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Speech Recognition Based System to Control Electrical Appliances

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Abstract-* Speech is one of the natural forms of communication. Recent development has made it possible to use this in the security system and controlling the devices. In speech recognition, the task is to use a speech sample to select the identity of the person that produced the speech from among a population of speakers. An important pre-processing step in Automatic Speech Recognition systems is to detect the presence of noise. It has been shown that accurate speech endpoint detection improves the isolated word recognition accuracy. Also, proper location of regions of speech reduces the amount of processing. This aspect is also important for mobile telephony. Sensitivity to speech variability, inadequate recognition accuracy, and susceptibility to impersonation are among the main technical hurdles that prevent the widespread adoption of speech-based recognition systems. Speech recognition systems work reasonably well with a quiet background but poorly under noisy conditions or in distorted channels. Such a mismatch in the training and testing has severely limited. The objective of this algorithm is the development of signal processing and analysis techniques that would provide sharply improved speech recognition accuracy in any type of noisy environments. Speech is a natural medium of communication for humans, and in the last decade various speech technologies like automatic speech recognition (ASR), Voice response systems and another similar system have considerably matured. The above systems rely on the clarity of the captured speech but many of the real-world environments include noise and reverberation that mitigate the system performance. The key focus of the project is on the effectiveness of ASR.

***Keywords*: ASR (Automatic Speech Recognition), GUI (Graphical User Interface), Speech Recognition**.

I. INTRODUCTION

The objective of this Project is the development of "Speech recognition based system to control electrical appliances" and the analysis techniques that would provide sharply improved speech recognition accuracy in any type of noisy environments. Speech is a natural medium of communication for humans, and in the last decade various speech technologies like automatic speech recognition (ASR), voice response systems etc. have considerably matured. Moreover in robotics the ASR is sharply arranging its place from the last pair of decades as it is the only one medium by which these human-like robots are getting converted in the humanoids. The above systems rely on the clarity of the captured speech but many of the real-world environments include noise and others that mitigate the system performance. So our main Goal here is to develop the MATLAB Based Automatic Speech Recognition System which also is able to decrease the noise level up to .8 db in some commands. Our goal also includes the more Users Friendly System GUI based So that every user doesn't need any major training sessions before using the system. To achieve the given Goal the following "Sub Objectives" have been formulated:-English Speech Recognition Software. The given software will be able to control various electrical appliances directly. Creating a Database for Storing English Commands using the Data Acquisition Tool. A user Friendly GUI (Graphical User Interface) system in MATLAB. The impact of changes in a speaker's vocal effort on the performance of automatic speech recognition has largely been overlooked by researchers and virtually no speech resources exist for the development and testing of speech recognizers at all vocal effort levels. This study deals with speech properties in the whole range of vocal modes – whispering, soft speech, normal speech, loud speech, and shouting. Fundamental acoustic and phonetic changes are documented. The impact of vocal effort variability on the performance of an isolated-word recognizer is shown and effective means of improving the system's robustness are tested. The proposed multiple model framework approach reaches a 50% relative reduction of word error rate compared to the baseline system. A new specialized speech database, BUT-VE1, is presented, which contains speech recordings of 13 speakers at 5 vocal effort levels with manual phonetic segmentation and sound pressure level calibration.

II. IMPLEMENTATION

How the technique works to recognize speech of a person and to control appliances:

a. Recording voice of two or three persons separately. These will be treated as inputs to the system and along will also see the frequency ranges of the inputs through plots. Following separate plots are showing frequency ranges of inputs.
b. Now the above ten voices we have taken including first three will be considered as our database for whole technique. And another database is created for Storing English Commands using the Data Acquisition.
c. The technique will consider the ten and first three voices simultaneously and runs the system to get results .Suppose first voice matches with one of from ten then following results are obtained.

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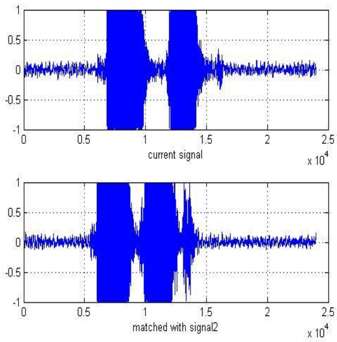
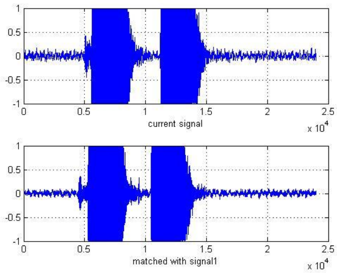
d. And if second from first matches with one of from ten then following frequency range is obtained.
e. In case no matches are found between first three and other ten, then result is no match found.

I. GRAPHS

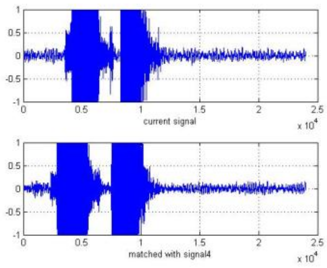
a.



b.



c.



d.

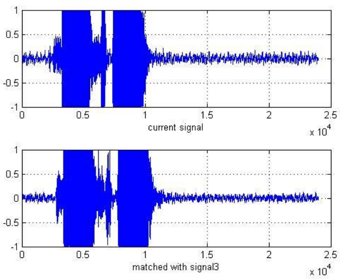
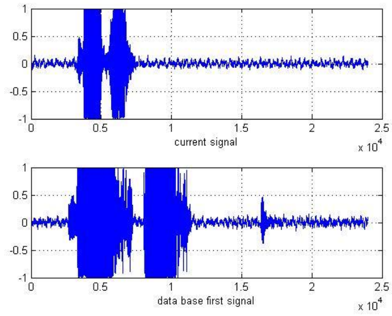


Fig 1: A, B, C, D Are Showing Respective Current Signals With Their Frequency Ranges and Showing Their Match Found From the Recorded Voices

e.



f.

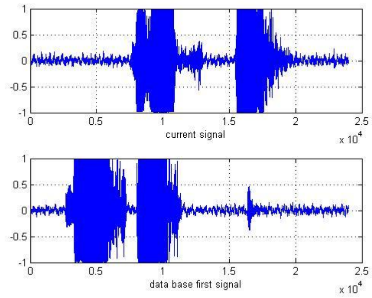


Fig 1: E and F are Showing the Current Signals With Their Frequency Range Whose Result is Match s Not Found.

III. CONCLUSION

The presented work showed the impact of varied vocal effort level on the performance of automatic speech recognition in all speech modes, ranging from whispering to shouting, it shows accuracy up to 98%. An isolated-word speech recognizer utilizing whole-word hidden Markov models with Gaussian mixture output distributions was used in the experiments. Be focused on a more precise classification of the level of speaker's vocal effort considering real-world situations (i.e. including additive noise, speaker's variable distance from the microphone, etc.). The contribution of a reliable VE classifier extends beyond the automatic speech recognition; other fields of speech processing could also greatly benefit from the knowledge of a speaker's VE level, e.g. speaker identification, psychological or forensic voice analysis, medical diagnostics of the vocal tract, etc.

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I, ARVINDER SINGH doing M.TECH (ECE) regular from CGC Landran (Mohali) and I have done my B.TECH in ECE from AIET FARIDKOT. My area of interest is image and speech processing.

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