CHAPTER 1

Overview of Speech Research

1.1 Introduction

Since the early days of computing, researchers are trying to design machines, which could understands human speech. The interest towards this goal is due to the very fact that - speech comes most naturally to everyone and perhaps the easiest way of communication. But at the same time, it is one of the most complex signals to handle. During the last few decades researchers are putting strong efforts in this area with a common consensus that voice recognition may become the next primary user interface. The output of Automatic Speech Recognition (ASR) Technology can be utilized in many applications with a diverse advantage to different professionals like Doctors, Engineers, Lowers, Teachers, and Students etc. Automatic speech recognition technology permits human speech signals to be used to carry out predefined activities. Once the system detects and recognizes a sound or string of sounds, the recognizer can be programmed to perform according to a predetermined goal.

Besides speaking face-to-face, speech can also be propagated by other means. In today's world, speech communications can be thought of in the area of telephony, recording systems (cassettes, CDs, DVDs, and their players), the internet, and many other frontiers of digital world. The design and operation of such systems requires knowledge of the characteristics of human speech in order to effectively and efficiently convey vocal content. Unlike telecommunication signals, which are designed in such a way that loading/unloading them with information can easily be done by a bunch of electronic equipments bundled into a simple device, it is very difficult to understand how a human brain sends information to another human brain[1]. Speaking and listening has come up as the primary means of communication among human although other means of communication techniques, such as **sign language**, are available. Speaking is achieved through speech. The basic building blocks of the speech of any language are a set of sounds named as **phonemes.** Without knowing the complexity of speech production, we learn this technique of communication since our early childhood. Even with differences in term of **accent, articulation, nasality, roughness, volume, pitch, pronunciation** and **speed**, we are still able to interpret the speech most of the time as long as the spoken language is the language that we are familiar with. Since the early childhood our brain learns a spoken language or speech unconsciously. Children learn the basic phonemes within the first year of their birth [2]. Gradually, they start to learn the meaning of words and subsequently followed by the development of their vocal tract until they start to understand words and able to pronounce them correctly. Further development continues until the child is able to utter sequences of words to form complete or semi-complete sentences. It is understood that the learning of correct grammar, adaptation to different speakers and environment and even learning of different languages is obvious in the life span of a human being.

With the advent of Speech Technology, the Scientists and Speech researchers are putting strong efforts to produce an efficient speech recognizer so that a natural human-machine interface would be possible and will replace the primitive interfaces, such as keyboard and mouse for the computer. This would certainly fuel the human-machine interface and interaction in terms of some valuable applications in our life, such as making jobs done easier and effective. For examples, language translation machine, smart-home controller and telephone directory assistance, agricultural assistance improves the quality of living for human. Because of the glamour of designing an intelligent machine that can recognize the spoken language, studies have been done in various fields to achieve this goal. To design a successful speech recognition system, expertise and knowledge from a wide range of disciplines are required. Some of the disciplines that have been widely applied to solve the speech recognition problems are: **Signal processing, acoustics, communication theory, computer science**, and **pattern recognition** etc. Since human brain is efficient in speech recognition, researches have been motivated to build brain-like computational methods.

1.2 Speech Production Mechanism

Human speech is produced by vocal organs as depicted in **Figure 1.1.** As shown in the figure, the main energy source is the **lungs** with the diaphragm. When speaking, the air flow is forced through the **glottis** between the **vocal cords** and the **larynx** to the three main cavities of the **vocal tract**, the **pharynx** and **the oral** and **nasal** cavities. From the oral and nasal cavities the air flow exits through the nose and mouth, respectively. The V-shaped opening between the vocal cords, called the **glottis**, is the most important sound source in the vocal system. The vocal cords may act in several different ways during speech. The most important function is to modulate the air flow by rapidly opening and closing, causing buzzing sound from which vowels and voiced consonants are produced.

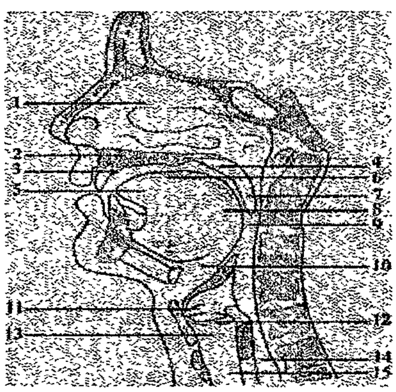

Figure. 1.1 The human vocal organs. (1) Nasal cavity, (2) Hard palate, (3) Alveoral ridge, (4) Soft palate (Velum), (5) Tip of the tongue (Apex), (6) Dorsum, (7) Uvula, (8) Radix, (9) Pharynx, (10) Epiglottis, (11) False vocal cords, (12) Vocal cords, (13) Larynx, (14) Esophagus, and (15) Trachea.

The fundamental frequency of vibration with men, women, and children is about 110 Hz, 200 Hz, and 300 Hz, respectively [82]. This might, of course, varies based on different factors like mass and tension. For example, with stop consonants e.g. **\b\, \d\, \g\ etc,** the vocal cords may act suddenly from a completely closed position in which they cut the air flow completely, to totally open position producing a cough like phenomenon or a glottal stop. On the other hand, with unvoiced consonants, such as in case of **/s/ or /f/,** they may be completely open. An intermediate position may also occur with for example phonemes like **/h/.**

Critically, the vocal system may be considered as a single acoustic tube between the glottis and mouth. Glottal excited vocal tract may be then approximated as a straight pipe closed at the vocal cords where the acoustical impedance Zg= ∞ and open at the mouth (Zm = 0). In this case the **volume-velocity transfer function, V (w),** [3, 4], of vocal tract can be represented as given in equation (1.1)below :

$$V(\omega) = \frac{Z_m}{Z_g} = \frac{U_m}{U_g} = \frac{1}{\cos(\frac{\omega l}{c})} \qquad \textit{(1.1)}$$

where,

l = length of the tube,
ω = frequency in radian,
C = sound velocity.

The denominator is zero at frequencies $F_i = w_i/2p$ $(i=1,2,3,...)$, where

$$\frac{\omega_i l}{c} = (2i-1)\frac{\pi}{2}, \text{ and } F_i = \frac{(2i-1)}{4l}, \qquad \textit{(1.2)}$$

If l=17 cm, $V(\omega)$ is infinite at frequencies F_i = 500, 1500, 2500,... Hz, which means resonances every 1 kHz starting at 500 Hz. If the length l is other than 17 cm, the frequencies Fi will be scaled by factor ***(17/l)***. *In such cases* the vocal tract may be approximated with two (**in case of vowels**) or three (**in case of consonants**) sections of tube where the areas of adjacent sections are quite different and resonances can be associated within individual cavities. The two tube model approximated for vowels and consonants are shown in the fig. 1.2 and Fig. 1.3 respectively. For example, with vowel /a/ the narrower tube represents the pharynx opening into wider tube representing the oral cavity. If assumed that both tubes have an equal length of 8.5 cm, formants occur at twice the frequencies noted earlier for a single tube. Due to acoustic coupling, formants do not approach each other by less than 200 Hz so formants F1 and F2 for /a/ are not both at 1000 Hz, but rather 900 Hz and 1100 Hz, respectively [4].

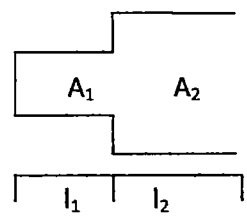

Figure 1.2 Two tube model for vocal tract (for vowels).

In the same way, the consonants can be approximated with a three-tube model shown in Figure 1.3, where the narrow middle tube represents the constriction of the vocal tract.

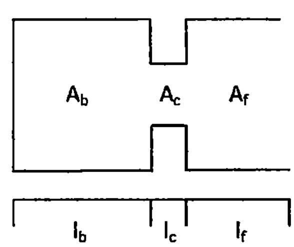

Figure 1.3 Three tube model for vocal tract (for consonants).

The back and middle tubes are half-wavelength resonators and the front tube is a quarter-wavelength resonator with resonances

$$\frac{ci}{2l_b}, \frac{ci}{2l_c}, \frac{c(2i-1)}{2l_f}, \quad \text{for } i = 1, 2, 3 \ldots.. \qquad (1.3)$$

where *lb*, *lc*, and *lf* are the length of the back, center, and front tube, respectively. With the typical constriction length of 3 cm the resonances occur at multiples of 5333 Hz and can be ignored in applications that use less than 5 kHz bandwidth [4].

Using the vocal organs, speech sounds are produced by two essential and one optional functional process namely, ***initiation***, ***articulation***, and/or ***phonation***.

1.2.1 Initiation

For speech sounds to occur air has to be present and it is usually provided by the lungs in our body. There are three types of initiation to the provision of air among all languages of the world [5]:

1. **Pulmonic,** which involve the lungs,
2. **Glottalic,** which involves the vocal cord or glottis, and
3. **Velaric,** which involves the tongue, and the velum or soft palate (located at the top inner part of the mouth).

1.2.2 Articulation

Articulation takes place to transform the airflow into acoustic elements that forms the different types of sounds. It can be performed by the **glottis, upper surface of the vocal tract, teeth, tongue, and/or lips**. Different methods of articulation exist for consonants and vowels. For consonants, air that flows from the initiation process is obstructed whereas for vowels, it remained relatively unobstructed. At most of these places of articulation, there are also various ways to articulate [6] such as:

Stop - where airflow is stopped by the articulators to prevent air from escaping the mouth.

1. **Nasal Stop (Nasal)** - Air is allowed to flow out of the nose by releasing the soft palate even though it is stopped in the oral cavity. Examples of

nasal stops are the beginning of English words '**me**' (bilabial closure), '**night**' (alveolar closure), and the end of word '**hang**' (velar closure). Another term used by phoneticians for nasal stops is ***"nasal"***.

2. **Oral Stop (Stop)** - Air is completely stopped in this case where no air flows out of the mouth or nose. In this case, air pressure is build up in the oral cavity and subsequently released in bursts. Examples of oral stops that occur at the beginning of English words are **p**it and **b**oy (bilabial closure), **t**ee and **d**ye (alveolar closure), **k**ite and **g**ird (velar closure). Words with oral stops at the end are 'te**d**' (alveolar stop) and 'da**n**' (alveolar nasal). The term *"stop"* commonly refers to oral stops.

- **Fricative** – Here, the turbulent airflow is created due to partial obstruction that arose from close proximity of two articulators.

  1. **Sibilants** - Sibilants are louder in intensity and have higher pitches. Examples of sibilants are **/s/** in '**s**ign' and **/z/** in '**z**oo'.
  2. **Non-sibilants** - They are softer and have lower pitches than sibilants. Some examples are **/th/** in '**th**ese' and **/f/** in '**f**it'.

- **Approximant** – They are similar to fricatives except that articulators are not as close as to producing a turbulent airflow. Examples of approximants are the beginning of '**y**ellow' and '**w**illow'.

- **Lateral (Approximant)** – Lateral approximant is produced by partial obstruction between one or both sides of the tongue and the roof of the mouth. Examples of laterals are the beginning of '**l**ie' and end of '**pale**'.

• **Affricate** - A stop followed by a fricative. An example is the beginning and end of '**church**' (palato-alveolar affricate).

• **Flap (Tap)** - A single tap by the tongue on the alveolar ridge. An example will be the middle of the word 'better' when it is pronounced quickly (more common in American English).

• **Trill (Roll)** - A repeating or trilling action of the 'r' sound. For vowels, the airflow is smoother than consonants where the obstruction is not as great. Articulation involves the tongue and the lips. There are three classes in which a vowel can be classified [6]:

1. **Position of tongue** - The position of the tongue's highest point within the mouth (e.g. **feet** (front), **the** (centre), and **good** (back)).
2. **Height of tongue** - The height of the body of tongue or the proximity between the tongue and the roof of the mouth (e.g. **beet** (high or close1), bit (mid-high or close-mid), **bed** (mid-low or open-mid), and **bad** (low or open)).
3. **Shape of lips** - How "rounded" are the lips (e.g. **feet** (unrounded), **hood** (rounded)).

1.2.3 Phonation

Phonation refers to the voicing of a sound which relates to the vibration of the **vocal cord** or **glottis.** Although phonation is optional in speech production, it occurs in a non-negligible fraction of speech sounds. Excluding whispers, all **English** and **Bodo** vowels are **voiced.**

1.3 Common Linguistic terms used in NLP

Some commonly used linguistic terms in NLP are as follows:

Syllable: Syllable is considered as the phonological building blocks of a word. It can influence the rhythm of a language, its prosody and stress. For example the word computer composed of three syllables namely **com**, **pu**, **ter** (tri-syllabic).

Phonemes: Phonemes are the linguistically contrastive or significant sounds (or sets of sounds) of a language. Such a contrast is usually demonstrated by the existence of **minimal pairs** or **Contrast in Identical Environment (C.I.E.)**. Minimal pairs are pairs of words which vary only by the identity of the **segment** (another word for a single speech sound) at a single location in the word (eg. **[mæt]** and **[hæt]**). If two segments contrast in identical environment then they must belong to different phonemes. A **paradigm** of minimal phonological contrasts is a set of words differing only by one speech sound. In most languages it is rare to find a paradigm that contrasts a complete **class** of phonemes (eg. all vowels, all consonants, all stops etc.).

Allophones: **Allophones** are the linguistically non-significant variants of each phoneme. In other words, a phoneme may be realized by more than one speech sound and the selection of each variant is usually conditioned by the phonetic environment of the phoneme. Occasionally, allophone selection is not conditioned but may vary from person to person and occasion to occasion (ie. **free variation**).

Morpheme: Morpheme is the smallest linguistic unit which bears a meaning or has a grammatical function. There are basically two types of morphemes they are **Free** and **Bound**. A morpheme is free if it is able to appear as a word by itself and it is bound if it can only appear as part of a larger, multi-morphemic word. Every morpheme is either Free or Bound. The Free morphemes are also referred as **roots**. Bound morphemes are also referred to as **affixes** which are popularly known as **prefixes, infixes**, and **suffixes**. In spoken languages morphemes are composed of **Phonemes** and in written language morphemes are composed of **Graphemes**.

Word : Word is a unit of language, consisting of one or more spoken sounds (syllable) or their written representation, that functions as a principal carrier of meaning. Words are composed of one or more morphemes and are either the smallest units susceptible of independent use or consist of two or three such units combined under certain linking conditions, as with the loss of primary accent that distinguishes "homework" from "home" and "work". Words are usually separated by spaces in writing, and are distinguished phonologically, as by accent, in many languages, and by tone in some languages like Bodo and Chinese.

Semantics : It is used as a technical term for the meaning of words and sentences.

Pragmatics: It is a technical term meaning, roughly, what the person speaking or writing actually meant, rather than what the words themselves mean. It also signifies how the contexts affect the meaning of the sentence.

Alveolar : It is a phone produced when the tongue touches the tooth ridge behind the teeth alveolus). The **'t'** sound' in English is an alveolar stop, produced by stopping and then releasing the air flow out of the mouth by closing the tongue onto the tooth ridge.

Fricative : If during the production of a phone, air is made to pass through a narrow passage, a **'friction'** sound or fricative is produced (i.e. a more-or-less 'hissing' sound). English examples are the 'f sound' in "fee" or the '**sh** sound' in "**she**".

Intonation : Intonation refers to changes in the tone or frequency of sounds during speech. For example, in English the tone usually falls at the end of a statement and rises at the end of a question, so that "You want some coffee" and "You want some coffee?" can be distinguished by tone alone. In some languages (e.g. Chinese, Thai, Bodo), sequences containing the same phones but with different intonation patterns correspond to different words.

1.4 Problem Statement

Keeping in view of the above findings and statements the present research problem is formulated as below:

A language may or may not exhibit word-level prominence, by which it is mean that one or more syllables within a word are more prominent than others. Moreover, syllables may be prominent because of either lexical or measurable stress. Whether lexically or measurable assigned, prominence at word level may be realized by one or a combination of multiple acoustic cues. Which cues are utilized is language specific. In English, for example, a stressed syllable of a word may have a pitch excursion and it may be louder than the unstressed syllables. Besides pitch and loudness, duration is another possible cue; a stressed syllable may be longer than unstressed ones. In the present study all these acoustic variables are examined to isolate this relevance in Bodo language.

1.5 Man-Machine Communication and Speech Input

The field of Speech Processing Technology can be broadly categorized into the following five categories:

a) **Speech Coding:** It is a technique of compressing the information into a voice signal so that it can be stored efficiently or transmitted over a channel at a smaller bandwidth [8]. Speech coding is very important as it compresses the original signal and finds the importance in the field of telecommunication and other types of network communications. The digital telephone answering machine also relies heavily on speech coding in which both voice prompt and voice message are compressed and stored in the machine's local memory.

b) **Speech Synthesis:** It is the process of replicating a speech signal to transmit a message from a device to a person [9]. A number of significant application areas usage it as for example, IVR systems used in Customer Care Service of Telecommunication, Remote Student Registration etc.

c) **Speaker Recognition:** This system can be defined as the process of identifying or verifying a speaker by using the voice characteristics of an individual. Speaker Recognition Technology is one of the areas where a computer system can give more accurate and timely response as compared to a human [9]. An application area of this system normally includes the systems which involves security. A commonly used application area is banking transaction taking place over voice communication.

d) **Speech Recognition:** Speech Recognition is a process by which a computer system understands the voice input given to it and response back. In this case, voice is considered as an input mechanism or command.

e) **Spoken Language Translation:** Spoken Language Translation seems to be the most complex process among all and has significant advantages [9]. Here, two speaker can communicate without the knowledge of each other's language. The computer system works as a two way interpreter. It accepts one language as input, understands the content within the message and synthesizes back to the second language.

1.6 Speech Recognition Technique

The goal of most of the speech recognition system is to design a model, which can recognize human speech. However, computer based systems do not yet have the capability and flexibility of understanding speech as human does. Human can recognize the sound of interest from a set of sounds audible concurrently. But in a computer based system, the other sounds which are not of interest will be considered as noise. That is why, improvement in the computer based systems is necessary to make the system as robust as possible.

In speech recognition systems, phonemes and syllables are the most popular sub-word units used [7, 8]. Automatic speech recognition systems are normally designed for three types of utterances. They are:

a) Isolated word recognition,

b) Connected word recognition, and

c) Connected speech recognition also termed continuous speech recognition

The first type of recognition is the easiest one. In this case speaker needs to pause automatically in between words. Connected word recognition is capable of analyzing a string of words spoken together, but at a limited speech rate. Connected speech recognition or continuous speech recognition allows the system to recognize normal conversational speech. In this type of system, the system is needed to be trained. Such systems are known as **Speaker Dependent System**. At the same time there are certain systems which are not required to be trained and are termed as Speaker Independent System. The main goal of the ASR is the understanding continuous speech, natural or conversional speech. However, in the case of conversional speech, which is spoken in a natural flow, it is very difficult to recognize the words or phrase as there is a very negligible pause or even no pause between words or phrase. So, to distinguish words or phrases, the recognizer need to apply the concept of "**guessing**", where the statistical analysis takes place to produce the most likely words or phrases to produce a correct sentence. A general system for training and recognition of continuous speech is depicted in the figure 1.4 [10].

1.7 Applications

It is one of the difficult and complicated tasks to determine when voice would be a preferred medium of human computer interaction [13]. Scientists and researchers however have identified a number of situations in which spoken communications with machines would be advantageous.

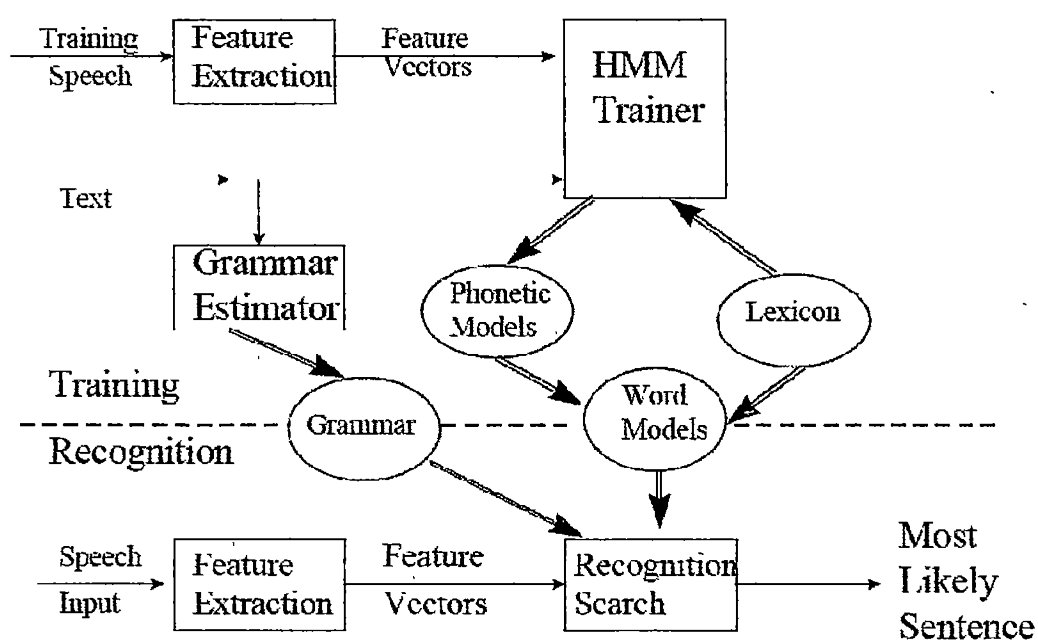

Figure 1.4: General system for training and recognition [9]

Some of the common situations are - when the user's hands or eyes are busy, a limited keyboard and/or screen is available, disabled users, when any individual is not acquainted with the computer system or when the natural language is preferred [13]. Spoken interaction with machines is a situation in which a user's hands' and/or eyes are busy performing another task. When users are able to use speech to communicate with a machine, they are free to pay attention to their other important tasks. It has been noticed that a highly accurate speech recognition systems where speech input can be given, leads to higher task productivity and accuracy [13]. Some of the important application areas are -

1.7.1 Application for the Blind

As the blind cannot communicate with a computer system, like a normal man do, speech synthesis application seems to be very useful for them. Earlier, before the development of synthetic speech system, to help the blind people, the content of the book were recorded in audio form, which can be listen by the blind. However, the main drawback appears, when the size of the book becomes large. It takes several months to record resulting in high expense. It is also easier to get information from computer with speech, using special **bliss symbol** keyboard, which is an interface for reading **baille** characters.

When the output from a speech synthesizer is listened for the first time, it may sound intelligible and pleasant. However, during longer listening period, single clicks or other weak points in the system may arise very annoying. This is called an annoying effect and it is difficult to perceive with any short-term evaluation method, so for these kinds of cases, the feedback from long-term users is sometimes very essential.

1.7.2 Applications for the Deafened and Vocally Handicapped

People who are born-deaf cannot learn to speak properly and people with hearing difficulties have usually speaking difficulties. Synthesized speech gives the deafened and vocally handicapped an opportunity to communicate with people who do not understand the sign language. With a talking head it is possible to improve the quality of the communication situation even more because the visual information is the most important with the deaf and dumb. A speech synthesis system may also be used with communication over the telephone line [11].

With keyboard it is usually much slower to communicate than with normal speech. One way to speed up this is to use the predictive input system that always displays the most frequent word for any typed word fragment, and the user can then hit a special key to accept the prediction. Even individual pre-composed phrases, such as greetings or salutes, may be used.

1.7.3 Educational Applications

Synthesized speech can be used also in many educational areas and situations. A computer with speech synthesizer can teach 24 hours a day and 365 days a year. It can be programmed for special tasks like spelling and pronunciation teaching for different languages. It can also be used with interactive educational applications.

Especially with people who are impaired to read (dyslexics), speech synthesis may be very helpful because especially some children may feel themselves very embarrassing when they have to be helped by a teacher [11]. It is also almost impossible to learn write and read without spoken help. With proper computer software, unsupervised training for these problems is easy and inexpensive to arrange.

A speech synthesizer connected with word processor is also a helpful aid to proof reading. Many users find it easier to detect grammatical and stylistic problems when listening than reading. Normal misspellings are also easier to detect.

1.7.4 Applications for Telecommunications and Multimedia

Speech synthesis also finds applications in the area of multimedia. Synthesized speech has been used for decades in all kind of telephone enquiry systems, but the quality has been far from good for common customers. Today, the quality has reached the level that normal customers are adopting it for everyday use.

Electronic mail has become very usual in last few years. However, it is sometimes impossible to read those E-mail messages. There may be no proper computer available or some security problems exist. With synthetic speech e-mail messages may be listened to via normal telephone line. Synthesized speech may also be used to speak out short text messages (**sms**) in mobile phones.

For totally interactive multimedia applications commonly known as Interactive Voice Response (IVR) [12] an automatic speech recognition system is also needed. The automatic recognition of fluent speech is still far away, but the quality of current systems is at least so good that it can be used to give some control commands, such as yes/no, on/off, or ok/cancel etc.

1.7.5 Other Applications and Future Directions

Speech synthesis, in principle, may be used in all kinds of man-machine interactions. For example, in warning and alarm systems synthesized speech may be used to give more accurate information of the current situation. Using speech instead of warning lights or buzzers gives an opportunity to reach the warning signal for example from a different room and also with proper understanding. Speech synthesizer may also be used to receive some desktop messages from a computer, such as printer activity or received e-mail.

In the future, if speech recognition techniques reach sufficient level, synthesized speech may also be used in language interpreters or several other communication systems, such as videophones, videoconferencing, or talking mobile phones. If it is possible to recognize speech, transcribe it into ASCII string, and then re-synthesize it back to speech, a large amount of transmission capacity may be saved. With talking mobile phones it is possible to increase the usability considerably for example with visually impaired users or in situations where it is difficult or even dangerous to try to reach the visual information. It is obvious that it is less dangerous to listen than to read the output from mobile phone for example when driving a car.

During last few decades the communication aids have been developed from talking calculators to modern three-dimensional audiovisual applications. The application field for speech synthesis is becoming wider all the time which brings also more funds into research and development areas.

1.8 Organization of the Thesis

The thesis is organized into seven chapters. **Chapter 1** gives an overview of Speech Research. Speech Production mechanism has been discussed in this chapter supported by mathematical formulation. The chapter also discusses about some commonly used linguistic terms in **Natural Language Processing (NLP)** in detail.

Chapter 2 is about literature review. The chapter begins with the speech synthesis process and its types. Different techniques which are commonly used in Speech Recognition Process have also been introduced in this chapter. The Speech Research works that have been carried out across the globe, starting from 1950's has been discussed in the remaining part of the chapter.

Chapter 3 introduces about Bodos and Bodo Language. The history of Bodo Script and different Bodo dialects has been elaborated in this chapter. The characteristic of Bodo phonemes along with their occurrences at different position (initially, medially and finally) is depicted in this chapter. Special characteristics of Bodo Language in terms of **Tone, Prosody** and **Intonation** are also discussed in this chapter.

Chapter 4 illustrates the acoustical properties of Bodo phonemes and words of CV, VC, CVC types. The vowels and words are studied in terms of **Mel-Frequency Cepstral Coefficients (MFCC)** in this chapter.

Chapter 5 is devoted to the study of Bodo vowels and words in terms of **Formant Frequency**. The vowels and words are studied with respect to **First Formant (F1)**, **Second Formant (F2)** and **Third Formant (F3).**

Chapter 6 is solely devoted to the study of Prosodic feather of Bodo Language. Acoustical correlation of Prosody is one of the important topic of discussion in this chapter. The syllable constituents and different types of syllables present in Bodo language and their acoustic study in terms of change of pitch across different syllables are studied in this chapter. Various methods (**CPD**, **SIFT**, **ACF**, **AMDF** etc) available for the study of pitch has been mentioned in this chapter. For the current study **ACF** and **AMDF** are used.

Chapter 7 presents the concluding remarks based on the present study. Finally, the possible benefits which can be acquired by further extending the present study have been discussed.