

Speech Based Answer Sheet Evaluation System

Sairaj Chittal¹, Rohini Sonawane², Asmita Patil³, Pankaj Mohite⁴, V.B Gaikwad⁵

[1] [2] [3] [4] Student, Dept. of Computer Engineering, Terna Engineering College, Nerul, Navi Mumbai, Maharashtra, India

[5] Associate Professor, Dept. of Computer Engineering, Terna Engineering College, Nerul, Navi Mumbai, Maharashtra, India

Abstract - Speech is the most effective form for interaction among humans. Human computer interface is the middle link which acts like an interpreter between the human user and the computer system. In this paper we present a direction in which the speech recognition technology can be used for evaluation of answer sheets. Real time answer sheet evaluation requires the examiner to manually evaluate the answer sheets and give marks accordingly. Our proposed system is an effort to minimize the drawbacks of the existing manual method of answer sheet evaluation, improve the performance, possibly prevent malpractices and to provide ease of usability over the existing system arrangement. Some concepts that are aimed to be used in this system are Hidden Markov's Model (HMM), Vector Quantization (VQ) as part of Google Speech API used for speech recognition and corresponding evaluation.

KEY WORDS: Speech recognition, answer sheet evaluation, HMM, VQ, ease of usability, malpractices

1. INTRODUCTION

Answer sheet evaluation is a major aspect of present day education system. Speech recognition is used to communicate with a particular machine using speech input. The speech recognition process is accomplished by converting the speech signal to a sequence of textual representation by way of various algorithms implemented. Modern day speech recognition systems are instantly capable of swearing out and understanding spoken language input for an ordination of up to thousands of words in real time operational environment.

Our aim is to develop a model system which can evaluate the scanned answer sheets through speech input provided by examiner. The marks and other evaluated data will be stored in appropriate tables in the database. The objective of this project is to make improvements over the existing system of paper evaluation by adding speech recognition functionality for evaluation. Our software system is aimed to correctly interpret and recognize the user speech input and accurately perform the evaluation process.

2. LITERATURE OVERVIEW

By the year 2001, the speech recognition by computer machines had a maximum accuracy of up to 70%. However,

this technology's progress seemed to be slowed down towards the remainder of the decade. Although recent research activities have contributed to substantial improvements in this area. Now various parallel processing methods which use combinations of Hidden Markov Model's (HMM's) and acoustic-phonetic approaches for the detection and correction of linguistic errors and for increasing the robustness of system for correct speech recognition even in a fairly noisy environment, have been discovered. The collection of these techniques together can be called as a Speech Recognition System.

SUMA SHANKARANAND describes the Enhanced Speech Recognition System as it consisted of two main modules namely, 1) Speaker Recognition and 2) Speech Recognition. She explains that the Speaker Identification feature extraction process, extracts data from the speech signal that is unique for each speaker. She further states that the Mel Frequency Cepstral Coefficient (MFCC) technique is practiced to create like a fingerprint i.e. a unique identification of the sound files. VQ Algorithm is used for feature extraction in both modules.[1]

PRACHI KHILARI describes the Automatic Speech Recognition (ASR) system. The ASR systems operate in two phases viz. 1) Training Phase and 2) Recognizing Phase. In the training phase, the system determines the patterns for presenting the different language sounds like words, phonemes and short phrases that are a function of the linguistic vocabulary. These patterns are stored in form of templates. In the recognizing phase, if an unknown pattern is identified by considering the set of references, it is stored for future referencing and processing. A proper database has to be created from the various domain words and syllables.[7]

3. EXISTING SYSTEM

The traditional system is a manual evaluation system where an examiner/evaluator grades a bunch of papers at hand manually. The chances of malpractices in this traditional system are more as examiner has direct access to answer sheet hardcopies. However, recent improvements have contributed to the use of a software arrangement for proper evaluation where an examiner sits in front of the computer system and grades the answer sheets which are in the configuration of scanned copies. In this existing system the evaluator has to manually enter the marks for evaluation of answer sheets in the software system. Even this system

requires more human efforts like more mouse clicks and also requires greater amount of constant precision of the examiner.

4. PROPOSED SYSTEM

To overcome the drawbacks of existing system we are proposing to construct a speech based answer sheet evaluation system that will use speech input of examiner for evaluation. The primary purpose here is to utilise the speech recognition technique to provide a better performance by the software system as well as have maximum level of transparency and ease of evaluation for the examiner, as compared to the existing system. The proposed system consists of two main actors, 1) Admin and 2) Examiner. Admin can add/remove the examiners and assign the subjects to the examiners. Admin has to upload the scanned answer sheets for the various subjects. Examiner can only evaluate the answer sheets of the subjects that he/she is assigned to. The software system is aimed to be able to correctly interpret and recognize the examiner's speech input via a microphone and accurately perform the evaluation and store the results in the corresponding tables in the database.

5. SCOPE

This software system can possibly be used for evaluating answer sheets of any domain. This scheme is in line with technical changes which are moving towards automation i.e. minimizing manual human efforts. Attempt to use maximum features provided by a computer system. Efficient utilization of speech recognition technology for the evaluation process. Possibility of improvement over traditional software system majorly in terms of ease of usability.

6. REQUIREMENTS

Hardware Specifications :

- Processor : AMD A4-6210 APU with AMD Radeon R3 Graphics.
- Installed memory (RAM) : 4 GB.
- System Type : 64-bit Operating System , x64 based processor
- Microphone

Software Specifications :

- Windows 10
- Netbeans 7.2 version+
- Eclipse Juno
- SQLYog Community Edition (MYSQL GUI)

7. SPEECH RECOGNITION SYSTEM

Speech recognition is the ability of a machine or program to identify words and idiomatic expressions present in spoken words or phrases and convert them to a machine-understandable format. Primary speech recognition software's have a fixed vocabulary of words and phrases, and it may simply identify these if they are uttered with more clarity. In our system we are using Google Speech API for speech recognition. Our system uses finite i.e. limited set of words, phrases and numeric speech inputs for evaluation purposes.

HMM method is used for speech recognition basically because it has an inherent statistical (mathematically precise) framework, the ease and availability of various training algorithms for correct estimation of parameters of the models from the finite training sets of speech data, the flexibility of the resulting recognition system where one can easily change the type, size and architecture of the models to suit the particular words, sounds etc., and the ease of implementation of the overall recognition system.[8].

The Automatic Speech Recognition of Google API is done using Deep Learning Neural Networking. It has a Global Vocabulary i.e. it can support as many as 80 languages. The speech to text conversion is done by applying set of powerful neural network models and in an easy to use API. The command and control is done through speech input, enabling application's microphone. Figure-1 depicts the basic working of Google speech API.

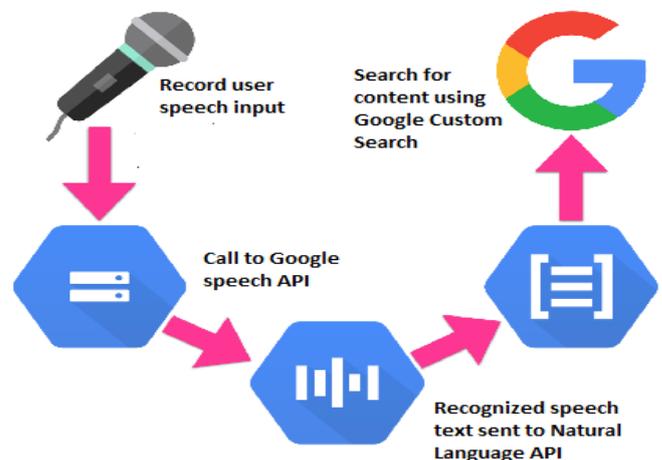


Figure-1: Google Speech API Working

8. SPEECH EVALUATION WORKFLOW OF PROPOSED SYSTEM.

The speech evaluation workflow diagram (Figure-2) is the description of how the important concepts of HMM and VQ work as a part of Automatic Speech Recognition for Google Speech API. HMM acts in the ASR technology block using acoustic models. Powerful neural network models are used for speech to text conversion. The main aim of VQ is to

generate an efficient codebook. It acts in the Speech Synthesizer block. Once the speech input is accurately recognized, corresponding evaluation action is performed by the system accordingly.

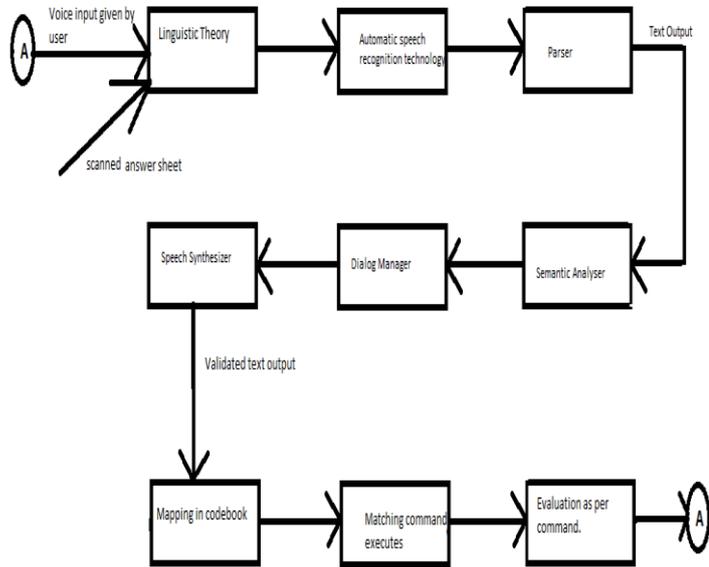


Figure-2: Evaluation by speech input

9. FUNCTIONAL MODULES OF PROPOSED SYSTEM

Input for system: Scanned Answer sheets (in pdf format), Speech input by examiner for evaluation purpose.

The following are the various functional modules of the proposed system.

Output of system: Evaluated answer sheets.

ADMIN MODULE :

Admin logs in to the system. Admin can perform following functions.

1) Upload scanned Answer sheets (in pdf format) & question paper.

2) Add/Remove Examiners and Edit their Details by choosing semester, branch and subject.

EXAMINER MODULE :

Examiner login to the system. Examiner can perform following functions.

1) Select the required subject if one or more subjects are assigned to him/her.

2) Evaluate answer sheets for the selected/assigned subject by giving marks either by speech input using a microphone or manually.

EVALUATION MODULE :

1) Examiner logs in and then selects the subject. The corresponding subject answer sheets get displayed to him in the evaluation window. The scanned answer sheets are in pdf format.

2) Evaluation of answer sheets can be done by giving speech input via a microphone. Also a provision for manual evaluation using drag and drop functionality is provided.

3) After evaluating the entire paper, he can submit it and the marks given will be stored in the appropriate tables in the database.

4) Next paper will automatically be loaded in the evaluation window and if no papers are available then appropriate message will be displayed to the examiner regarding the same.

10. SYSTEM FLOWCHART

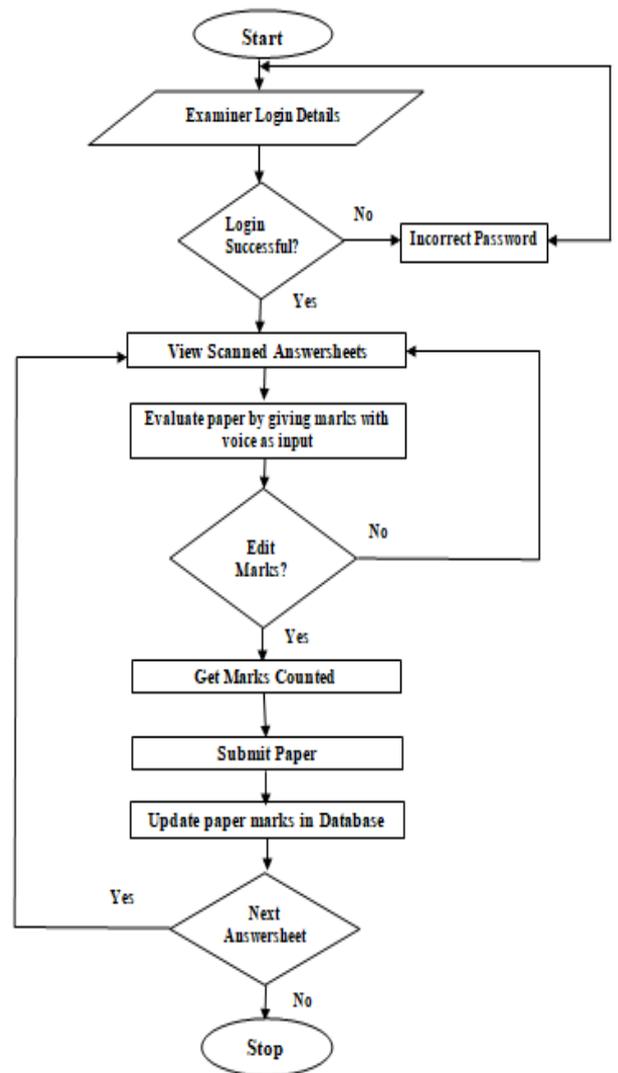


Fig -3: Flowchart of proposed system

11. CONCLUSIONS

To summarize in a nutshell, we have used speech recognition technology by taking speech as input via a microphone for answer sheet evaluation in our proposed software system. This can be applied as an additional functionality over the traditional evaluation system and can possibly reduce the malpractices and possibly offer more accuracy. Keeping in mind about the move towards automation, our system is in conformity with it and can prove to be beneficial as minimal manual human efforts are required. Tested accuracy of correct speech recognition of our proposed system has been found decent enough i.e. about 80-85%.

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