

FIR FILTER FOR MAKHRAJ RECOGNITION SYSTEM

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TABLE OF CONTENTS

CHAPTER	ELEMENTS	PAGE
	TITLE	
	DECLARATION THESIS STATUS	ii
	DECLARATION BY SUPERVISOR	iii
	DECLARATION	iv
	DEDICATION	v
	ACKNOWLEDGEMENT	vi
	ABSTRACT	vii
	ABSTRAK	viii
	TABLE OF CONTENTS	ix
	LIST OF FIGURES	xi
	LIST OF TABLES	xiii
	LIST OF ABBREVIATIONS	xiv
	LIST OF APPENDICES	xv
CHAPTER 1	INTRODUCTION	
	1.1 Overview	1
	1.2 Objective	2
	1.3 Scope of Project	3
	1.4 Problems Statement	3
CHAPTER 2	LITERATURE REVIEW	
	2.1 Introduction	4
	2.2 Signal Processing	4
	2.3 Speech Pre-processing	5
	2.4 Digital Filtering Fundamentals	17
	2.4.1 Types of filter	18
	2.5 Makhraj Alphabet	22

2.6	Previuos Research	24
2.7	MATLAB	26
CHAPTER 3	METHODOLOGY	
3.1	Introduction	27
3.2	Loading Speech	29
3.3	Pre-processing Process	30
	3.3.1 Normalization	31
	3.3.2 FIR filtering	32
	3.3.3 Windowing	34
	3.3.4 Output Enhancement	35
3.4	Matching Part	36
CHAPTER 4	RESULT AND DISCUSSION	
4.1	Introduction	37
4.2	Loading Speech	38
4.3	Normalization	38
4.4	FIR Filtering	41
4.5	Windowing	44
4.6	Output Enhancement	46
4.7	Matching part	47
CHAPTER 5	CONCLUSION AND RECOMMENDATION	
5.1	Conclusion	49
5.2	Future Recommendation	50
REFERENCES		51
APPENDIX A		

LIST OF FIGURES

FIGURE NO.	TITLE	PAGE
2.1	The principle of the comb filter, (a) Spectrum with Noise and (b) The shape of comb filter	6
2.2	Typical output of the non-overlapping moving frame	10
2.3	Typical output of the non-overlapping moving sample	10
2.4	LMS Algorithm block diagram	11
2.5	Time-domain comparison of input signal with the output of LMS Block (Red is corrected waveform)	14
2.6	Variation of Signal Noise to Ratio (SNR) with increasing noise in the input (LMS) (Red is corrected waveform)	14
2.7	Time domain comparison of input signal and the output of Spectral Subtraction Block. (Red is corrected waveform)	15
2.8	Variation of SNR with increasing noise in the input (Spectral Subtraction)	15
2.9	Bandpass filter specifications	20
2.10	Analog filter and digital filter for frequency response and step response	21
3.1	Flow chart for a speech recognition	28
3.2	MATLAB coding for loading speech	29
3.3	Flow chart for a pre-processing process	30
3.4	MATLAB coding for normalization	31

3.5	Feature selection for real lowpass to bandpass Transformation	32
3.6	MATLAB coding for FIR filtering using discrete sequence data	33
3.7	MATLAB coding for windowing	35
3.8	MATLAB coding for SNR	36
4.1	The reference of speech an alphabet \cup (ba) from human Voice	38
4.2	Example of the speech from human voice Original signal of speech, (a) Alphabet \cup (ba) respondent 1, (b) Alphabet \cup (ba) respondent 2 and (c) Alphabet \cup (ba) respondent 3.	39
4.3	The speech an alphabet \cup (ba) from human voice with noisy signal from air conditioner.	40
4.4	Discrete sequence for lowpass filter	41
4.5	Frequency domain representation of speech with noise	42
4.6	Frequency response for lowpass filter	42
4.7	Frequency for lowpass the alphabet \cup (ba) to remove noise	43
4.8	Bands in frequency domain an alphabet \cup (ba), (a) Low band (b) High band and (c) Band only	45
4.9	Comparison signal between (a) The original of alphabet \cup (ba) with noise and (c) The alphabet \cup (ba) after filtered	46

LIST OF TABLES

TABLE NO.	TITLE	PAGE
1	Letter of Hijaaan or Hijayah	22
2	Signal Noise to Ratio of the output	47

LIST OF ABBREVIATIONS

ADC	-	Analog-to-Digital Converter
ATC	-	Adaptive Transform Coders
CELP	-	Codebook Excited Linear Prediction
DAC	-	Digital-to-Analog Converter
DSP	-	Digital Signal Processing
DFT	-	discrete fourier transform
FIR	-	Finite Impulse Response
GUI	-	Graphical User Interface
IIR	-	Infinite Impulse Response
LMS	-	least mean square
LPC	-	Linear Predictive Coding
MATLAB	-	MATLAB software
MFCC	-	cepstral coefficients and Mel-Frequency Cepstral Coefficients
PCM	-	Pulse Code Modulation
SBC	-	Subband Coders
SNR	-	Signal Noise to Ratio
2-D	-	Two Dimensional

LIST OF APPENDICES

APPENDIX	TITLE	PAGE
A	Source Code	54