

English Speakers

English Mohd Ali
(1040610467) Investigation of Robust Speech Feature Extraction Techniques for Accents Classification of Malaysian

A thesis submitted in fulfillment of the requirements for the degree of Doctor of Philosophy

> School of Mechatronic Engineering UNIVERSITI MALAYSIA PERLIS

# **UNIVERSITI MALAYSIA PERLIS**

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### THESIS DECLARATION

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#### LIST OF ABBREVIATIONS

AAC Automatic Accent Classification

AE American English

ANN Artificial Neural Network

ANOVA Analysis of Variance

ASR Automatic Speech Recognition

BrE British English
BruE Brunei English

BS Baseline

CALL Computer-assisted Language Learning

CRD Completely Randomized Design

CRs Classification rates

dB Decibels

DOE Design of Experiments

DWT Discrete Wavelet Transform

FF-MLP Feed forward Multilayer Perceptron

FIR Finite Impulse Response
FIS Fuzzy Inference System

Formants Formant frequencies

GMM Gaussian Mixture Model

GSTs Global Statistical Thresholds

HMM Hidden Markov Model

Hz ( Hertz

ICA Independent Component Analysis

IWs Isolated Words

KNN K-nearest Neighbors

LDA Linear Discriminant Analysis

LPC Linear Prediction Coefficients

MalE Malaysian English

MAP Maximum-a-Posteori

MBSE Mel-band Spectral Energy

MFCC Mel-frequency Cepstral Coefficients

MLLR Maximum Likelihood Linear Regression

MMS Maximum-mean subtraction normalization

mse Mean-squared errors

msec milliseconds

MVN Mean and variance normalization

PCA Principal Component Analysis

PD Pronunciation Dictionary

PPRLM Parallel Phone Recognition Language Modeling

PRLM Phone Recognition Language Modeling

QMF Quadrature Mirror Filters

RFE Recursive Feature Elimination

RP Received Pronunciation

SBS Statistical Band Selection

SFFs Spectral Feature Fusions

SgE Singapore English

SNR Signal-to-Noise Ratio

STE Short-time Energy

STFT Short-time Fourier Transform

STs Sentences

SVM Support Vector Machine

SVM-RFE Support Vector Machine-Recursive Feature Elimination

ΓΤS Text-to-Speech

V-UV Voiced-Unvoiced

ZCR Zero-crossing Rate

### LIST OF SYMBOLS

$\alpha$	Learning rate of ANN
	· ·

β Momentum rate of ANN

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## Kajian terhadap Teknik Penyarian Sifat Pertuturan Lasak untuk Pengelasan Loghat Bagi Penutur Bahasa Inggeris Berbangsa Malaysia

#### **ABSTRAK**

Sistem pengecaman pertuturan automatik (ASR) bukanlah suatu topik baru dalam pemprosesan pertuturan dan interaksi manusia-mesin. Ianya telah dikaji lebih daripada lima dekad lepas. Walau bagaimanapun, loghat kekal sebagai cabaran besar berkait rapat dengan kepelbagaian bahasa dalam isu-isu ASR masakini yang menggambarkan perbezaan pertuturan dalam sebutan dan intonasi seseorang yang mempunyai pelbagai perbezaan latar-belakang dari segi sosiolinguistik. Terdapat banyak dan pelbagai literatur yang telah mendedahkan kesan negatif daripada pelbagai loghat sebagai penyebab kemerosotan prestasi ASR. Walaupun loghat Bahasa Inggeris telah menjadi jenis loghat paling banyak dikaji kerana diangkat sebagai bahasa yang paling penting dan berprestij, MalE yang merupakan versi baru didalam 'New Englishes' dikalangan penutur bukan ibunda masih belum diterokai. Dalam produk pasaran ASR pada masa kini, MalE dianggap sebagai versi yang seragam secara konvensional walaupun tanggapan ini dipertikaikan oleh ramai sarjana dan penyelidik yang menganggap MalE sebagai penuturan yang terhasil daripada implikasi setempat kepelbagaian etnik. Kajian persepsi yang lepas telah melaporkan kemungkinan tinggi mengesan identiti etnik daripada penuturan Singapore English (SgE) dan Brunei English (BruE) yang boleh dijadikan perbandingan yang sesuai dengan MalE melalui ujian pendengaran. Pada masa ini, tiada kajian yang telah dilakukan untuk mengenal pasti asal usul etnik dari sampel penuturan MalE menggunakan pelbagai teknik analisis pertuturan dan algoritma pembelajaran mesin untuk pengelasan automatik yang lebih dapat dipercayai, standard dan tepat melalui kaedah eksperimen. Kajian ini merupakan satu cubaan untuk mengisi jurang tersebut dan untuk tujuan ini, satu pangkalan data baru loghat MalE telah dibina. Kajian ini merangsang sebutan jenis IWs dan STs daripada para pelajar universiti (lelaki dan perempuan) yang terdiri daripada tiga etnik utama di negara ini iaitu Melayu, Cina dan India yang mewakili para penutur berpendidikan tinggi menggunakan perkataan yang sensitif kepada loghat, dipilih daripada kajian yang lepas. Reka bentuk sistem yang dicadangkan terdiri daripada pra- pemprosesan, penyarian sifat dan pengelasan. Selain daripada pra-pemprosesan asas, kajian ini mencadangkan integrasi dengan sistem inferens kabur untuk segmentasi asas frem suara kepada bergetar-tidak bergetar (FIS V-UV) turut menyumbang kepada pelaksanaan sistem keseluruhan yang lebih baik berbanding sistem pengelasan loghat (AAC) konvensional. Satu kaedah baru yang dicadangkan yang dinamakan sebagai ambang statistik global (GSTs) untuk membina fungsi keahlian masukan-masukan tenaga pendek masa (STE) dan kadar lintasan sifar (ZCR) dalam segmentasi FIS V-UV telah mengurangkan jumlah frem yang perlu diproses di peringkat penyarian ciri. Keputusan eksperimen menunjukkan keberkesanan AAC-terbantu FIS V-UV yang dicadangkan menggunakan GSTs dengan peningkatan tertinggi dalam kadar ketepatan sebanyak 7.70% dan pengurangan frem sebanyak 24.26% berbanding AAC konvensional. Pada peringkat kedua, ciri-ciri akustik berkait rapat dengan loghat daripada tiga etnik dibangunkan melalui beberapa kaedah analisis bank-penuras, model saluran vokal, analisis hibrid dan analisis paduan. Daripada lapan vektor sifat yang diuji ke atas pangkalan data MalE, perihalan statistik tenaga spektrum jalur-Mel (MBSE), analisis komponen utama-berubah MBSE (disingkatkan sebagai PCA-MBSE), dua teknik hibrid ombak kecil diskret berubah diperolehi pekali ramalan