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MALAY STATISTICAL PARAMETRIC SPEECH SYNTHESIS WITH INTELLIGIBILITY IMPROVEMENT USING ARTIFICIAL INTELLIGENCE

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Dedicated to my mum and dad,

my brother and sisters,

and my beloved friends.

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ABSTRACT

Speech synthesis is important nowadays and could be a great aid in various applications. So it is important to build a simple, reliable, light-weight, ease of use speech synthesizer. However, conventional speech synthesizers require tedious human efforts to prepare high quality recorded database, and the intelligibility of synthetic speech may decrease due to the appearance of polyphone (character with more than 1 pronunciation) because the speech synthesizer may not contain the definition of the polyphones. Moreover, the ready speech synthesizers in market are mostly built in Unit Selection method, which is large in database size and relying on Malay linguist knowledge. In this study, statistical parametric speech synthesis method has been adopted using lab speech and free speech data harvested online. The intelligibility improvement has been achieved using Active Learning and Feedforward Neural Network with Back-Propagation. The amount of training data used remained the same throughout this study. The result was evaluated using perception test. The listening test showed that the intelligibility of synthetic speech has been improved about 20%-30% using the artificial intelligence technique. Volunteers were invited to take part in Active Learning experiment. The result showed no controversy between the result done by volunteers and the correct answer. In conclusion, a light-weight Malay speech synthesizer has been created without relying on Malay linguist knowledge. Using free source as training data can ease the human effort in preparing training database and using artificial intelligence technique can improve the intelligibility of synthetic speech under the same amount of training data used.

ABSTRAK

Sintesis ucapan adalah penting pada hari ini dan boleh menjadi bantuan yang besar untuk pemulihan masalah menghasilkan ucapan. Jadi adalah penting untuk membina pensintesis yang mudah, boleh dipercayai dan mudah alih. Walau bagaimanapun, pensintesis ucapan konvensional memerlukan banyak usaha manusia untuk menyediakan data rakaman, dan kejelasan ucapan sintetik mungkin berkurangan akibat kemunculan *polyphone* (watak dengan lebih daripada 1 sebutan) dalam perkataan yang berbeza kerana pensintesis ucapan tersebut mungkin tidak mengandungi definisi maklumat *polyphone*. Selain itu, pensintesis ucapan yang terdapat dalam pasaran kebanyakannya dibina dengan kaedah Pemilihan Unit, menyebabkan saiz pangkalan data yang besar dan bergantung kepada pengetahuan ahli bahasa Melayu. Dalam kajian ini, statistik parametrik kaedah sintesis ucapan telah digunakan menggunakan sumber bebas yang boleh didapatkan daripada internet secara percuma. Peningkatan kejelasan telah dicapai dengan menggunakan beberapa teknik *Artificial Intelligence (AI)* seperti *Active Learning (AL)* dan *Feedforward Neural Network (FNN)* dengan *Back-Propagation (BP)*. Jumlah data latihan yang digunakan adalah tetap sama sepanjang kajian ini. Keputusan ini telah dibandingkan dengan data terlatih yang direkodkan. Ujian menunjukkan bahawa kejelasan ucapan sintetik telah bertambah kira-kira 20% - 30% menggunakan teknik AI tersebut. Sukarelawan-sukarelawan telah dijemput untuk mengambil bahagian dalam eksperimen pembelajaran aktif. Hasilnya menunjukkan tiada sebarang kontroversi antara penutur asli dan berbilang sukarelawan. Kesimpulannya, ucapan pensintesis Melayu yang ringan telah dicipta tanpa bergantung kepada pengetahuan ahli bahasa Melayu. Dengan menggunakan sumber bebas sebagai data latihan boleh mengurangkan usaha manusia dalam penyediaan data latihan dan menggunakan teknik AI boleh meningkatkan kejelasan ucapan sintetik di bawah jumlah data latihan yang sama.

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LIST OF ABBREVIATIONS

AL	-	Active Learning
BIC	-	Bayesian Information Criterion
BP	-	Back-Propagation
CVC	-	Consonant Vowel Consonant
CV	-	Consonant Vowel
DRT	-	Diagnostic Rhyme Test
EM	-	Expectation-Maximization
FNN	-	Feedforward Neural Network
GMM	-	Gaussian Mixture model
HMM	-	Hidden Markov Model
HTK	-	Hidden Markov model Toolkit
LLR	-	Log Likelihood Ratio
MFCC	-	Mel-frequency Cepstral Coefficient
MOS	-	Mean Opinion Score
MRT	-	Modified Rhyme Test
PDF	-	Probability Distribution Function
QBB	-	Query-by-Bagging
QBC	-	Query-by-Committee
SM	-	Standard Malay
STRAIGHT	-	Speech Transformation and Representation using
	-	Adaptive Interpolation of weiGHTed spectrum
SUS	-	Semantically Unpredictable Sentences
VAD	-	Voice Activity Detection
WER	-	Word Error Rate

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CHAPTER 1

INTRODUCTION

1.1 Background Study

Speech synthesis is a method of converting written text into spoken speech (Sproat *et al.*, 1995; Dutoit and Stylianou, 2003; Dutoit, 1997). This process is also known as Text-to-Speech (TTS) generation. It is a reversion of speech recognition (Rabiner, 1989) which recognizes speech and transcribes the speech into text. From time to time, the evolution of speech synthesis has made speech synthesizers robust and reliable in handling many applications such as telephony services, screen readers for the blind or visually impaired, navigation systems and many more (Lemmetty, 1999). For medical purposes, this technique could provide a substitute for mute people to communicate with other people. A famous example of a person with a speech disability is the theoretical physicist Stephen Hawking (Larsen, 2005). He is almost entirely paralyzed and uses synthetic speech to communicate with others. In order to build a high quality speech synthesizer, the development should take care of the following aspects:

1. **Naturalness** (Taylor, 2009). People are sensitive to speech, not only by the words spoken but how the person speaks. Mechanical or robotic synthetic voices are annoying and irritating after a long time listening to that type of voice. Therefore, one of the goals for a speech synthesizer is to generate natural sounding speech.
2. **Intelligibility** (Benoit *et al.*, 1996). The key significance of a speech synthesizer is to deliver messages. A good speech synthesizer can replace human efforts and take over many areas of speech. There is no point building a speech synthesizer if it produces speech that we cannot understand. Therefore, speech intelligibility is an important factor to be considered when making high quality speech synthesizers.

3. **Able to produce novel speech** (Taylor, 2009). Normally the quality of speech synthesizers depend on the condition of the training data. The way to design and produce a high quality training database is highly sophisticated. However, a good speech synthesizer should be able to speak any novel words beyond the training data. It is less practical if the speech synthesizer is only able to speak utterances within the training corpus. Moreover, the uttered novel words should also be natural and intelligible to listeners.

In short, a speech synthesis system should be efficient, be able to produce intelligible speech, and sound natural for novel words (Tabet and Boughazi, 2011).

With the improvement of computer technology nowadays, speech synthesis has evolved from knowledge-based into data-based (Black *et al.*, 2007). Speech synthesizer can be built from a sufficient amount of human speech data. One of the example of data-based speech synthesizers is Statistical Parametric Speech Synthesis. It is a data-based speech synthesis method and it has gained more and more attentions recently. It models the data of parametric representations of natural speech and generates similar sounding speech segments during synthesis. This is in contradiction to the Unit Selection method (Conkie, 1999) which keeps the speech data unmodified and generating synthetic voices using natural speech data. However, experiments have shown that the synthetic voice generated using the Statistical Parametric Speech Synthesis method is natural and intelligible. In the Blizzard Challenge 2005 (Bennett and Christina, 2005) and 2006 (Clark *et al.*, 2006), a common speech database was provided to participants to build synthetic voices. The results showed that the synthetic speech generated using the Statistical Parametric Speech Synthesis method was preferred due to its naturalness. The synthetic speech was intelligible and understandable to the listeners and it was proven using the Word Error Rate (WER) score (Zechner and Waibel, 2000). This result has shown that Statistical Parametric Speech Synthesis is capable to synthesize good quality speech.

Besides, Statistical Parametric Speech Synthesis also offers several advantages which increases its flexibility and extends speech technology:

1. Unit Selection chooses a finite unit from its database. It may face a problem of choosing inadequate examples. This can be viewed as a lack of database coverage. However, Statistical Parametric Speech Synthesis generates speech using statistical data. Therefore, it has better acoustic space coverage than the Unit Selection method and a wider range of units are available.

2. The Statistical Parametric Speech Synthesis method stores the statistical data of the acoustic model whereas the Unit Selection method stores real speech segments. Therefore, the Statistical Parametric Speech Synthesis method can achieve a smaller footprint than the Unit Selection method. For example, the footprint of voices of Nitech HMM-Based Speech Synthesis System in Blizzard Challenge 2005 is less than 2MB (Zen *et al.,* 2007).

3. The Statistical Parametric Speech Synthesis method is more robust than the Unit Selection method. This is because the real speech database of the Unit Selection method may suffer from noise and fluctuation disturbances due to the recording surroundings and the recording of a real human's speech may not practically cover all the phonetic possibilities. However, research has shown that Statistical Parametric Speech Synthesis method can resolve these problems (Yamagishi *et al.*, 2008).

4. The representation of speech in the Statistical Parametric Speech Synthesis method is statistical data of the spectrum, duration and excitation. Therefore these parameters can be separately modified and monitored.

5. The voice characteristics, emotions and speaking styles of synthetic speech can be transformed into Statistical Parametric Speech Synthesis. This is the key flexibility of this method. The transformation can be done by utilizing adaptation (Masuko *et al.*, 1997), eigenvoice (Kuhn *et al.*, 2000), interpolation (Yoshimura *et al.*, 1997) and multiple regression (Miyanaga *et al.*, 2004).

6. Statistical Parametric Speech Synthesis uses statistical principles that are defined in mathematical frameworks. The tuning parameters are lesser than the Unit Selection method which requires manual tuning and settings for various control.

1.2 Problem Statement

In order to build a reliable speech synthesizer especially targeted to Malaysian, the following problems should be considered.

1. The available speech synthesizers in the market are mostly in English. There are not many Malay speech synthesizers ready for Malaysian. The available Malay speech synthesizers are larger in file size (>25MB) (Tan, 2009; Lim, 2013) which is not peactical to be used in light-scale embedded system (Kim *et al.*, 2006).

2. The process of preparing training data in building a speech synthesizer is

sophisticated and cumbersome. It involves gathering words from sources, constructing suitable scripts which includes all the phonemes in the Malay language, the recording of scripts and the recording of sessions should be conducted in a high quality recording studio. It is expensive to construct a real speech database over a long period of time.

3. Conventionally, to build a speech synthesizer requires the knowledge of language expert to precisely draw the boundary of every phoneme because phonemes are the basic synthesis unit for a speech synthesizer. However, consulting a language expert adds extra workload and it is expensive to do so.
4. The intelligibility of synthetic speech is the main concern in every speech synthesizer. Most of the speech synthesizers might face the problem of low intelligibility especially in synthesizing words which are not found in database.

The aim of this research is to solve the aforementioned problems and create a reliable Malay speech synthesizer. Several techniques have been applied to resolve the problems and it will be explained in Chapter 3 and 4.

1.3 Objectives

This study is aiming to solve the related problems in building a speech synthesizer. Therefore, the objectives are:

1. To build a Malay speech synthesizer with a low footprint (data size).
2. To alleviate the problem of preparing database in Statistical Parametric Speech Synthesis System by including free data harvested online.
3. To exclude the dependency of linguist in building speech synthesizer.
4. To improve the synthetic speech intelligibility using Active Learning (AL) and Feedforward Neural Network (FNN) with Back-Propagation (BP) while the same amount of training data was used.

The block diagram of this study is shown in Figure 1.1.

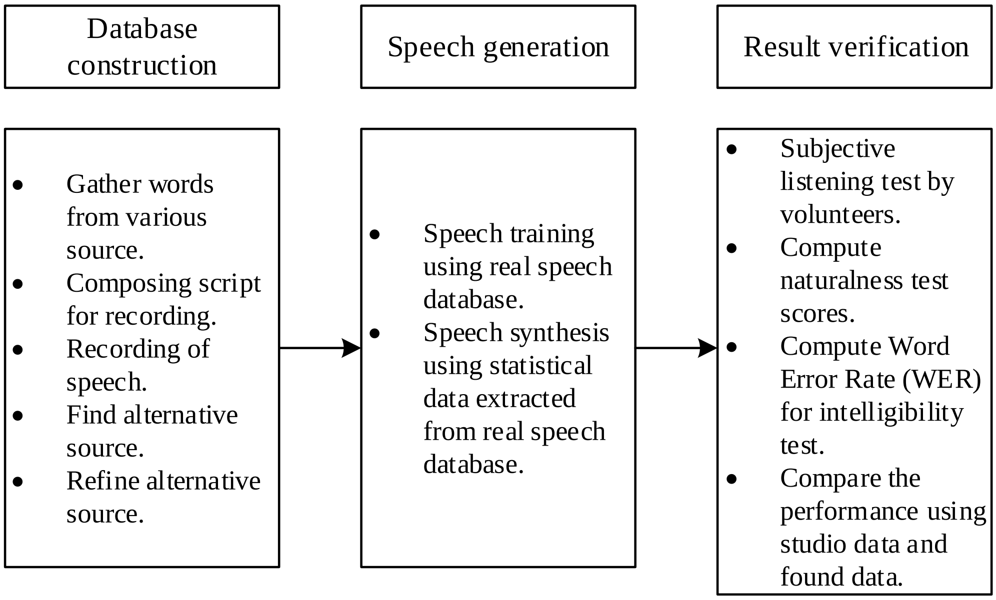

Figure 1.1: Mapping of procedures in this study

1.4 Scope of the Study

This study follows several scopes and they are:

1. The Malay speaking style used in this study is Standard Malay (SM) (Seman and Jusoff, 2008) which is the usual Malay speaking style spoken by Malaysians. No other accents like Kelantan Malay, Ulu Muar Malay and so on were used throughout this study. The reason Standard Malay is going to be used is to make the speech synthesizer suitable to be used in almost every area of speech, for example, speech rehabilitation, education, or any speech emitting devices like computer and smart phones. Standard Malay is also easily understandable by almost every Malaysian.
2. The invited speaker for the recording of the database is a Malay adult native speaker. This is to ensure the database contained the correct Malay pronunciations and that the voice is mature. Correct pronunciation can improve the synthetic speech intelligibility, therefore the synthetic speech would be easily understood.

3. The free training data harvested online is clear in pronunciation, low in background noise and no overlapped with any other voices or music.
4. The synthetic speech synthesized in this study would be in normal reading style. No any other voice tone would be incurred like happy, sad or angry emotions.

1.5 Thesis Organization

Chapter 1 briefly introduced the background of the study. It gave a basic overview on speech synthesis technologies and briefly talked about the state-of-the-art Statistical Parametric Speech Synthesis. It also presented the problem statements, objective and the scope of this study.

Chapter 2 provided a literature review of this study. It included a basic overview on the Malay language. The history of the speech synthesizer was introduced in this chapter in a timeline fashion. Comparisons between state of the art speech synthesizers were also discussed. A decision was made on which type of speech synthesizer was used in this research and the reasons. The technical review on statistical parametric speech synthesizer which was used in this thesis was presented from the basic model applied in this method until how it produces synthetic speech sounds. A brief discussion on how speech synthesizer can help people was presented within this chapter. The evaluation methods available were overviewed and only one evaluation approach was selected based on the suitability and effectiveness. How the result was statistically compared was also introduced in this chapter.

Chapter 3 is the Methodology used in this study. It involved how the training database was constructed, how the free source was obtained online, how the modifications were done to the found data, how the Artificial Intelligence techniques (Feedforward Neural Network with Back Propagation and Active Learning) was applied, how the front end processing was conducted, how the speech training and speech synthesis works, and how the listening test was carried out to test the quality of synthetic speech.

Chapter 4 is the Result and Discussion section. It showed the accuracy of classifiers trained with Feedforward Neural Network with Back Propagation and Active Learning. It also presented the listening test result of both Naturalness Test and Intelligibility Test in the experiments involving Found Data, Feedforward

Neural Network with Back Propagation and Active Learning. The total footprint or total file size of the speech synthesizers was displayed in detail. The significant difference test result was also calculated and compared and this chapter was concluded with discussions for all the experiments and benchmark with other Malay speech synthesizers.

Chapter 5 outlined the conclusion and explained the contributions of this study. The future work was also presented in the end of this chapter.

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