Improvement of Sound Quality in Vehicular Hands-Free Communication Systems

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ABSTRACT

For a vehicular hands-free communication system, the sound quality of communication is usually degraded by background noise such as engine noise and aerodynamic noise which are known to be detrimental to system performance. In this report, a novel adaptive filtering algorithm and an integrated system for acoustic echo and noise cancellation are presented. The proposed system includes adaptive noise cancellation, line enhancer, and echo cancellation which are based on a novel variable step-size affine-projection algorithm (VSS APA). The proposed VSS APA filtering algorithm is a combination of a variable step-size least-mean-square (VSS LMS) and an affine-projection algorithm (APA). The input signal in the APA is structured as a matrix, which is more complete than the vector structure in other algorithms. As a result, the estimation of the weight vector has orthogonality, thus allowing quick and accurate error corrections. However, the step-size of the APA is invariable. The choice of the step-size affects the overall performance of the algorithm. The variable step-size of the proposed VSS APA can adjust itself automatically to reach an optimum step-size according to the operations of the system at different times. To understand and verify the effectiveness of the proposed system, a performance evaluation and comparison were conducted to compare the proposed algorithm and various traditional adaptive filtering algorithms in this application. The results demonstrated that the VSS APA has an effective performance and convergence in sound quality improvement of hans-free communication systems.

Keywords : Echo cancellation, Adaptive filter; Hands-free communication system, Digital signal processor, VSS APA

Table of Contents

COVER AUTHORIZATION LETTERS.		iii ABSTRACT
(CHINESE)	V ABSTRACT (ENGLISH)	vi
ACKNOWLEDGMENT	viii TAE	BLE OF
CONTENTS	vii LIST OF FIGURES	ix LIST
OF TABLES	xiv ABBREVIATIO	NS & SYMBOLS
xv CHAPTER 1 INTROD	UCTION 1.1 Introduction of this Study	
Literature Review		of this Thesis
5 CHAPTER 2 ADAPTIVE F	ILTERING 2.1 Adaptive Filter	9
2.2 Adaptive Echo Cancellation	11 2.3 Adaptive No	bise Cancellation
12 2.4 Adaptive Line Enhance	ment	14 2.5 Combined Structure of Vehicular
Hands-free Communication System15 CH	APTER 3 ADAPTIVE ALGORITHMS A	ND RESEARCH METHODS 3.1 LMS
Algorithm	17 3.2 NLMS Algorithm	٦
21 3.3 RLS Algorithm		.23 3.4 VSS LMS Algorithm
25 3.5 VS	S APA	27 CHAPTER 4
SIMULATION RESULTS AND DISCUS	SION 4.1 Test of Convergence Speed	
Results of Adaptive Echo Cancellation		otive Noise Cancellation
43 4.4 Results of Adaptive Line Enh	nancement50 C	HAPTER 5 CONCLUSIONS
	63 REFERENCES	65

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