

Physics Department Faculty of Science Ain Shams University

Design and Implementation of a Stand-alone Voice Recognition System

A Thesis

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By

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Abstract

For the past several decades, the biometric security systems varied between those based on behavioral and others based on physiological features of people. One of the behavioral biometric security systems is that based on voice features of people and is called speaker recognition system. As a result of the advance of machine learning and computer technology, speaker recognition has rapidly evolved and has become very popular in the recent years. It is being intensively researched and found many applications where it saves lot of troubles which appear as a result of using other biometric security systems, so it is considered a high security system.

Many methods were presented for the aim of designing an automatic speaker recognition system and other methods were presented for the aim of developing reliable speaker recognition systems. In general, any speaker recognition system involves four basic steps: a) data base formation, b) pre-processing, c) features extraction, and d) classification or matching of extracted features.

In this work we have developed a method for speaker recognition using the English Language Speech Database for Speaker Recognition (ELSDSR) database which is compose of audio files for training and others for testing. The developed

method starts by pre-processing the training audio files of the database. Then the Wavelet Packet Transform (WPT) is employed on the pre-processed files for feature extraction purposes. For the excessive number of features provided with the WPT, the energy corresponding to each WPT node is calculated to reduce the dimensionality of the wavelet coefficients by removing redundant features and to form features vectors. The features vectors are sent to the Feed Forward Back-propagation Neural Network (FFBPNN) system.

Experimental results showed the effectiveness of the developed method by using the test audio files. Our results have also showed that the rate of correct recognition of the developed method is about 100% when using the training files and 95.7% when using a one testing file for each speaker from the ELSDSR database. The proposed method showed efficiency better than the well-known Mel Frequency Cepstral Coefficient (MFCC) and the Zak transform.

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List of Abbreviations

ADALINE ADAptive LInear Neuron
ANN Artificial Neural Network
ANNs Artificial Neural Networks
ART Adaptive Resonance Theory

ASRS Automated Speaker Verification System

CWT Continuous Wavelet Transform

Db Daubechies

db20 Daubechies 20-tap

DFT Discrete Fourier Transform

DTFT Discrete Time Fourier Transform **DWT** Discrete Wavelet Transforms

ELSDSR English Language Speech Database for Speaker

Recognition

FFBPNN Feed-Forward Back-Propagation Neural Network

FFT Fast Fourier Transform

IDWT Inverse Discrete Wavelet Transform
IMM Informatics and Mathematical Modeling

LMS Least Mean Square

MADALINE Multiple- ADAptive LInear NEuron
MFCCs Mel Frequency Cepstral Coefficients

MRA Multi-Resolution Analysis

RBF Radial Basis Function

SIS Speaker Identification System

SOM Self-Organization Map

SRS Speaker Recognition System
STFT Short Time Fourier Transform

SVM Support Vector Machines
SVS Speaker Verification System
WPT Wavelet Packet Transform