

بسم الله الرحمن الرحيم



-Call 4000





شبكة المعلومات الجامعية التوثيق الالكتروني والميكروفيلم





جامعة عين شمس

التوثيق الإلكتروني والميكروفيلم

قسم

نقسم بالله العظيم أن المادة التي تم توثيقها وتسجيلها علي هذه الأقراص المدمجة قد أعدت دون أية تغيرات



يجب أن

تحفظ هذه الأقراص المدمجة يعبدا عن الغبار













بالرسالة صفحات لم ترد بالأصل





AIN SHAMS UNIVERSITY FACULTY OF ENGINEERING

Electronics and Communications Department

Adaptive Filter Design and Implementation For Acoustic Noise Cancellation

A Thesis submitted in partial fulfillment of the requirements of a Master of Science degree in Electrical Engineering Electronics Engineering and Electrical Communications Department

Mohamed Salah Mahmoud Hassan

by

Bachelor of Science degree in Electrical Engineering
Electronics Engineering and Electrical Communications Department
Faculty of Engineering, Ain Shams University, 2011

Supervised By

Prof. Dr. Mohamed Dessouky
Assoc. Prof. Bassant Abdelhamid

Cairo, 2020

Faculty of Engineering Ain Shams University Electronics and Communication Engineering Department

Thesis title: "Adaptive Filter Design and Implementation For Acoustic Noise Cancellation"
Submitted by: Mohamed Salah Mahmoud Hassan
Degree: Master of Science in Electrical Engineering

Examiners' Committee:

Arab Academy for Science, Technology & Maritime Transport Faculty of Engineering, Prof. Dr. Mohamed Wathek El-Kharashi Ain Shams University, Faculty of Engineering, Computer & Systems Dept. Prof. Dr. Mohamed Amin Dessouky Ain Shams University, Faculty of Engineering, Electronics & Communications Dept.	Prof. Dr. Khaled Ali				
Prof. Dr. Mohamed Wathek El-Kharashi Ain Shams University, Faculty of Engineering, Computer & Systems Dept. Prof. Dr. Mohamed Amin Dessouky Ain Shams University, Faculty of Engineering,	Arab Academy for Science, Technology &				
Prof. Dr. Mohamed Wathek El-Kharashi Ain Shams University, Faculty of Engineering, Computer & Systems Dept. Prof. Dr. Mohamed Amin Dessouky Ain Shams University, Faculty of Engineering,	Maritime Transport				
Ain Shams University, Faculty of Engineering, Computer & Systems Dept. Prof. Dr. Mohamed Amin Dessouky Ain Shams University, Faculty of Engineering,	Faculty of Engineering,				
Ain Shams University, Faculty of Engineering, Computer & Systems Dept. Prof. Dr. Mohamed Amin Dessouky Ain Shams University, Faculty of Engineering,					
Faculty of Engineering, Computer & Systems Dept. Prof. Dr. Mohamed Amin Dessouky Ain Shams University, Faculty of Engineering,	Prof. Dr. Mohamed Wathek El-Kharashi				
Dept. Prof. Dr. Mohamed Amin Dessouky Ain Shams University, Faculty of Engineering,	Ain Shams University,				
Prof. Dr. Mohamed Amin Dessouky Ain Shams University, Faculty of Engineering,	Faculty of Engineering, Computer & Systems				
Ain Shams University, Faculty of Engineering,	Dept.				
Ain Shams University, Faculty of Engineering,					
Faculty of Engineering,	Prof. Dr. Mohamed Amin Dessouky				
V	Ain Shams University,				
Electronics & Communications Dept.	Faculty of Engineering,				
	Electronics & Communications Dept.				

Statement

This thesis is submitted as a partial fulfillment of Master of Science in Electrical Engineering, Faculty of Engineering, Ain shams University. The author carried out the work included in this thesis, and no part of it has been submitted for a degree or a qualification at any other scientific entity.

Mohamed	Salah	Mahmoud	d Hassan
			Signature
		Dat	e: / /2020

Researcher Data

Name: Mohamed Salah Mahmoud Hassan

Date of Birth: 22/08/1989

Place of Birth: Dakahlia, Egypt

Last academic degree: Bachelor of Engineering

Field of specialization: Electrical Engineering

University issued the degree: Ain Shams University

Date of issued degree: 2011

Current job: Senior Digital Compression Engineer at the

Egyptian Satellite Company

Abstract

To minimize the noise level in speech signals, many adaptive filters, such as Least Mean Square (LMS) and Normalized Least Mean Square (NLMS), are utilized to reach the steady state. The filter weights are adapted based on specific functions to enhance the Signal to Noise Ratio (SNR) of the system output with fast convergence speed.

This thesis proposes a new LMS-based variable step size algorithm to reduce the noise level in the corrupted speech. The new algorithm is called Regularized Square Root Absolute Error LMS (R-SRAE-LMS). R-SRAE-LMS switches between two different variable step size algorithms. The first algorithm is the approximated solution of the Regularized NLMS (RNLMS) which has the ability to reach the steady state very fast. The second algorithm is the Square Root LMS (SRLMS) which is able to achieve high output SNR value by eliminating the residual noise samples.

This new algorithm is compared with other variable step size algorithms in speech enhancement. The results show that the new proposed algorithm improves the SNR at the filter output. It has fixed SNR improvement over a wide range of input SNR. Over this range, the proposed algorithm has more stable performance measures and achieves the highest convergence speed and the lowest steady state error. It is also able to get very small misalignment values between the filter weights and the targeting channel.

Moreover, this thesis proposes a detailed design and Field Programmable Gate Array (FPGA) implementation of the proposed adaptive algorithm. The design of the adaptive filter is divided into a forward path and two feedback paths. The device utilization, operating frequency and the power consumption after a complete implementation process are also presented. It shows remarkable results compared to other variable step size based adaptive filter designs.

Keywords: Adaptive Filter Design, NLMS, Variable Step Size, Acoustic Noise Cancellation, FPGA Implementation, Interference Cancellation.