

2. Mosayyebpour, S.; Lohrasbipeydeh, H.; Esmaeili, M.; Gulliver, T.A.; "Time delay estimation via minimum-phase and all-pass component processing," *Proc. IEEE Int. Conf. Acoustics, Speech and Signal Process. (ICASSP), 2013.*
3. Lohrasbipeydeh, H.; Mosayyebpour, S.; Gulliver, T.A.; "SINGLE HYDROPHONE PASSIVE ACOUSTIC SPERM WHALE RANGE AND DEPTH ESTIMATION." *Proc. IEEE Int. Conf. Acoustics, Speech and Signal Process. (ICASSP), 2013.*
4. Mosayyebpour, S.; Sheikhzadeh, H.; Gulliver T.A.; Esmaeili, M.; "Single-Microphone LP Residual Skewness-Based Approach for Inverse Filtering of Room Impulse Response." *IEEE Transaction on Audio, Speech and Language processing vol. 20, no. 5, pp. 1617--1632, July 2012.*
5. Mosayyebpour, S.; Keshavarz, A.; Bigiuash, M.; Gulliver, T.A.; Esmaeili, M.; "Speech-Model Based Accurate Blind Reverberation Time Estimation Using an LPC Filter." *IEEE Transaction on Audio, Speech and Language processing, vol. 20, no. 6, pp. 1884--1893, Aug. 2012.*
6. Mosayyebpour, S.; Lohrasibpeydeh, H.; Gulliver, T.A.; Esmaeili, M.; "Single Source Time Delay Estimation using Two Microphones in a Noisy Reverberant Environment." *IEEE Transaction on Audio, Speech and Language processing (in revised process).*
7. Mosayyebpour, Saeed.; Mosayyebpour, Sina; Gulliver, T.A.; Tzanetakis, G.; Esmaeili, M.; "Time Delay Estimation based on Logarithm Phase Difference in Reverberant and Time Varying Environments." *IEEE signal processing letter (in revised process).*
8. Mosayyebpour, S.; Gulliver, T.A.; Esmaeili, M.; "Single-microphone speech Enhancement by skewness maximization and spectral subtraction." *International Workshop on Acoustic Signal Enhancement (IWAENC), pp. 1-4, Sep. 2012.*



University
of Victoria

Graduate Studies

PROGRAMME

The Final Oral Examination
for the Degree of

DOCTOR OF PHILOSOPHY

(Department of Electrical and Computer Engineering)

Saeed Mosayyebpour

2010 Amirkabir Univ. of Technology MSc (Communication Eng.)
2009 Amirkabir Univ. of Technology BSc (Communication Eng.)
2007 Amirkabir Univ. of Technology BSc (Electronic Eng.)

"Robust Single-Channel Speech Enhancement and Speaker Localization in Adverse Environments"

Wednesday, April 23, 2014

1:00pm

Engineering/Computer Science Building, room 467

Supervisory Committee:

Dr. T. Aaron Gulliver, Department of Electrical and Computer Engineering, UVic (Co-Supervisor)

Dr. Morteza Esmaeili, Department of Electrical and Computer Engineering, UVic (Co-Supervisor)

Dr. Wu-Sheng Lu, Department of Electrical and Computer Engineering, UVic (Member)

Dr. George Tzanetakis, Department of Computer Science, UVic (Outside Member)

External Examiner:

Dr. Ivan J. Tashev, Principal Software Architect, Microsoft Research

Chair of Oral Examination:

Dr. Martha McGinnis-Archibald, Department of Linguistics, UVic

Abstract

In speech communication systems such as voice-controlled systems, hands-free mobile telephones, and hearing aids, the received signals are degraded by room reverberation and background noise.

This degradation can reduce the perceived quality and intelligibility of the speech, and decrease the performance of speech enhancement and source localization. These problems are difficult to solve due to the colored and nonstationary nature of speech signal, and features of the Room Impulse Response (RIR) such as its long duration and nonminimum phase response. In this dissertation, we focus on two topics of speech enhancement and speaker localization in noisy reverberant environments.

A two-stage speech enhancement method is presented to suppress both early and late reverberation in noisy speech using only one microphone. It is shown that this method works well even in highly reverberant room. Experiments under different acoustic conditions confirm that the proposed blind method is superior in terms of reducing early and late reverberation effects and noise compared to well-known single-microphone techniques in the literature.

Time Difference Of Arrival (TDOA)-based methods usually provide the most accurate source localization in adverse condition. The key issue for these methods is to accurately estimate the TDOA using the smallest number of microphones. Two robust Time Delay Estimation (TDE) methods are proposed which use the information from only two microphones. One method is based on adaptive inverse filtering which provides superior performance even in highly reverberant and moderately noisy condition. It also has negligible failure estimation and thus makes it the most reliable method in real environment.

This method has high computational complexity due to its estimation in the first stage for the first microphone. As a result, it cannot be applied for high time-varying condition and real-time processing. Our second method improves this problem by

introducing two effective preprocessing stages for the conventional Cross Correlation (CC)-based methods. The results obtained in different noisy reverberant conditions including a real and time-varying room demonstrate that the proposed methods are superior compared to the conventional TDE methods.

Awards, Scholarships, Fellowships

2013 – Excellence in Teaching Award, *University of Victoria*
2012 – Charles S. Humphrey Graduate Student Award, *University of Victoria*
2011-2012 – Fellowship Award, *University of Victoria*
2007 – Scholarship of Electrical Engineering Department, *Amirkabir University of Technology*

Presentations

1. Saeed Mosayyebpour “*Time delay estimation via minimum-phase and all-pass component processing*” *IEEE Int. Conf. Acoustics, Speech and Signal Process, Vancouver, British Columbia, Canada. May 2013 (poster)*
2. Saeed Mosayyebpour “*Single hydrophone passive acoustic sperm whale range and depth estimation*” *EEE Int. Conf. Acoustics, Speech and Signal Process, Vancouver, British Columbia, Canada. May 2013 (poster)*
3. Saeed Mosayyebpour “*Single-microphone speech Enhancement by skewness maximization and spectral subtraction*” *International Workshop on Acoustic Signal Enhancement, Vancouver, British Columbia, Canada. September 2012 (Poster)*

Publications

1. Mosayyebpour, S.; Esmaili, M.; Gulliver T.A.; "Single-Microphone Early and Late Reverberation Suppression in Noisy Speech" *IEEE Transaction on Audio, Speech and Language processing vol. 21, no. 2, pp. 322--335, Feb 2013.*