruction , the pc will ask speaker's name and training GMM model for them in any case speakers' voice are imported. And after training process completed, the pc can test the speaker who's now speaking. Using mel-frequency cepstral coefficients: the most important thing to know about speech input is that the sounds generated by an individual are filtered by the form of the vocal movement including the movement of tongue, lips etc. This shape determines what sound comes out. Packages of spectrum of short time power are made and therefore the job of MFCC is to accurately represent this package. In learning, we take out features of human voice and build models for every person. The elements we used here are called Mel-Frequency Cepstral Coefficients (Abbreviated as MFCCs) and therefore the model we build for each and every person is named gaussian mixture model. (Abbreviated as GMM). These two model train the data accurately and makes it ready for testing. The testing time is efficient and mostly depends on the hardware being used.

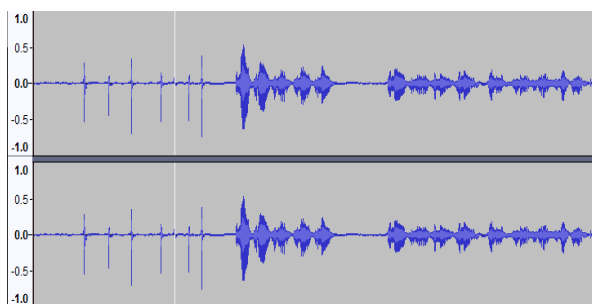


Fig.1 Input waveform

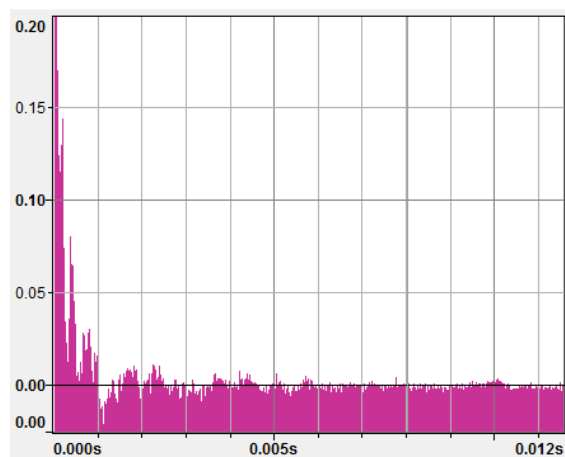


Fig.2 Cepstrum

```

In [1]: runfile('D:/text_independent/Text-Independent-Speaker-Identification-System-master/Code/main.py', wdir='D:/text_independent/Text-Independent-Speaker-Identification-System-master/Code')

Do you wanna test Now? [Y/N] Y
1
2
3
4
5
6
7
8
jee anand
    
```

Fig.3 Terminal output

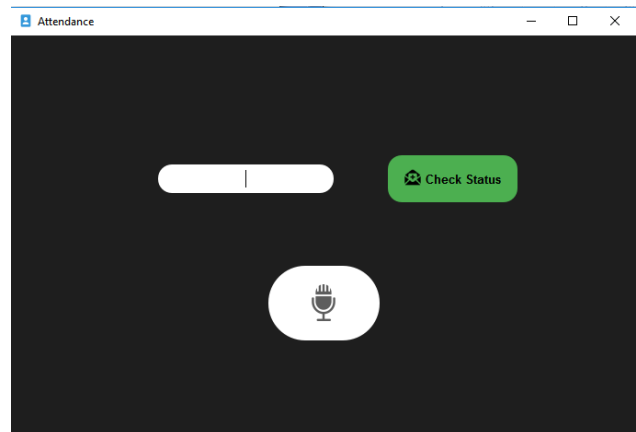


Fig.4 General UI

The Student has got to tap on the buttons shown in fig.4 to start out the backend process of the program. Once that's done the interface asks the scholar if he wants to check his voice, after that the scholar has got to represent his voice 10 seconds for complete accuracy. The system will take a while counting on the specs it has. If the voice matches that of the student's entry in the system then he will be marked as present. If it doesn't work then the system will ask the scholar to talk again.

Input of voice is taken through Audacity which is a tool used to record and export file in .wav format which is lossless to a great extent. It also helps in cleaning the input voice to remove unwanted noises which indeed helps in optimizing the solution. Keeping the test file as clean as possible is the main goal in doing all the mentioned above methods.

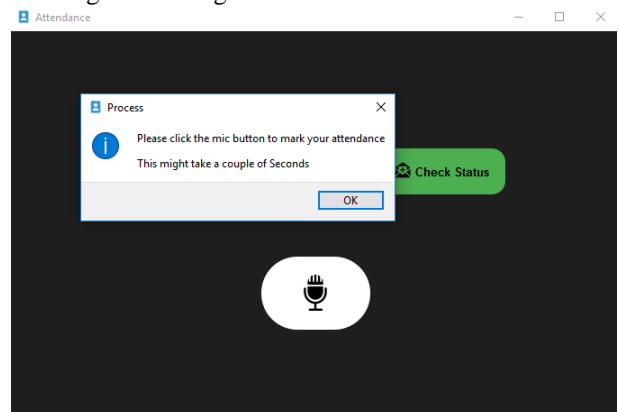


Fig.5 UI Alerts

[voice-6: ee8](#) > Student

Student

```

-- Jeff: 'Absent'
-- John: 'Absent'
-- Omkar: 'Present'
-- Vikas: 'Absent'
    
```

Fig.6 Database structure

After the attendance of a particular day has been registered in the admin database the data from each subject is reset to absent for the next day. If any student is having difficulty in marking the attendance they will be given three chance .But if all the three chances fail then they will not be allowed the mark the attendance because it will be assumed that the student is trying to mark someone else’s attendance and a notification will be shared to the teacher of the class.

V. RESULTS AND DISCUSSIONS

By using the above proposed system and the result there is a clear extraction of data from the voice input. Depending on the quality of the input taken during initial stage the accuracy changes. As of now the accuracy of this propose system is 80% which can be increased in the future. We can also develop a mobile application which will help in managing the time complexity. The program can be used in embedded systems by using python user interface builder that is the Qt software.

VI. CONCLUSION

The result expected from the proposed system promises an efficient, convenient and time-saving alternative for the students and teachers of schools, colleges etc. The proposed system is to reduce the effort for attendance, to save the time of taking attendance, to provide security for attendance. For these the system features and required research can be improved.

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