

**A NOVAL APPROACH  
TO  
ADAPTIVE NOISE CANCELLATION  
FOR  
SPEECH SIGNAL  
USING  
WAVELET BASED GRAZING  
ESTIMATION OF SIGNAL METHOD**

**Dr. S. Manikandan**

*Professor*

Department of Electrical & Electronics Engineering

**Balaji Institute of Technology & Science**

**Warangal, Telegana, INDIA.**

# **A NOVAL APPROACH TO ADAPTIVE NOISE CANCELLATION FOR SPEECH SIGNAL USING WAVELET BASED GRAZING ESTIMATION OF SIGNAL METHOD**

Copyright © : Dr. S. Manikandan  
Publishing Rights © : VSRD Academic Publishing  
*A Division of Visual Soft India Pvt. Ltd.*

**ISBN-13: 978-93-86258-57-1**  
**FIRST EDITION, JUNE 2017, INDIA**

*Typeset, Printed & Published by:*  
**VSRD Academic Publishing**  
*(A Division of Visual Soft India Pvt. Ltd.)*

**Disclaimer:** The author(s) are solely responsible for the contents of the papers compiled in this book. The publishers or its staff do not take any responsibility for the same in any manner. Errors, if any, are purely unintentional and readers are requested to communicate such errors to the Editors or Publishers to avoid discrepancies in future.

All rights reserved. No part of this publication may be reproduced, stored in a retrieval system or transmitted, in any form or by any means, electronic, mechanical, photocopying, recording or otherwise, without the prior permission of the Publishers & Author.

*Printed & Bound in India*

**VSRD ACADEMIC PUBLISHING**  
*A Division of Visual Soft India Pvt. Ltd.*

## **REGISTERED OFFICE**

154, Tezabmill Campus, Anwarganj, KANPUR–208003 (UP) (IN)  
Mb: 99561 27040, Web: [www.vsrdpublishing.com](http://www.vsrdpublishing.com), Email: [vsrdpublishing@gmail.com](mailto:vsrdpublishing@gmail.com)

## **MARKETING OFFICE (NORTH INDIA)**

Basement-2, Villa-10, Block-V, Charmwood Village, FARIDABAD–121009 (HY)(IN)  
Mb: 98999 36803, Web: [www.vsrdpublishing.com](http://www.vsrdpublishing.com), Email: [vsrdpublishing@gmail.com](mailto:vsrdpublishing@gmail.com)

## **MARKETING OFFICE (SOUTH INDIA)**

340, FF, Adarsh Nagar, Oshiwara, Andheri(W), MUMBAI–400053 (MH)(IN)  
Mb: 99561 27040, Web: [www.vsrdpublishing.com](http://www.vsrdpublishing.com), Email: [vsrdpublishing@gmail.com](mailto:vsrdpublishing@gmail.com)

## P R E F A C E

This Text book is very helpful for Electrical, Mechanical, Electronics Engineers to develop Noise control applications in aircraft, Machines and headphone devices. In this book is introduced a new concept of Noise reduction application with simple graphical methods by using wavelet based adaptive algorithms. It reduced the complexity of algorithms writing and implementation problems. This is very helpful to write thesis and implementation of Real Time applications in all the scholars. It is mainly used aircraft applications during very high noise problems. This Research content having for Literature Survey, Neural Network Concepts, DSP Processor, Matlab 7.0 Implementation is used for designing of ANC. It is easily understood the Research concepts from low level to High level Scholars.

 *Dr. S. Manikandan*



# ACKNOWLEDGEMENT

It gives me a great pleasure to complete and submit this thesis entitled “**A Novel Approach To Adaptive Noise Cancellation For Speech Signal Using Wavelet Based Grazing Estimation Of Signal Method**””, within a stipulated time. It has been my earnest effort to contribute for the future generations in the educational and research pursuits in the field of Electronics and telecommunication and hope that this work may open door for years to come.

I am immensely grateful to the Chancellor **Mr. Vinod Tibrewala**, JJT University, Rajasthan, India for giving me the opportunity to undergo this highly specialized and unique course and also express my thanks to vice chancellor **Prof. D. D. Agarwal**, JJT University, Rajasthan, India for his invaluable support to complete this research work.

To complete this small piece of work, I have many helps, suggestions and guidance from various corners of the country. I fervently express my gratitude to my supervisor, **Dr. N. Rajasekar**, Professor, School of Electrical Engineering, VIT University, Vellore, under whose precious guidance, support and encouragement this work is being completed smoothly. He helped me in various ways from the very beginning and always ready to guide me at no cost. I am indebted to thank him for his invaluable support.

I cannot find words, enough to thank **Mr. P. Gopinath**, Personal secretary to the chancellor JJT University, Rajasthan, who has helped me at all stages of my work

by providing me with timely criticism and recommendations, especially during the last few weeks in their busy schedule of works.

I would like to thank **Administrative and Technical Staff Members** of the JJT University, Rajasthan, who have been kind enough to advise and help in their respective roles.

I am indebted to record my sense of appreciation to **Mr. J. Ilango**, Chairperson, King College of Technology, Namakkal, Tamilnadu, who provided me all possible helps and supports throughout my research work.

Last, but not least, I would like to dedicate this thesis to my family, my wife **Dr. S. Subathradevi** and my childrens **M. Arun karthick, M. Shivani**, for their love, patience, and understanding they allowed me to spend most of the time on this thesis.

I offered my thanks to the Almighty God for giving me good health and other blessings to carried out this work. He is the supreme guide to my life and the glory for the glory of the Lord forever and ever.

*Dr. S. Manikandan*

# **LIST OF ABBREVIATIONS**

<b>ABBREVIATION</b>	<b>FULL FORM OF ABBREVIATION</b>
DSP	Digital Signal Processors
ANC	Active Noise Cancellation
FXLMS	Filtered-X Least Mean Square
IIR	Infinite Impulse Response
FIR	Finite Impulse Response
RLS	Recursive Least Squares
FFT	Fast Fourier Transform
LMS	Least Mean Squares
ANN	Artificial Neural Network
HVAC	Heating, Ventilation and Air Conditioning
EVM	Evaluation Module
SNR	Signal to Noise Ratio
PSNR	Peak Signal to Noise Ratio

## **ABSTRACT**

This thesis introduces the reducing the content of noise present in the received Speech signals for wireless communication medium by using Wavelet based Grazing Estimation of Signal (WGES) Method. The received signal is corrupted due to mixing of white Gaussian noise. This proposed method is designed based on the superposition principle with eight possible cases. By conducting multiple possible cases of signal movement the noise signal is moved to opposite direction of original signal. This output is cascaded with wavelet transforms techniques with compare the available control algorithms output error signals. Compared to other available control algorithms the proposed method is Simple to implement, yields good performance and converges quickly. This proposed technique is implemented using Matlab software and DSP processor .This computer output simulation results confirm the effectiveness of our proposed algorithm.



# CONTENTS

<b>CHAPTER 1</b>	
<b>INTRODUCTION.....</b>	<b>1</b>
<b>CHAPTER 2</b>	
<b>LITERATURE SURVEY FOR ACTIVE NOISE CONTROL SYSTEMS.....</b>	<b>5</b>
<b>2.1 INTRODUCTION.....</b>	<b>7</b>
2.1.1 CURRENT APPLICATIONS.....	8
2.1.2 PERFORMANCE EVALUATION & PRACTICAL CONSIDERATIONS.....	8
2.1.3 ANC SYSTEM PROPERTIES .....	9
<b>2.2 BROAD – BAND FEED FORWARD ANC .....</b>	<b>9</b>
<b>2.3 FILTERED – XLMS ALGORITHM .....</b>	<b>12</b>
2.3.1 DERIVATION OF THE FXLMS ALGORITHM .....	12
2.3.2 ANALYSIS OF FXLMS ALGORITHM .....	14
2.3.3 LEAKY FXLMS ALGORITHM .....	15
<b>2.4 NARROW – BAND FEED FORWARD ANC.....</b>	<b>17</b>
2.4.1 WAVEFORM SYNTHESIS METHOD.....	17
2.4.2 ADAPTIVE NOTCH FILTER .....	18
<b>2.5 SINGLE CHANNEL FEEDBACK ANC SYSTEM .....</b>	<b>21</b>
<b>2.6 MULTIPLE CHANNEL ANC .....</b>	<b>24</b>
<b>2.7 NLINE SECONDARY – PATH MODELING .....</b>	<b>25</b>
<b>2.8 CONCLUSION.....</b>	<b>26</b>
<b>CHAPTER 3</b>	
<b>PROPOSED METHOD TO ACTIVE NOISE FEED FORWARD CONTROL SYSTEMS USING DELTA RULE ALGORITHM.....</b>	<b>27</b>
<b>3.1 DESIGN OF ANC SYSTEM USING LMS ALGORITHM .....</b>	<b>29</b>
<b>3.2 DESIGN OF ANC SYSTEM USING RLS ALGORITHM.....</b>	<b>33</b>
<b>3.3 DESIGN OF ANC USING ARTIFICIAL NEURAL NETWORKS .....</b>	<b>34</b>
3.3.1 DELTA RULE ALGORITHM .....	35
<b>3.4 SIMULATION AND RESULTS .....</b>	<b>36</b>

3.5	CONCLUSION.....	38
-----	-----------------	----

## **CHAPTER 4**

	<b>PROPOSED METHOD TO ACTIVE NOISE CONTROL SYSTEM FOR REAL-TIME NOISE REDUCTION USING THE TMS320C5416 PROCESSOR.....</b>	<b>39</b>
--	--	-----------

4.1	MATHEMATICAL MODELING OF ADAPTIVE NOISE FILTER .....	42
4.2	IMPLEMENTATION OF ANC USING TMS320C5416 PROCESSOR.....	44
4.3	REAL TIME IMPLEMENTATION OF ANC USING LMS ALGORITHM IN MATLAB .....	45
4.4	CONCLUSION.....	48

## **CHAPTER 5**

	<b>BY USING TMS 320C5402 DSP PROCESSOR CREATE THE DESIGN OF ACTIVE NOISE CANCELLATION FOR SPEECH SIGNAL .....</b>	<b>49</b>
--	---	-----------

5.1	DESIGN OF FEEDBACK ACTIVE NOISE CONTROL .....	52
5.2	DESIGN OF PRACTICAL SETUP .....	53
5.3	SETUP FOR SPEAKER MICROPHONE .....	54
5.4	DESIGN OF LEAST SQUARE ALGORITHM .....	54
5.5	DUCT SYSTEM SECONDARY PATH MODELING .....	55
5.6	ANALYSIS OF ADAPTIVE NOISE CANCELLATION .....	58
5.7	CANCELLATION FOR RESULTS AND ANALYSIS.....	59
5.8	CONCLUSION.....	60

## **CHAPTER 6**

	<b>PROPOSED METHOD TO ADAPTIVE NOISE CANCELLATION FOR SPEECH SIGNALS USING WAVELET BASED GES METHOD.....</b>	<b>61</b>
--	--	-----------

6.1	PROBLEM STATEMENT.....	64
6.2	MATHEMATICAL MODELING OF GRAZING ESTIMATION OF SIGNAL METHOD.....	64
6.3	PROPOSED METHOD FOR POSSIBLE CASES.....	66

<b>6.4</b>	<b>PROPOSED GES ALGORITHM .....</b>	<b>73</b>
6.4.1	EMBEDDING GES WITH WAVELET TRANSFORM TECHNIQUE [WGES] .....	75
<b>6.5</b>	<b>SIMULATION RESULTS AND DISCUSSION .....</b>	<b>75</b>
<b>6.6</b>	<b>CONCLUSION .....</b>	<b>82</b>
 <b>CHAPTER 7</b>		
	<b>CONCLUSION AND FUTURE WORK .....</b>	<b>83</b>
<b>7.1</b>	<b>CONCLUSION .....</b>	<b>85</b>
<b>7.2</b>	<b>FUTURE WORK .....</b>	<b>86</b>
 <b>CHAPTER 8</b>		
	<b>REFERENCES AND PAPER PUBLICATIONS .....</b>	<b>87</b>
<b>8.1</b>	<b>REFERENCES .....</b>	<b>89</b>
<b>8.2</b>	<b>PAPER PUBLICATIONS .....</b>	<b>98</b>
8.2.1	PAPERS PUBLISHED IN IEEE XPLORE .....	98
8.2.2	PAPER PUBLICATIONS IN INTERNATIONAL JOURNALS.....	98
 <b>CHAPTER 9</b>		
	<b>APPENDIX.....</b>	<b>101</b>
<b>9.1</b>	<b>"C" CODE FOR LMS ALGORITHM .....</b>	<b>103</b>
<b>9.2</b>	<b>THE ASM CODE GENERATED FOR TMS320C5402 KIT .....</b>	<b>104</b>

# LIST OF FIGURES

Fig. No.	TITLE	Page No.
2.1	Signal – channel broad band feed forward ANC system in a duct	9
2.2	System identification of Active Noise Control System	10
2.3	Simplified block diagram of Active Noise Control system	11
2.4	Block Diagram of ANC system using the FLXMS algorithm	13
2.5	Equivalent diagram if Fig.4 for slow adaptation and $\hat{S}(z) = S(z)$	14
2.6	Block diagram of ANC system with feedback	16
2.7	ANC with acoustic feedback neutralization	16
2.8	Equivalent diagram of waveform synthesis method using impulse train input and neglecting secondary path effects	17
2.9	Signal – frequency adaptive notch filter	19
2.10	Single-frequency ANC system using the FXLMS algorithm	20
2.11	Block diagram of signal frequency active noise equalizer	21
2.12	Block diagram of basic active noise Control system	21

<b>Fig. No.</b>	<b>TITLE</b>	<b>Page No.</b>
2.13	Wideband adaptive feedback ANC system using the FXLMS algorithm	22
2.14	Block diagram of adaptive predictor	23
2.15	Hybrid ANC system with combination of feedback ANC and feed forward ANC	23
2.16	Hybrid ANC system using the FIR feed forward ANC with the FXLMS algorithm	23
2.17	Structure of a multiple – channel acoustic ANC system with J reference inputs, K secondary sources, and M error sensors	24
2.18	Block diagram of an adaptive multiple channel feed forward ANC system with feedback paths	25
2.19	Block diagram for Real-Time secondary-path modeling technique	26
3.1	Block diagram of the ANC control system using LMS algorithm	30
3.2	Input Gaussian Noise	36
3.3	Residual noise of LMS Algorithm	37
3.4	Residual noise of RLS Algorithm	37
3.5	Residual noise of Delta Rule Algorithm	37
4.1	Simple Active Noise detection model	42
4.2	Critical Issues in the design of an Active Noise Filter	43

<b>Fig. No.</b>	<b>TITLE</b>	<b>Page No.</b>
4.3	Simulation result for Error signal	46
4.4	Simulation result for Adaptive Filter using a delay time of two counts	46
4.5	Noise signal in time domain at an error microphone	47
4.6	Noise signal in noise power spectrum at an error microphone	47
5.1	Block diagram of Adaptive Filter System Using FXLMS Algorithm	53
5.2	Block Diagram in Duct Systems	54
5.3	Block diagram of Adaptive Filter Using Least Mean Square	55
5.4	Block diagram of System Identification Using Offline	56
5.5	Using step sizes $d$ & $\mu$ values	57
5.6	Results in Cancellation result on pure tone noise at 130 Hz	58
5.7	Results in Cancellation result on pure tone noise at 110+120+130 Hz	59
5.8	Results in Cancellation result on Pure Tone Noise at 100+110+120+130 Hz	60
6.1	Depiction of Estimated Signal	72
6.2	Block diagram of Grazing Estimation Method	74

<b>Fig. No.</b>	<b>TITLE</b>	<b>Page No.</b>
6.3	Block diagram of Wavelet Denoising	75
6.4	System performance in time domain	76
6.5	System performance in Frequency Domain	77
6.6	Represents the gain in the PSNR	77
6.7	Analysis of proposed method using Ding sound	81
6.8	Analysis of proposed method using S10mwb Speech signal	81

## LIST OF TABLES

<b>Table No.</b>	<b>TITLE</b>	<b>Page No.</b>
4.1	Total noise reduction in the Error Microphone	48
6.1	Comparisons of various methods for s1omwb speech signal	79
6.2	Comparisons of various methods for ding sound signal	79
6.3	Comparisons of various methods for ding sound signal.	80