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Survey on Speech Recognition

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Abstract: *speech is the basic means of communication. The main aim of this paper is to provide speech recognition system survey for different Indian languages and to discuss different methods and soft ware's used in different stages of recognition of speech system that includes: classification of speech, Modes of recognition, Classification of speech recognition vocabulary, modeling of the system, different feature extraction techniques, mathematical models, different speech classifiers, language models and performance evaluation*.

Keywords: MFCC, HMM, DTW, PLDA, LPC, RASTA

I. INTRODUCTION

Speech is the basic media for communication and exchange of information from one person to another. People, who are physically handicapped and blind, are unaware of operating the machines i.e. computer. Hence operating and communicating with machines using speech in the known language is the main aim to the above mentioned problem statement. Speech to text conversion type of communication helps those people who are not able to hear to communicate easily. Speech signal is converted into text or to a written equivalent of the message information by a process known as Speech Recognition.

An acoustic speech signal is mapped into its corresponding text is called Automatic speech. Mapping of an acoustic speech signal to some form of abstract meaning of the speech is called as Automatic speech understanding. Automatic speech recognition (ASR) is one of the recent advancements in the field of speech processing. Following speech classification and recognition is important to mention as it forms the basis for further analysis.

A. Classification of Speech

i) Isolated Words: The isolated words require brief pause between each spoken word.

ii) Connected Words: connected words does not require pause between two utterances.

iii) Continuous Speech: a small silence exists in between the utterance of continuous speech.

iv) Spontaneous Speech: speech is not planned or arranged but is done because someone suddenly wants to do them.

B. Modes of Recognition

i) Speaker Dependent: a system that can respond only with trained speakers are called speaker dependent. It allows for very large vocabulary.

ii) Speaker Independent: a system that responds to both trained and untrained speakers are called speaker independent.

iii) Speaker-adaptive: A system that is adaptive to the changes is called speaker adaptive.

C. speech vocabulary Classification

1. Small Size speech Vocabulary: varies from 1 to 100 of words.

2. Medium Size speech vocabulary: varies from 101 to 1000 words.

3. Large Size vocabulary: varies from 1001 to 10,000 of words.

4. Very-large speech vocabulary: for more than 10,000 words.

5. Out of vocabulary: mapping the unknown word with the word "unknown" (or its local language equivalent).

II. LITERATURE REVIEW ON INDIAN LANGUAGES

Recent developments related to automatic speech recognition (ASR) as presented by Ranu Dixit et.al., used speech recognition using stochastic approach [1]. Deep and Wide Multiple layers in ASR was proposed by Nelson Morgan and have focused on the use of Hidden Markov Model (HMM) for processing of multiple layers for decoding of word sequences. Large vocabulary ASR has some standard approaches such as VTLN, LDA, HLDA, MMI, MPE, MLLR and Tandem approaches. This paper also describes some methods developed on small vocabulary tasks for noisy speech to provide large gains and large vocabulary tasks for high SNR speech. This paper includes topics only related to acoustic model [2].Li Deing et.al., proposed Machine learning paradigms for speech recognition. The paradigms discussed in this paper are related to Machine Learning are Generative learning based on GMM-HMM and Discriminative learning are referred to as Bayesian Minimum Risk classifiers uses the model MEMMs. It also includes supervised, un-supervised, semi-supervised and active learning are discussed in ASR and its applications. Deep learning and learning with sparse representations are also discussed related to ASR [3].A large-vocabulary speech recognition (LVSR) was proposed by George E. Dahl et.al., that describes recent advances for phoneme recognition using deep belief networks. A pre-trained deep neural network hidden Markov model (DNN-HMM) hybrid architecture that trains the DNN to produce a distribution over senones is discussed in this paper [4].

Some of the works related to English ASR are as follows in which Vaibhavi Trivedi showed that for translating the speech waveform into a set of feature vectors is implemented on Isolated words for digits from (0 to 9) using Mel Frequency Cepstral Coefficients (MFCC) technique for speech recognition [5].A new approach for Isolated Word speech is carried out for Text Dependent Speaker using HMM was proposed by Ms. Rupali S Chavan et.al., HMM training is performed by MFCC feature parameter and these feature parameter is also used for parameterization of speech. These feature parameters are necessary for HMM training. Parameter estimation & optimization is done using forward backward algorithm with EM principle for HMM modeling. Division of feature dataset into smaller parts and then means are calculated by using K means algorithm. MATLAB 7.9 is used for implementation of speech system. For speaker independent mode 60 speech samples of selected words is tested for trained speakers. For noisy environment a recognition accuracy of 92%, 92% & 88% respectively is obtained [6]. S.Aparna et.al., proposed Backoff N-Gram Modelling for Android Application for Speech Recognition. An overview of speech recognition using an application related to the Google search is implemented in this paper. The algorithm is implemented based on Map Reduce/SS (Spectral Subtraction) using HMM and Gaussian Mixture models [7].In an another paper by B. Raghavendhar Reddy et.al., showed Android Platform application for voice to Text Conversion. The system recognizes the voice online that gets connected to Google's server. English language is implemented using Hidden Markov model for Speech recognition. Further the work can be extended for implementation of different languages for modeling of speech recognition [8].

Some of the works related to Kannada ASR are as follows in which Shiva kumar K.M et.al., used CMU Sphinx for conversion of speech to text. Speech corpus of thousand sentences was carried out in this paper. Research work is carried out for sentences with four to ten word lengths. The experiments were conducted for Kannada Language for speech to text conversion with limited speech corpus of thousand sentences [9].M.A.Anusuyaet.al., proposed Vector quantization technique for Kannada speech recognition for Speaker Independent system. The vector quantization algorithm is used for the clustering and for improving the efficiency has been highlighted. PRAAT software is used to acquire the speech signal [10]. Multivariate Bayesian Classifier was presented by Prashanth Kannada Guli et.al., for Speech Recognition in Kannada for Phoneme modeling. This approach is very much different from traditional methods. Phoneme recognition building is done using this system. The Gaussian Mixture Modeling and various acoustic phonetic features can be extracted by further extending the work [11]. Kannada Phoneme Recognition was proposed by Kavya.B.M et.al., in which Audacity software is used to record phonemes. For Performance Analysis MFCC and LPC Techniques are used. The database consists of 5 Kannada phonemes. The work can be further expanded to all the phonemes. So that a complete Kannada phoneme recognition system can be built this further helps in building word recognition system [12].

Gurudath K P et.al., proposed Isolated Digits Recognition in Kannada Language. The method used for feature extraction is MFCC and HMM model is used for pattern matching [24]. Concatenated words/digits for continuous speech can be used for future enhancement of the system [13]. Acoustic Phonetic Characteristics of Kannada Language was proposed by Hemakumar G. A text corpus for fifteen million is studied. Simple and most frequently used Kannada words of about four hundred and forty five were chosen. Praat and pearl software were used. Spectrographic analysis, gives a better insight about the nature of sounds [14]. In another work proposed by Hemakumar G et.al., for Speaker Dependent Continuous Kannada Speech Recognition Using HMM. About 20 different unique Kannada sentences are used for operating simple mobile sets. For training the system each of these sentences were recorded for 10 times for one male speaker. Similarly 3 times recording was done for testing one

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male speaker. The accuracy is about 87.76% and implementation was done using Mat lab [15]. They used LPC for feature extraction for Speaker Independent Isolated Kannada Word Recognizer. 294 unique Kannada words were used and trained on 10 speakers. The accuracy of 98.29% for tested data and 91.66% for unknown speaker data [16]. Discrete Wavelet Transform (DWT) and PCA is used for Implementation. MFCC coefficients of the speech and DWT are calculated. This can be implemented for other languages as well [29].

Preeti Saini et.al., proposed a system using HTK for Hindi language for Automatic Speech Recognition. The main objective is to build a Hindi Language speech recognition system. The system is implemented using Hidden Markov Model Toolkit (HTK). Isolated speech Recognition for isolated words is done using acoustic word modeling. The database consists of 113 trained Hindi words taken from 9 speakers. The result gave an overall of 96.61% and 95.49% of accuracy [17]. Isolated & Connected Words for Hindi Language was proposed by Annu Choudhary et.al., using Hidden Markov Model Toolkit. The system is implemented using a statistical approach such as (Hidden Markov model toolkit). The system is trained for 100 distinct Hindi words for Hindi Language . The system has an overall system accuracy of 95% for an isolated word is 95% and 90% for connected words [18].

Development of Automatic Kannada Speech Recognition System by Akshata K Shinde, Anjali H R, Deepika N Karanth, Gouthami K, and Vijetha T S. proposed a system that allows individual to record their voices to a computer/ laptop interfaced in a manner such that, any variations or disturbances, resembles regular human conversation. Automatic speech recognition is a technology that recognizes human speech by system in all suitable conditions. The work provides an overview of Automated Speech Recognition for Kannada language. To improve the performance of ASR many trends and studies are made to work efficient with the aid of many research scholars. The steps for implementing simple speech recognition system are training, testing, pre-processing filter, end-point detection, feature extraction strategies, speech classifiers, and overall performance assessment. The recognition device is implemented using Hidden Markov Tool kit for Kannada language [2]. **Development of Kannada Speech Corpus for Continuous Speech Recognition** by Anand H. Unnibhavi and D. S. Jangamshetti et.al., gives an idea for development of speech corpus for kannada language for impartial speaker for continuous speech recognition. Kannada Speech corpus plays a vital role in production of Automated Speech Reputation (ASR) and Text-To-Speech (TTS) synthesis. The speech corpus is developed for the age group of 21 to 45 years. Kannada Speech corpus for ASR system is done by means of gathering textual content corpus such that information is recorded similar to the text corpus accompanied by transliteration (phonetic illustration of the textual content corpus) and subsequently a pronunciation dictionary is advanced [32].

Kalith et.al., proposed Automatic speech recognition system for Tamil language. Carnegie Mellon University developed this system and this system is based on CMU Sphinx. The training is carried for both speaker dependent and speaker independent and has been adapted back ground noise [19]**.** Ms.Vimala.C et.al., proposed Speech Recognition System for Tamil Language using HMM for Speaker Independent system. N Pushpa et.al., proposed

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Back Propagation Neural Network and Semi-Supervised Training for Speech processing Of Tamil Language. Implementation has been done using HMM and achieved a word accuracy of 88% for trained and test data spoken by the speakers. The system performance is evaluated on the Word Error Rate (WER) that gave 0.88 WER [20].Speech Processing Of Tamil Language with Back Propagation Neural Network and Semi-Supervised Training was done by N.Pushpaet.al., .Multilayer feed forward network was used for speech recognition system for individually spoken word in Tamil language. The implementation was carried out using four types of filters namely preemphasis, median, average and Butterworth band stop filter in order to remove the background noise and to enhance performance of the speech signal. MSE and PSNR values are used to measure the overall performance of these filters. The work can be further extended for more number of isolated and continuous words 21].

P.Vijai Bhaskar et.al., proposed Telugu Speech Recognition system using Hidden Markov Model Toolkit (HTK). The system data base consists of trained data for continuous Telugu speech taken from male speakers [22]. In another work carried by Cini Kurian et.al., for Malayalam language using PLP Cepstral Coefficient for Continuous Speech. Hidden Markov Model (HMM) is used for pattern recognition. The system consists of 21 male and female trained data speakers in the age group of 19 to 41 years. A word recognition accuracy of 89% and 83% of sentence recognition accuracy for continuous speech data of independent speakers. The performance evaluation was done using with bigram and trigram language models [23].

SabuKamini proposed Speech Conversion to Devanagari Script**.** This paper proposes use of phonetic model to convert speech to Devanagari script. MFCC features are used for feature extraction [24]. Kanika Garg et.al., proposed different Noise Reduction Techniques for Automatic Speech Recognition Systems. Noise estimation, removal and speech enhancement techniques are summarized in this paper. Gamma tone filters instead of conventional Weiner filters and Line Enhancers are used. Spectral subtraction technique showed good results [26]. Anoop.V et.al., proposed Performance Analysis of Speech Enhancement Methods Using Adaptive Algorithms and Optimization Techniques. An optimization technique is used for comparison of the performance of the conventional adaptive filter. De-noising of the speech signal is evaluated in terms of SNR. Different noises at different levels are added to a speech signal 7]. Shreya Naranget.al., proposed Speech Feature Extraction Techniques. The different feature extraction techniques such as probabilistic Linear Discriminate Analysis (PLDA), Linear Predictive Coding (LPC), Mel-frequency cepstrum (MFFCs), RASTA filtering are used in this paper. The paper summarizes the different techniques used for feature extraction used in speech recognition system [30].

Speech Recognition Using Euclidean Distance was proposed by Akanksha Singh et.al., Processing of digital speech signal and algorithm for voice recognition algorithm is very difficult for instant and accurate automated voice recognition era. The voice is a signal of countless statistics. A right away of evaluating and synthesizing the complex voice signal is because of an excessive amount of facts contained within the signal. Consequently, the digital signal procedures consisting of function extraction and function matching are introduced to represent the

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voice signal. This paper describes a technique of speech reputation with the aid of the use of the Mel-Scale Frequency Cepstral Coefficients (MFCC) extracted from speech signal of spoken phrases. A weighted Euclidean distance is used for Verification. MFCC approach uses the software platform MATLABr2010b for implementation of speech recognition [33]. **Speech to text conversion for visually impaired person using µ law companding** by Suraj Mallik, Rajesh Mehra. The paper describes the implementation of speech recognition using DSP and textual content conversion gadet. Audio signal that is speech is considered as the desired means of exchanging the information of an individual, the author represents conversion of textual data by voice orientated command i.e. it uses software program to analyze the data which is accepted as input. Correlation and µ law companding techniques are used for comparison. Voice recognition is done using MATLAB. The characteristic keys are used to save the voice command. The real time speech input obtained is further analyzed by the speech recognition system and then the required part of the speech sample is filterd, extracted and tested with the sample that is stored as data base. MATLAB is used for conversion of the message into corresponding text. [34].

Speech Feature Extraction Techniques: A Review by Shreya Narang1, Ms. Divya Gupta2**.** The system accepts the speech and conversion process is carried out. The survey paper discusses the current trends of speech recognition and the speech recognition techniques and also describes a short explanation of the 4 stages in the speech recognition technology are labeled. The paper gives a description of 4 different feature extraction techniques such as Linear Predictive Coding (LPC), Mel-Frequency Cepstrum (MFCCs), RASTA filtering and Probabilistic Linear Discriminate Analysis (PLDA). The paper summarizes the different feature extraction techniques used in speech recognition system [35].

Zero crossing rate and Energy of the Speech Signal of Devanagari Script by D.S. Shete et al.,. In the analysis of speech, the speech signal is classified into voiced and unvoiced by few techniques and indicators. The two main techniques usually employed for splitting the voiced- voiceless elements of speech from a speech signal are energy and zero crossing rate (ZCR). For calculating the zero crossing rate and energy the speech samples are divided into segments and then the speech samples are classified as voiced and unvoiced components of speech. From the obtained results we consider that for voiced part the zero crossing rate are low for excessive for unvoiced speech. For voiced speech the energy is high and for unvoiced part the energy is low. For separation of voiced and unvoiced these two methods provide greater effectiveness [36]. **Automatic Identification of Silence, Unvoiced and Voiced Chunks in Speech** by Poonam Sharma and Abha Kiran Rajpoot. The main aim of the work is to segment the speech signal automatically into silence, voiced and unvoiced regions that can be very useful in increasing the overall performance and accuracy of popularity systems. Zero crossing rate, brief time power and essential frequency are the parameters taken to identify the speech signal. The above parameters is checked for performance evaluation and accuracy of 96.61% is achieved for four extraordinary speakers [37].

A Review on Speech to Text Conversion Methods by Miss.Prachi Khilari and Prof. Bhope V. P. This paper gives a top-level view of appreciation and main technological perspective of the essential development of speech to text conversion and additionally offers evaluation technique evolved in every stage of classification of speech to text conversion. In this paper, they have developed an online speech-to-textual content system. The aim of the paper is to recapitulate and match up to different speech recognition systems in addition to procedures for the speech to textual content conversion [38].

Speech to text and text to speech recognition systems-A review by Ayushi Trivedi, Navya Pant, Pinal Shah ,Simran Sonik and Supriya Agrawal. Nowadays, communication is the important means to communicate with people. Passing on records, to the proper individual, and within the proper way is very critical, not simply on a corporate level, but additionally on a non-public level. The sector is moving closer to digitization, for the method of conversation i.e. smart phone calls, emails, text messages etc. So that we can serve the purpose of powerful conversation among two events without hindrances, many applications have come to image, which acts as a mediator and assist in efficiently carrying messages in speech of textual content, or speech signals over miles of networks. In this paper, they have used different techniques and algorithms that might be implemented for speech to text conversion functionalities [40].

III. Modeling of the system

1. Pre-Processing: In speech an analog signal is converted to digital and it is necessary for sampling and also for recognition of the speaker. The steps involved in preprocessing process are noise removal, voice activity detection, preemphasis, framing and windowing.

2. Feature Extraction: a set of features are extracted to find the correlation of the uttered speech signal is known as feature extraction.

3. Pattern Classification: Pattern recognition is the science of making inferences from perceptual data, using tools from statistics, probability, computational geometry, machine learning, signal processing, and algorithm design.

1. Acoustic-phonetic approach: Acoustic phonetics is a subfield of phonetics which deals with acoustic aspects of speech sounds.

2. The pattern-matching approach: Pattern recognition is a branch of machine learning that focuses on the recognition of patterns and regularities in data although it is in some cases considered to be nearly synonymous. There exists two methods namely template approach and stochastic approach

a. Template based approaches: to find the best match the set of pre-recorded words (templates) is compared with Unknown speech.

b. Stochastic Approach**:** the uncertain or incomplete information is modeled using the probabilistic models.

3. Dynamic time warping: Dynamic time warping is an algorithm for measuring similarity between two sequences which may vary in time or speed. DTW can be applied to video, audio, and graphics. DTW is a method that calculates an optimal match between two given sequences (e.g. Time series) with certain restrictions.
4. Knowledge based approaches: knowledge-based systems use uncertainty techniques, based on logic (logic which permits treatment of incorrect and partial information and/or statistical methods). In a conventional system it is usually required to specify correct and complete information.
5. Statistical based approaches: Methods of collecting, summarizing, analyzing, and interpreting variable numerical data based on Hidden Markov Models, or HMM a statistical learning procedure.
6. Learning based approaches: it involves machine based learning such as neural networks.
7. The artificial intelligence approach: a machine mimics human such as learning and problem solving is termed as "artificial intelligence".

IV. Matching Techniques

Some of the techniques used for matching a known word using an unknown word in speech recognition are

1. Whole-word matching: the incoming digital-audio signal is compared against a prerecorded template of the word.
2. Sub-word matching: it identifies the phonemes and then performs the word matching using pattern recognition on those sub words.

V.Language Models Used While Designing ASR Model

For large vocabulary speech recognition language model is used. The language models provide us information on words and their relations.

VI. Evaluating the Performance of ASR Systems*: The* performance of the system is evaluated in terms of accuracy and speed. Accuracy is measured using Word Error Rate (WER), speed is measured in terms of the real time factor.

VII. FUTURE RESEARCH AREAS

Based on the survey, most of the work is carried out on many Indian languages. A challenging task is automatic segmenting of Indian languages into phonemes. It is considered that most of the researchers worked on English language only few research works has been done related to Kannada language.The problem is challenging because there are about 250 basic, modified and compound character shapes. The Structures of the characters have very complex orthography and Kannada script is an inflectional language.

VIII.CONCLUSION

In this paper, speech classification, different Modes of Recognition, Classification of different speech recognition vocabulary and different challenges and problems present in ASR are specified. The authors feel that designing of new models are required to achieve the solutions for different identified problems.

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