SPEECH ENHANCEMENT HYBRID OF WIENER FILTER AND HAAR WAVELET ALGORITHM USING DSP

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LIST OF ABBREVIATIONS

DSP	digital Signal Processing
PSNR	Peak Signal to Noise Ratio
SSA	Signal Subspace Approach
MMSE	Minimum Mean Square Error
MSE	Mean Square Error
MMSESTSA	Minimum Mean Square Error Short Time Spectral Amplitude
FFT	Fast Fourier Transform
SS	Spectral Subtraction
DFT	Discrete Fourier Transform
IDFT	Inverse Discrete Fourier Transform
KLT	Karhunen Loeve Transform
WT	Wavelet Transform
LMS	Least Mean Square
CMS	Cepstral Mean Subtraction
ANC	Adaptive Noise Cancellation
MISO	Multiple Input and Single Output
DOA	Direction of Arrival
LPC	Linear Predictive Coding
LMMSE	Linear Minimum Mean Square Error
STDFT	Short Time Discrete Fourier Transform
SBC	Single Board Computer
FPGA	Field Programmable Gate Array
LE	Logic Element

ALM	Adaptive Logic Module
TSNR	Two Step Noise Reduction
HRNR	Harmonic Regeneration Noise Reduction
DD	Decision Direct
PSD	Power Spectral Density
UWB	Ultra Wide Band
PC	Personal Computer

Peningkatan Ucapan Hibrida Wiener Filer dan Haar Wavelet Algoritma Menggunakan DSP

ABSTRAK

Bentuk komunikasi yang paling biasa, semula jadi dan berkesan antara manusia adalah ucapan. Pada umumnya, semasa komunikasi ucapan banyak bunyi yang tidak diingini telah rosak dan membuat isyarat ucapan lemah. Oleh kerana ini meningkatkan atau menimbulkan isyarat ucapan menjadi sangat penting. Dalam projek ini, hibrida peningkatan pertuturan penapis wiener dan algoritma wavelet haar akan melakukan merendahkan bunyi yang tidak diingini yang bermain dalam isyarat pertuturan. Algoritma akan dilaksanakan pada pemprosesan pemprosesan Isyarat Digital terapung dengan menggunakan bahasa pengaturcaraan C. MATLAB digunakan untuk mendapatkan koefisien proses. Objektif utama projek ini adalah untuk meningkatkan isyarat pertuturan dengan menggunakan hibrid wiener penapisan dan pemampatan dengan haar wavelet. Dalam penapisan penggunaan teknik TSNR dan HRNR mengurangkan bunyi bising dan menghasilkan semula isyarat harmonik dalam ucapan. Kemudian dengan menggunakan algoritma transformasi Haar wavelet untuk memampatkan bunyi. Dengan menggunakan pengukuran PSNR untuk membandingkan hasilnya. Hasilnya dapat membuka ruang untuk kajian masa depan yang baru untuk mencuba menerapkan proses lain seperti pengenalan ucapan dan klasifikasi suara.

Speech Enhancement Hybrid of Wiener Filer and Haar Wavelet Algorithm Using DSP

ABSTRACT

The most common, natural and effective form of communication between human is speech. Generally, during speech communication a lot of unwanted noise were corrupted and make speech signal weak. Due to this enhancing or de-noising to speech signal become very important. In this project the speech enhancement hybrid of wiener filter and haar wavelet algorithm will perform degrading the unwanted noise which play in speech signal. The algorithm will implement on the floating point Digital Signal Processing processor by using c programming language. MATLAB is used to get the coefficients of process. The main objective of this project is to enhance the speech signal by using hybrid of wiener filtering and compressing with haar wavelet. In the filtering the use of TSNR and HRNR techniques reduced the noise and regenerated the harmonic signal in speech. Then by using Haar wavelet transform algorithm to compress the noise. By using PSNR measurement to compare the result. This result can open the doors for new future studies to try for applying other process like speech recognition and sound classification.

CHAPTER 1

INTRODUCTION

1.1 Background

The most fundamental ways to communicate between human is the speech. Nowadays, a wide use of mobile phones and teleconferencing system were very common in our daily life. Moreover, the support of hearing aids and other processing performance such as tele broadcasting. Although, there has high communication system still have some impacts mainly is noise. The purpose of noise detection and reduction becomes popular in speech signal processing research world. With that, reducing unwanted noise can make better in communication in our daily life. To do the process of speech enhancing there are many approaches have been proposed to achieve a good sound enhancement system. There are various ways to perform speech enhancement task such as applying filtering methods, transforms techniques and so on. Filtering methods were applied to reduce the noise from signal and there are many forms of techniques in filtering in this propose by using wiener filtering method. In transform technique, numerous techniques can apply to attain a good performance of speech enhancement process, in this propose will be used with wavelet transform techniques. Fast transform methods and wavelet-based algorithm are some examples of wavelet transform methods. (J K Lee & Chang D Yoo, 2003). There are various ways to calculate the coefficient of the related filter and wavelets in this proposed will be used Matlab and c programming languages. And the very next step is implementing on embedded processors to monitor the result. In the part of embedded processors mostly applied are Field Programming Gate Array (FPGA) based processors, Single Board Computer (SBC) based and Digital Signal Processing (DSP) based processors. In this work will be applied on floating point DSP processor due to the excellent performance in signal processing. The final step is evaluating the outcome result performance and Peak Signal to Noise Ratio measurement will be taken to compare the result outcome performance.

1.2 Problem Statement

Speech enhancement hybrid wiener filter with wavelet is useful for providing noise reduction and other problems which is related to work with sound systems such as mobile phones and telecommunication systems, speech recognition and also for hearing aids. Furthermore, speech enhancing can extend in applying in finding natural resources without harming environment like searching oil and gas, and also useful in detecting body organisms related to medical problems for examples detecting heartbeat, lung working condition and so on.

There are various experiments and works which are related to enhancing speech has done in past. Speech enhancement for noise reduction, detection of heart beat condition and radio broadcasting are some examples of the work done. The main problem is to reduce the noise from the desire speech signal with minimum speech distortion rate when the communication done by the use of electronic devices and mobile phones some kind of distortion also call alteration waveform of the information signal is formed and need to decreased the rate of distortion to obtain the good performance. With the use of Signal Subspace approach, Spectral Restoration with MMSE short-time, and spectral amplitude estimator (MMSESTSA) and Speech-Model-Based some problems like a clear narrowband of noise still remains in spectrum, even the estimation of noise is correct.

Most of the speech processing techniques, speech has to be dealing with unwanted background noise, which is lead to requirement of front-end speech enhancement and the most basically divided in two main categories single-microphone and multi-microphone techniques which are briefly explain in chapter 2. In this proposed will be used single channel with wavelet transform based. A wavelet transform is very useful method for de-nosing signal which can provide a proper model of speech signal for de-noising application in both time and frequency domain. (Hamid Sheikhzadeh, Hamid Reza Abutaleb, 2001)

From above discussion, it can be seen that the speech enhancement has been done in various algorithms. Most of the process, the algorithm used were complicated and sometimes cannot get proper results in data collection and processing. In this proposed method will be hybrid wiener filter with wavelet algorithm is simpler than previous and efficient for speech enhancing. In a way of processing and data collection this proposed method can do successfully.

1.3 Objective of the Research

The main objective of this project are

1. To enhance and reduce the noise from speech signal and implement on floating point DSP processor.
2. To develop the enhancement system by using adaptive wiener filter hybrid with haar wavelet algorithm.
3. To validate the result Peak-Signal-to-Noise-Ratio measurement will be used.

1.4 Research Scope

The experiment is focus to attain the improve speech enhancement and noise reduction from speech. To achieve the relevant coefficients using the Haar wavelet algorithms combine with Adaptive Wiener filter. And calculate the peak signal noise ratio (PSNR) which has needed to use to compare the test result. The aim of using wiener filter is to perform better in noise reduction process and Haar wavelet will applied to de-noise the speech signal to achieve the expected output. After that, the MATLAB programming was used to compile the coefficient which is from the algorithms and filter to produce the output waveform. After the compilation with Matlab the coefficient and variables were set in C programming to implement on floating point digital signal processing (DSP) processor board to test the waveform and signal processing compilation process. Finally, the output result data was compared with the Peak Signal to Noise Ratio (PSNR) measurement to make the conclusion of the performance of the experiment.

1.5 Research Contribution

In this research, to remove the noise from speech which acts as a disturbance in any form of communication and also degrade the quality of signal information, a new approach of wavelet transform method is proposed to de-noise the above condition before the function of extraction based on Discrete Wavelet Transform for noisy condition. The feature extracted after de-noising are found to have lesser effects of additive noise speech signal and to benefit from its localization property in the time and frequency domains. Design and formula derivation of set of wavelet transform and adaptive wiener filter which will use in MATLAB program to attain the proper output coefficient value for implementation in digital signal processing board. To make a final decision for the performance of the test need to compare the result with peak signal to noise ratio measurement to know how the performance of experiment works.

1.6 Thesis Organization

The organization of this thesis includes five chapters as a follow:

In chapter 1, introducing of the approach on Digital Signal Processing and some research background. Motivations for research and problem statements are also defined. A goal, objectives and scopes of research are stated clearly. Finally, research contributions are discussed.

In chapter 2, introducing of the process of how signal enhancement is going through. This chapter introduce a brief related work that were accomplished in this area of signal processing and enhancing. The advantages of using DSP system out of others computer processing platforms.

In chapter 3, describing and implementation signal enhancement methods to achieve the desire results of the project.

In chapter 4, presenting the outcome results, discussion and a detail explanation of investigation process and implementation process.

In chapter 5, making the conclusions that derived from of the entire implementation of the system. One more is the future work also discussed in this chapter.

CHAPTER 2

LITERATURE REVIEW

2.1 Introduction

Nowadays, telecommunication and mobile phones and internet data connection were very popular among human daily life. To have a better communication in sound system, signal processing like sound detection, noise filtering, and speech enhancing process were become essential to perform.

In this chapter, the process and methods which have already done will be discussed. First of all, speech enhancing is called improve the speech quality with the lowest noise ratio. To do the speech enhancing the speech signal is corrupted by noise and then apply the algorithm to reduces the noise and evaluate the performance of the algorithm with subjective test method or objective test methods. In this project will be used objective test methods PSNR and cepstral distance. The different between subjective test and objective test is the objective test is more user friendly than subjective test. There are various forms of noise which are corrupted in speech signal such as white noise, color noise, random noise and so on. To decrease the noise from the speech signal algorithms were used to applied in numerous ways. Among the techniques the speech enhancement by using single channel and multi-channel enhancement methods were the most mainly methods, inside that two main techniques a lot of forms of algorithms and methods were included to apply.

To conclude, the final step is to apply on embedded processor such as field programmable gate arrays (FPGA) processors, digital signal processing (DSP) processors and so on. Choosing processor to perform the task is also important. In this proposed will be used DSP floating point processor TMS32C6713.

2.2 Speech Enhancing

The suppression of addition noise has been an area of concentrating in speech enhancement field for the past decade. The additive noise can easily to concern with convolutive-noise or nonlinear disturbances from the signal processing point of views Moreover, when speech is pausing as a result of the burst nature of speech, it can be able to perceive the noise by itself with large amount of value. Time when speech is nonstationary, and the human ear cannot be able to accept a simple mathematical error criterion, process of speech enhancement becomes a special case for signal estimation. Due to this, subjective measurements of intelligibility and quality are need to perform.

The main objective of the speech enhancement is to degrade the intensity of the noise, while preserving the overall structure such as pitch, format and intelligibility of speech. There are numerous forms of applications such as mobile voice communications, hearing aids and human-machine interfaces - and also verity of methods such as spectral magnitude estimation, wiener filtering and microphone arrays. Focusing on decreasing noise to improve listener more comfort and to upgrade the understandability of the acoustic signal. In this work, using the combination of spectral subtraction and wiener filtering frameworks. In the Wiener filtering process Two-Step Noise Reduction Technique and Harmonic Regeneration Noise Reduction Technique will be used. The most advantage of performing these two techniques can improve noise reduction performance and has the ability to preserve speech onsets and offsets and can remove successfully the unwanted reverberation effects of the typical direction decision approach. (DevyaniS.Kulkarni, RatnadeepR.Deshmukh and PukhrajP.Shrishrimal, 2016) (Yariv Ephraim and David Malah ,1984). The following diagram show the basic step for enhancing speech.

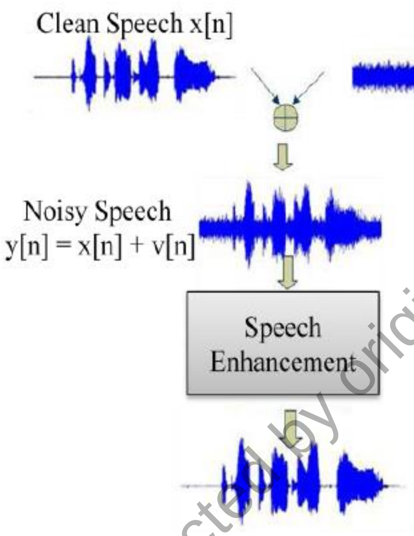

Figure 2.1: Basic steps of speech enhancement system. (Ganga Prasad,2011)

2.3 Type of Noise and Methods for Removing

In general, when capturing, storing, transmitting from one to another, processing and doing conversion signal can suffer noise also called unwanted signal

In our communication system today, noise plays as a main disturbance. There have some noise sources which impact the speech signal. As a result of this, the processes like de-noising, noise-cancellation were playing in main part for signal processing. There are three main categories in noise types and the methods used for removing them are as shown in following Fig. 2.2

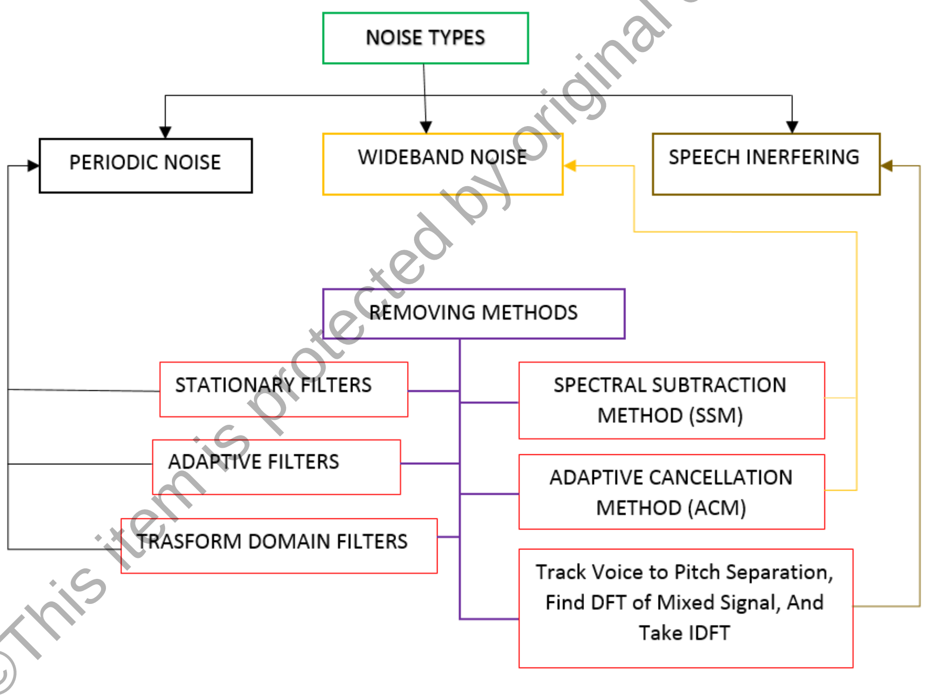

Figure 2.2: Noise Types and Methods to Perform Removing.