A Review on Speech Recognition Methods

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***Abstract*—** Voice recognition is the identification of a speaker on the basis of the characteristics of voices. For this, features of speech patterns that differ between individuals are used to achieve the objective. In this paper speaker recognition system are discussed. Implementation of speaker's voice recognition system with MATLAB makes possible use of voice for real life applications. This paper provides a brief review of different DSP based techniques applied for speech recognition.

***Keywords* –** *DSP; Speech recognition; text independent; voice; speech.*

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I. Introduction

Speech patterns reflect different traits of speaker and his voice such as size and shape of the throat and mouth, voice pitch, speaking style etc. So there are basically two methods - text-independent and text-dependent methods. In a text-independent method, system functions on characteristics of speaker's speech irrespective of what speaker is saying. In a text-dependent method, the system functions on the speaker's identity based on his speaking one or more specific phrases, like passwords, card numbers, PIN codes, etc.

A comparative study between classification techniques of speech recognition with different point of view is shown here. Speech processing is one of the most important application areas of digital signal processing [1]. Aphasia is a common adult language disorder acquired after a stroke, head injury, tumor, etc. Therefore, in this paper a semi-automated Aphasia diagnosis and classification framework employing feature extraction and pattern matching techniques of the digital signal processing (DSP) is described. The proposed scheme evaluates the acoustic properties; time consumed, and speech characteristics for each language component i.e. naming, repetition, and comprehension. The naming and repetition tasks utilize DSP techniques. The proposed solution is highly scalable since it determines the diagnosis based on acoustic properties instead of the language characteristics It occupies computational analysis of the speech signals, so it reduces the subjectivity of the manual diagnosis process, also increasing the efficiency and accuracy. Finally, it distinguishes two sub types of Aphasia i.e. Anomic Aphasia and Wernicke's Aphasia. The results showed the efficiency improvement achieved by replacing the live auditory model with pre-recorded auditory model [2].

In [4], Paper gives an overview about the FPGA implementation and design flow. The main goal of [4] is to ensure the quality of the speech. An Optimized Speech Enhancement Algorithm (OSEA) is used to improve speech intelligibility and quality. This work is implemented through a Discrete Wavelet Transform (DWT) and Kalman filter methods. To analyze speech quality, various noisy types for each SNR level are used. The results were then compared with a wavelet based speech enhancement using objective assessments like SNR, NRMSE and PESQ. The final results gave a better speech quality .All tests were implemented on a TMS320C6416 fixed –point digital signal processor for real-time requirements [3]. This system is basically for the people that are suffering from stuttering and speech fluency disorder. To modify the original voice and present it back to his ear. This system does modification to the voice pitch frequency and reduces background noise. The output signal produces an altered feedback voice. The hardware implementation on a low-cost DSP processor gives a real-time computational performance and low energy consumption [4].

In this paper the main goal is to ensure different methods to overcome difficulties are being faced by people having pathologic voices. This paper gives the conditions that affect voice patterns of patients [5].

The neural prosthesis that has been done in early 2015 has excellent results. Since the first implementation it is seen that an adequate response when this device is implemented in human beings. About 120000 deaf persons, around 6000 from them are able to produce nearly normal language. It has become an important measure to indicate that signal processing helped in the development of appropriate techniques for extracting electrical stimuli from the speech signal. In this paper the implementation of 3 different strategies used in cochlear implants is done. A digital signal processor from Texas Instruments named TMS320C6416 is used to implement these strategies. The results obtained from the DSP of the

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speech signal are mostly very efficient. [6]

This paper is an overview for the speaker identification. Like fingerprint Biometric is also physical characteristic unique to each individual. We have also seen that the problems of speech recognition and machine learning are growing in these days. This paper shows different approaches and algorithms to find out the most efficient model for speaker recognition. The proposed system is absorbing effective version of voice biometric [7].

In this study, the feasibility of utilizing a Bayesian-based method for multi-scale signal decomposition called Bayesian Residual Transform (BRT) is done for the purpose of physiological signal processing. In BRT, a signal is modeled as the summation of residual signals; each characterizing information is taken from the signal at different scales. A deep cascading framework is introduced as a realization of the BRT. Signal-to-noise ratio (SNR) analysis using electrocardiography (ECG) signals is used to illustrate the utility of using the BRT for suppressing noise in physiological signals. Results in this study shows that it is feasible to utilize the BRT for processing physiological signals for tasks such as noise suppression [8].

The methods for automatic lungs sound recognition system has really improved in these few years a lot, the development of digital signal processing technology has helped researchers to develop better quality of these methods. [9] describe the all sensors, processing techniques and data set for extraction the lungs sound. In [10] a well-known problem, i.e. extraction of original signal from mixed signal has been introduced. In this some example is presented in which this type of problem is solved with MATLAB.

In [11] paper consists of neural network based speech recognition. There is two different type of neural network which is mainly of Feed-forward Neural Network (NN) and a Radial Basis Functions. Neural networks can be considered as a most powerful speech signal classifiers. By using this we can recognize small set of words in simplified form. It was found that neural networks are a powerful speech signal classifiers and can recognized small set of words. It is found that Mel Frequency Cepstral Coefficients are a very efficient tool for the pre-processing stage and which can provide good and accurate result. By using both the neural network which is Multilayer Feed-forward Network and the Radial Basis Functions are giving a accurate and good results when Mel Frequency Ceptstral Coefficients are used to find out [11].

In [14] scaly neural network is used to recognize the speech. A small sets of words are “word, file, open, print, exit, edit, cut, copy, paste, doc1, doc2” these are 11 vocabulary words are established. The above used words can be divided into two parts testing and training. The above process proved to be 79.5-88% successful, which is quite good. In this the tested data and trained data both are different network which are presented.

This research paper based on MATLAB and main focused are based on how we can implement voice recognition system by using MATLAB without making them so complex. This paper shows how MATLAB implemented algorithm provides good result in speech recognition in real world environment. Through this project and with experience in trial and error, the student are able to make more useful, better and more powerful system which can be based on automatic speech recognition system. What the knowledge he gained can easily utilize in making a project. By which he can make a better project and also this project will help him in future research [15].

Generally communication are used to interact from one person another but different mode such as involuntary repetitions and prolongation of sounds, syllables, words or phrases, and involuntary silent pauses or blocks in communication. This paper gives overview of work done by various researcher on automatic stuttering recognition system. If we do classification of speech disorder it seems to be difficult and complicated. However some classification techniques which is effective and easily recognized are used. Some works earlier had been which involving different step in recognizing speech from speech samples. This paper also compares the different research. Stuttering which means disorder of speech communication. In last 2 decades, many attempts are made and researches being done on stuttering recognition. Under this there are 3 major classifiers i.e., ANNs, HMMs and SVM. Each classifier provides different accuracies where HMM provide highest accuracy [12].

In today era speech technologies play an important role. This technology is commercially and easily available for a different uses. These technologies make machines respond correctly and it provides valuable services. This paper gives overview of implementation of speech recognition using DSP. Speech is a most natural and efficient way to exchange information for human begins. A computer to reach the goal of natural human- computer communication. Speech recognition has been developed to convert speech input into other form means from speech to text and after converting into text from it is converted into output means from text to speech [16].

Speech is the most basic and important form of communication for interaction with anyone. It can be done by many way may be interact with computers via speech or by using devices. This can be done by developing a Speech Recognition Application in which computer can easily identify the words that a person speaks into a microphone and convert that words into written text in a accurate . As a result, it can be consider a tough task. According to present situation speech Recognition is performed through which computer can identify

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words spoken by the person and convert that into text accordingly. Hence, this gives an interaction between human and computer. In doing this process accuracy plays the major role [13].

As we see in daily life the use of speech recognition in mobile phone, As it shown the DSP based software change voice command into alphabetical symbol .the use of speech recognition makes life better and more easier, but the speech recognition method is not only use in mobiles but also in various field of technology such as car, various home appliances etc. It use various method to transform speech into symbolic form of alphabet. The invention generally relates to data communication and in particular to a two way wireless communication device that utilize network based speech recognition argument to local user [18].

Concerned is speech recognition that reference speech information is extracted from a plurality of speech recognition dictionaries in a hierarchical structure to compare between extracted reference speech information and an inputted speech thereby recognizing the speech. Reference speech information representative of hierarchical-level skipping is prepared in a predetermined speech recognition dictionary so that, when recognizing an input corresponding to the reference speech information representative of hierarchical-level skipping, speech recognition is carried out by extracting a part of speech recognition dictionary belonging to a lower hierarchical level of the reference speech information being compared [17].

This review paper states the art of implementation of different method of DSP on automatic speech recognition. The graphs of various signals which can be sensed by human when compared with the word recognition performance in a syllable oriented continuous speech. The continuous speech consists of the many similar monosyllabic words and thus emphasis was on the ability to retain those words .this comparison was done on the face of combined as well as distributed word and with the duration variations. For each parameter we observed the patterns and these were generated using an efficient dynamic warping method. Parameters were set on the mel-frequency cepstral, linear prediction cepstral or a set of reflection coefficients. The values of the mel frequency cepstral coefficients shows the superior performance as they represented the more better patterns for speech.

Voice recognition can be divided into voice identification and verification, and into text dependent and text independent methods. Voice recognition is the process of automatically identifying whose voice is this on the basis of information in form of speech waves. So we can easily identify the persons accessing systems with the help of voice control in various sectors like banking transactions, telephone shopping, voice mail, security control for confidential information areas. Voice recognition and voice adaptation research conducted separately many times, but it was not of any use to achieve best performances in both areas as it has not been necessarily realistic .however if we say about text prompted voice recognition then it’s very necessary to create the specific model that contains information about both phenomenon and the voice [21].

Voice recognition methods are divided into text-independent and text dependent methods. In a text independent method, speaker model consists characteristics of voice of the person and it’s identifying what one is saying. In a text dependent method, the identification of the speaker's identity is based on person speaking specific words. The process of searching and developing an efficient vice recognition system has been very difficult as it faces many challenges due to the highly variant in input voice signals, reason of these variations is speaker himself. There are other factors too which cause change in these voice signals as people voice changes with time, and condition of health also changes the voice of person. There are some other challenges too which made this technology more interesting and wide [20].

A model can be formed comprising of speaker identification system and word identifier. The speaker identification system determines the speaker. The system model is capable of automatically identifying the speaker speech and even it is able to generate that speech. The speaker identification system is also capable to discrete the speech which associates the unknown speaker with most probable speech.in order to identify the speaker the application such as DTW and HMM speech recognizer are used. Results obtained for the model with DTW word recognizer are quite better. The experiment was performed with DTW word recognizer with inputs as isolated word digits. The results obtained for this experiment perfectly identifies the speaker. As our system discrete the words of speech this improved the performance of the system as in this case it was quite easy to identify the digits [23].

In modern era, no one wants to reveal his identity due to security purposes. So, Speech can be used for the identification of person because every person has different speech characteristic. Thus with the different information in speech waves we can easily identify the speaker. Here, speech recognition system has been designed using Vector Quantization (VQ). This system can be used to detect speaker for controlling access to services such as voice dialing, telephone shopping, information services, voice mail, security control for confidential information areas. In this, the speakers were told to say their nick name and as there were different names for each speaker so it is text dependent speaker recognition. On the basis of the VQ distortion value with database speech we identify the speaker [25-90]. As we input the speech then the system calculates the VQ distortion values and based on that with lowest value identification is done. Here, every speaker was modeled by a codebook of 32 vectors using LBG (Linde, Buzo and Gray) splitting algorithm. For this system we have considered the ideal conditions as no noise effect was considered and we know that noise will decrease the sufficiency of the system even there are others factors too which can reduce the performance of the system such as environmental conditions, quality of microphones, silent parts of speech, overall signal energy, different equipment's used conditions whether good or defected etc. can affect the recognition process. In this system if we use only few number of speakers then we may get approximately 100% recognition [19].

From the research point of view, the speech enhancement is very wide field. It can be used to increase the intelligibility, quality improvement etc. This can be done with the various techniques offered by the DSP. We have to made assumption for that our system should be fully self-contained and it does not rely on the feedback given by the recognizer. Thus, it is the only limitation of the signal processing system. Speech enhancement systems can be divided into two categories that is single and multi-microphone .In single microphone systems input signal to noise ratio over frequency is positive for this interference required is stationary. While in case of the multi microphone systems we need the some knowledge about the place of the desired source.

Spectral subtraction and delay are the simplest methods for single and multichannel processing. No doubt that at the same time there are various more powerful and complex methods, complexity also increases the cost of the improved quality. So further developments should be made on the decrease in cost and increase in computational power [22].

we observed the performance of three common nonlinear time warping algorithms from the point of recognition, accuracy and computational efficiency .complexity of alphabets and digits were varied in order to reach to the limit of this method and founded that the asymmetric dynamic algorithm by Itakura gives good performance than the other two methods. Limitations of this algorithm is that segments uttered received equal treatment, although perceptually important cues encoded in the signal were different for different segments. This algorithm has to view as a time alignment method. The performance of this method by the use of variables with combination of alpha digit task is yet to be observed [24].

VI. CONCLUSION

Speech recognition is one of the hot topics of these days, as it contributes to many security and civil applications. In this paper, various methods of speech recognition have been illustrated. Focusing on the prominent works in this field, a review of these techniques is presented in this paper.

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