

Efficient human voice recognition system



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Declaration

We, HABBA AAABROO SHERDIL, HAFSA RAFIQUE, MUHAMMAD TAJDAR ALAM, HAFIZ MUHAMMAD AUN SABIR, hereby declare that this project report is our own and original work and that it has not been presented to any other University for similar or any other degree award.

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Dedicated to

“Teacher of universe, Holy Prophet Muhammad (P.B.U.H)”

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Table of Contents

Acknowledgement	v
List of Figures:	viii
Abbreviations	ix
Values or Ranges:	ix
Abstract:	x
Chapter 1	1
Introduction	1
Principle of voice recognition:	4
Over View:.....	4
Chapter 2	5
Literature Review.....	5
CHAPTER: 3	11
Problem statement:.....	11
CHAPTER: 4	13
Implementation Model:.....	13
CHAPTER: 5	15
Implementation strategy.....	15
Database.....	16
: Digital processing	16
: Filtering.....	17
: Aligning	19
: Amplitude normalization	19
: Removing unvoiced part:.....	20
: Zero part removal.....	21

: Data Matrix Equalization:.....	22
: Feature Extraction:.....	23
Featuers:	23
: Spectrogram of Voice Signal.....	26
: MFCC (Mel Frequency Cepstral Coefficient):.....	27
: Feature matching.....	29
CHAPTER: 6	31
Applications:	31
CHAPTER: 7	34
Conclusion:	34
CHAPTER: 8	36
Results:.....	36
CHAPTER: 9	38
MATLAB Code	38
References	56

List of Figures:

Figure 1: Verification system.....	3
Figure 2: Identification system	3
Figure 3: Various Technique for Feature Extraction	7
Figure 4: MFCC Block Diagram	8
Figure 5: Comparison b/w Euclidean Distance and DTW.....	9
Figure 6: Block Diagram	14
Figure 7: Butterworth Filter	17
Figure 8: Oder of Filters.....	18
Figure 9 : Filtering	18
Figure 10: Aligning and Amplitude Normalization.....	20
Figure 11: Removing Unvoiced Part or Noise.....	21
Figure 12: Zero part removal	21
Figure 13: After Processing	22
Figure 14: Voice Signal	23
Figure 15: Energy (voiced and unvoiced).....	24
Figure 16: Zero crossing	24
Figure 17: Showing high and low pitch.....	25
Figure 18: Showing high and low pitch.....	25
Figure 19: Showing Spectrogram of voice signal (aaaa eeee).....	26
Figure 20: MFCC.....	27
Figure 21 : Mel Filter Bank	28
Figure 22 : Mapping of two sequences.	29
Figure 23: Two MFCC spectrums are very close to each other.....	30
Figure 24: Results	37

Abbreviations

MFCC.....	Mel frequency cepstral coefficients
DTW	Dynamic Time Warping
VRS	Voice Recognition System
ADC	Analog to Digital Conversion
ASR	Automatic Speech Recognition
DCT	Discrete Cosine Transform
DC.....	Direct Current

Values or Ranges:

Range of human hearing sound frequencies 20 Hz to 20 kHz.

Range of human speech sound frequencies 100 Hz to 7,000 Hz

Threshold value -0.02 TO 0.02 to remove unvoiced part.

Threshold value in DTW is $0 < D < 300$

Abstract:

The usefulness of identification of a person from the uniqueness of his/her voice is increasing with growing importance of automatic information processing. This report reviews the voice characteristics and voice identification techniques which are used in recognizing people by their voices. Voice is a random signal with continuous acoustic variability's hence a proper digital processing algorithm has been selected for voice recognition system to extract required information from the voice sample.

Voice features are the tool on the basis of which voice recognition can be done. In this report, the implementation of VRS (Voice Recognition System) is done using MATLAB. MFCC (Mel frequency cepstral coefficients) and DTW (Dynamic Time Warping) are two algorithms which are adopted for voice feature extraction and matching.

These techniques make it possible to use human voice to recognize the identity of a person and after verification enable the control access to database services, security control for confidential information areas, providing voice recognition ability in robots.

Chapter 1

Introduction

Speaker verification and voice recognition technologies include a reference speaker authentication / identification from the fixed recognition. Speaker recognition applications through the use of this technique using the rapid development of the human voice. As for the area used for authentication, then allow management and voice dialing, voice mail and e-mail, telephone later to get along as claimed service database to obtain the right to land related services, statistical services, and security management of private information, evidence and a lot of programs Additional charges. [8]