ISSN 2229-5518

Survey about Speech Recognition and Its Usage for Impaired (Disabled) Persons

Ananthi. S, Dhanalakhsmi. P

Abstract— Speech Recognition enables the operating system to convert spoken words to written text which is achieved by Speech to Text method (STT). Speech synthesis enables the operating system to convert written text into spoken words which is achieved by Text to Speech (TTS). Speech recognition based on Acoustic model and Language model (Pronunciation Model). Usual Method used in Speech Recognition (SR) is Hidden Markov Model (HMM), Dynamic Time Warping (DTW) algorithm and Artificial Neural Network (ANNs). The widely used technique for Speech Recognition is HMM. These Speech Recognition and Speech Synthesis methods are especially very useful for Disabled persons like Hearing Disability and visually impaired, Speech impaired and Co-ordination or dexterity impairment.

Index Terms— Artificial Neural Network, Automatic Speech Recognition (ASR), Disability, Dynamic Time Warping (DTW), Hidden Markov Model (HMM), Performance, Speech Recognition (SR)

1 Introduction

Speech is one of the most important tools for communication between human and its environment. Speech Recognition is the process of analyzing the entire phrase in order to promptly and effectively provides the appropriate outcome [1]. Speech recognition (SR) is the translation of spoken words into text. It is also known as automatic speech recognition (ASR), or computer speech recognition [2]. Some SR systems use "training" where an individual speaker reads sections of text into the SR system. These systems analyze the person's specific voice and use it to fine tune the recognition of that person's speech, resulting in more accurate transcription. Systems that do not use training are called "Speaker Independent" systems. Systems that use training are called "Speaker Dependent" systems [3]. The term voice recognition refers to finding the identity of "who" is speaking, rather than what they are saying. Recognizing the speaker can simplify the task of translating speech in systems that have been trained on specific person's voices or it can be used to authenticate or to verify the identity of a speaker as part of a security process [4].

Voice recognition is a technology that automatically recognizes human voices using computers. In general, "voice recognition" means comprehending the input voice and generating text strings. In voice recognition, both linguistic and acoustic information play an important role [5]. Linguistic is study of Language, whereas Acoustic is the study of sound. There are many obstacles to accurate voice recognition, such as tonal differences between individuals and background noise.

- *Ms. Ananthi. S is currently pursuing PhD Program in Computer Science and Engineering in Annamalai University, Chidambaram, Tamil Nadu, India. E-mail: ananthi68@gmail.com.*
- *Dr. P. Dhanalakshmi is currently Working as an Associate Prof. Dept of Computer Science and Engineering in Annamalai University, Chidambaram, Tamil Nadu, India. E-mail: abi_dhana@rediffmail.com.*

Natural Language is the only speech recognition engine all over the world which manages voice chat dialogues with capability of changing the language during the conversation. Normal Human Language speech is also known to be as Natural Language speech. Natural Language Speech Recognition is an innovative technological platform for natural language processing and intelligent dialogue management. Natural Language Speech Recognition engine allows you to communicate with your clients virtually, in real time, by chat [5].

Speech Recognition Engine has the ability to answer questions and effectively provide appropriate answers as if it was an actual live human. Other technologies use simple key word detection. Natural Language Speech Recognition actually analyzes the entire phrase in order to promptly and effectively provides the appropriate outcome. Natural Language Speech Recognition has the ability to adapt to the callers request and make real time adjustments by using built in logic. This technology allows you to treat your customer's requests 24\7.

Speech synthesis is the artificial production of human speech. A computer system used for this purpose is called a speech synthesizer, and can be implemented in software or hardware [5],[6]. A text-to-speech (TTS) system converts normal language text into speech; other systems render symbolic linguistic representations like phonetic transcriptions into speech. Synthesized speech can be created by concatenating pieces of recorded speech that are stored in a database [6]. Systems differ in the size of the stored speech units; a system that stores phones or diaphones provides the largest output range, but may lack clarity. For specific usage domains, the storage of entire words or sentences allows for high-quality output. Alternatively, a synthesizer can incorporate a model of the vocal tract and other human voice characteristics to create a completely "synthetic" voice output.

In General there are three usual methods in speech recognition: Dynamic Time Warping (DWT), Hidden Markov Model (HMM) and Artificial Neural Networks (ANNs) [1]..

2 HISTORY

Long before electronic signal processing was invented, there were those who tried to build machines to create human speech. Some early legends of the existence of "speaking heads" involved Gerbert of Aurillac (d. 1003 AD), Albertus Magnus (1198–1280), and Roger Bacon (1214–1294). In the 1930s, Bell Labs developed the vocoder, which automatically analyzed speech into its fundamental tone and resonances. From his work on the vocoder, Homer Dudley developed a manually keyboard-operated voice synthesizer called The Voder (Voice Demonstrator), which he exhibited at the 1939 New York World's Fair.

The Pattern playback was built by Dr. Franklin S. Cooper and his colleagues at Haskins Laboratories in the late 1940s and completed in 1950. There were several different versions of this hardware device but only one currently survives. The machine converts pictures of the acoustic patterns of speech in the form of a spectrogram back into sound. Using this device, Alvin Liberman and colleagues were able to discover acoustic cues for the perception of phonetic segments (consonants and vowels).

In 1779, the Danish scientist Christian Kratzenstein, working at the Russian Academy of Sciences, built models of the human vocal tract that could produce the five long vowel sounds (in International Phonetic Alphabet notation, they are [a:], [e:], [i:], [o:] and [u:]). This was followed by the bellows-operated "acoustic-mechanical speech machine" by Wolfgang von Kempelen of Pressburg, Hungary, described in a 1791 paper. This machine added models of the tongue and lips, enabling it to produce consonants as well as vowels. In 1837, Charles Wheatstone produced a "speaking machine" based on von Kempelen's design, and in 1857, M. Faber built the "Euphonia". Wheatstone's design was resurrected in 1923 by Paget.[7]

In 1982, Kurzweil Applied Intelligence and Dragon Systems released speech recognition products. By 1985, Kurzweil's software had a vocabulary of 1,000 words—if uttered one word at a time. Two years later, in 1987, its lexicon reached 20,000 words, entering the realm of human vocabularies, which range from 10,000 to 150,000 words. But recognition accuracy was only 10% in 1993. Two years later, the error rate crossed below 50%.

Dominant systems in the 1980s and 1990s were the MITalk system, based largely on the work of Dennis Klatt at MIT, and the Bell Labs system, the latter was one of the first multilingual language-independent systems, making extensive use of natural language processing methods.

Dragon Systems released "Naturally Speaking" in 1997, which recognized normal human speech. Progress mainly came from improved computer performance and larger source text databases. The Brown Corpus was the first major database available, containing several million words. In 2006, Google published a trillion-word corpus, while Carnegie Mellon University researchers found no significant increase in recognition accuracy.

3 PERFORMANCE

The performance of speech recognition systems is usually evaluated in terms of Accuracy and Speed. In other words Speed and Accuracy are the criterion for measuring the performance of Speech Recognition[7].

3.1 Accuracy

Accuracy is usually rated with word error rate (WER), whereas speed is measured with the real time factor. Other measures of accuracy include Single Word Error Rate (SWER) and Command Success Rate (CSR). However, speech recognition (by a machine) is a very complex problem. Vocalizations vary in terms of accent, pronunciation, articulation, roughness, nasality, pitch, volume, and speed. Speech is distorted by a background noise and echoes, electrical characteristics [8].

Accuracy of speech recognition varies with the following:

1. Vocabulary size and confusability
2. Speaker dependence vs. independence
3. Isolated, discontinuous, or continuous speech
4. Read vs. spontaneous speech
5. Adverse conditions

- **Vocabulary is hard to recognize if it contains confusable words:**

For e.g. The 26 letters of the English alphabet are difficult to discriminate because they are confusable words (most notoriously, the E-set: "B, C, D, E, G, P, T, V, Z"); an 8% error rate **is considered good for this vocabulary.**

- **Speaker dependence vs. independence:**

A speaker-dependent system is intended for use by a single speaker. A speaker-independent system is intended for use by any speaker, more difficult.

- **Isolated, Discontinuous or continuous speech:**

With isolated speech single words are used, therefore it becomes easier to recognize the speech. With discontinuous speech full sentenced separated by silence are used, therefore it becomes easier to recognize the speech as well as with isolated speech. With continuous speech naturally spoken sentences are used, therefore it becomes harder to recognize the speech, different from both isolated and discontinuous speech.

- **Read vs. Spontaneous Speech:**

When a person reads it's usually in a context that has been previously prepared, but when a person uses spontaneous speech, it is difficult to recognize the speech because of the disfluences (like "uh" and "um", false starts, incomplete sentences, stuttering, coughing, and laughter) and limited vocabulary.

- **Adverse conditions:**

Environmental noise (e.g. Noise in a car or a factory) Acoustical distortions (e.g. echoes, room acoustics) Speech recognition is a multi-leveled pattern recognition task.

Word Accuracy:

Word accuracy metric is used to evaluate speech recognizers.

The percentage of Word Accuracy (WAcc) is measured by

$$\% \text{ WAcc} = 100 - (\%\text{WER})$$

Where,

WAcc - Word Accuracy.
WER – Word Error Rate.

The Value of WAcc can be Negative. It depends on the Percentage of WER.

3.2 Speed

Real time factor is parameter to evaluate speed of speech recognition [7]. If it takes time P to process an input of duration I, the real time factor is defined as

$$RTF = \frac{P}{I}$$

For e.g., Real time factor is 2 if it takes 6 hours of computation time to process a recording of duration 3 hours. RTF $\leq$ 1 implies real time processing [7].

4 ALGORITHMS

Generally there are three usual methods in speech recognition: Dynamic Time Warping (DWT), Artificial Neural Networks (ANNs) and Hidden Markov Model (HMM) [2].

4.1 Dynamic time warping

Dynamic time warping (DTW) is a technique that finds the optimal alignment between two time series if one time series may be warped non-linearly by stretching or shrinking it along its time axis [9]. This warping between two time series can then be used to find corresponding regions between the two time series [9], [10]. In Speech Recognition Dynamic time warping is often used to determine if two waveforms represent the same spoken phrase. This method is used for time adjustment of two words and estimation their adjustment of two words and estimation their difference [10]. In a speech waveform, the duration of each spoken sound and the interval between sounds are permitted to vary, but the overall speech must be similar. Main Problem of this system is little amount of learning words high calculating rate and large memory requirement [9].

4.2 Hidden Markov model

Hidden Markov Models are finite automates, having a given number of states; passing from one state to another is made instantaneously at equally spaces time moment. At every pass from one state to another, the system generates observations, two processes are taking place: the transparent one, represented by the observations string (feature sequence), and the hidden one, which cannot be observed, represented by the state string [11], [12]. Main point of this method is timing sequence and comparing method [1].

A hidden Markov model (HMM) is a statistical Markov model in which the system being modeled is assumed to be a Markov process with unobserved (hidden) states. An HMM can be considered as the simplest dynamic Bayesian network. The mathematics behind the HMM was developed by L. E. Baum and coworkers. It is closely related to an earlier work on optimal nonlinear filtering problem (stochastic processes) by Ruslan L. Stratonovich, who was the first to describe the forward-backward procedure [11].

In a regular Markov model, the state is directly visible to the observer, and therefore the state transition probabilities are the only parameters. In a hidden Markov model, the state is not directly visible, but output, dependent on the state, is visible. Each state has a probability distribution over the possible output tokens. Therefore the sequence of tokens generated by an HMM gives some information about the sequence of states. Note that the adjective 'hidden' refers to the state sequence through which the model passes, not to the parameters of the model; even if the model parameters are known exactly, the model is still 'hidden'[12],[13].

Hidden Markov models are especially known for their application in temporal pattern recognition such as speech, handwriting, gesture recognition, part-of-speech tagging, musical score following, partial discharges and bioinformatics[11],[12],[13]. A hidden Markov model can be considered a generalization of a mixture model where the hidden variables (or latent variables), which control the mixture component to be selected for each observation, are related through a Markov process rather than independent of each other [12].

1) Description in terms of urns:

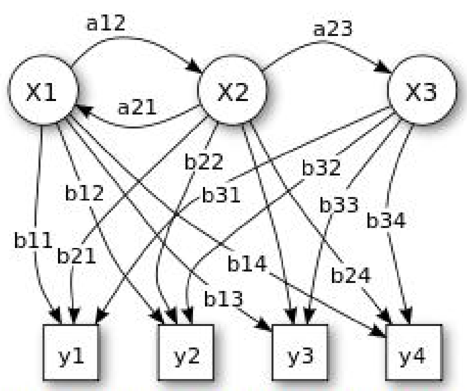

Figure 1. Probabilistic Parameters of HMM(Example

Where,
x — states
y — possible observations
a — state transition probabilities
b — output probabilities

In its discrete form, a hidden Markov process can be visualized as a generalization of the Urn problem: A genie is in a room that is not visible to an observer. In this hidden room there are urns X1, X2, X3, ... each of which contains a known mix of balls, each ball labeled y1, y2, y3, The genie chooses an urn in that room and randomly draws a ball from that urn. It then puts the ball onto a conveyor belt, where the observer can observe the sequence of the balls but not the sequence of urns from which they were drawn. The genie has some procedure to choose urns; the choice of the urn for the n^{th} ball depends only upon a random number and the choice of the urn for the $(n-1)^{th}$ ball. The choice of urn does not directly depend on the urns chosen before this single previous urn; therefore, this is called a Markov process. It can be described by the upper part of Fig.1.

The Markov process itself cannot be observed, and only the sequence of labeled balls can be observed, thus this arrangement is called a "hidden Markov process", where one can see that balls y1, y2, y3, y4 can be drawn at each state. Even if the observer knows the composition of the urns and has just observed a sequence of three balls, e.g. y1, y2 and y3 on the conveyor belt, the observer still cannot be sure which urn (i.e., at which state) the genie has drawn the third ball from. However, the observer can work out other details, such as the identity of the urn the genie is most likely to have drawn the third ball from[11],[13].

2) Architecture

The diagram below shows the general architecture of an instantiated HMM. Each oval shape represents a random variable that can adopt any of a number of values. The random variable x(t) is the hidden state at time t (with the model from the above diagram, $x(t) \in \{x1, x2, x3\}$). The random variable y(t) is the observation at time t (with $y(t) \in \{y1, y2, y3, y4\}$). The arrows in the diagram (often called a trellis diagram) denote conditional dependencies.

From the diagram, it is clear that the conditional probability distribution of the hidden variable x(t) at time t, given the values of the hidden variable x at all times, depends only on the value of the hidden variable $x(t-1)$: the values at time $t-2$ and before have no influence. This is called the Markov property. Similarly, the value of the observed variable y(t) only depends on the value of the hidden variable x(t) (both at time t).

In the standard type of hidden Markov model considered here, the state space of the hidden variables is discrete, while the observations themselves can either be discrete (typically generated from a categorical distribution) or continuous (typically from a Gaussian distribution). The parameters of a hidden Markov model are of two types, transition probabilities and emission probabilities (also known as output probabilities). The transition probabilities control the way the hidden state at time t is chosen given the hidden state at time t-1.

The hidden state space is assumed to consist of one of N possible values, modeled as a categorical distribution. (See the section below on extensions for other possibilities.) This means that for each of the N possible states that a hidden variable at time t can be in, there is a transition probability from this state to each of the N possible states of the hidden variable at time t+1, for a total of N2transition probabilities. Note that the set of transition probabilities for transitions from any given state must sum to 1. Thus, the N2 matrix of transition probabilities is a Markov matrix. Because any one transition probability can be determined once the others are known, there are a total of N (N-1) transition parameters.

ISSN 2229-5518

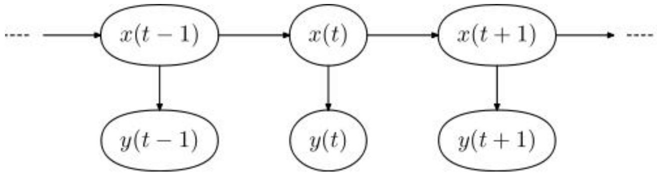

Figure 2. Possible States in HMM

In addition, for each of the N possible states, there is a set of emission probabilities governing the distribution of the observed variable at a particular time given the state of the hidden variable at that time. The size of this set depends on the nature of the observed variable. For example, if the observed variable is discrete with M possible values, governed by a categorical distribution, there will be M-1 separate parameters, for a total of N(M-1) emission parameters over all hidden states. On the other hand, if the observed variable is an M-dimensional vector distributed according to an arbitrary multivariate Gaussian distribution, there will be M parameters controlling the means and M(M+1)/2 parameters controlling the covariance matrix, for a total of emission parameters.

$$N(M + \frac{M(M+1)}{2}) = NM(M+3)/2 = O(NM^2) \qquad (1)$$

(In such a case, unless the value of M is small, it may be more practical to restrict the nature of the co-variances between individual elements of the observation vector, e.g. by assuming that the elements are independent of each other, or less restrictively, are independent of all but a fixed number of adjacent elements).

4.3 Artificial Neural Networks

Artificial Neural Networks (ANNs) are utilized in wide ranges for the parallel distributed processing, distributed memories, error stability and pattern learning distinguishing ability [14]. The complexity of all these the systems increased when their generality rises. The biggest restriction of two first methods is their low speed for searching and comparing the models. But ANNs are faster, because output is resulted from multiplication of adjusted weights in present input [15].

ANNs performance depends on their pattern classification method. So Before feeding this speech data to the ANN for between itself should give a relatively high value than the correlation to the other digits. A high value in correlation signifies the similarity and a smaller value signifies that there is a marked difference in data allowing the ANN to easily classify the data [14], [16].

5 PEOPLE WITH DISABILITIES

People with disabilities can benefit from speech recognition software is used to automatically generate a closed captioning of conversation such as discussions in conferences rooms, class rooms etc., [17].

Speech Recognition is also very useful for people who have difficulty using their hands, ranging from mild repetitive stress injuries to involved disabilities that preclude using conventional computer input devices [17]. In fact, people who used the keyboard a lot and developed RSI became an urgent early market for speech recognition. Speech Recognition is used in deaf telephony, such as voicemail to text, relay services and captioned telephone [18]. Individual with learning disability that has problems with thought-to-paper communication (essentially they think of an idea but it is processed incorrectly causing it to end up differently on paper) can benefit from the software [17].

6 ROLE OF SPEECH RECOGNITION FOR IMPAIRED PERSONS

6.1 Captioned Telephones

A new method for people who are Hard-of-Hearing, Oral Deaf or Late–Deafened to make phone calls is called a captioned telephone (also called captioned relay or CapTel). It is a telephone that displays real-time captions of the current conversation [18].

Figure 3. Captioned Telephone

The captions are typically displayed on a screen embedded into the telephone base. A captioned telephone may also be called a CapTel, which is the main brand name for a captioned telephone. A CapTel can also function exactly like a VCO by switching the device to VCO mode, for example, to communicate with an HCO user directly, without relay.

6.2 Blind Impairment

People who unable to see or difficulty in seeing is called as blind or visual impairment. Blind people use many products that are speech enabled such as talking watches, talking calculators and talking computers. Talking scales, talking compasses and talking thermometers are also available. Talking computers use screen reading software (sometimes referred to as Text-to-Speech programs) to have the machine read to blind people. They also use products with Braille feedback, such as Braille watches and Braille writing devices.

6.3 Visually Impaired People

Visually impaired people are the people who have eye problems but still have some sight, have computers which have enlarged screens so that images and text are much clearer to read [17].

6.4 Deaf or Hearing Impairment

Deaf or hearing impairment peoples who is unable to hear or difficulty in hearing. Technologies to assist the Deaf and hearing impaired include closed captions on television and the TTY/TDD phone service. Also, blinking lights and vibration devices also help to enhance hearing impaired functioning in a predominantly hearing world [18]. Some technologies not specifically designed as adaptive have become popular with the Deaf: for example, devices such as text-messaging-equipped cellular phones and Black Berry e-mail devices are almost ubiquitous among young Deaf people. The hard of hearing may also use speech recognition software which converts spoken words into text which can either be read from a screen or converted into sign language/Braille [19].

6.5 Speech impairment

It is inability to speak, or difficulty speaking and being understood Speaking impaired people, those who have lost the ability to speak but can still hear, include but are not limited to people who have had strokes, other brain injury, or injury to the vocal cords through surgery or other insult. Computers may provide speech through speech synthesis (text-to-speech programs), and text-messaging-equipped mobile phones are also popular. A famous individual who has speech impairment and uses a keypad to communicate with other people is Stephen Hawking. Augmentative communicators give people who are non-verbal or speech impaired a "voice" enabling them to communicate through messages pre-recorded by others [17].

6.6 Co-ordination or dexterity impairment

Co-ordination or dexterity impairment means difficulty using hands or arms; for example, grasping or handling a stapler or using keyboard/mouse Mobility impairment: difficulty leaving the bed or moving around; for example, moving from one office to another or up and down stairs. This is a sip-and-puff device which allows a person with substantial disability to make selections and navigate computerized interfaces by controlling inhalations and exhalations. People with limited manual mobility have software which enables non-manual methods of computer use, such as eye-driven keyboarding or speech recognition software. Robotic arms are also in development and a number of low-fi assistive devices are available such as jelly buttons, head dobbers and sip-and-puff devices [18].

6.7 Other Disability

Other Disabilities including learning disabilities, developmental disabilities and all other types of disabilities. Dyslexia is perhaps the most common example of an "other disability" found in the workplace and in schools. A variety of software is available to enable persons with dyslexia to deal more effectively with reading and writing tasks. Scan and read software (ex: Kurzweil or E-Text Reader) allows people/students with reading disabilities to view the text on the computer screen as it "reads" aloud and highlights the word/sentence as it moves along. This allows students who cannot read efficiently to tackle reading assignments with speed and confidence. The software program Kurzweil can be very expensive, but for a student who has serious difficulty decoding the words on a page, it can be a great asset. Speech recognition software (such as Dragon Naturally Speaking) can be used to help students with writing disabilities write text to the computer without the worry of spelling phonetically. It can record just the way the person speaks the sentence. However, several hours of training are needed for the user, and the computer program must "learn" to recognize the speech patterns of the user.

7 Conclusion

Speech Recognition is widely used in industrial software market [20]. Researchers, working on the very promising and challenging field of Speech Recognition are bearing towards the ultimate goal i.e., Natural Conversation between Human beings and machines, are applying the knowledge from areas 99of Acoustic-Phonetics, Speech Perception, Artificial Intelligence etc., The challenges to the recognition performance of SR are being provided concrete solution so that the gap between recognition capability of machine and that a human being can be reduced to maximum extend [7]. Main goal of SR is to design a network which would be able to do continuous speech recognition on a larger vocabulary [20]. An attempt has been made through this paper give a comprehensive survey and uses of speech recognition. Speech Recognition mainly focused on Blind or visually impaired and Deaf or Hearing impaired.

References

[1] Aggarwal, R.K. and Dave, M., "Acoustic Modelling Problem for Automatic Speech Recognition System: Conventional Methods (Part I)", International Journal of Speech Technology (2011) 14:297–308.

[2] Song Yang, Meng Joo Er, and Yang Gao. "A High Performance Neural-Networks-Based Speech Recognition System", IEEE

trans.pp.1527,2001.

[3] Kimberlee A. Kemble "An Introduction to Speech Recognition",Voice Systems Middleware Education IBM Corporation.

[4] Digital Signal Processing Mini Project." An Automatic Speaker Recognition System", 14 June 2005.

[5] Samoulian, A., "Knowledge Based Approach to Speech Recognition", 1994.

[6] "ECE3Speech Recognition." A Simple Speech Recognition Algorithm.15 April 2003. 1 July 2005.

[7] Wiqas Ghai & Navdeep Singh "Literature Review on Automatic Speech Recognition"International Journal of Computer Applications (0975 – 8887) Volume 41– No.8, March 2012 42

[8] Jain, R. And Saxena, S. K., "Advanced Feature Extraction & Its Implementation In Speech Recognition System", IJSTM, Vol. 2 Issue 3, July 2011.

[9] "Isolated Word, Speech Recognition using Dynamic Time Warping." 14 June 2005.

[10] "Speech Recognition by Dynamic Time Warping.", 20 April 1998. 06 July 2005.

[11] Corneliu Octavian DUMITRU, Inge GAVAT. "Vowel, Digit and Continuous Speech Recognition Based on Statistical, Neural and Hybrid Modelling by Using ASRS_RL ". EUROCON 2007, The International Conference on "Computer as Tool", pp.858-859.

[12] Gavat, O.Dumitru, C. Iancu, Gostache, "Learning strategies in speech Recognition", Proc. Elmar 2005, pp.237-240, june 2005,Zadar, Croatia.

[13] J. Bilmes: "A Gentle Tutorial on the EM Algorithm and its Application to Parameter Estimation for Gaussian Mixture and Hidden Markov Models", Technical Report, University of Berkeley, ICSI-TR-97-021. April 1998. 01 August 2005.

[14] Bahlmann. Haasdonk. Burkhardt. "speech and audio recognition" IEEE trans. Vol 11. May 2003.

[15] Edward Gatt, Joseph Micallef, Paul Micsllef, Edward Chilton. "Phoneme Classification in Hardware Implemented Neural Networks "IEEE trans, pp.481, 2001

[16] "Speech Recognition Based on Statistical, Neural and Hybrid Modelling by Using ASRS_RL ". EUROCON 2007, The International Conference on "Computer as Tool", pp.858-859.

[17] Leitch D., MacMillan T., Year III Final Research Report on the Liberated Learning Project "How Students with Disabilities Respond to Speech Recognition Technology in the University Classroom", July 2002.

[18] Language: "Implications for Deaf Readers". Journal of Deaf Studies and Deaf Education 5(1). Winter 2000. 32-50.

[19] Dutch Television Programs: "A Text Linguistic Approach". Journal of Deaf Studies and Deaf Education 10(4). Fall 2005. 402-416.

[20] Meysam Mohamad pour, Fardad Farokhi, "An Advanced method for Speech Recognition", *World Academy of science, Engineering and Technology 49 2009.*

S. Ananthi received her B.E Degree in Computer Science from Dhanalakshmi Srinivasan Engineering College, Perambalur and M.E Degree in Computer Science from Annamalai University, Chidambaram. Currently, she is pursuing her PhD in Computer Science and Engineering in Annamalai University, Chidambaram. She had presented a paper about Stegnography in National Conference on Multimedia Signal Processing, NCMSP' 2011 in Annamalai University on 2011. She had participated in the National Conference on Confluence of Multidisciplinary in Engineering Fields NCCME'12. She had published a paper in International Journal of Engineering and Innovative Technology (IJEIT). She had published a paper in International Organization of Scientific Research (IOSR) International Journal on the topic of Stegnography. Her research interests include speech processing, cryptography and stegnography.

Dr. P. Dhanalakshmi received her Bachelor's degree in Computer Science and Engineering from Government College of Technology, Coimbatore in the year 1993. She received her M.Tech degree in Computer Applications from the reputed Indian Institute of Technology, New Delhi under the Quality Improvement Programme in the year 2003. She completed her Ph. D in Computer Science and Engineering from Annamalai University in the year 2011. She joined the services of Annamalai University in the year 1998 as a faculty member and is presently serving as Associate Professor in the Department of Computer Science & Engg. She has published 11 papers in international conferences and journals. She is guiding several students who are pursuing doctoral research. Her research interests include speech processing, image and video processing, pattern classification and neural networks.