Isolated to Connected Tamil Digit Speech Recognition System Based on Hidden Markov Model

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Isolated to Connected Tamil Digit Speech Recognition System Based on Hidden Markov Model

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Abstract: Speech recognition technology has improved with time to enhanced Human Computer Interaction (HCI).This paper proposed a system for isolated to connected Tamil digit speech recognition system using CMU Sphinx tools. The connected speech recognition important in many application such as voice-dialling telephone, automated banking system automated data entry, pin entry etc. the proposed system is tri phone based, small vocabularies, speaker specific and speaker-independent. The most powerful Mel Frequency Cepstral Coefficient (MFCC) feature extraction techniques are used to train the acoustic feature of speech database. The probabilistic Hidden Markov Model (HMM) is used to model the speech utterance. And the Viterbi beam search algorithm is used in decoding process. The system tested with random digit (0 to 100) in a various condition shows optimum result 96.7% recognition rates for speaker specific and 54.5% recognition rate for speaker independent in connected word recognition. We use CMU sphinx speech recognition tools to construction of speech recognizer.

Keywords: HCI, MFCC, Tamil digits, Features extraction, Hidden Markov Models, ASR.

I. INTRODUCTION

The role of the speech recognition is to recognize of the words uttered by speaker. Making computers in understanding human speech has enhanced the human community. Alternatively, human voice is the one of the best media for communicating with computers rather than typing and clinking. Also human voice is a better interface when it comes to illiterate people rather than graphical user interface objects. ASR research is being done throughout the world for improving the human and computer interaction. As a result the studies attempts to develop a human and machine interface in which humans and machine can communicate in an unskilful way. This leads to a future development of vocally interactive computers. The speech recognition technology is the challenges to achieve due to the human speech variability and complexity.

A. The Problem of Tamil Speech Recognition

Recognizing human speech is still a problem for modern day computers. One of the reasons for this to the speech variability present in any human's spoken utterance and language nature. In spite of this and other complexities found in speech recognition. The researchers are trying to solve the problem in their laboratories. Still the researchers engaged in speech technology are mystified how the natural human vocal and auditory system works. Modelling the exact blueprint of these both human systems is the current challenge in speech laboratories. Through this only, a well understanding could be made of the construction and the perception process related to human communication. While building a hundred percent reliable continuous speech recognizers for a lager vocabulary for a particular language has been the ultimate challenges for the speech recognition scientists. Although there are many different technologies and theories applied for this process, none provides hundred percent reliabilities. Furthermore some scientists have predicted that recognizing human speech without any constrains is not possible and cannot be realize at any stage. Nowadays the most popular input media for any machine is via keyboard or pointing device, but in future microphone can be one of input media. Then microphone will become an essential input device just like the use of a mouse with the keyboard. You can work with the computer without mouse and work with only keyboard, but the mouse is the quite comfortable. A very similar impact would occur in near future, where the system will always have some sort of device to speak into.

B. Classification of Automatic Speech Recognition (ASR)

ASR development is the process of building a system for finding equivalent acoustic features to a given string of words. Based on the type of speech utterance ASR can be classified as isolated, connected, continuous or spontaneous. Based on the speaker class. ASR can be classified in to speaker dependent and speaker independent. Based on the size of the vocabulary used, ASR can be classified in to small, medium, large and very large or out of vocabulary systems [5].

C. Features used in this research - MFCC

Sound is a wave form which is created by vibrating objects and propagated through a medium from one location to another. A wave can be explained as a disturbance that travels through a medium from one location to another location [7]. The sound wave could be visualized as sequence of changes in air pressure. The frequency and amplitude are two key characteristics of the sound wave. The number of times a signal repeats itself is denoted as frequency or cycle and is measured in Hertz (Hz). The amount of exert air pressure variation is denoted as amplitude. In addition to these two characteristics the waveform contains two key perceptual properties of the frequency and amplitude which are the pitch and the loudness of the sound signal. The pitch of a sound is the perceptual correlate of frequency and the loudness of a sound is the perceptual correlate of the power, it is related to the square of the amplitude. There are number of feature used in speech recognition to represent the waveform, such as Linear Predictive Coding (LPC), Cepstrum and Perceptual Linear Prediction (PLP). In these the Cepstrum features have been widely used than others because of its nature to represent variation of the frequency and the higher recognition rates it provides for speech recognizers. Cepstral features are used in this research to represent the feature set from the waveform. The types of cepstral feature used are in the form of Mel Frequency Cepstral Coefficients (MFCC) [6]. These cepstrum coefficients are the result of a cosine transform of the real logarithm of the short time energy spectrum expressed on a Mel-frequency scale. In MFCC, the main advantage of its uses Mel frequency scaling which is very approximate to the human auditory system.

D. The Hidden Markov Model.

The Hidden Markov Model (HMM) is one of the most powerful and leading statistical approaches, which has been applied for many years. The basic theory of HMM was published in a series of classic papers by Baum and his colleagues in the late 1960s and early 1970s which was then implemented for speech recognition applications by Baker at Carnegie Mellon University (CMU) and by Jelinek and his colleagues at IBM in the 1970s [16].

The joint probability that O is generated by the model M moving through the state sequence X is calculated simply as the product of the transition probabilities and the output probabilities. So for the state sequence X in Figure 1.

$$P(O, X|M) = a_{12} b_2(o_1) a_{22} b_2(o_2) a_{23} b_3(o_3) \ldots$$

However, in practice, only the observation sequence O is known and the underlying state sequence *X* is hidden. This is why it is called a Hidden Markov Model.

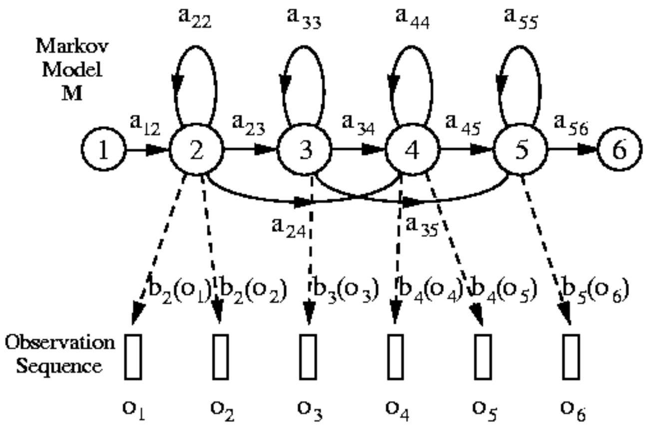

Figure 1. The Markov Generation Model

D. Tamil Language Features

Tamil is a Draviden language speaking about 77 million people all over the world. It is a 15th largest language among the world languages. Spoken predominantly in south India Sri Lanka and Malaysia, it is an official language in India and Sri Lanka. A phone is a fundamental unit of a language in speech. Tamil is a phonemic script. Tamil prosody includes 12 vowels, 18 consonants, 216 compound characters and 1 Aythm altogether 247 letters in standard Tamil alphabet. An allophone is a variant sound of standard phoneme. Including all allophones the Tamil language has almost 100 sounds [17].

E. Aims and Objectives

In a nutshell the main aim of this research is to design and implement a Tamil digit speech recognition system based on Hidden Markov Model (HMM) using Sphinx tools. By achieve the following sub objectives, the system is capable of recognizing and responding both isolated and connected digit speech input.

1. Build the speech recognizer for isolated word of Tamil digit.
2. Build the speech recognizer for isolated to connected word of Tamil digit.

The main activities would be to feed the spoken utterance into the computer via a microphone for speech training of the system. And test the system with spoken utterance to evaluate the recognition result. The result log file shows the Word Error Rate (WER) and Sentence Error Rate (SER). Simply to reduce the complexity of the overall work several limitations have been enforced on the recognition mode. The type of recognition is isolated word and connected word of Tamil digit speech recognition with an identified vocabulary of nearly thirty words.

II. RELATED WORK

Speech recognition work era stated in 1970's Itakura performed speech recognition by calculating the prediction residual for 200 words. Rabiner in 1980's was proposed a Dynamic Time Wrapping (DTW) algorithm for connecting word recognition [13]. Wilpon in 1990's was automatic English word recognition using HMM [1]. In 2009 Musmita Sharma proposed a platform for speech corpus generation by an adaptive LMS filter and LPC cepstrum as part of ANN speech recognition. Further Guido Aversano et al [2], proposed a new text-independent method for phoneme segmentation this method included both new preprocessed and new segmentation algorithms. The result shows 74% of the phoneme transition with approximately zero number of insertions. In 2011, V. Radha et al [3], proposed an isolated word recognition system for Tamil spoken language using back propagation neural network based on LPCC features. The result achieved minimum Mean Square Error (MSE) for Tamil speech signal is 0.00515 / 0.01.

III. TAMIL DIGIT SPEECH RECOGNITION SYSTEM

A. Scope

The Table 1 Shows ten distinct Tamil digits ie, Zero to Ten. Speakers could speak any of these ten digits in isolation which is presented under the isolated word speech recognition. The Table2 shows connected second decimal place Tami digit words ten to hundred. Although the training data were recorded in a studio environment which is clean and noise free, however the system was tested with data recorded in the same environment.

B. Statistical Approach of the Speech Recognition.

Figure 1 schematically shows how the speech recognition problem to be solved by applying statistical theory.

If someone speaks it produce speech utterances. It can be represented as speech vector O, which is also called observation. Observation O is incorporates with time t, so speech vector O_t is given.

$$O = O_1, O_2, O_3, \ldots O_t \qquad (1)$$

Let us define a sentence as a string of words w

$$W = w1 , w2 , w3 , \ldots, wn \qquad (2)$$

The words defined here are based upon orthography,

Where W' is the required string of words (sentence) and L represents the set of all

$$W' = \underset{W \in L}{\operatorname{argmax}} P(W|O) \qquad (3)$$

sentences in the language. As there is no direct way of calculating this, we may simplify it by using the Bayes' Rule, defined as:

$$P(x|y) = \frac{P(y|x)P(x)}{P(y)} \qquad (4)$$

So applying it on Equation we get

$$W' = \underset{W \in L}{\operatorname{argmax}} \frac{P(O|W)P(W)}{P(O)} \qquad (5)$$

Probability in the denominator is not a simple one to calculate. However, as it is only interested in the maximum value it can remove as the common denominator to give:

$$W' = \underset{W \in L}{\operatorname{argmax}} P(O|W)P(W) \qquad (6)$$

It is said that the most probable sentence W given some observation sequence O can be computed by taking the product of two probabilities: the P(W) or the prior probability comes from the language model , while the P(O|W) or the observation likelihood is computed by the acoustic model. So the Automatic Speech Recognition System is composed of a trainer which trains the P

(O|W) and P (W) using a particular data set. The trained system can then be used to decode (recognize) input speech O, to give a string of words are output. Figure1 depicts this procedure.

C. System Architecture and Algorithm.

Figure II. Shows proposed system architecture of Tamil speech recognizer this project should be based on this architecture. This architecture based on CMU Sphinx [19] speech recognition system, the components of the Sphinx system that we used for training and recognition. In other words, Sphinx Trainer [19] used for training and Sphinx Decoder [19] used for the decoder. This architecture comprises into two faces they are called pre-processing and post-processing. Therefore the work can be categories the following.

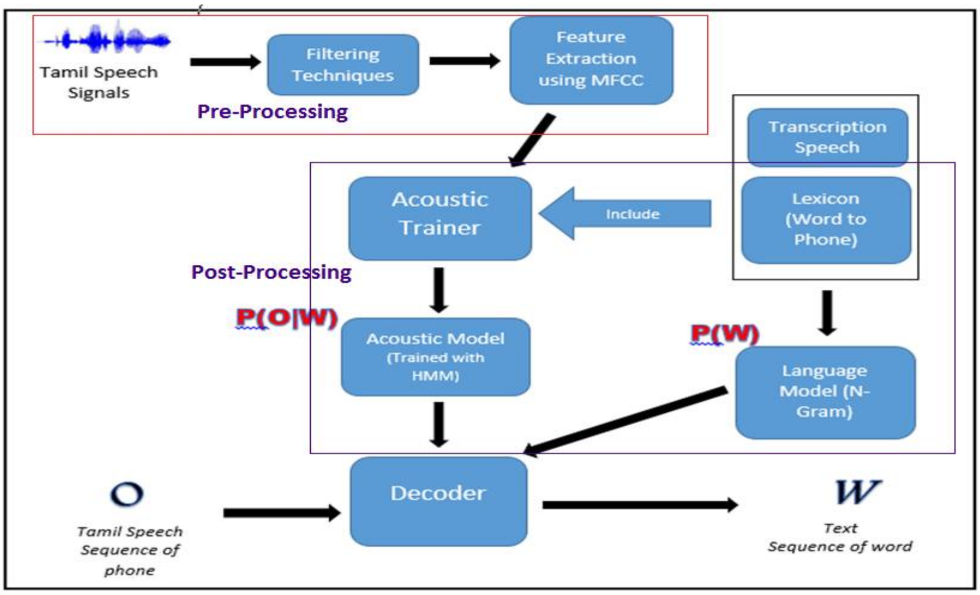

Figure II. Proposed ASR System Architecture for Tamil Digit Speech Recognizer

Pre-processing

The front end part involves speech file segmentation and feature extraction. The speech files are split using Audacity Version 1.2.6 (open source) and each speech unit labeled in order to identify. Finally these speech word or units are hand transcribed for sending acoustic training face. The segmented speech files are requiring to be in .NIST or .WAW format. These files are subject to feature extraction techniques called Mel Frequency Cepstral Coefficients (MFCCs) using which, the system will be trained. The files we used in .nist format, single channel, sampled at 16000 sample per second at 16 bit per sample in little Endean format.

Post-processing

Post-processing is back end part of the system it's comprises the following task

- Acoustic Training
- Language Model
- Decoding

Acoustic Training – The acoustic training means acoustic modeling, using training data the statistical HMM model will be creating and updating. The training process of acoustic model used by Sphinx Train tools which consists of a set of programs, each responsible for a well defined task. The input of the Sphinx Train requires the following components.

- The acoustic signals
- The corresponding transcript file

- A language dictionary (lexicon word to phone mapping)
- A filler dictionary (non speech sound units)

Language model – The n-gram language model can be generated by online language modeling utilities if the unique word count is below 5000 words, or else can use Statistical Language Modeling (SLM) toolkit, which works in Linux environment. The SLM toolkit is more useful as it allows more options for generating the language model.

Decoding - This is the actual recognition face where the speech signal would be identified as the particular word in the vocabulary or reject indicating a miss match the word. The decoder also consists of a set of programs, which have been compiled to give a single executable that will perform the recognition task, given the right inputs. The inputs that need to be given in to the trained acoustic models, a language model, a language dictionary, a filler dictionary, a model index file and the set of acoustic signals that need to be recognized. The data to be recognized are commonly referred to as test data.

IV. Data Collections and Experimental Setup

Speech sample collections were done mostly concerned with recording various speech sample of each Tamil digit (isolated and connecting words) by different speakers. There are two format of samples were collected one in text corpus another one in speech corpus. When collecting speech sample there must consider four main factors that affect the training set vectors that are used to train the HMM. That factor includes who are specking, the speaking condition, transducers and transmission system and the speech units.

A. Text Corpus

There are 30 samples words were selected for training the speech recognizer. These 30 words were coded by following the phonetic rule. The International Phonetic Alphabet IPA [18] explains the phonetic rules for sound unit belongs to all languages used in the world. The Unicode character encoding explains how to code Tamil alphabet mapping to Unicode character mapping. and the selected 30 Tamil digit encoded with the online tools available [18] Kandupidi Tamil Unicode editor explain in Table I & II. Since Sphinx speech recognizer does not support Unicode character and case sensitive to capital letters, we change Unicode character in to ASCII representation [17] of Tamil character shows in table I & II.

B. Speech Corpus

There are 10 selected speakers (5 males and 5 females) were uttered each word in isolated digit and connecting digits. The Tamil digits speech recognition system's speech sample collection was done in totally noise free environments. The speech sample of 10 speakers were collected in a clean environment using mobile phone sound recorder device name is Sumsung Galaxy J1. Our studio was our room condition without any noise from such as air-condition, fan and insect. Recorded sample transfer to a PC for preprocessing. Linux based sound editing software called Audacity used for segmenting large file into small sound units and corresponding digits.

The Tamil digits speech recognition system's main speech units are isolated words and their prefix called connected word which includes in continues speech recognition. In other words the objectives of the system are to recognize the words that belong to isolate and then some extent to continuous speech recognition. The speech sample was recorded in two faces, one is set of ten distinct Tamil digits as shown in Table 1 and another face is set of connected word as a prefix of isolated word as shown in Table II. Therefore, this system is mainly an isolated and assists to some extent as continues word recognizer.

Each word uttered by every speaker and their record were saved for further processing. There are two modes of speech sample were collected namely first speaker dependent mode (i.e., the same set of speakers were used in both the training and testing phases) and second speaker independent mode (i.e., the speaker used in training are different from tossed used in testing).

Table 1. Isolated Tamil Digit with its Unicode format an ASCII representation

Digit	Tamil	Unicode	ASCII
0	பூச்சியம்	puuchchiyam	PUUCCIYAM
1	ஒன்று	onRu	ONNNRRU
2	இரண்டு	iraNdu	IRANNTTU
3	மூன்று	muunRu	MUUNNNRRU
4	நான்கு	n-aanku	NAANNNKU
5	ஐந்து	ain-thu	AINTU
6	ஆறு	aaRu	AARRU
7	ஏழு	Ezhuu	EELLLUU
8	எட்டு	eddu	ETTTTU
9	ஒன்பது	onpathu	ONNNPATU
10	பத்து	paththu	PATTU

Table II. Connecting Tamil Digit with its Unicode format and ASCII representation

Digit	Tamil	Unicode	ASCII
1_	பதின்	pathin	PATINNN
19	பத்தொன்பது	paththonpathu	PATTONNNPATU
20	இருபது	irupathu	IRUPATU
2_	இருபத்தி	irupaththi	IRUPATTI
30	முப்பது	muppathu	MUPPATU
3_	முப்பத்தி	muppaththi	MUPPATTI
40	நாற்பது	n-aaRpathu	NAARRPATU
4_	நாற்பத்தி	n-aaRpaththi	NAARRPATTI
50	ஐம்பது	aimpathu	AIMPATU
5_	ஐம்பத்தி	aimpaththi	AIMPATTI
60	அறுபது	arupathu	ARRUPATU
6_	அறுபத்தி	arupaththi	ARRUPATTI
70	எழுபது	ezhuupathu	ELLLUPATU
7_	எழுபத்தி	ezupaththi	ELLLUPATTI
80	எண்பது	eNpathu	ENNPATU
8_	எண்பத்தி	eNpaththi	ENNPATTI
90	தொண்ணூறு	thoNNuuRu	TONNNNUURU
9_	தொண்ணூற்றி	thoNNuuRRi	TONNNNUURRRRI
100	நூறு	n-uuRu	NUURRU

Table III. Overall Word Recognition Rate (%) of the Tamil Digit Speech Recognition System

No of Sound file for	No of Sound	Isolated Word Recognition Rate	Connected Word Recognition Rate

training	files for testing	Speaker specific mode	Speaker independent mode	Speaker specific mode	Speaker independent mode
300 (1 to 9)	30	98.6 %	64.80 %		
300 (10 to 100)	30			96.70 %	54.5 %

V. EXPERIMENTAL RESULTS AND DISCUSSION.

The database consists of 11 distinct words isolate (digits) and 19 connected words of 10 male and female speakers. It also contains 300 sound files used for training and testing the Isolated Words Recognition to connected words module and 28 sound files for testing the Continuous Speech Recognition module in clean and noisy environments for both multi-speaker and speaker-independent modes. Recognition rate given by Word Error Rate (WER) of the trained HMM is defined as follows:

$$WER = \frac{S + D + I}{N} \tag{7}$$

Where,

S is the number of substitutions,

D is the number of deletions,

I is the number of the insertions,

N is the number of words in the reference.

OR word accuracy (*WAcc*) is calculated as

$$WAcc = \frac{N-S-D-I}{N} = 1 - WER \tag{8}$$

Table III. Shows overall WER (%) of the Tamil Digit Speech Recognition System. Experimental results of the combination of MFCC and HMM algorithms in the Tamil digits speech recognition system are acceptable, but could be improved further to obtain higher accuracy rates. Table IV shows a comparison of recognition rates (%) for current speech recognition researches and systems together with feature extraction and classification techniques used.

Table IV. Comparison of Recognition Rate(%) with Current Speech Recognition Research

Reference	Features Extraction	Features Classification Techniques	Recognition Rate (%)
[4]	MFCC	HMM	Speaker specific 88.82 Speaker Independent 92.06
[5]	MFCC	VQ	88.88
[6]	MFCC	VQ	70 to 85
[7]	MFCC	HMM	92
[8]	MFCC	VQ	57 to 100

[9]	MFCC (Clean) MFCC (Noisy)	HMM (clean) HMM (Noisy)	86 28 to 78
[10]	LPC	VQ and HMM	62 to 96
[11]	MFCC PLC	------ ------	33 to 45 30 to 40
[12]	MFCC	HMM	90
[15]	MFCC	ANN	80.95, Isolated Word
Our System Experiment Results	MFCC	HMM	Isolated speaker specific 98.6 Isolated speaker independent 64.8 Connected speaker specific 96.7 Connected speaker independent 54.5

VI. Conclusion

The above results shows recognition rates of speaker specific mode performed better than the speaker independent mode tested in noiseless environment. Also, it is observed that the isolated word recognition in both speaker mode is higher than that connected word recognition rate. And there is a small difference in isolated to connected digit recognition rate. This research expectation is to achieve the same result in both isolated and connected digit recognition. Although it is believed that the recognition rates achieved in this research are comparable to other systems and researches of the same domain, however, more improvements need to be made specially increasing the training and testing speech data. The more speech data used for training the in a system, the better and higher the system's performance can be obtained.

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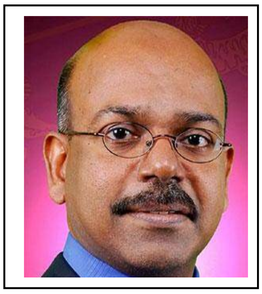

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