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# **Real-Time Adaptive Noise Cancellation for Automatic Speech Recognition in a Car Environment**

Ziming Qi

2008

# **Real-Time Adaptive Noise Cancellation for Automatic Speech Recognition in a Car Environment**

A thesis presented in partial fulfillment of the requirements for the degree of

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Ziming Qi

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# Table of Contents

Table of Contents.....	I
Table of figures.....	IV
List of Tables.....	VIII
Abstract.....	IX
List of Abbreviations and Acronyms.....	X
Nomenclature.....	XI
Acknowledgements.....	XIII
Declaration.....	XIV
1 INTRODUCTION.....	1
1.1 RESEARCH OBJECTIVE.....	1
1.2 SPEECH ENHANCEMENT APPROACH.....	3
1.2.1 VOICE ACTIVITY DETECTION APPROACH.....	3
1.2.2 ADAPTIVE NOISE CANCELLATION APPROACH IN THIS THESIS.....	4
1.2.3 ADAPTIVE WIENER FILTER APPROACH.....	4
1.3 CONTRIBUTIONS TO KNOWLEDGE.....	5
1.4 PERFORMANCE WITH FAVORABLE EFFECTS.....	5
1.5 STRUCTURE OF THIS THESIS.....	6
2 LITERATURE REVIEW.....	7
2.1 INTRODUCTION.....	7
2.2 ACOUSTIC BEAMFORMING.....	8
2.2.1 OVERVIEW.....	8
2.2.2 CONVENTIONAL “DELAY AND SUM” ACOUSTIC BEAMFORMER.....	9
2.2.3 FAR-FIELD AND NEAR-FIELD ACOUSTIC WAVEFRONTS.....	11
2.2.4 ADAPTIVE ACOUSTIC BEAMFORMING.....	12
2.2.5 ADAPTIVE ALGORITHM FOR BEAMFORMING.....	15
2.2.5.1 Recursive Least Square algorithm.....	15
2.2.5.2 Least Mean Square algorithm.....	16
2.2.5.3 Normalized least mean square algorithm.....	17
2.2.5.4 Normalized Least Mean Forth algorithm.....	19
2.2.6 ROBUST ACOUSTIC ADAPTIVE BEAMFORMING.....	22
2.3 VOICE ACTIVITY DETECTION.....	23
2.3.1 TIME DELAY ESTIMATION.....	23
2.3.2 MAGNITUDE SQUARED COHERENCE.....	24
2.4 COCKTAIL PARTY EFFECT AND SOLUTION.....	26
2.4.1 COCKTAIL PARTY EFFECT.....	26
2.4.2 ADAPTIVE DIGITAL FILTER.....	27
2.4.2.1 Finite impulse response filter.....	27
2.4.2.2 Infinite impulse response filter.....	28
2.4.3 WIENER FILTER.....	30
2.5 SPEECH ENHANCEMENT IN CAR NOISE ENVIRONMENTS.....	32
2.5.1 OVERVIEW.....	32
2.5.2 VOICE ACTIVITY DETECTION IN A CAR.....	34
2.5.2.1 Beamforming applications in automotives.....	34

2.5.2.2	Detect a geometrical zone with three-microphone beamforming.....	34
2.5.2.3	Speech enhancement for automotive speech recognition in a car .....	36
2.5.3	SPEECH ENHANCEMENT FOR VOICE COMMUNICATION IN CAR .....	39
2.5.3.1	An Adaptive filter in a car.....	39
2.5.3.2	An Wiener filter in a car .....	45
2.6	SUMMARY .....	47
3	PROBLEM DEFINITION AND RESEARCH ENVIRONMENTAL SET-UP .....	49
3.1	PROBLEMS OF SPEECH ENHANCEMENT IN A CAR.....	49
3.2	APPROACHES ON SPEECH ENHANCEMENT IN A CAR .....	49
3.2.1	THREE-MICROPHONE VAD SWITCH AND NLMS FILTER .....	50
3.2.2	WIENER FILTER IN 3 MICROPHONE ARRAY .....	53
3.3	OVERVIEW OF SYSTEM BUILD-UP .....	53
3.4	RESEARCH ENVIRONMENTAL SET-UP.....	54
3.4.1	THREE-MICROPHONE DATA ACQUISITION IN CAR .....	54
3.4.2	SIGNAL CONDITIONING IN A CAR.....	55
3.4.3	DIGITAL SIGNAL PROCESSING HARDWARE AND SOFTWARE .....	58
3.5	SUMMARY .....	58
4	REAL-TIME ADAPTIVE NOISE CANCELLATION IN A CAR.....	59
4.1	THREE-MICROPHONE BEAMFORMER IN A CAR .....	59
4.1.1	INTRODUCTION .....	59
4.1.2	THREE MICROPHONE BEAMFORMING VOICE ACTIVITY DETECTION WITH ADAPTIVE NOISE CANCELLATION .....	59
4.1.3	THREE-MICROPHONE ADAPTIVE NOISE CANCELLATION .....	60
4.1.4	A THREE-MICROPHONE VAD .....	61
4.1.5	SUMMARY AND DISCUSSION.....	66
4.2	ADAPTIVE WIENER FILTER IN A CAR .....	67
4.2.1	INTRODUCTION .....	67
4.2.2	ADAPTIVE WIENER FILTER.....	67
4.2.3	MATRIX INVERSION METHOD .....	73
4.2.4	AUTOMATIC SPEECH RECOGNITION .....	74
4.2.5	SUMMARY AND DISCUSSION.....	76
5	EXPERIMENTS .....	77
5.1	OVERVIEW .....	77
5.2	EXPERIMENTS – THREE MICROPHONE BEAMFORMING IN A CAR .....	79
5.2.1	THREE-MICROPHONE VAD IN A CAR.....	79
5.2.2	NLMS ADAPTIVE NOISE CANCELLATION IN 3-MICROPHONE BEAMFORMING IN A CAR ...	83
5.2.2.1	The 3-microphone noise canceller in a 2 speech environment .....	87
5.2.2.2	Definition of Noise canceller valid zone.....	89
5.2.2.3	A single noise environment: Driver’s voice and a stationary white noise.....	91
5.2.2.4	A single noise environment: Driver’s voice and a second speech.....	93
5.2.3	AN ASR WITH 3-MICROPHONE VAD AND NLMS ANC .....	95
5.2.4	COMPARISON OF NLMF AND NLMS ANC.....	96

5.3	EXPERIMENTS – ADAPTIVE WIENER FILTER IN A CAR .....	100
5.3.1	CASE STUDY ONE: COMPARISON OF ADAPTIVE WIENER FILTER UPDATE METHODS .....	101
5.3.2	CASE STUDY TWO: ANALYSIS OF AVERAGE POWER IN THE SPECTROGRAMS.....	104
5.3.3	CASE STUDY THREE: TEST IN RS-07 SPEECH RECOGNITION KIT .....	107
5.3.3.1	The inputs to the speech recognition successful rate without an adaptive Wiener filter. .....	107
5.3.3.2	The inputs to the speech recognition successful rate with an adaptive Wiener filter .... .....	110
5.3.3.3	Results of ASR successful rate with an AWF or without an AWF .....	112
5.3.4	CASE STUDY FOUR: TEST IN TEMPLATE MATCHING ASR IN LABVIEW .....	113
5.3.5	CASE STUDY FIVE: THE SIZE OF THE WIENER FILTER MATRIX.....	117
5.3.6	CASE STUDY SIX: WIENER FILTER IN UNWANTED SPEECH FROM DIFFERENT LOCATIONS ... .....	122
5.4	SUMMARY.....	127
6	CONCLUSIONS AND FUTURE WORK.....	129
6.1	THE IMPROVEMENTS ON THE OTHER RESEARCHES.....	129
6.2	MAJOR FINDINGS AND CONTRIBUTIONS.....	129
6.3	SUGGESTIONS FOR FUTURE WORK.....	130
7	REFERENCE.....	132

## Table of figures

Figure 1-1 Three-microphone beamforming in car .....	1
Figure 1-2 Research objective .....	2
Figure 1-3 Hybrid noise cancellation approach.....	3
Figure 1-4 Structure of this thesis.....	6
Figure 2-1 Block diagram for a Real-Time Adaptive Acoustic noise cancellation for Automatic Speech Recognition in a Car Environment.....	7
Figure 2-2 A 2-microphone beamforming.....	9
Figure 2-3 Delay-and-sum beamformer .....	10
Figure 2-4 Griffiths-Jim beamformer .....	13
Figure 2-5 RLS adaptive filter.....	15
Figure 2-6 LMS adaptive filter as noise canceller block diagram.....	16
Figure 2-7 NLMS adaptive filter as noise canceller block diagram.....	18
Figure 2-8 Chen and Moir's three-microphone system .....	35
Figure 2-9 Overview of adaptive beamformer history .....	47
Figure 3-1 non- stationary noise and interference .....	49
Figure 3-2 A hybrid system with acoustic beamforming VAD and an AWF .....	50
Figure 3-3 A desired zone is defined with 3 microphone.....	52
Figure 3-4 Plan view of 3-microphone VAD valid zone in 3D.....	52
Figure 3-5 A modified adaptive Wiener filter with two microphone.....	53
Figure 3-6 Automobile environment layout .....	54
Figure 3-7 Three microphones in a car.....	55
Figure 3-8 Pre-amplifier .....	56
Figure 3-9 Anti-alias filter.....	56
Figure 3-10 frequency respond.....	57
Figure 3-11 Pre-amplifier and anti-alias filter.....	57
Figure 3-12 A block diagram of DSP hardware and software .....	58
Figure 4-1 Overview of three-microphone VAD controlled three-microphone noise canceller .....	60
Figure 4-2 Three-microphone noise canceller block diagram.....	61
Figure 4-3 Three-microphone VAD Block diagram .....	64
Figure 4-4 A defined active zone .....	64
Figure 4-5 A Wiener filter.....	68

Figure 4-6 A Wiener filter with pre-computed W matrix .....	69
Figure 4-7 W Matrix Calculation for single microphone case .....	70
Figure 4-8 matrix updated during speech or noise period .....	71
Figure 4-9 In-car test plan .....	72
Figure 4-10 A simple ASR block diagram .....	75
Figure 5-1 Overview of experiments .....	77
Figure 5-2 Eight testing points .....	79
Figure 5-3 An overview of testing unit in a car .....	80
Figure 5-4 Speech waveforms .....	81
Figure 5-5 NLMS adaptive filter as noise canceller block diagram .....	84
Figure 5-6 Three-microphone noise canceller block diagram .....	85
Figure 5-7 Definition of a noise canceller valid zone around microphone 1 .....	85
Figure 5-8 The 3-microphone noise canceller in a speech and unwanted speech environment .....	87
Figure 5-9 Refers to Figure 5-6, driver's voice enabled VAD ( $E = 1$ ) and then disabled VAD ( $E = 0$ ) .....	88
Figure 5-10 A test for the noise canceller valid zone .....	89
Figure 5-11 White noise source testing waveforms .....	90
Figure 5-12 Numbering the test points in a frequently moving noise sources environment (Driver's voice and a second voice) .....	91
Figure 5-13 Voice in the VAD valid zone activates the VAD, whilst a white noise source comes from test points 1, 2 and so on .....	92
Figure 5-14 Moving second voice (white noise) results .....	92
Figure 5-15 Numbering the test points in a frequently moving noise sources environment (Driver's voice and a second voice) .....	93
Figure 5-16 $E = 1$ (as in Figure 6-6), other speech arrives via test points 1, 2 and 3 .....	94
Figure 5-17 An ASR with 3-microphone VAD and NLMS ANC .....	95
Figure 5-18 testing plan .....	96
Figure 5-19 Testing of NLMF when $A=0.1$ .....	97
Figure 5-20 Testing of NLMF when $A=0.2$ .....	97
Figure 5-21 Testing of NLMF when $A=0.3$ .....	97
Figure 5-22 Testing of NLMF when $A=0.4$ .....	97
Figure 5-23 Testing of NLMF when $A=0.5$ .....	98
Figure 5-24 Testing of NLMF when $A=0.6$ .....	98



Figure 5-25 Testing of NLMF when $A=0.7$ .....	98
Figure 5-26 Testing of NLMF when $A=0.8$ .....	98
Figure 5-27 Testing of NLMF when $A=0.9$ .....	99
Figure 5-28 Testing of NLMS .....	99
Figure 5-29 Experimental setup .....	101
Figure 5-30 The $W$ matrix is updated during a noise period only or a speech activity period. (a) Wiener filter output (b) Waveform of Signal + noise from Microphone 2 (c) Waveform of noise from Microphone 1 .....	102
Figure 5-31 $W$ matrix is updated in real-time. (a) Wiener filter output (b) Waveform of signal + noise from Microphone (c) Waveform of noise from Microphone 1 .....	102
Figure 5-32 Spectrograms of the filtering process. The horizontal axis represents time (in seconds) and the vertical axis is frequency (in Hz) (a) clean speech “open the door” (b) Filtered speech (c) recording of Microphone1 (d) recording of Microphone2 (e) Intensity scale in dB.....	103
Figure 5-33 In-car test plan .....	104
Figure 5-34 Test results in a stationary car with engine and car radio on.....	105
Figure 5-35 The spectrograms of the waveforms in Figure 6-34. The horizontal axis represents time (in second) and the vertical axis is frequency (in Hz). (a) Spectrogram at Microphone1. (b) Spectrogram at Microphone2. (c) Spectrogram of filtered signal. (d) Intensity scale in dB.....	106
Figure 5-36 Test without an adaptive Wiener filter .....	107
Figure 5-37 records of “open the door” in low noise environment when the car radio and engine does not turn on.....	108
Figure 5-38 Records of speech “open the door” in music environment by Microphone 1	108
Figure 5-39 Records of speech “open the door” in music environment by Microphone 2	109
Figure 5-40 Records of speech “open the door” in unwanted speech environment by Microphone 1 .....	109
Figure 5-41 Records of speech “open the door” in unwanted speech environment by Microphone 2.....	110
Figure 5-42 Test with AWF.....	110
Figure 5-43 Filtered speech “open the door” in music environment.....	111
Figure 5-44 Filtered speech “open the door” in unwanted speech environment.....	111
Figure 5-45 The spectrograms of the waveforms of a female driver’s speech “right” and “left”. The horizontal axis represents time (in second) and the vertical axis is frequency (in	

Hz). (a) Spectrogram at Microphone 1. (b) Spectrogram at Microphone 2. (c) Spectrogram of filtered signal. (d) Intensity scale in dB .....	114
Figure 5-46 The spectrograms of the waveforms of a male driver’s speech “left” and “right”. The horizontal axis represents time (in second) and the vertical axis is frequency (in Hz). (a) Spectrogram at Microphone 1. (b) Spectrogram at Microphone 2. (c) Spectrogram of filtered signal. (d) Intensity scale in dB .....	114
Figure 5-47 Design layout of hybrid noise canceller .....	117
Figure 5-48 Comparing of Frame size at spectrum of filtered speech in grey scale intensity .....	119
Figure 5-49 The spectrograms of filtered speech whilst driver’s speech is incoming with simultaneous unwanted speech from radio loud-speakers or a passenger, in grey scale intensity.....	124
Figure 5-50 Summary of experiment.....	127

## List of Tables

Table 5-1 SNR improvement in different test zones .....	82
Table 5-2 Power of microphones inputs and noise canceller's output.....	90
Table 5-3 SNR results of white noise test .....	90
Table 5-4 Result of AST successful rate in different source positions in 100 tests .....	95
Table 5-5 SNR Analysis at average power.....	106
Table 5-6 Results of test with AWF or without AWF.....	112
Table 5-7 Average power in dB result of cross-correlation between fingerprint "right" and 5 records of "right" or 5 records of "left" from a female driver .....	115
Table 5-8 Average power in dB at result of cross-correlation between fingerprint "right" and 5 records of "right" or 5 records of "left" from a male driver .....	115
Table 5-9 Average power in dB at result of cross-correlation between fingerprint "right" from a female driver and 5 records of "right" or 5 records of "left" from a male driver .....	116
Table 5-10 Average power in dB at result of cross-correlation between fingerprint "right" from a male driver and 5 records of "right" or 5 records of "left" from a female driver .....	116
Table 5-11 Average Power for different orders of W matrix .....	120
Table 5-12 SNRs for different W matrix dimension obtained at an 11025 Hz sample rate .....	120
Table 5-13 Average Signal Power Samples with reference to the seating positions referred to in Figure 5-49.....	125
Table 5-14 Improved SNR with reference to Figure 5-49.....	125

## Abstract

This research is mainly concerned with a robust method for improving the performance of a real-time speech enhancement and noise cancellation for Automatic Speech Recognition (ASR) in a real-time environment. Therefore, the thesis titled, “Real-time adaptive beamformer for Automatic speech Recognition in a car environment” presents an application technique of a beamforming method and Automatic Speech Recognition (ASR) method. In this thesis, a novel solution is presented to the question as below, namely:

How can the driver’s voice control the car using ASR?

The solution in this thesis is an ASR using a hybrid system with acoustic beamforming Voice Activity Detector (VAD) and an Adaptive Wiener Filter.

The beamforming approach is based on a fundamental theory of normalized least-mean squares (NLMS) to improve Signal to Noise Ratio (SNR). The microphone has been implemented with a Voice Activity Detector (VAD) which uses time-delay estimation together with magnitude-squared coherence (MSC). An experiment clearly shows the ability of the composite system to reduce noise outside of a defined active zone. In real-time environments a speech recognition system in a car has to receive the driver’s voice only whilst suppressing background noise e.g. voice from radio. Therefore, this research presents a hybrid real-time adaptive filter which operates within a geometrical zone defined around the head of the desired speaker. Any sound outside of this zone is considered to be noise and suppressed. As this defined geometrical zone is small, it is assumed that only driver's speech is incoming from this zone. The technique uses three microphones to define a geometric based voice-activity detector (VAD) to cancel the unwanted speech coming from outside of the zone. In the case of a sole unwanted speech incoming from outside of a desired zone, this speech is muted at the output of the hybrid noise canceller. In case of an unwanted speech and a desired speech are incoming at the same time, the proposed VAD fails to identify the unwanted speech or desired speech. In such a situation an adaptive Wiener filter is switched on for noise reduction, where the SNR is improved by as much as 28dB.

In order to identify the signal quality of the filtered signal from Wiener filter, a template matching speech recognition system that uses a Wiener filter is designed for testing. In this thesis, a commercial speech recognition system is also applied to test the proposed beamforming based noise cancellation and the adaptive Wiener filter.

## List of Abbreviations and Acronyms

<b>ANC</b>	Adaptive Noise Canceller
<b>ASR</b>	Automatic Speech Recognition
<b>AWF</b>	Adaptive Wiener Filter
<b>BSS</b>	Blind Source Separation
<b>DOA</b>	Direction of Arrival
<b>DFT</b>	Discrete Fourier Transform
<b>DS</b>	Delay and Sum
<b>DSP</b>	Digital Signal Processing
<b>EOD</b>	Estimation of Direction
<b>EOZ</b>	Estimation of Zone
<b>FFT</b>	Fast Fourier Transform
<b>FIR</b>	Finite Impulse Response
<b>GCC</b>	Generalized Cross Correlation
<b>GPS</b>	Global Positioning System
<b>GSC</b>	Generalized Sidelobe Canceller
<b>IDFT</b>	Inverse Discrete Fourier Transform
<b>IFFT</b>	Inverse Fast Fourier Transform
<b>IIR</b>	Infinite Impulse Response
<b>LabVIEW</b>	Laboratory Virtual Instrument Engineering Workbench
<b>LMS</b>	Least Mean Square
<b>MSC</b>	Magnitude Squared Coherence
<b>NLMS</b>	Normalized Least Mean Squares
<b>NLMF</b>	Normalized Least Mean Forth
<b>RLS</b>	Recursive Least Square
<b>SNR</b>	Signal-to-Noise Ratio
<b>TDOA</b>	Time Difference of Arrival
<b>VAD</b>	Voice Activity Detector

## Nomenclature

$\mu$	Step-size parameter for LMS
$\mu_n$	A modified input dependent step size for NLMS
$\mathbf{h}_n$	Tap weight vector at time $n$ of LMS or NLMS
$\hat{\mathbf{h}}_n$	Instantaneous estimate of the tap weight vector at time $n$
$E[\cdot]$	Expectation operator
$\mathbf{R}_{xx}(k)$	Discrete autocorrelation function of the input signal $x_n$
$\mathbf{R}_{xd}(k)$	Discrete cross-correlation function between $x_n$ and the desired response $d_n$
$Z$	Z-transform operator
$\Phi_{xx}(z)$	Z-transform auto power spectrum of the input signal $x_n$
$\Phi_{xd}(z)$	Z-transform cross power spectrum between the input signal $x_n$ and a desired response $d_n$
$\mathbf{R}$	$E[X_n X_n^H]$ , autocorrelation vector of tap input vector $\mathbf{x}_n$
$\mathbf{P}$	$E[X_n d_n^*]$ , cross-correlation vector between the tap input vector $\mathbf{x}_n$ and the desired response $d_n$
$\mathbf{x}_n^T$	Transposition input vector $\mathbf{x}_n$ at time $n$
$\mathbf{x}_n^H$	Hermitian transposition input vector $\mathbf{x}_n$ at time $n$
$\delta(t)$	Dirac delta function
$\hat{S}_{x_i x_i}(i)$	
$\psi_g(f)$	General frequency weighting function
$\mathbf{R}_{d'x'}^{(g)}(\tau)$	Generalized cross correlation function between $d'(t)$ and $x'(t)$
$\hat{\gamma}_{dx}(f)$	Coherence estimate between $x_d(t)$ and $x_x(t)$
$ \gamma_{dx}(f) ^2$	Magnitude squared coherence function
$\lambda_{\max}$	The largest eigenvalue of the tap input auto correlation matrix $\mathbf{R}$
$\beta$	Forgetting factor

$G_{x_1x_2}(f)$	Cross-spectrum of $x_1(t)$ and $x_2(t)$ at frequency $f$
$G_{x_1x_1}(f)$	Auto spectral density functions of $x_1(t)$ at frequency $f$
$G_{x_2x_2}(f)$	Auto spectral density functions of $x_2(t)$ at frequency $f$
$\gamma_{x_1x_2}(f)$	coherence between two zero-mean stationary random processes $x_1(t)$ and $x_2(t)$ , at frequency $f$
$d_{\max}$	Maximum desired time-difference of arrival (TDOA) between two microphones
$C_{\min}$	Minimum desired MSC (with empirical meaning) to prevent reverberant speech from being detected as desired speech

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## Declaration

I declare that the thesis is based on my own research work under the supervision of Dr. T. J. Moir during the Ph.D. study in School of Engineering and Advanced Technology, Massey University at Albany.

The research work has produced a book chapter, journal papers, conference proceedings and presentations during the Ph.D. study. The content of this thesis therefore contains theory, procedure, application and experimental outputs from the research papers published during the research period as listed below.

### Book Chapter

Qi, Z and Moir T (2008), An Adaptive Wiener Filter for Automatic Speech Recognition in a Car Environment with Non-Stationary Noise. In S. Mukhopadhyay & G. S. Gupta (Eds.), *Smart Sensors and Sensing Technology*: Springer-Verlag.

### Refereed journal paper

Qi, Z., & Moir, T. J. (2006). Automotive 3-microphone Noise Canceller in a Frequently Moving Noise Source Environment. *International journal of signal processing*, 3 (4), 298-304.

### Refereed conference proceedings

Qi, Z., & Moir, T. (2008). Automotive speech control in a non-stationary noisy environment. Paper submitted at the 15th International Conference on Mechatronics and Machine Vision in Practice, Auckland, New Zealand.

Qi, Z. & Moir, T. (2007). A Design of Automotive Voice Recognizer Using LabVIEW. Paper was accepted to The 14th Electronics New Zealand Conference (ENZCon). Wellington New Zealand.

Qi, Z., & Moir, T. J. (2007). An Adaptive Wiener Filter for an Automotive Application with Non-Stationary Noise, Paper was accepted to 2<sup>nd</sup> International Conference on Sensing Technology 2007. Palmerston North, New Zealand.

Qi, Z., & Moir, T. J. (2005). An Automotive three-microphone Voice Activity Detector and noise canceller, *2005 International Conference on Intelligent Sensors, Sensor Networks and Information*. Melbourne, 5 - 8 December, Melbourne.

Qi, Z., & Moir, T. J. (2005). A geometrical active zone voice activity detector in car, *2nd IIMS Post-Graduate Conference*. Auckland, 27 October.

Non-refereed journal paper

Qi, Z., & Moir, T. J. (2005). Automotive three-microphone Voice Activity Detector and noise-canceller. *Research Letters in the Information and Mathematical Sciences*, 7 (July 2005), 147-156, Institute of Information and Mathematical Sciences, Massey University, Auckland, New Zealand.

Candidate's signature:

A handwritten signature in black ink, appearing to be 'Ziming Qi', written in a cursive style.

Candidate's name: Ziming Qi      Date: 18-08-2008

