Urdu Text to Speech Synthesizer

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(June, 2018)

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Urdu Text to Speech Synthesizer

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Evaluation of M. Phil. Thesis

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We are satisfied with the thesis and performance of the candidate in the examination and are of the opinion that she fulfills the requirements as set in the rules and regulations for the M.Phil. degree in Computer Science at the University of the Punjab.

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Abstract

Text to speech synthesis system is system which takes raw text as input and converts it into speech signal. This is done by concatenation of small speech segments called phonetic strings of words or Statistical parametric speech synthesis which uses parameters to describe speech. In this technique, model is learned from speech data using Hidden Markov model (HMM) or Deep Neural Networks (DNN).

This paper describes development of Festival TTS system based Urdu text to speech system using Hidden Markov model (HMM). It describes Urdu text preprocessor system used to process numbers, dates and time text in input data and how Festvox voice package is generated for Urdu. In the end, evaluation of system is conducted using DRT, MRT and MOS tests to get performance of the system.

Keywords: Text to Speech, Urdu Text Preprocessor, Hidden Markov model, Festival, Festvox

Acknowledgements

Computational modeling is branch of computer science which deals with multiple disciplines. It assists other domains in understanding complex systems and phenomena by providing theory, tools and technology to model and simulate related systems and phenomena. In complex systems, behavior of an individual can have butterfly effect and can become root cause of an emergent phenomenon. Interaction of drivers with each other and surrounding environment forms the dynamics of traffic flow. Hence global effects of a traffic flow depend upon behavior of a single driver. In this research.

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Chapter 1

Introduction

1.1 Speech synthesis

Speech is the most important medium of conveying opinions and expressing feelings and thoughts. Human convert their thoughts into speech by using words, phrases and sentences in order to communicate [1]. Speech is produced when air is exhaled by the lungs and vibrations are produced by air, these vibrations get a proper waveform shape by glottal cords and vocal tract. Text to Speech synthesis is the process of conversion of raw text into speech signals. It works by concatenation of small segments of recorded speech called phonemes [2]. Speech data is obtained by first recording natural speech by using some type of recording systems and is converted to digital form. The digital data is sampled and stored in computer, after that passed back to analog signals and is converted back to speech [3].

Text to speech systems are becomming important as they can be used in machines to effectively transmit information to human using artificial speech as information exchange through computers has become the integral part of new era. Visually impaired people usually suffer while using computer technology when there is no assistant or computer is not enough interactive which makes text to speech systems necessity of modern life. These systems increase the degree to which blind people can interact with sighted people [4] and could boost up their hope to survive in this world gracefully [5]. Many applications of speech synthesis are emerging such as machines that read for blinds, aids for handicaps, computers that interact with user through speech. For all these applications a text to speech that convert text to speech are used [6].

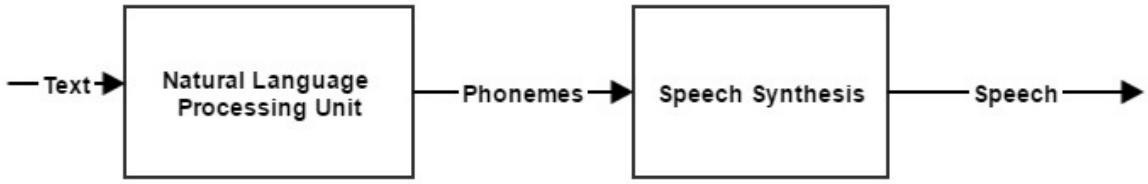

Figure 1.1: TTS Block Diagram

The TTS system comprises of two main stages. One is called Natural language Processing (NLP) and other is called Speech Synthesis (SS). This is shown in figure 1.1.

In NLP unit, text is first converted into string of letters and then word boundaries are marked by tokenizer. This is called normalization of text. Normalized data is then converted into phonetic strings with the help of letter to sound rules after which syllabifier marks syllable boundaries. Sound change rules are applied on the syllabified data. Language modeling techniques are also applied for finding context in which a specific word is used. As human has tendency to recognize basic rules for his native language, it is easy to judge context of a word in a sentence and what should be correct pronunciation of that word with respect to its context. For example, it can be guessed easily that "پل" in a sentence is used for پل (moment) or پل (bridge) for any native speaker of Urdu. But for computer, we need to mark context of each word according to data. Last stage of NLP is stress intonation marker which adds stress and intonation to the text. Speech Synthesis unit converts symbolic information received from NLP unit into audible speech with the help of different Digital Signal Processing techniques. The quality of speech synthesis system is detected by naturalness and intelligibility of the speech.

In digital world, there are some people who can read and understand different languages and some who can't understand languages except their own language. Speech to text conversion system can also provide a facility to exchange information between people speaking different languages [2]. TTS systems are also needed to reduce the extinction of minority languages. As minority languages of the world are facing challenge of extinction considerable efforts are going on from last few years for their survival. Fon language is spoken in Republic of Benin and some other regions of Africa and it is also facing challenge of extinction [7]. The Xitsonga is spoken in more than three African countries. TTS system of such languages will help lot of people of different literacy level [8]. Urdu is national language of Pakistan and it is spoken by more than 100 million people across the world

[9]. A Text-to-Speech (TTS) system for Urdu will be very helping for visually impaired, handicapped and illiterate people.

For human, the task of speech synthesis is not difficult one as they have basic knowledge of their language but for computer some other method has to be implemented for this task. When we talk about TTS systems speech types and procedure for synthesis, strategies or modules used to develop systems etc. are important to consider. Different types of speech exist such as isolated word (process single word at a time), connected words (isolated words but separated with least gap), continuous speech (permit client to talk while computer is processing content) and spontaneous speech (deals with variety of words that are used rarely) as well as two types of speaker model were presented independent and dependent of clients or speaker specifications. Vocabulary is also characterized according to size such as small vocabulary, medium vocabulary, large vocabulary, very large vocabulary and out-of-vocabulary. Below are the major speech generation techniques.

1.2 Types of speech synthesis

For the process of speech synthesis, three types of techniques are used.

1.2.1 Formant synthesis

In Formant synthesis, speech waveform is generated using concatenation of sine wave with the help of some algorithms to model a source of sound [10]. All speech parameters are changed periodically in order to get speech waveform. Some set of rules are also used to generate speech due to which this technique is also called rule based speech synthesis. But it is very difficult to accurately describe the process of speech generation in set of rules therefor the speech generated by this technique is not very natural but intelligible.

1.2.2 Concatenative synthesis

In concatenative synthesis, small units are selected from carrier sentences which are joined to form speech of complete sentence. These small units are called phonemes. These are the units which collectively describe correct pronunciation of a word. This process is easy as compared to previous one as number of such phonemes are limited for any language.

For English, there are 44 such phonemes. Similarly in Urdu, there are 44 consonants, 8 long vowels, 7 long nasal vowels, 3 short vowels and many diphthongs [11]. This reduce distortion but it can decrease the naturalness. That's why the derived synthetic speech may not resemble the donor speaker in training database [12].

1.2.3 Statistical parametric speech synthesis

Statistical parametric speech synthesis is another approach which uses parameters to describe speech. In this technique, model is learned from speech data. This technique works better than concatenative technique [13].

1.3 Quality of speech synthesis system

Intelligibility and naturalness is the measure of quality of the synthesized speech [14]. There are lots of experimentation over naturalness of voice as a result of TTS systems. In today's world, different segments are recorded and then concatenated for completing a message. A collection of speech words is collected and maintained in database by using a reader who reads large series of text. In these kind of systems, to maintain the consistency the speaker speaks in a single style and keep in mind the distance from microphone and other factors to avoid the inconsistency. This type of TTS system is not required at all as the need is to have a system which can be expressive and convey message with proper expressions and styles. Work is performed to build a system that can convey the message according to the needs of the users. A single style of communication can lead towards wrong messages and can cause other problems of misunderstandings. For example, it is not appropriate to convey a good news and bad news in a same style and manner. Similarly, it is not acceptable to ask a question in neutral way of communication [15]. Multiple techniques like linear regression and neural networks were applied to get the improved results. Concatenation techniques are applied to get fully expressive and stylish messages for end users. By using concatenation technique, users can customize, add styles and expression through provided Speech Synthesis Markup Language (SSML) [15]. Timing of events in speech is also important as timing of events in speech signals are affected by some contextual factors like phone identity factors. These factors make it difficult to control timing of events [16]. There are some approaches

which have been proposed to control timing of events like linear regression [17] and tree regression [18]. A new technique is proposed in [16] where timing of events is controlled by multi-dimensional Gaussian distribution based Hidden Markov model.

1.4 Architecture

Text to speech is a way of communication and transferring information using words and styles of speaking [15]. It has two processes which are text processing and speech generation. In text processing, the given input text is processed so that to get appropriate chain of phonemic units. Speech generator takes these units as input and converts them into synthetic speech by selection of a unit from large corpus TTS system for small database is easier to implement but not in good quality [19]–[21]. Different researchers and developers used different strategies and architecture to develop TTS system. In [22], raw text is converted into intelligible speech signals by following two sub processes called High-level synthesis and Low-level synthesis. High-level synthesis converts text into phonetic strings and Low-level synthesis converts these strings into speech signals [22]. In [23], TTS system is divided into three modules.

1. Natural language processing
2. Text parameterization
3. Speech synthesis

Natural Language processing unit converts text into phonetic strings. The second and third stages use these phonetic strings and convert them into speech signals. This is shown in figure 1.2.

In [24], TTS system is implemented by following four modules in sequence.

1. Text analysis
2. Word pronunciation
3. Phonetic interpretation
4. Speech signal generation

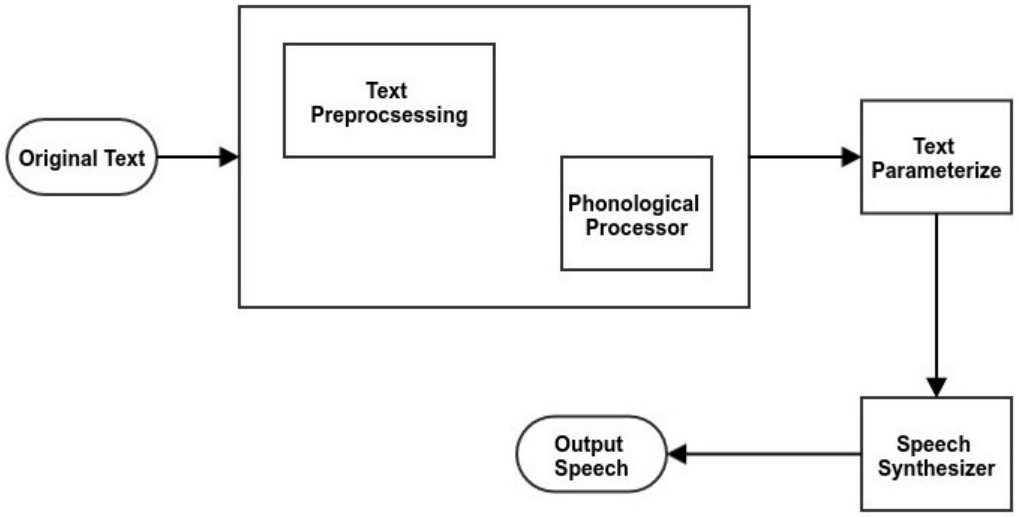

Figure 1.2: Architecture of TTS

In text analysis, the input text is segmented into sentences and then dividied into words. These words are then categorized according to their syntactic and contextual meaning. The numbers and abbreviations are also processed in this step. In word pronunciation process, words are represented by respective phonetic notations by using word pronunciation dictionary. In phonetic interpretation the duration of phonetic segments, pitch and accents are assigned. Signal generation component of TTS system takes output from all above processes and generate a signal of speech using a function. In [25], Text-to-Speech system is divided in two parts. One is called Natural Language Processing unit and other is called Speech Synthesis unit. Natural Language Processing unit preprocess text and converts it into phonetic strings. These phonetic strings are then marked by stress marker and passed to speech synthesis unit which converts it into speech signals.

Chapter 2

Related Work

Text to speech synthesis is not a new field and people have been working on this field before electronic signal processing techniques. In beginning, people tried to build machines with mechanical devices which were used to create human sound. After the development of computers, better systems were built using different techniques. In [14], basic speech synthesizing technique are discussed which works by concatenation of small recorded speech segments called phonemes to form complete speech. Each word is first divided into syllables and then corresponding voice signal for each syllable is concatenated in order to get pronunciation for whole word. This concatenated word has some delay between pronunciations of each syllable which is removed and as a result of which final pronunciation of that word is obtained. Problems like Text Preprocessing, Pronunciation and Prosody make this system difficult. In Text Preprocessing, digits and abbreviations are converted to full words. Other problem is the guessing correct pronunciation of a word. For example, word "lives" has different pronunciation in "He lives in Lahore" and "He saved two lives". To create naturalness in sound, stress and intonation are applied to the input text which is also a very complex task. A TTS system for Azerbhaijani language is developed using concatenative synthesis in [5] where small recordings were concatenated to make speech waveform.

Two sub parts of text-to-speech system text analysis and word pronunciation were discussed in [24]. TTS system is divided in 4 sub parts: text analysis, word pronunciation, phonetic interpretation and signal generation. Text analysis step includes division of text into sentences, words, phrases and expansion of abbreviations etc. Text analysis is required to get correct context and pronunciation of each word in a sentence. In text analysis,

sentence division and parts of speech tagging is done by using heuristic solutions [26] and dynamic programming respectively. For word pronunciation, dictionary based approach was used for about 99.9% words and for 0.1% words, letter to sound rules were followed.

A technique based on letter to sound rules is also developed in [27]. Total 329 letter to sound rules have been created. These rules take text as input and translate it into phonetic alphabet which in turn converted to synthetic sound. This system produces about 97% correct pronunciation of phonemes. The paper also describes software and hardware requirements and overall performance statistics of this system. The dataset is developed by extracting 50,000 words from standard corpus, Corpus of present-day edited American English i.e. Brown corpus [28]. The system gave accuracy of about 93%. A more improved system was proposed in [6], [29]. In [6], a TTS system is developed for English and simple numerical and algebraic expressions. The system is rule based system having 500 letter to sound rules. However, it can use pronunciation dictionary of 1500 words for exceptions. The system interface provides facility of selection of voice types (Male or Female). The word boundaries and probable position of phrases and clauses is analyzed by syntactic analyzer. The phonemes of word are then passed to synthetic speech synthesizer that converts phoneme to sound by following extensive set of rules and rules for consonant-vowel transitions. The whole process is divided in two sub processes, text analysis which is conversion of text to corresponding linguistic representation that comprises of phoneme, stress and boundaries and positions of respective vowel or consonant and durational phenomena such as pauses, interaction between segments [30] and speech synthesis in which sequence of phonemes are converted to speech sound with the help of some set of rules. The abbreviations in input data are converted to their respective text and if dictionary does not have their respective words, the abbreviation is pronounced as a word. After preprocessing, the words are exposed to letter to sound rules. If an unstressed function appears in text and there is no rule for it then it is passed to pronunciation dictionary. The results show that about 95% rules were successful when executed. The syntactic analyzer then determines the structure of sentence according to its pauses and boundaries. The phoneme to speech rules are divided into two components, phonological that provide information of stress and rhythm and duration of words in a sentence. All these outputs are then passed to synthesizer that produces synthetic sound. Synthesizer is simple version of synthesizer proposed in [31].

In [29], a text to speech system is presented which require some hardware resources as well with the enhancements in microprocessor, memory and signal processor technology. This TTS system can be put into portable form and can be used anywhere with different systems. A higher level language is developed in laboratory that can be easily used for linguistic processes. These enhancements and developments in hardware and software level made transforming of text to speech at 250 wpm rate. This combination of hardware and software is tested against several applications used for handicaps. The main purpose of this system is to make a framework that will be used for conversion of any language text to speech.

In [12], Whistler which is a Text-to-Speech engine is developed by using prosody and concatenative speech parameters that were extracted through use of probabilistic learning methods. Whistler engine produce speech which appears to be very much real. This system can also help to build TTS system for other languages.

A formant and concatenative synthesis is developed in [32] where small segments of phonemes were concatenated to form whole speech. A training database was used which contains about 6,000 phonetically balanced sentences recorded in natural style. The technologies used in whistler can considerably facilitate the process of creating generic TTS system for new speech style. This engine supports Microsoft Speech API [33] and requires less than 3 MB memory.

In [34], linear regression and unit selection based speech synthesis is designed using ATR Japanese database. In this algorithm, raw text is converted to phonetic strings and against each phoneme, best candidate unit from a huge database of speech units is selected with *Viterbi* search. By concatenating these units, target waveform is generated. The units in database can be considered as state transition network where each unit represents a different state. The cost of the system depends on target cost and cost of the concatenation of units. Each phoneme and unit is represented by a multi-dimensional feature vector. The target cost is measured by the weighted difference of target and candidate feature vector. Similarly, concatenation cost is also weighted sum of sub-cost of concatenation. Cost function can be trained in two different ways. In Weight Space Search method, units are searched with *Viterbi* and distance between constructed waveform and natural waveform is minimized. In Regression Training, linear regression is used to choose best unit from the list of all possible units for a given phoneme.

Statistical parametric speech synthesis is another approach which uses parameters to describe speech [35]. This technique works better than concatenative technique on smaller data. On larger data, concatenative synthesis can produce better quality speech. In this technique, Hidden Markov model (HMM) or model firmly related to HMM are used for training model over given data. HMM based statistical parametric speech synthesis has gained popularity because of its ability to produces high quality speech automatically with parametric flexibility, less data and resources [19].

A multi-dimensional Gaussian distribution based Hidden Markov model based statistical parametric speech synthesis system was developed in [36]. In this technique, decision tree based context clustering is used for duration models clustering. The contextual factors are also considered with phone identity factors. Mel-cepstral coefficients are calculated and model is trained by these coefficients. State of the context dependent HMMs is clustered using decision tree based context clustering technique [37]. In state duration modeling, multi-dimensional Gaussian distributions are used to model Hidden Markov model. Duration models are clustered after estimation using decision tree based clustering techniques. By traversing decision tree, all contexts can be searched. Contextual factors which effects timing of events in speech are also taken into account and resultant speech shows that it has good quality and natural timing. For testing of the system, 450 sentences of Japanese are used for training of system. Sampling of speech signal is done at 16 kHz. Feature vector is composed of 25 mel-cepstral coefficients. There were 3030 states and 2984 distributions in output of the system. The listening tests show that synthesized speech has good quality and it has natural timing even if speaking rate is changed to some degree.

A similar system is designed in [16] using HMM and evaluated by taking input from Japanese database. The parameters in this system are generated with Hidden Markov model. The state sequence fully or partially is hidden due to which iterations are performed for parameter generation and forward-backward algorithm is used for the situation where state sequence is provided. This algorithm, from multi-mixture HMMs, can generate clear formant structure.

A HMM and unit selection based system is proposed in [38] where model is trained with speech database after which excitation and spectral parameters are extracted. These extracted parameters are modeled by context dependent HMMs. To find accurate model parameters, decision tree based context clustering is used. The parameters are then used to

generate speech signals. These parameters can be used to control speech characteristics. In [39], HMM and rule based approaches are applied on voices taken from e-learning courses and online lessons for dataset creation and tested by generating voices and given as input to students to interpret it.

Corpus based approach for Expressive Prosody Modeling is applied in [15] where manually produced dataset was used. To evaluate the synthesized speech and expression, the output is given for testing to 32 native English speakers. Test is performed with different types of sentences like bad news, good news and for yes/no and the accuracies we get are 70.2%, 80.3% and 84% respectively.

HMM based approach is used in [8] to construct a speech synthesizer for Xitsonga which is an African language. The dataset used here consists of phone set of consonants and vowels. These sets are used to prepare a set of letter to sound rules to be used in speech synthesis system. The main tool used for speech synthesis is HTS toolkit [40] with other software that support to setup complete environment for text-to-speech conversion. The technique used in this study is Hidden Markov model because the statistical parametric speech synthesis based on HMM can be used to synthesize speech waveform without requiring huge dataset for training. The system received acceptability of 92.3%.

TTS system for Fon language is designed in [7] using Multisyn algorithm [41] which consists of Natural Language Processing (NLP) and Digital Signal Processing (DSP) modules. NLP consists of segmentation, Letter-to-Sound conversion and back-off rules module. Back-off rules are applied when input text contains some characters that are not in us know characters. DSP module than choose required unit from database of units are concatenate them to form complete speech signals.

In [42], hybrid text to speech converter is developed by concatenating benefits of HMM based TTS system and waveform based TTS system. For developing it, an audio phoneme library is used. The main edge of developed system over other is that it produced more human like voice/speech. The experiments were taking in matlab and a phoneme library is developed that consists of audio files and dictionary of words with their phoneme. Sentence is taken as input then model parsed it into words. The system analyzes each word, gets its phoneme and combines all phonemes and plays the sound. Waveform for each sound is also presented. This sub-ban speech synthesized approach is obtained by this combination of models that improved the quality of synthesized speech. The current model is not good

enough, in future system will be improved to get better controls.

A speech synthesis system is introduced in [43] which uses context-dependent Hidden Markov Model for defining set of subphone units. This system uses context-dependent HMM for defining set of subphone units. These subphone units are then used in concatenation synthesizer. The training data is one hour recorded speech which is used for getting required parameters. TD-PSOLA waveform concatenation synthesizer is then used to generate pronunciation using these parameters. The synthesized speech imitates the voice of the speaker used to record the preparing database. This system uses automatic statistical processes to extract segments of speech from large speech carpus. Desired sentence is produced by concatenation of small segments of speech. Hidden Markov model is trained and used for segmentation of speech database into HMM-state-sized units. A decision tree is constructed by using phonetic context labels which is used for clustering of the training speech into acoustically self-comparable grouped states. This process helps to find most important context effects. The string to be converted into speech is first converted into sequence of phonetic strings which then with the help of decision tree is converted to speech segments which are used to generate final speech signals. Modified Rhyme Tests [44] were used to compare system with other. Six listeners were used with each give an answer sheet, and they have to mark word from list of provided words which is played during test. The MRT error rate for test was 5.0% and standard error rate was 0.47%. Hidden Markov model is used for training of the model. The dataset used for training of model is recorded speech. Four datasets are used in which are termed as M2, M3, F1 and F2 where M stands for male and F stands for female. Six listeners evaluate output produced by model. Error rate is used as measure of performance. The MRT error rate for test was 5.0% and standard error rate was 0.47%. In future, segment selection algorithm used in the system can be improved where segments in each state would be available in speech synthesis process. Dynamic programming can be used to find optimal segment sequence.

[45] present a TTS system which is based on HMM which comprises dynamic features. The statistical parametric speech synthesis system can change voice characteristics of speech by speaker adaptation technique [46] and speaker interpolation technique [47]. The Hidden Markov Model statistical parametric text-to-speech system can model speech parameters like spectrum or excitation with the help of context-dependent HMM and construct speech signals. Version 2.0 of already build HMM based text to speech system (HTS) toolkit is

presented in [20]. HMM based speech synthesis system can be used to build speech synthesis system with small dataset for training [48] but the quality of that speech will not be equal to recorded speech.

A more advance technique is Neural Networks based technique as it works better than Hidden Markov model based technique. Time domain Neural Networks with database containing sounds of words called phonemes is used in [49]. The basic flow of the system involves speech recording, speech labeling, voice coder and input processing using Time Delay Neural Network. The figure 2.1 shows the block diagram of system.

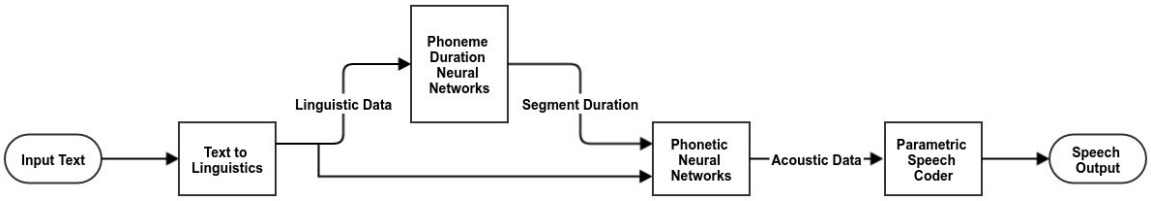

Figure 2.1: Time Domain Neural Networks based TTS System

Neural Networks based techniques are used to learn features automatically during training along with the combination of various techniques like linear regression and Neural Networks in [50]. Dataset of all previous Blizard Challenges [51] and afterwards up to 2013 were used. Model was evaluated using 5-fold cross validation. The given model gave 0.11% and 0.17% error for LR+LR and LR+NN respectively. Neural Networks is also used in [52] with dataset consisting of 328 hours which was collected in voice of native 1506 speakers. The model is tested by giving 20 sets of randomly selected from evaluation set and asked them to rate output of each set between 0 and 100.

Deep Neural Networks is applied in place of Hidden Markov model in [53] as HMM based system cannot model complicated context dependencies. Deep Neural Networks (DNN) can cover limitations in HMM based system and can also outperform HMM based system.

Recurrent Neural Networks (RNN) is applied in [54] by using the Bidirectional Long Short Term Memory (BLSTM) with dataset consisting of 5000 training utterances and 200 utterances for testing the system. Whole recording was done in voice of the female native speaker. Objective and subjective evaluation measures are used to find distortion between natural and synthesized speech and quality respectively which shows that hybrid system

is better as it gave 44%, 59% and 55% accuracy whereas the Neural Networks, HMM and DNN gave 29%, 22% and 20% accuracy respectively. In [55] Recurrent Neural Networks (RNN) postfilters are used for speech synthesis.

Researchers have also been working on TTS system for Urdu language for many years. A bi-lingual text-to-speech synthesis system for Urdu and Sindhi is designed in [56] using bilingual hybrid knowledge based approach by using concatenated synthesis method which is capable of providing high quality Urdu and Sindhi speech. This system can be further expanded to include sensitive and visual text-to-speech (VTTS) policies in future.

In [57], an HMM based speech synthesis system is developed for Urdu. The speech corpus is created by recording 1 hour and 15 minutes of speech containing 989 sentences in total. In training phase, context level and prosody level parameters are extracted from recorded sentences e.g. counts, position, distances, stress and phone utterance information. F0 excitation parameter and mel-cepstral coefficients are calculated using RAPT [58]. The F0 is modeled using frequency distributions discrete for unvoiced and continuous for voiced regions. These HMM models are then clustered using decision trees. In text analysis, numerals and abbreviations in input text are preprocessed and converted to full textual forms. The date/time and numeric notations are processed using regular expressions (rule based component) and abbreviation are converted to text by finding their corresponding words from dictionary. This stage is followed by diacritic restoration stage which used dictionary develop by CRULP [59] to restore diacritics. After this, G2P converter which follows guidelines of [60] is used to convert grapheme to phoneme. In synthesis module, input text is labeled and then by speech synthesis algorithm, it generates speech features which are passed to filter and obtain speech signal. The model uses 36 consonants and 10 vowels. All speech recordings in a corpus are converted to their phoneme representation and saved in pronunciation dictionary after checking by author. After this, STRAIGHT vocoder is used to estimate speech parameters and generation of speech waveform. 680 questions were gathered by using spectrum and context features, speech parameter were generated using maximum likelihood criteria. For evaluation of system, the author listened to synthetic speech himself and found that it is not intelligible but can be improved in future work.

In [61], a TTS system for Urdu is developed by using HTS toolkit and Urdu Qaida of grade 2 and 4. This system consists of two processes, text analysis and synthesis of

speech. Here feature extraction and calculation of mel-cepstral coefficient is done using technique mentioned in [62], text processing is done using [22] and process of synthesis using process described in [16]. The HTS toolkit is available for English, Japanese and Portuguese languages. For Urdu, certain modifications are needed which involve creation of context level labels and questions file for Urdu phoneme set. Frequently speaking Urdu words were identified by using greedy algorithm and question files are made to deal with the issue of data sparsity as in a model, only a certain amount of examples can be handled during training phase. If we look on to the contextual level, it is observed that multiple contextual occurrences exist for a single phoneme. To deal with this problem, a clustering method is used to cluster similar acoustic words. The whole training set is placed into single cluster and then split on the basis of each question and the question which minimizes the objective function is selected. In the evaluation process, experiment is performed by using 200 frequent Urdu words and native Urdu speakers. Testing of the system shows that the system gives output which is intelligible but not very natural. The reason behind this is data used in training phase consists of full sentences rather than words. Performance with respect to naturalness can be improved by using words instead of sentences because of clarity and length of word. It is found that 92.5% words are correctly identified. The system has taken 66 phonemes but for better performance at least 270 examples should provide the system during training phases.

Natural Language Processing unit is very importing unit in speech synthesis system as it handles all language related issues. This unit performs a list of steps for its complete operation which contains tokenization, semantic tagging, string generation, syllabification, stress and intonation marking etc. [11], [25] discuss such unit for Urdu language. This unit is divided in two parts called as pre-processing and phonological processing unit. Pre-processing unit converts number, date and time into their respective literal strings. For example, 100 and 5-11-2002 will be converted into سو and پانچ نومبر دو ہزار دو respectively. Special symbols like $ and are also handled in pre-processing unit. Last stage of pre-processing unit is grapheme into phoneme converter. Phonological processing unit contains syllable marker which marks syllable boundaries and stress and intonation markers mark stress and intonation.

[11] discussed consonantal and vocalic sounds for Urdu Language in detail. In [23], Phonological Processing unit for Urdu language is discussed in detail. This module applies

letter to sound rules, syllabification to the normalized text. This is followed by stress and intonation marker. Statistical based part of speech tagger for Urdu language is discussed in [63]. This is done by calculating probability of each word given a particular tag. Unigram model assign tag for each token that has the maximum probability. Conditional probability for given tags against each word using maximization principle as used in [64], [65]. The model is evaluated by comparison of Unigram, Bigram and Backoff experiments with small and large tag sets. t-test, POS accuracy are used to measure performance. Bigram model considered maximum likelihood principle keeping an eye on the context of text. Backoff model was used to blow away sparse problems.

Problems in Urdu segmentation are discussed for Urdu in [66]. Clause boundary identification is discussed in [67] using classifier and clause markers in Urdu language using conditional Random Field as a classifier.

2.1 Discussion

Raw text can be converted into speech by concatenation of small units of speech from a huge single-speaker speech database. Huge database makes it possible to produce more natural sound. TTS system development can be based on rules for generation of speech but this method can take intensive labor and rules are difficult to be general so that they can be used for other languages as well. In prosody modeling, linguistic rules are used [4], [68] but speech produce by these rule based prosody models felt to be robotic. So for naturalness of voice, large units are used. This method not only improved naturalness but also decreased required time to generate new voice and also made the synthetic speech similar to original donor speaker.

The best approach for speech synthesis until now is considered to selection synthesis but it has certain limitation that is it need large database of recording which is very expensive and not feasible for certain languages [34], [69], [70]. Statistical parametric speech synthesis is becoming popular and being used for number of languages like English [38], Chinese [71], Arabic [72], Croatian [73] and Urdu [57]. The advantage of parametric over selection is that it does not require saving original signal for synthesis due to which database is small for this approach [74]. Basic Text to Speech or TTS system focus over conversion of text to voice using multiple techniques [13]. Different synthesis model has been developed but

HMM is becoming popular from last few years. HMM based statistical parametric speech synthesis become very popular in last few years [53]. There are multiple tools for TTS but freely available tools mostly use 2 techniques i.e.

1. Hidden Markov Model based speech synthesis called SPSS
2. Simple waveform concatenation.

SPSS technique is attractive although its results are comparatively not amazing. In recent years, the use of statistical modeling in speech recognition system has increased a lot and most of these systems are using Hidden Markov Model for acoustic modeling of the system [75]. These systems enable us to construct models with large amount of data that is difficult to analyze manually. This technique can be applied on the process of speech synthesis. This type of system can be used to run on different data, voices and languages [75]. There are many speech synthesis systems which can generate high quality speech, but they still cannot generate speech with different speaking style and voice because large amount of speech data is required in order to get these characteristics. This can be achieved by using HMM based speech synthesis system [38]. Statistical learning techniques can be used to build a speech synthesis system. These systems can be trained and voice characteristics of original speaker can be produced in synthesized speech. This type of system can be built with Hidden Markov model and its performance can be improved by techniques like context-dependent modeling and environment adaptation techniques [16]. It has many advantages like ability to change voice characteristics and robustness which will be very difficult in concatenative speech synthesis. But it has some limitations like inefficiency in handling complicated context ascendance [53].

Another popular speech synthesis technique is unit selection where small units of recorded speech are concatenated in order to synthesize speech waveform. This technique is capable of generating high quality speech signals but for getting various characteristics of synthesized speech, a huge database is required. On the other hand, HMM based statistical parametric text-to-speech system can generate speech signals with various voice characteristics without requiring huge database [20]. A lot of research work on TTS system has been done using HMM techniques but the output voices produced by these kinds of systems look unnatural sometimes. It is surprising that by the time this should be improved a lot but there are still existing problems and drawbacks that decrease the performance of TTS system, as

compared to other simpler concatenation based TTS systems. Through literature review it is easy to say that HMM system do over smoothing which cause unnaturalness for TTS System output. There is no proper study which can prove this hypothesis so [13] present the reasons for this unnatural behavior [13].

Many text to speech systems have been purposed and each have its own pros and cons. For example, waveform based model is good enough to produce human like sound but it requires large database. In rule based techniques, most of the time rules updating is required and novelty is too much difficult with traditional rules. Similarly, with concatenation of phonemes, it is also difficult to bring novelty and handle new and unseen words [49], [76]. Neural Networks can be used to improve results of speech synthesis system [55]. From recent years, Neural Networks are being used as acoustic models [77], [78]. There exists a wide research over the correlation between acoustic modeling and linguistic features in late 90s [79]. Now more focus is not Neural Networks based techniques. Neural Networks easily map linguistic features to acoustic models using feed forward approach [53], [80]–[82].

Chapter 3

Methodology

Text to speech synthesis convert raw text into speech waveform. This process is divided in two sub modules. One module performs analysis and preprocessing of data and other transforms processed data into sound signals. This is shown in 3.1

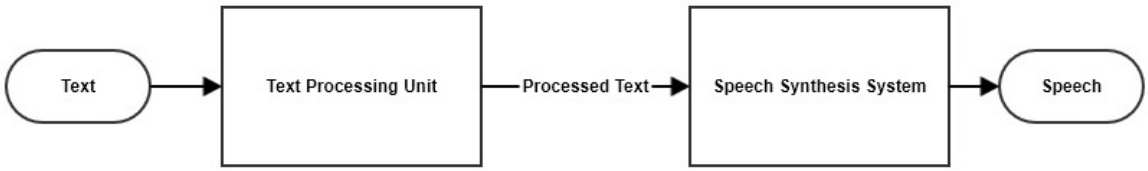

Figure 3.1: TTS Sub Modules

3.1 Text Processing Unit

Text processing unit is responsible for processing of text before it is sent to speech synthesis system. It finds numbers, dates and time in input data and converts it into format acceptable by speech synthesizer. This module consists of following sub modules.

1. Special Character Processor
2. Semantic Tagger
3. Text Generator
4. Text Formatter

3.1.1 Special Character Processor

Raw text may contain special characters such as punctuation marks. These characters help to understand context of a word but are not converted into sounds. We remove all such characters from text before further processing. Another processing which is done is conversion of all arabic numerals like ۱, ۲ and ۳ into their corresponding characters like 1, 2 and 3 because it is easy to further process text after it has been converted into same type of numerals.

3.1.2 Semantic Tagger

The purpose of Semantic Tagger is to identify numbers, dates and time from input data and give them proper tags. All numbers in input data are converted into arabic numerals of type 1, 2 and 3 in previous step. There can be multiple form of numbers, dates and time. These forms are explained below.

1. Dates in following format
    a. 12/11/2018 or 12/11/18 with different separaters like "/" or "-" or "."
    b. 12 دسمبر 2012
    c. 12 دسمبر
2. Number in following format
    a. Whole Numbers such as 123
    b. Floating point numbers such as 12.3
3. Time in following format
    a. 12:12
    b. 12:12:12

In table 3.1, regex used for identification of these numbers, dates and time are shown.

Regex	Type	Example
(\d+(?:\.\d+)?)	Integer or Floating point number	123 or 12.312
\d{1,2}:\d{1,2}(?::\d{1,2})?	Time with or without seconds	5:12 or 5:12:10
\d{1,4}[./-]\d{1,4}[./-]\d{1,4}	Date with separator like "/" or "-" or "."	12-10-2018 or 12/10/18
%s \d{4}	Dates with month name in Urdu. This is checked by replacing %s with each month name separately.	12 دسمبر

Table 3.1: Regular Expression for Semantic Tagger

Word	Converted Text
123	ایک سو تئیس
1231	ایک ہزار دو سو اکتیس
123.1234	ایک سو تئیس اعشاریہ ایک دو تین
12345	بارہ ھزار تین سو پینتالیس
1234567	بارہ لاکھ چونتیس ھزار پانچ سو ستاسٹھ
987654321	اٹھانوے کروڑ چھہتر لاکھ چون ھزار تین سو اکیس
143.159874	ایک سو تینتالیس اعشاریہ ایک پانچ نو آٹھ سات چار

Table 3.2: Number Conversion example

3.1.3 Text Generator

Semantic tagger will return all numbers, dates and time from text with each one marked as date/time/number. Text generator part will take each word and generates Urdu text according to its tagging. Each tagged number is handled by specific text converter. These converters are listed below.

1. Number to text converter
2. Date to text converter
3. Time to text converter

3.1.3.1 Number to Text Converter

This unit will deal with whole numbers, fractional numbers and decimal numbers. In table 3.2, example conversions are shown.

Both the integer and fractional parts of floating point number are handled differently. The algorithm for integral number is described below.

This algo will return list of factored numbers and factors. For 1213, the result of this algorithm will be [1, 1000, 2, 100, 13]. The integer to Urdu mapping for number 0 to 100 and number like 1000, 100000 etc are stored in CSV file. The factored number list will then

Date	Converted Text
12/10/15	دس دسمبر دو هزار پندره
12.10.15	دس دسمبر دو هزار پندره
12-10-15	دس دسمبر دو هزار پندره
12.10.1989	دس دسمبر انیس سو نواسی

Table 3.3: Example Date Conversions

be converted into text by using integer to Urdu mapping. In table B.1, Urdu mapping of each possible factor is listed. Each number in fractional part of floating pointing number is replaced by their respective mapping from above list. Both the mappings of integral and fractional part of number are then joined to get complete text of input number. This module is very important module as it is also used in date and time conversion.

3.1.3.2 Date to Text Converter

The date to text Converter deals with date in following formats

- 12/12/2012
- 12/12/12
- 12.12.2012
- 12.12.12
- 12-12-2012
- 12-12-12
- 12 دسمبر 2012

All dates will be converted into common format e.g. بارہ دسمبر دو ہزار بارہ . Some example conversions are shown in table 3.3.

The word tagged as date is first processed to get year, month and day of the month. Day and year are then passed to number to text converter and month is converted to its corresponding mapping. This mapping is saved in CSV file. This is shown in table B.2. In Urdu, in dates, we have different notation for year e.g. 1980 in number is spoken as ایک ہزار نو سو نواسی (One thousand nine hundred and eighty nine) but in dates, it is spoken as انیس سو نواسی

Time	Converted Text
1:12:15	ایک بج کر بارہ منٹ اور پندرہ سیکنڈ
7:45	سات بج کر پینتالیس منٹ

Table 3.4: Example Time Conversions

(Nineteen hundred and eighty nine). This is also handled during the process of date to text conversion.

3.1.3.3 Time to Text Converter

Time can occur with seconds or without seconds in text. It is written in 1:11:12 or 1:12 format. All words tagged as time will be converted into text by separating hour, minutes and seconds from time. Each value will be converted into Urdu text by using number to text converter. All these values are combined to make complete time text. Table 3.4 shows example conversions.

3.1.4 Text Formatter

The purpose of formatter is to replace all number, dates and time with their corresponding Urdu text returned by Text generator. This process is performed in following order

1. Word tagged as dates are replaced by their corresponding text
2. Word tagged as time are replaced by their corresponding text
3. Word tagged as number are replaced by their corresponding text.

The output of all above processes will be the text which will only contain Urdu text which can now pass to the speech synthesizer which will convert it to speech signals.

3.2 Speech Synthesis System

3.2.1 Tools

Below are the tools used for the process of Speech Synthesis of Urdu.

1. Speech Tools Library of Edinburg

2. Festvox

3. SPTK

4. Festival

3.2.1.1 Speech Tools Library of Edinburg

The Edinburgh Speech tools is collection of utilities used for speech processing. These utilities cover major tasks such that reading and writing speech waveforms, parameter files(F0 and LPC etc). The speech tools also include executable programs which can be used in user defined programs.

3.2.1.2 Festvox

Festvox is a tool which can be used to build synthetic voices. This includes scripts for building voice in other languages.

3.2.1.3 Speech Signal Processing Toolkit (SPTK)

SPTK stands for Speech Processing Toolkit. As name suggests, this tool is for processing speech signals in UNIX systems.

3.2.1.4 Festival

Festival is a speech synthesis system developed in Centre for Speech Technology Research (CSTR) which is a multi-platform framework for building speech synthesis system. This system is designed in such a way that it can be used for following purposes

1. Improvement in speech synthesis system

2. Developing speech synthesis applications

One of the main thing that makes Festival very useful is scripting language which is based upon Scheme programming language. This can be used to manage parameters and flow of control in Festival.

3.2.2 Process

The process of speech synthesis is based on statistical parametric speech synthesis. The statistical parametric speech synthesis is model based speech synthesis in which model is trained using training data. Training data consists of recorded speech and their corresponding labels. In this method, speech is elaborated with parameters which are defined by statistics. This is why it is called as statistical parametric speech synthesis.

The CLUSTERGEN statistical parametric speech synthesis is type of synthesis in which model is trained and used for synthesis in Festival Speech Synthesis system.

3.2.2.1 Preparing Data

For training purpose, Phonetically Rich Urdu Speech Corpus [83] is used. This data consists of recordings of 708 phonetically rich sentences, 10,101 tokens with 5,656 unique words. Total duration of recording is 70 minutes.

3.2.2.2 Data Labeling

Data is labeled in specific format which is required by FestVox for training purpose. The is labeled in following format.

(c1 "نیلم نے سالگرہ پر ہیڈ سیسمو گراف اسود قریشی کے ماتھے پر اینٹھن اور غم کی آتشیں رو محسوس کی")

Where c1 is the name of recording file and text between quotation marks is corresponding label of that recording.

3.2.2.3 Training Data

The whole data is further divided in 10:1 ratio in training and test set respectively.

3.2.2.4 Urdu to Hindi Transliteration

The underlying system of Urdu TTS is Hindi TTS system. So all the alphabets are mapped in their corresponding Hindi alphabets. In this way, text is first converted into corresponding Hindi text using that mapping and then it is converted into sound. Mapping of each Urdu word with Hindi in this system is shown in table B.3.

3.2.2.5 Data Labeling

The first stage of training is to label speech database using HMM labeler. We are using EHMM labeler which is provided in FestVox. In this process, context dependent models are trained using Baum-Welch. This labeler works in 8 steps. Prompt files are extracted from utterance structure of Festival.

- Unique sequence of phones are extracted and stored.
- List of wav files is collected for feature extraction.
- From wav files, cepstral coefficients (LPCCs and MFCCs) are extracted.
- From cepstral coefficients, deltas and delta-delta features are generated.
- By using generated features and wav files list, features vectors are modified.
- Phones list generated in step 2 and wav file list is used to modify prompt list.
- Hidden Markov model is trained using Baum-Welsh algorithm till difference in the average likelihood is less than 0.001.
- Labels are generated according to training data.
- Integer indices of labels are converted into phone names

3.2.2.6 Building Utterance Structure

Utterance is the essential building unit of Festival. It shows relation between bunch of items where each item relates to word, syllable or segment etc. Below are the some of the relations used in building utterance structure.

- **Text**: It consists of strings to be processed and features of that string.
- **Token**: Token means each word in a sentence is separated by some language specific separator.
- **Word**: A small unit of speech which can be pronounced with the help of letter to sound rules of a language.
- **Phrase**: Phrase means group of words forming a part of a sentence.

- **Syllable**: Syllables are units which when combined with vowels form complete pronunciation of a word.
- **Segment**: Segment consists of list of phones.
- **SylStructure**: This is a tree structure which is formed with word, syllable and segment.
- **IntEvent**: These are array of syllable related intonation events.
- **Intonation**: Intonation means rise and fall in speech signals.

3.2.2.7 Coefficient Extraction

Coefficient extraction is the process of extracting parameters like F0, mcep and voicing coefficients using SPTK. This is done by generating F0 and mcep coefficient. These parameters are then combined to make final parameter files. This is a lengthy process which can take lot of time depending on size of training data.

3.2.2.8 Building the Model

All the data generated above is used to train and build HMM-state duration model. This process works in following steps.

1. Statenames Generation
2. Parametric Model generation
3. Duration model generation for statenames

This resulting model can be used to perform text to speech synthesis process.

Chapter 4

Experiments and Results

The purpose of a Text to Speech system is to build a system which is capable of generating voice as close to human voice as possible. The generated voice should be intelligible so that people can easily understand generated voice. To find quality of generated sound, every speech synthesis system is evaluated. The evaluation process can be subjective as well as objective. In subjective evaluation, system is evaluated using human users while in objective evaluation, different algorithms are used. For the process of evaluation, native speaker of specific language is required. For our system, we only focused on subjective testing.

4.1 Subjective Testing

There are many type of subjective tests. Some of them are listed below.

- Diagnostic Rhyme Test (DRT)
- Modified Diagnostic Rhyme Test (M-DRT)
- Naturalness Test
- Intelligibility Test
- Usability Test

4.1.1 Diagnostic Rhyme Test (DRT)

This test is for Indicative and relative assessment of the understandability of single starting consonants. This is conducted with words which are similar in sound but differ with each other in initial consonants [84]. User has to listen speech generated by system of a specific word and identify that spoken word from list of words. The result of this test is the percentage of words correctly identified.

4.1.2 Modified Diagnostic Rhyme Test (M-DRT)

This test is to check demonstrative and relative assessment of the coherence of single last consonants. This is conducted using words which are similar in sound but differ with each other in last consonants [44]. User has to listen speech generated by system of a specific word and identify that spoken word from list of words. The result of this test is the percentage of words correctly identified.

4.1.3 Naturalness Test

This test is conducted to find out to which extent generated voice is close to human voice. This is conducted by rating generated voice from 1 to 5. User will play some voice and will give synthesized speech some value from 1 to 5 according to his understanding of the speech.

4.1.4 Intelligibility Test

This test is conducted to find out to which extent generated voice is understandable. This is conducted by rating generated voice from 1 to 5. User will play some voice and will give synthesized speech some value from 1 to 5 according to his understanding of the speech.

4.1.5 Usability Test

This test is conducted to find out to which extent generated voice can be used for blind or non-blind people. This is conducted by rating generated voice from 1 to 5. User will

play some voice and will give synthesized speech some value from 1 to 5 according to his understanding of the speech.

4.2 Evaluation

For the process of evaluation, we selected list of 64 words and 8 sentences. Words are selected on the sound and first and last words in order to use in Diagnostic Rhyme Test (DRT) and Modified Diagnostic Rhyme Test (M-DRT).

4.2.1 Methodology

An evaluation form is designed which have three sections.

4.2.1.1 Diagnostic Rhyme Test (DRT) Section

This section has 8 questions. In each question, user will play recording of some words which are converted to sound using our TTS system. These words are tested through following carrier sentence.

نیچے دیے گئے الفاظ میں سے حرکت پے نشان لگائیں

User will have to select that word from list of words which have same sound but different first character.

4.2.1.2 Modified Diagnostic Rhyme Test (M-DRT) Section

This section has 8 questions. In each question, user will play recording of some words which are converted to sound using our TTS system. These words are tested through following carrier sentence.

نیچے دیے گئے الفاظ میں سے حرکت پے نشان لگائیں

User will have to select that word from list of words which have same sound but different last character.

4.2.1.3 Mean Opinion Score (MOS) Section

In this section, user will have to play a sentence and user will rate converted text from 1 to 5. User will have to rate them on the basis of following properties.

Test	Score
Diagnostic Rhyme Test (DRT)	0.95
Modified Diagnostic Rhyme Test (M-DRT) Section	0.88
Naturalness	3.24
Intelligibility	3.42
Usability	3.49

Table 4.1: Evaluation Result

- **Naturalness**: How much converted sound is close to sound produced by a human?
- **Intelligibility**: How conveniently the word was recognized
- **Overall**: How do you rate this sound overall? Is this system is usable to use for blind people?

System is evaluated by 47 (33 males and 14 female) native Urdu speakers who carefully listened and evaluated system. Each listener evaluated each question separately after listening it.

4.3 Results

During the evaluation of this system, it is observed that most of the words which have same sound are easily recognizable. Almost 90% of the such words are correctly identified by users. The result shows that output of the system is recognizable and intelligible but not very natural. Table 4.1 shows the complete result of evaluation.

Chapter 5

Conclusion

A Text to Speech system for Urdu is build using Festival and Festvox. We used recorded speech of 70 minutes which consists of over 700 sentences. It contains almost 10000 tokens and over 5500 unique words. This data is divided in testing and training data in 10:90 proportion and used for training of the model to be used in synthesize speech. We used lexicon of Hindi for Urdu in order to build this system. For this, Hindi to Urdu transliteration system is also used. A text preprocessing unit is also developed which process any number, dates or time string in input text.

The resulting system is tested by 47 native Urdu speakers and found to be intelligible but not very natural. The synthesized words having same sound but having different first or last character are also recognizable.

The synthesized speech is not very natural mostly because system is developed using Hindi lexicon and letter to sound rules. Some character in Hindi are spoken differently as compared to Urdu. For example, غ in Hindi is spoken as گ. Apart from this, the size of training data also effects quality of synthesized speech. A larger speech corpus can be used to improve quality of speech.

Appendix A

Figures

Appendix B

Tables

Number	Mapping
.	اعشاریہ
0	زیرو
1	ایک
2	دو
3	تین
4	چار
5	پانچ
6	چھ
7	سات
8	آٹھ
9	نو
10	دس
11	گیارہ
12	بارہ
13	تیرہ
14	چودہ
15	پندرہ
16	سولہ
17	سترہ

Number	Mapping
18	اٹھارہ
19	انیس
20	بیس
21	اکیس
22	بائیس
23	تئیس
24	چوبیس
25	پچیس
26	چھبیس
27	ستائیس
28	اٹھائیس
29	انتیس
30	تیس
31	اکتیس
32	بتیس
33	تینتیس
34	چونتیس
35	پینتیس
36	چھتیس
37	سینتیس
38	اٹھتیس
39	انتالیس
40	چالیس
41	اکتالیس
42	بیالیس
43	تینتالیس
44	چوالیس
45	پینتالیس
46	چھیالیس
47	سینتالیس

Number	Mapping
48	اڑتالیس
49	انچاس
50	پچاس
51	اکیاون
52	باون
53	ترپن
54	چون
55	پچپن
56	چھپن
57	ستاون
58	اٹھاون
59	انسٹھ
60	ساٹھ
61	اکسٹھ
62	باسٹھ
63	تریسٹھ
64	چونسٹھ
65	پینسٹھ
66	چھیاسٹھ
67	ستاسٹھ
68	اٹھاسٹھ
69	انھتر
70	ستر
71	اکھتر
72	بھتر
73	تھتر
74	چوہتر
75	پچھتر
76	چھہتر
77	ستتر

Number	Mapping
78	اٹھتر
79	اناسی
80	اسی
81	اکاسی
82	بیاسی
83	تراسی
84	چوراسی
85	پچاسی
86	چھیاسی
87	ستاسی
88	اٹھاسی
89	نواسی
90	نوے
91	اکانوے
92	بانوے
93	ترانوے
94	چورانوے
95	پچانوے
96	چھیانوے
97	ستانوے
98	اٹھانوے
99	ننانوے
100	سو
1000	ھزار
100000	لاکھ
10000000	کروڑ
1000000000	ارب
100000000000	کھرب

Table B.1: Number Mappings

Number	Mapping
january	جنوری
february	فروری
march	مارچ
april	اپریل
may	مئی
june	جون
july	جولائی
august	اگست
september	ستمبر
october	اکتوبر
november	نومبر
december	دسمبر
1	جنوری
2	فروری
3	مارچ
4	اپریل
5	مئی
6	جون
7	جولائی
8	اگست
9	ستمبر
10	اکتوبر
11	نومبر
12	دسمبر

Table B.2: Month Mapping

Hindi Character	Urdu Mapping	Character Detail
ँ	ں	Noon Ghunna
ऄ	َ	Arabic Zabar or Fatha
न	ً	Arabic Fathatan

Hindi Character	Urdu Mapping	Character Detail
अ	ا	Alif
ओ	ا	Alif
अ	ء	Hamza
अ	ٔ	Hamza Above
अ	ع	Ain
आ	آ	Alif Madda
इ	ِ	Arabic Kasra or Zair
ई	ی	Yeh
उ	ُ	Arabic Damma or Paish
◌ُ	ؤ	Waw with hamza above
ऊ	ؤ	Waw with hamza above
ऋ	رِ	Reh with Zair
ए	ے	Baree Yeh
ऐ	آے	Aaey
ओ	ؤ	Waw with hamza above
औ	آو	Aao
क़	ق	Qaf
क	ک	Kaaf
ख	کھ	Khay
ख़	خ	Khay
ग	گ	Gaaf
घ	گھ	Ghaa
घ	چ	Chay
छ	چھ	Chhay
ज	ج	Jeem
झ	جھ	Jhay
ञ	یاں	Yaan
ट	ٹ	Tay
ठ	ٹھ	Thay
ड	ڈ	Daal

Hindi Character	Urdu Mapping	Character Detail
ढ	ڈھ	Dhaal
ण	ڈاں	Daan
त	ت	Tay
त	ط	Toain
थ	تھ	Thay
द	د	Dal
ध	دھ	Dhal
न	ن	Noon
प	پ	Pay
फ	پھ	Phay
ब	ب	Bay
ब	بھ	Bhay
म	م	Meem
र	ر	Ray
ल	ل	Laam
ळ	ّ	Arabic Shadda
व	و	Wow
श	ش	Sheen
स	ث	Say
स	س	Seen
स	ص	Saad
ह	ح	Hay
ह	ہ	Gol Heh
ह	ھ	Heh
ग़	غ	Ghain
ज़	ذ	Zaal
ज़	ز	Zay
ज़	ظ	Zoain
ज़	ژ	Zay
ज़	ض	Zaad

Hindi Character	Urdu Mapping	Character Detail
ड़	ڑ	Rhay
ढ़	ڑھ	Rhay
फ़	ف	Fay
य़	ۓ	Hamza Choti Yeh
०	0	Zero
१	1	One
२	2	Two
३	3	Three
४	4	Four
५	5	Five
६	6	Six
७	7	Seven
८	8	Eight
९	9	Nine
०	۰	Arabic Zero
१	۱	Arabic One
२	۲	Arabic Two
३	۳	Arabic Three
४	٤	Arabic Four
५	٥	Arabic Five
६	٦	Arabic Six
७	۷	Arabic Seven
८	۸	Arabic Eight
९	۹	Arabic Nine

Table B.3: Hindi to Urdu Character Mappings

Bibliography

[1] B. Mumtaz, S. Urooj, S. Hussain, and E. U. Haq, "Break index (bi) annotated speech corpus for urdu tts," in *Coordination and Standardization of Speech Databases and Assessment Techniques (O-COCOSDA), 2016 Conference of The Oriental Chapter of International Committee for*, IEEE, 2016, pp. 22–27.

[2] P. Khilari and V. Bhope, "A review on speech to text conversion methods," *International Journal of Advanced Research in Computer Engineering & Technology*, vol. 4, no. 7, 2015.

[3] B. G. Greene, J. S. Logan, and D. B. Pisoni, "Perception of synthetic speech produced automatically by rule: Intelligibility of eight text-to-speech systems," *Behavior Research Methods, Instruments, & Computers*, vol. 18, no. 2, pp. 100–107, 1986.

[4] D. H. Klatt, "Review of text-to-speech conversion for english," *The Journal of the Acoustical Society of America*, vol. 82, no. 3, pp. 737–793, 1987.

[5] . Aida–Zade, C. Ardil, and A. Sharifova, "The main principles of text-to-speech synthesis system," *International Journal of Signal Processing*, vol. 6, no. 1, pp. 13–19, 2010.

[6] D. Klatt, "The klattalk text-to-speech conversion system," in *Acoustics, Speech, and Signal Processing, IEEE International Conference on ICASSP'82.*, IEEE, vol. 7, 1982, pp. 1589–1592.

[7] T. K. Dagba and C. Boco, "A text to speech system for fon language using multisyn algorithm," *Procedia Computer Science*, vol. 35, pp. 447–455, 2014.

[8] N. Baloyi, "A text-to-speech synthesis system for xitsonga using hidden markov models," PhD thesis, University of Limpopo (Turfloop Campus), 2012.

[9] (). Top 30 languages by number of native speakers, [Online]. Available: http://www.vistawide.com/languages/top_30_languages.htm.

[10] (). Speech synthesis - formant synthesis, [Online]. Available: https://en.wikipedia.org/wiki/Speech_synthesis#Formant_synthesis.

[11] A. M. Saleem, H. Kabir, M. K. Riaz, M. M. Rafique, N. Khalid, and S. R. Shahid, "Urdu consonantal and vocalic sounds," *CRULP Annual Student Report*, 2002.

[12] X. Huang, A. Acero, J. Adcock, H.-W. Hon, J. Goldsmith, J. Liu, and M. Plumpe, "Whistler: A trainable text-to-speech system," in *Spoken Language, 1996. ICSLP 96. Proceedings., Fourth International Conference On*, IEEE, vol. 4, 1996, pp. 2387–2390.

[13] T. Merritt and S. King, "Investigating the shortcomings of hmm synthesis," in *Eighth ISCA Workshop on Speech Synthesis*, 2013.

[14] N. Swetha and K. Anuradha, "Text-to-speech conversion," *Int J Adv Trends Comput Sci Eng*, vol. 2, no. 6, pp. 269–278, 2013.

[15] E. Eide, A. Aaron, R. Bakis, W. Hamza, M. Picheny, and J. Pitrelli, "A corpus-based approach to expressive speech synthesis," in *Fifth ISCA Workshop on Speech Synthesis*, 2004.

[16] K. Tokuda, T. Yoshimura, T. Masuko, T. Kobayashi, and T. Kitamura, "Speech parameter generation algorithms for hmm-based speech synthesis," in *Acoustics, Speech, and Signal Processing, 2000. ICASSP'00. Proceedings. 2000 IEEE International Conference on*, IEEE, vol. 3, 2000, pp. 1315–1318.

[17] N. Kaiki, "Linguistic properties in the control of segmental duration for speech synthesis," *Talking Machines: Theories, Models, and Designs*, pp. 255–263, 1992.

[18] M. D. Riley, "Tree-based modeling of segmental durations," *Talking machines*, pp. 265–273, 1992.

[19] A. W. Black, H. Zen, and K. Tokuda, "Statistical parametric speech synthesis," in *Acoustics, Speech and Signal Processing, 2007. ICASSP 2007. IEEE International Conference on*, IEEE, vol. 4, 2007, pp. IV–1229.

[20] H. Zen, T. Nose, J. Yamagishi, S. Sako, T. Masuko, A. W. Black, and K. Tokuda, "The hmm-based speech synthesis system (hts) version 2.0.," in *SSW*, Citeseer, 2007, pp. 294–299.

[21] A. A. Raj, T. Sarkar, S. C. Pammi, S. Yuvaraj, M. Bansal, K. Prahallad, and A. W. Black, "Text processing for text-to-speech systems in indian languages.," in *SSW*, 2007, pp. 188–193.

[22] H. Kabir, S. R. Shahid, A. M. Saleem, and S. Hussain, "Natural language processing for urdu tts system," in *Multi Topic Conference, 2002. Abstracts. INMIC 2002. International*, IEEE, 2002, pp. 58–58.

[23] S. Hussain, "Phonological processing for urdu text to speech system," *Yadava, Y, Bhattarai, G, Lohani, RR, Prasain, B and Parajuli, K (eds.) Contemporary issues in Nepalese linguistics*, 2005.

[24] M. Y. Liberman and K. W. Church, "Text analysis and word pronunciation in text-to-speech synthesis," *Advances in speech signal processing*, pp. 791–831, 1992.

[25] H. R. Basit and S. Hussain, *Text processing for urdu tts system*, Poster presentation in Conference on Language and Technology 2014 (CLT 14), Karachi, Pakistan, 2014.

[26] M. D. Riley, "Some applications of tree-based modelling to speech and language," in *Proceedings of the workshop on Speech and Natural Language*, Association for Computational Linguistics, 1989, pp. 339–352.

[27] H. S. Elovitz, R. W. Johnson, A. McHugh, and J. E. Shore, "Automatic translation of english text to phonetics by means of letter-to-sound rules," NAVAL RESEARCH LAB WASHINGTON DC, Tech. Rep., 1976.

[28] H. Ku, W. Francis, *et al.*, "Computational analysis of present-day american english," 1967.

[29] R. Carlson, B. Granstrom, and S. Hunnicutt, "A multi-language text-to-speech module," in *Acoustics, Speech, and Signal Processing, IEEE International Conference on ICASSP'82.*, IEEE, vol. 7, 1982, pp. 1604–1607.

[30] D. H. Klatt, "Synthesis by rule of segmental durations in english sentences," *Frontiers of Speech Comm. Res.*, pp. 287–299, 1979.

[31] ——, "Software for a cascade/parallel formant synthesizer," *the Journal of the Acoustical Society of America*, vol. 67, no. 3, pp. 971–995, 1980.

[32] X. Huang, A. Acero, H. Hon, Y. Ju, J. Liu, S. Meredith, and M. Plumpe, "Recent improvements on microsoft's trainable text-to-speech system-whistler," in *Acoustics, Speech, and Signal Processing, 1997. ICASSP-97., 1997 IEEE International Conference on*, IEEE, vol. 2, 1997, pp. 959–962.

[33] (). Microsoft research's speech technology group web page, [Online]. Available: https://www.microsoft.com/en-us/research/?from=http%3A%2F%2Fresearch.microsoft.com%2Fresearch%2Fsrd.

[34] A. J. Hunt and A. W. Black, "Unit selection in a concatenative speech synthesis system using a large speech database," in *Acoustics, Speech, and Signal Processing, 1996. ICASSP-96. Conference Proceedings., 1996 IEEE International Conference on*, IEEE, vol. 1, 1996, pp. 373–376.

[35] S. King, "A beginners' guide to statistical parametric speech synthesis," *The Centre for Speech Technology Research, University of Edinburgh, UK*, 2010.

[36] T. Yoshimura, K. Tokuda, T. Masuko, T. Kobayashi, and T. Kitamura, "Duration modeling for hmm-based speech synthesis," in *Fifth International Conference on Spoken Language Processing*, 1998.

[37] J. J. Odell, "The use of context in large vocabulary speech recognition," PhD thesis, University of Cambridge, Mar. 1995.

[38] K. Tokuda, H. Zen, and A. W. Black, "An hmm-based speech synthesis system applied to english," in *IEEE Speech Synthesis Workshop*, 2002, pp. 227–230.

[39] H. D. Harashima, "Review of "voicetext"," *Electronic Journal of Foreign Language Teaching*, vol. 3, no. 1, pp. 131–135, 2006.

[40] (2002). Hmm based synthesis system version 2.2, [Online]. Available: http://hts.sp.nitech.ac.jp/.

[41] R. A. Clark, K. Richmond, and S. King, "Multisyn: Open-domain unit selection for the festival speech synthesis system," *Speech Communication*, vol. 49, no. 4, pp. 317–330, 2007.

[42] M. B. Ganai and E. J. Arora, "Text-to-speech conversion," 2016.

[43] R. E. Donovan and P. C. Woodland, "Improvements in an hmm-based speech synthesiser," in *Fourth European Conference on Speech Communication and Technology*, 1995.

[44] A. S. House, C. E. Williams, M. H. Hecker, and K. D. Kryter, "Articulation-testing methods: Consonantal differentiation with a closed-response set," *The Journal of the Acoustical Society of America*, vol. 37, no. 1, pp. 158–166, 1965.

[45] T. Masuko, K. Tokuda, T. Kobayashi, and S. Imai, "Speech synthesis using hmms with dynamic features," in *Acoustics, Speech, and Signal Processing, 1996. ICASSP-96. Conference Proceedings., 1996 IEEE International Conference on*, IEEE, vol. 1, 1996, pp. 389–392.

[46] M. Tamura, T. Masuko, K. Tokuda, and T. Kobayashi, "Speaker adaptation for hmm-based speech synthesis system using mllr," in *the third ESCA/COCOSDA Workshop (ETRW) on Speech Synthesis*, 1998.

[47] T. Yoshimura, K. Tokuda, T. Masuko, T. Kobayashi, and T. Kitamura, "Speaker interpolation for hmm-based speech synthesis system," *Acoustical Science and Technology*, vol. 21, no. 4, pp. 199–206, 2001.

[48] X. Huang, A. Acero, H.-W. Hon, and R. Reddy, *Spoken language processing: A guide to theory, algorithm, and system development*. Prentice hall PTR Upper Saddle River, 2001, vol. 1.

[49] O. Karaali, G. Corrigan, I. Gerson, and N. Massey, "Text-to-speech conversion with neural networks: A recurrent tdnn approach," *arXiv preprint cs/9811032*, 1998.

[50] T. Yoshimura, G. E. Henter, O. Watts, M. Wester, J. Yamagishi, and K. Tokuda, "A hierarchical predictor of synthetic speech naturalness using neural networks.," in *INTERSPEECH*, 2016, pp. 342–346.

[51] K. T. Alan W Black Simon King. (Jan. 2009). The blizzard challenge 2009, [Online]. Available: https://synsig.org/images/archive/9/94/20090121161941!Blizzard_2009_full.pdf.

[52] Z. Wu, O. Watts, and S. King, "Merlin: An open source neural network speech synthesis system," *Proc. SSW, Sunnyvale, USA*, 2016.

[53] H. Ze, A. Senior, and M. Schuster, "Statistical parametric speech synthesis using deep neural networks," in *Acoustics, Speech and Signal Processing (ICASSP), 2013 IEEE International Conference on*, IEEE, 2013, pp. 7962–7966.

[54] Y. Fan, Y. Qian, F.-L. Xie, and F. K. Soong, "Tts synthesis with bidirectional lstm based recurrent neural networks," in *Fifteenth Annual Conference of the International Speech Communication Association*, 2014.

[55] P. K. Muthukumar and A. W. Black, "Recurrent neural network postfilters for statistical parametric speech synthesis," *arXiv preprint arXiv:1601.07215*, 2016.

[56] A. A. Shah, A. W. Ansari, and L. Das, "Bi-lingual text to speech synthesis system for urdu and sindhi," in *National Conf. on Emerging Technologies*, 2004, pp. 20 126–130.

[57] Z. Ahmed and J. P. Cabral, "Hmm-based speech synthesiser for the urdu language," in *Spoken Language Technologies for Under-Resourced Languages*, 2014.

[58] W. B. Kleijn and K. K. Paliwal, *Speech coding and synthesis*. Elsevier Science Inc., 1995.

[59] (). Center for research in urdu language processing, [Online]. Available: http://www.cle.org.pk/software/ling_resources.htm.

[60] S. Hussain, "Letter-to-sound conversion for urdu text-to-speech system," in *Proceedings of the workshop on computational approaches to Arabic script-based languages*, Association for Computational Linguistics, 2004, pp. 74–79.

[61] O. Nawaz and T. Habib, "Hidden markov model (hmm) based speech synthesis for urdu language," in *Conference on Language & Technology (CLT)*, 2014.

[62] T. Fukada, K. Tokuda, T. Kobayashi, and S. Imai, "An adaptive algorithm for mel-cepstral analysis of speech," in *Acoustics, Speech, and Signal Processing, 1992. ICASSP-92., 1992 IEEE International Conference on*, IEEE, vol. 1, 1992, pp. 137–140.

[63] W. Anwar, X. Wang, L. Li, and X.-L. Wang, "A statistical based part of speech tagger for urdu language," in *Machine Learning and Cybernetics, 2007 International Conference on*, IEEE, vol. 6, 2007, pp. 3418–3424.

[64] S. Bird, E. Klein, and E. Loper, "Introduction to natural language processing," *University of Pennsylvania*, 2007.

[65] J. Carlberger and V. Kann, "Implementing an efficient part-of-speech tagger," *Software: Practice and Experience*, vol. 29, no. 9, pp. 815–832, 1999.

[66] N. Durrani and S. Hussain, "Urdu word segmentation," in *Human Language Technologies: The 2010 Annual Conference of the North American Chapter of the Association for Computational Linguistics*, Association for Computational Linguistics, 2010, pp. 528–536.

[67] D. Parveen, R. Sanyal, and A. Ansari, "Clause boundary identification using classifier and clause markers in urdu language," *Polibits*, no. 43, pp. 61–65, 2011.

[68] J. Pierrehumbert, "Synthesizing intonation," *The Journal of the Acoustical Society of America*, vol. 70, no. 4, pp. 985–995, 1981.

[69] A. W. Black and P. Taylor, "Chatr: A generic speech synthesis system," in *Proceedings of the 15th conference on Computational linguistics-Volume 2*, Association for Computational Linguistics, 1994, pp. 983–986.

[70] A. W. Black, "Unit selection and emotional speech," in *Eighth European Conference on Speech Communication and Technology*, 2003.

[71] Y. Qian, F. Soong, Y. Chen, and M. Chu, "An hmm-based mandarin chinese text-to-speech system," in *Chinese Spoken Language Processing*, Springer, 2006, pp. 223–232.

[72] O. Abdel-Hamid, S. M. Abdou, and M. Rashwan, "Improving arabic hmm based speech synthesis quality," in *Ninth International Conference on Spoken Language Processing*, 2006.

[73] S. Martincic-Ipsic and I. Ipsic, "Croatian hmm based speech synthesis," in *Information Technology Interfaces, 2006. 28th International Conference on*, IEEE, pp. 251–256.

[74] H. Zen, K. Tokuda, and A. W. Black, "Statistical parametric speech synthesis," *Speech Communication*, vol. 51, no. 11, pp. 1039–1064, 2009.

[75] R. E. Donovan and P. C. Woodland, "A hidden markov-model-based trainable speech synthesizer," *Computer speech & language*, vol. 13, no. 3, pp. 223–241, 1999.

[76] J. F. Pitrelli, "Tobi prosodic analysis of a professional speaker of american english," in *Speech Prosody 2004, International Conference*, 2004.

[77] Z.-H. Ling, S.-Y. Kang, H. Zen, A. Senior, M. Schuster, X.-J. Qian, H. M. Meng, and L. Deng, "Deep learning for acoustic modeling in parametric speech generation: A systematic review of existing techniques and future trends," *IEEE Signal Processing Magazine*, vol. 32, no. 3, pp. 35–52, 2015.

[78] H. Zen, "Acoustic modeling in statistical parametric speech synthesis–from hmm to lstm-rnn," *Proc. MLSLP*, 2015.

[79] G. Cawley and P. Noakes, "Lsp speech synthesis using backpropagation networks," in *Artificial Neural Networks, 1993., Third International Conference on*, IET, 1993, pp. 291–294.

[80] H. Lu, S. King, and O. Watts, "Combining a vector space representation of linguistic context with a deep neural network for text-to-speech synthesis," in *Eighth ISCA Workshop on Speech Synthesis*, 2013.

[81] Y. Qian, Y. Fan, W. Hu, and F. K. Soong, "On the training aspects of deep neural network (dnn) for parametric tts synthesis," in *Acoustics, Speech and Signal Processing (ICASSP), 2014 IEEE International Conference on*, IEEE, 2014, pp. 3829–3833.

[82] L.-H. Chen, T. Raitio, C. Valentini-Botinhao, Z.-H. Ling, and J. Yamagishi, "A deep generative architecture for postfiltering in statistical parametric speech synthesis," *IEEE/ACM Transactions on Audio, Speech and Language Processing (TASLP)*, vol. 23, no. 11, pp. 2003–2014, 2015.

[83] (). Phonetically rich urdu speech corpus, [Online]. Available: http://www.cle.org.pk/software/ling_resources/phoneticallyrichurduspeechcorpus.htm.

[84] W. D. Voiers, "Diagnostic evaluation of speech intelligibility," *Speech intelligibility and speaker recognition*, 1977.