ETHNIC RECOGNITION SYSTEM FOR MALAY LANGUAGE SPEAKERS USING GAMMATONE FREQUENCY CEPSTRAL COEFFICIENTS PITCH (GFCCP) AND PATTERN CLASSIFICATION

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To my beloved mother who taught me to trust in Allah and believe in hard work. To my husband and children who have always stood by me and understand my difficulties in completing this thesis.

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ABSTRACT

Malaysia is a multi-racial country consisting of many ethnic groups such as the Malay, Chinese, Indian, and Bumiputera, also known as a multilingual society. The Malay language is a non-tonal language, which does not need lexical stress. The study on recognizing the speaker's ethnicity is important as it has many potential and useful applications such as improving the interaction between robots and humans, audio forensic, telephone banking, and electronic commerce. Feature extraction, voice textindependent, and variability coverage are issues related to speaker recognition systems. The research focused on establishing a novel method, Gammatone Frequency Cepstral Coefficients and pitch (GFFCP) coupled with the K-Nearest Neighbours (KNN) and the voice text-independent system were used to identify the speaker's ethnicity. The speech corpus consisted of a collection of readings of Malay texts by both genders with ages ranging from 10 to 48 years old and classified into three ethnic groups: Malay, Chinese, and Indian. GFCC and Mel Frequency Cepstral Coefficients (MFCC) were used to represent the human auditory system. Pitch was added to MFCC and GFCC, as it contributes to the differences in the human voice and is difficult to imitate. The use of Naïve Bayes, Support Vector Machine (SVM), and KNN as classifiers was to quantify the pattern classification performance. The dataset used the hold-out validation methods (80% training, 20% testing) to split the data for training and testing. The system's performance was assessed based on the validation and prediction accuracy. The results revealed that the GFCCP obtained the highest validation and prediction accuracy from the KNN classifier. The validation accuracy was 100%, 99.6%, and 99.2% for 12, 24, and 34 speakers, respectively, while the prediction accuracy was 89.98%, 73.56%, and 72.36% for 12, 24, and 34 speakers, respectively. An important finding in the study is that the combination of the pitch with MFCC and GFCC provided better accuracy, with the latter performing better than the former, compared with those of MFCC and GFCC alone under noisy conditions.



ABSTRAK

Malaysia merupakan negara berbilang kaum yang terdiri daripada pelbagai etnik seperti Melayu, Cina, India, dan Bumiputera, dan dikenali sebagai masyarakat berbilang bahasa. Bahasa Melayu merupakan bahasa *non-tonal*, yang tidak memerlukan tekanan leksikal. Kajian pengecaman etnik penutur penting kerana berpotensi dan berguna dalam aplikasi untuk meningkatkan interaksi antara robot dan manusia, forensik audio, perbankan telefon, dan perdagangan elektronik. Pengekstrakan ciri, bebas teks suara dan liputan kebolehubahan antara isu yang berkaitan dengan sistem pengecaman penutur. Penyelidikan ini menumpukan kepada mewujudkan kaedah baru, di mana Gammatone Frequency Cepstral Coefficients dan nada (GFFCP) ditambah dengan K-Nearest Neighbours (KNN) menggunakan sistem bebas teks suara untuk mengenal pasti etnik penutur. Korpus pertuturan terdiri daripada koleksi bacaan teks Melayu oleh kedua-dua jantina dengan umur antara 10 hingga 48 tahun dan diklasifikasikan kepada tiga kumpulan etnik: Melayu, Cina, dan India. GFCC dan Mel Frequency Cepstral Coefficients (MFCC) digunakan kerana mewakili sistem pendengaran manusia. Nada ditambah kepada MFCC dan GFCC, kerana ia dapat membezakan suara manusia dan sukar ditiru. Penggunaan Naïve Bayes, Mesin Vektor Sokongan (SVM), dan KNN sebagai pengelas bertujuan mengukur prestasi pengelasan corak. Set data menggunakan kaedah hold-out (80% latihan, 20% ujian) untuk memisahkan data latihan dan ujian. Prestasi dinilai berdasarkan ketepatan pengesahan dan ramalan. Keputusan menunjukkan GFCCP memperoleh ketepatan pengesahan dan ramalan tertinggi daripada pengelas KNN. Ketepatan pengesahan adalah 100%, 99.6%, dan 99.2% untuk 12, 24, dan 34 penutur, masing-masing, manakala ketepatan ramalan ialah 89.98%, 73.56% dan 72.36% untuk 12, 24, dan 34 penutur, masing-masing. Penemuan penting kajian ialah gabungan GFCC dan MFCC dengan nada memberi ketepatan lebih baik, berbanding MFCC dan GFCC sahaja dalam situasi hingar.



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LIST OF ABBREVIATIONS

ABI	-	Accents of British Isles
ADAM	-	Advanced Development Autonomous Machine
AI	-	Artificial Intelligence
ANN	-	Artificial Neural Network
ASR	-	Automatic Speaker Recognition
AUC	-	Area Under the Curve
CER	-	Character Error Rate
CLSP	-	Centre for Language and Speech Processing
CNN	-	Centre for Language and Speech Processing Convolutional Neural Network
DA-DNN7L	-	Data Augmentation Deep Neural Network 7 Layers
DBN	-	Deep Belief Network
DCT	-	Discrete Cosine Transform
DFT	-	Discrete Fourier Transform
DL	115	Deep Learning
DL DNA ERP	US	Deep Learning Deoxyribonucleic Acid
	5	
DNA ERP	US	Deoxyribonucleic Acid
DNA DNA	US	Deoxyribonucleic Acid Deep Neural Architecture
DNA DNA DNN		Deoxyribonucleic Acid Deep Neural Architecture Deep Neural Network
DNA DNA DNN DT-CWPT		Deoxyribonucleic Acid Deep Neural Architecture Deep Neural Network Dual-Tree Complex Wavelet Packet Transform
DNA DNA DNN DT-CWPT DTW	- - - - -	Deoxyribonucleic Acid Deep Neural Architecture Deep Neural Network Dual-Tree Complex Wavelet Packet Transform Dynamic Time Warping
DNA DNA DNN DT-CWPT DTW DWPT	- - - - - -	Deoxyribonucleic Acid Deep Neural Architecture Deep Neural Network Dual-Tree Complex Wavelet Packet Transform Dynamic Time Warping Discrete Wavelet Packet Transform
DNA DNA DNN DT-CWPT DTW DWPT ECG	- - - - - -	Deoxyribonucleic Acid Deep Neural Architecture Deep Neural Network Dual-Tree Complex Wavelet Packet Transform Dynamic Time Warping Discrete Wavelet Packet Transform Electrocardiogram
DNA DNA DNN DT-CWPT DTW DWPT ECG EEG	- - - - - - -	Deoxyribonucleic Acid Deep Neural Architecture Deep Neural Network Dual-Tree Complex Wavelet Packet Transform Dynamic Time Warping Discrete Wavelet Packet Transform Electrocardiogram Electroencephalogram
DNA DNA DNN DT-CWPT DTW DWPT ECG EEG EER	- - - - - - - -	Deoxyribonucleic Acid Deep Neural Architecture Deep Neural Network Dual-Tree Complex Wavelet Packet Transform Dynamic Time Warping Discrete Wavelet Packet Transform Electrocardiogram Electrocardiogram

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	EM	-	Expectation Maximization
	ERB	-	Equivalent Rectangular Bandwidth
	ERICA	-	ERATO Intelligent Conversational Android
	FAR	-	False Acceptance Rate
	FCM	-	Fuzzy C-Means
	FFT	-	Fast Fourier Transform
	FIR	-	Finite Impulse Transform
	FN	-	False Negative
	FNR	-	False Negative Rate
	FP	-	False Positive
	FPR	-	False Positive Rate
	FRR	-	False Rejection Rate
	FVQ	-	Fuzzy Vector Quantization
	FVQ2	-	Fuzzy Vector Quantization2
	GFCC	-	Gammatone Frequency Cepstral Coefficient
	GFCCP	-	Gammatone Frequency Cepstral Coefficient Gammatone Frequency Cepstral Coefficient Pitch Gaussian Mixture Model
	GMM	-	Gaussian Mixture Model
	GMM-JFA	-	Gaussian Mixture Model-Joint Factor Analysis
	GMM-UBM	-	Gaussian Mixture Model-Universal Background Model
	HASR	-	Human Assisted Speaker Recognition
8-11	НММ		Hidden Markov Model
	HRI	72	Human-Robot Interaction
	IoT	-	Internet of Things
	KNN	-	K-Nearest Neighbour
	LID	-	Language Identification
	LPC	-	Linear Prediction Coding
	LPCC	-	Linear Prediction Cepstral Coefficient
	MFCC	-	Mel-Frequency Cepstral Coefficient
	MFCCP	-	Mel-Frequency Cepstral Coefficient Pitch
	MGFCC	-	Modified GFCC
	ML	-	Machine Learning
	MLAN	-	Multi-level Adaptive Network
	MLP	-	Multi-Layer Perception
	MNN	-	Modular Neural Network

NB	-	Naïve Bayes
NICO	-	Neuro-Inspired Companion
NIST	-	National Institute of Standards and Technology
NIST 2003	-	National Institute of Standards and Technology 2003
PD	-	Partial Discharge
RCC	-	Real Cepstral Coefficient
RNN	-	Recurrent Neural Network
ROC	-	Receiver Operating Characteristics
SNR	-	Signal to Noise Ratio
SRE	-	Speaker Recognition Evaluation
STE	-	Short-Term Energy
STFT	-	Short-Time Fourier Transform
SVM	-	Support Vector Machine
TIMIT	-	Texas Instruments and Massachusetts Institute of
		Technology
TN	-	Technology True Negative True Positive True Positive Rate
ТР	-	True Positive
TPR	-	True Positive Rate
VQ	-	Vector Quantization
WER	-	Word Error Rate
WPT	- 6'	Wavelet Packet Transform
ZCR	U D	Zero-Crossing Rate

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CHAPTER 1

INTRODUCTION

1.1 **Background of the Study**

Biometrics is widely used to identify and authenticate individuals trustworthily and promptly through unique biological characteristics. As shown in Figure 1.1, biometrics can be classified into physiological and behavioural categories (Porta et al., 2021; Rousan and Intrigila, 2020).

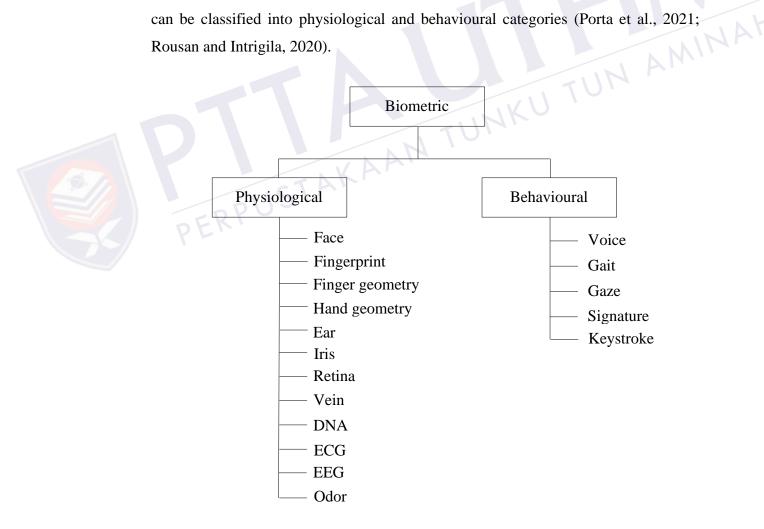


Figure 1.1: Types of biometrics: physiological and behavioural

The former refers to features identified through the five senses, i.e., sight, sound, smell, taste, and touch. For example, face, fingerprint, iris, retina, vein, ECG, odour, etc. The latter is usually based on how people conduct themselves, including voice, gait, gaze, signature, and keystroke (Rousan and Intrigila, 2020).

Biometric technology has various characteristics, by which we can distinguish their applications. Table 1.1 compares the most used biometric types based on the characteristics of biometric technology such as distinctiveness, complexity, universality, quantifiability, performance, comparison, collect capacity, acceptance, cost, and use.

 Table 1.1: A comparison of biometric types based on the characteristics of biometric (Rousan and Intrigila, 2020)

Biometric Identifier	Distinctiveness	Complexity	Universality	Quantifiability	Performance	Comparison	Collect Capacity	Acceptance	Cost	Use
Fingerprint	М	L	Н	Н	М	Н	н	Н	М	н
Iris	Н	L	Н	н	Н	Н	н	Н	н	М
Facial	М	L	Н	Н	м	М	н	н	М	М
Palm	м	Н	Н	н	м	м	L	L	н	М
Ear	М	Н	н	н	L	L	L	L	н	L
Footprint	М	Н	М	М	L	L	L	L	н	L
Finger vein	Н	Н	н	L	н	Н	L	L	н	L
Voice	М	н	Н	М	М	М	L	L	Н	L
Signature	L	н	н	н	L	L	м	Н	L	L
Keystroke dynamics	L	м	М	L	L	L	L	L	н	L



H = High; M = Medium; L = Low

Based on the information in the table, it can be deduced that voice is one of the useful technologies. Furthermore, a study by Sharma (2019) asserted that voice is a useful biometric because it provides comparable and much higher levels of security. In addition, the study by Zheng and Li (2017) stated that voice could be used to differentiate people because each person's voice has some unique characteristics. Before going any further, it is vital first to understand the essential characteristics of the voice.

In general, any sound produced by humans to communicate meanings, ideas, opinions, etc., is called the voice. In a more specific term, voice is any sound produced by vocal fold vibration, which occurs when air is under pressure from the lungs (Zhaoyan, 2016). Voice is the most natural communication tool used by humans. It conveys the speaker's traits, such as ethnicity, age, gender, and feelings. Lungs, larynx, pharynx, nose, and various parts of the mouth are all involved in producing voice

(Holmes and Holmes, 2002), as shown in Figure 1.2. A voice's features are dependent on its pace or speed, volume, pitch level, and quality, while articulation rate and speech pauses rely on the speaker's speaking style (Sujiya and Chandra, 2017).

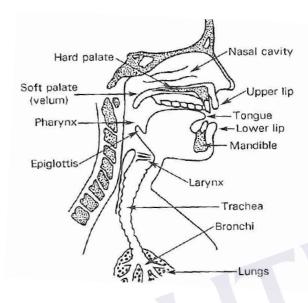


Figure 1.2: Diagrammatic cross-section of a human head showing vocal organs (Holmes and Holmes, 2002)



In speech processing, speaker and speech recognition are the two applications commonly used by researchers to analyse uttered speech (Sharma, 2019). Before delving further into the concept of speaker recognition, it is vital to understand the difference between speaker recognition and speech recognition. Although the terms 'speaker recognition' and 'speech recognition' have often been used interchangeably, they are different. Speech recognition is concerned with the spoken words, while speaker or voice recognition aims to recognise/identify the speaker rather than the words.

Speech recognition is helpful for people with various disabilities, such as those with physical disabilities who find typing the words difficult, painful, or impossible, and those who have difficulties recognising and spelling words, such as people with dyslexia. Since speech recognition deals with converting audio into text, its effectiveness depends heavily on the language and the text corpus (Sharma, 2019).

On the other hand, speaker recognition is to identify the person who is speaking. Speaker recognition scans the features of the speech uttered by an individual, which is distinctive due to their physiology and behavioural patterns. Pitch, speaking style, and accent are some features that contribute to the differences. Speaker recognition technology has been used in various applications, such as biometrics, security, and even human-computer interaction. Table 1.2 summarises the differences between speaker recognition and speech recognition in terms of several features: recognition, purpose, focus, and application.

Features	Speaker Recognition	Speech Recognition				
Recognition	Recognises who is speaking by measuring voice pattern, speaking style, and other verbal traits.	Recognises what is being said and converts them into text.				
Purpose	To identify the speaker.	To identify and digitally record what the speaker is saying.				
Focus	Biometric aspects of the speaker, such as pitch, intensity, etc., to recognise them.	Convert the vocabulary words of what is being said by the speakers into digital texts.				
Application	Voice biometrics.	Speech to text.				

Table 1.2: Speaker recognition vs speech recognition



Malaysia is a multi-racial country consisting of many ethnic groups such as the Malay, Chinese, Indian, and Bumiputera, which can further be classified as Iban, Kadazan, Melanau, Murut, Bidayuh, and Bajau (Nagaraj et al., 2009). Malaysia is also a multilingual society with hundreds of languages that more than a million native speakers speak (Lim, Huspi, and Ibrahim, 2021). The speech sound is concerned with phonetics, whereas phonology involves language functions. Malay is the national language, while English is the second language in Malaysia. The various ethnic groups speak both languages in Malaysia, but they might pronounce the same word slightly differently without affecting the meaning. Accents in a particular language are common in speech, especially when the language is spoken by non-native speakers (Juan, Besacier, and Tan, 2012).

Since Malay and English are the two important languages in Malaysia that began from British colonization, thus the comparison between these two languages is made in terms of vocals and diphthongs, place, and manner of articulations. There are six vocals, 27 consonants, and three diphthongs in the Malay sound system, whereas there are 12 vocals, 24 consonants, and eight diphthongs in the English sound system (Alam, Zilany, and Davies-Venn, 2017). According to Kristin Denham and Anne Lobeck, there are seven important places of articulation in English, i.e., bilabial, labiodental, dental, alveolar, palatal, velar, and glottal. Whereas Malay phonology has labio-velar and no labiodentals and dental sounds (Azmi et al., 2016). As for the manner of articulation, Malay and English phonologies have six manners with voiced and unvoiced pronunciation. In Malay, they are plosive or affricate, fricative, nasal, trill, approximant, and lateral, while in English, they are stop, fricative, affricate, nasal, approximant, and glide.

Humans have long dreamed of creating robots that can socially interact just like

1.2 Research Motivation

humans interact with each other. Applications based on social robots, which are a kind of humanoid robots, have recently emerged as a platform with huge potential in the field of human-robot interaction (HRI). Sophia, Jia Jia, ERICA, Nadine, Pepper, and NICO are some examples of humanoid robots that have been enhanced with humanlike traits to improve the communication between robots and humans. If Nadine, a sitting robot designed as a companion for the elderly or children with special needs (Indramalar, 2016), Pepper is another personal humanoid robot that is used in Japan by pre-school children to help them study English at home and at retail stores to greet customers and provide information about products and services (Tanaka, Isshiki, and Takahashi, 2015). Unfortunately, those mentioned social humanoid robots can only converse in English despite being developed by researchers from China and Japan. Since each language reflects the culture of the particular social group, a humanoid robot must be sensitive to the pitch and intonation of each language for it to interpret correctly and give an appropriate response when communicating with users. ADAM, the Malaysian humanoid robot, currently converses only in English. It would be great if ADAM could interact with Malaysian people in the Malay language. It is the country's national language and a common language spoken by various ethnic groups. The Malay language is also commonly spoken in the region, such as in Indonesia, Singapore, Brunei, and South Thailand.

The pitch period refers to the interval of periodic motion caused by vocal cord vibration when an individual is uttering. Thus, it represents the vocal cords' speed



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APPENDIX E

LIST OF PUBLICATIONS

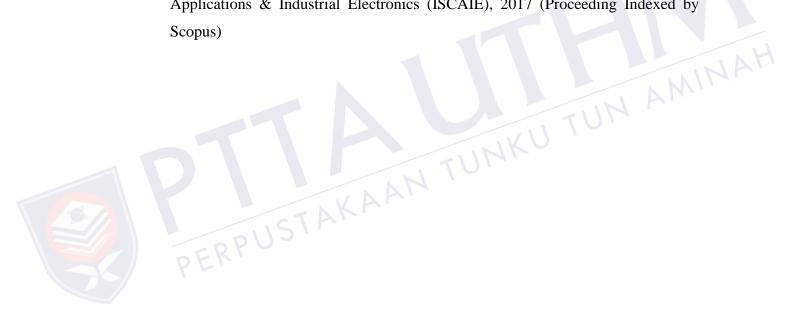
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APPENDIX F

VITA

The author was born on January 12, 1975, in Penang, Malaysia. She went to Sekolah Sultan Mohamad Jiwa, Sungai Petani, Kedah, Malaysia for her secondary school. She pursued her degree at the University Sciences of Malaysia, Penang, and graduated with the degree of Bachelor of Computer Science (Hons) in 1999. Upon graduation, she worked as a lecturer at Institute Teknologi Tun Abdul Razak, Matriculation Centre, Langkawi, Malaysia. She then enrolled at the University Utara Malaysia, Kedah, in 1999, where she was awarded the M.Sc. (Information Technology) in 2001. After that, she taught Computer Programming and other Information Technology courses for the Computer Science Department at Tunku Abdul Rahman College (Penang Branch), Malaysia. After working for four years in Penang, she worked as the Academic Coordinator at Infusion Solution for nearly six years before becoming a permanent academician at Universiti Tun Hussein Onn Malaysia (UTHM) in 2010. In 2015, she was admitted into the PhD program in Computer Engineering, Faculty of Electrical and Electronic Engineering, UTHM. She became a part-time student for four years before converting to full-time status in 2019 after being fully sponsored by UTHM.

