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Simulation of IN-CAR Voice Filtering Using MATLAB

A Research Submitted In Partial Fulfillment for the Requirement of The Degree of B.Sc. (Honors) In Electronics Engineering

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صدق الله العظيم

To our beloved Mothers

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ABSTRACT

Nowadays, technology advanced rapidly in many approaches such as car environment. This project presents simulation of in-car voice module, which proposed to aim control the car using voice system. This module communicate with driver by human speech. It informs driver about the state of car equipment and recognizes driver voice commands. This feature makes it easy to control a lot of car equipmentby human voice and leads to deal with cars in the same way of dealing with humans in addition, it makes driving easy as talking and use voice conversation with cars.The proposed system canbe used also for speaker verification to protect car against stealing.The driving environment surrounded with a lot of noise that should overcome to get a perfect recognition of voice.This noise takes many representations differ with the situation of the traffic, for all this MATLAB software is used for signal processing using different techniques to enhance the speech signal by designing filters capable for suppressing noise as possible.This research give a background about the digital processing techniques used for voice enhancement such as frequency domain spectral analysis and different filtering approaches to make a decision about which one give the most proper results.

المستخلص

في الوقت الحاضر تتقدم التكنولوجيا بسرعة كبيرة في عديد من المجالات. ولأهمية السيارات في حياتنا اليومية كان لابد لها من تواكب هذا التطور التكنولوجي، ويعرض هذا البحث تصميم لوحدة صوت تستخدم في السيارات. هذا التصميم يسمح للتواصل الصوتي بين السائق والسيارة واعلام السائق عن حالة السيارة والمعدات وستلام الأوامر من السائق هذه الميزة تجعل من السهل السيطرة على كثير من معدات السيارة بواسطة صوت الانسان والتعامل مع السيارات بنفس الطريقة التي يتعامل بها البشر مع بعضهم بالإضافة الى ذلك فانه يجعل القيادة سهلة كسهولة التحدث باستخدام محادثة صوتية مع السيارة وكذلك يمكن استخدام هذا النظام للتحقق من هوية السائق كنوع من حماية السيارة ضد السرقة.

تحاصر بيئة القيادة الكثير من الضوضاء التي يجب التغلب عليها للحصول على تعرف كامل للصوت دون أي تشويش. هذا الضجيج يأخذ مختلف التمثيلات مع اختلاف الوضع المروري او المناطق استخدم برنامجالماتلاب في معالجة الشارات المدخلة باستخدام تقنيات مختلفة لتعزيز إشارة الكلام من خلال تصميم مرشحات قادرة على قمع هذا الضجيج قدر الإمكان.

هذا البحث يعطي خلفية عن تقنيات المعالجة الرقمية المستخدمة لتعزيز الصوت مثل التحليل الطيفي في مجال التردد وطرق ترشيح مختلفة لاتخاذ القرار حول أي منهم يعطي أكثر النتائج المناسبة ولتطبيق هذا المشروع عمليا هناك الكثير من التوصيات او التطويرات المستقبلية للمشروع.

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List of Abbreviations

PCM	Pulse Code Modulation
FIR	Finite Impulse Response
IIR	Infinite Impulse Response
DSP	Digital Signal Processing
LPC	Linear Predictive Coding
HMM	Hidden Marcov Model
ANN	Artificial Neural Network
SNR	Signal-to-Noise Ratio
DFT	Discrete Fourier Transform
FFT	Fast Fourier Transform
PSD	Power Spectral Density
AR	Autoregressive
STFT	Short Time Fourier Transform
GPS	Global Positioning System
INS	Inertial Navigation System

Chapter One

Introduction

1.1 Background

The most popular topics in today's life are the role of technology in facilitating people life and make everything easy to control. It has known that, human contact with each other using voice. Indeed, speech is the most natural way to communicate for them and it would be so intelligent if people could use their voice to communicate with cars [1].

Speech recognition is the process of automatically recognizing a certain word spoken by a particular speaker based on a specific characteristic of voice included in speech waves. The specifications of the voice recognition system that used to generate a speech recognizer machine in car are sampling frequency and number of bits [2].

Analysis voice in frequency domain provides an alternative description of the voice in which the time axis replaced by a frequency axis. The time domain represent the number of samples over time per second, which is equal to (number of samples/sampling frequency **[Fs]**).In the frequency domain, representing sounds in a frequency by amplitude and/or phase diagram in order to determine the amplitude of each point for filtering and power determination purposes[3].

so it shows the number of samples in the signal over the amplitude and the difference between samples equal (1/sampling frequency **[Fs]**) conversion done by using spectral analysis different methods used to convert the signal from time to frequency domain such as, the Fourier transform methods and power spectral estimation methods or the spectrogram representation [19].

Generally, filter is an equipment that used to remove the frequencies in certain parts, to improve the magnitude, phase, or group delay in some other part(s) of the spectrum of a signal and for noise cancelation and phrases detection from a certain statements. In this project different types of filters will be used in order to get high quality and perfect noise cancelation and words detection and extraction such as (Adaptive filter, wiener filter, FIR and IIR filter) [26].

In this thesis, Audio Jet recording program used to record the voice with Pulse Code Modulation (PCM) characteristics, sampling frequency [Fs] equals to 22050KHZ and number of bits [N] is 16bits. Number of bits determines the number of possible digital values that each sample can take it[5]. In addition, different noise environments are simulated using MATLAB in order to represent the real environment of the driver which it contains this different type of noise. Here, Different types of filter are simulated such as the matched filter to detect the presence of a waveform (or pulse) of known structure buried in additive noise. Finally, the proposed system elaborated including a simulation of the designed system using DSP processor.

1.2 Problem Statement

It has known that cars are very important in daily basis. Recently, car companies are looking forward for more friendly, safety and easier car controller device with low cost using voice recognition concepts. This proposed device should more friendly with all people even the handicapped users.

1.3 Proposed Solution

The in car voice control system has been proposed to as controller system that can be found in the cars to give a control commands by using voice only. Where, this device makes the controllable of the car easier and accurate for all can control everything by using only the voice without losing control whileimplement in car voice device with low cost.

1.4 Aim and Objectives

The main aim of this project is to simulate the in car voice recognition control system using different DSP methods.

The objective of this work is to investigate the possibility DSP filtering techniques in speech recognition over a noisy environment, which is decrease the level of the performance of the voice recognizer.

Noise robustness is one of the most problems challenging the voice recognizer.

1.5 Research Methodology

There are two sections in this project: theory part and algorithm simulation part of in car voice system using MATLAB software program. Introduction of different signal processing techniques for this proposed system. In addition, explanation of different methods of noise cancellation. Presenting the time and frequency domain analysis. In this thesis, the MATLAB software used to develop a voice recognition system and plotting all the related graphs.

The main idea of this voice recognition system is a design a device to convert a spoken word in well-known language translated as a command for machines .Where, the input is the voice, the system identifies a spoken word, and the result is the executing of the command directly.

This project presents the analysis of matching process of the given command by extracting a small word of a command sentence and converting it into digitized sound waves. Where, the system made decision for the submitted data after being compared using different filters such as IIR filter, wiener filter. Furthermore, time domain and frequency domain analysis shown by using MATLAB software. Finally, the proposed system elaborated including a simulation for the designed system using MATLAB.

1.6 Thesis Outlines

This thesis contains five chapters divided as follows:

Chapter 2: Presents an overview of the voice specifications parameters behind the voice digitalization.

Chapter 3: Discuss the design of the matched filter. In addition, frequency domain analysis introduced.

Chapter 4: Discuss different DSP filtering techniques for the development of the speech recognition system. These filters are IIR band-pass filter, wiener filter and the system block diagram.

Chapter 5: Discuss the conclusion and the recommendations for the future work related to this project. Finally, all consideration is engaged to the result and performance of the speech recognizer.