Acoustic Signal Processing Algorithms for Reverberant Environments

Terence Betlehem

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Department of Information Engineering
Research School of Information Sciences and Engineering
The Australian National University

Declaration

The content of this thesis are the result of original research and has not been submitted for a postgraduate degree at any other university or institution. Much of this work has either been published or submitted for publications as journal papers and conference proceedings. Following is a list of these papers.

Journal Publications

- T. Betlehem and T. D. Abhayapala, "Theory and Design of Sound Field Reproduction in a Reverberant Room," *J. Acoust. Soc. Amer.*, vol. 117, no. 4, pp. 2100-2111, 2005.
- T. Betlehem and T. D. Abhayapala, "Robustness of Equalization in a Reverberant Room," *IEEE Trans. Speech and Audio*, (to be submitted).

Conference Proceedings

- T. Betlehem and R. C. Williamson, "Acoustic Beamforming Exploiting Directionality of Human Speech Sources," in *Proc. IEEE Int. Conf. Acoust., Speech and Signal Processing* (ICASSP'03), pp. 365-368, April, 2003.
- T. Betlehem and T. D. Abhayapala, "Spherical Harmonic Analysis of Equalization in a Reverberant Room," in *Proc. IEEE Int. Conf. Acoust., Speech and Signal Processing* (ICASSP'04), pp. 689-672, April, 2004.
- T. Betlehem and T. D. Abhayapala, "A Modal Approach to Sound Field Reproduction in Reverberant Rooms," *Proc. IEEE Int. Conf. Acoust., Speech and Signal Processing* (ICASSP'05), pp. 289-292, April, 2005.

The research presented in this thesis has been performed jointly with Dr. Thushara D. Abhayapala and Prof. Robert C. Williamson. Approximately 70% of this work is my own.

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Abstract

This thesis investigates the design and the analysis of acoustic signal processing algorithms in reverberant rooms. Reverberation poses a major challenge to acoustic signal processing problems. It degrades speech intelligibility and causes many acoustic algorithms that process sound to perform poorly. Current solutions to the reverberation problem frequently only work in lightly reverberant environments. There is need to improve the reverberant performance of acoustic algorithms.

The approach of this thesis is to explore how the intrinsic properties of reverberation can be exploited to improve acoustic signal processing algorithms. A general approach to soundfield modelling using statistical room acoustics is applied to analyze the reverberant performance of several acoustic algorithms. A model of the underlying structure of reverberation is incorporated to create a new method of soundfield reproduction.

Several outcomes resulting from this approach are: (i) a study of how more sound capture with directional microphones and beamformers can improve the robustness of acoustic equalization, (ii) an assessment of the extent to which source tracking can improve accuracy of source localization, (iii) a new method of soundfield reproduction for reverberant rooms, based upon a parametrization of the acoustic transfer function and (iv) a study of beamforming to directional sources, specifically exploiting the directionality of human speech.

The approach to soundfield modelling has permitted a study of algorithm performance on important parameters of the room acoustics and the algorithm design. The performance of acoustic equalization and source tracking have been found to depend not only on the levels of reverberation but also on the correlation of pressure between points in reverberant soundfields. This correlation can be increased by sound capture with directional capture devices. Work on soundfield reproduction has shown that, though reverberation significantly degrades the performance of conventional techniques, by accounting for the reverberation it is possible to design reproduction methods that function well in reverberant environments.

Symbols and Terms

$\lceil\cdot\rceil$	ceiling operator
$\lfloor\cdot\rfloor$	floor operator
$[\cdot]^*$	complex conjugate of a matrix
$[\cdot]^T$	transpose of a matrix
$[\cdot]^H$	complex conjugate transpose of a matrix
$\lvert\cdot\rvert$	magnitude of a complex number
$\angle\cdot$	phase of a complex number
$\lVert\cdot\rVert$	Euclidian norm of a vector
$\boldsymbol{x}\cdot\boldsymbol{y}$	dot product between two vectors
$E\{\cdot\}$	expectation operator
$\Pr\{\cdot\}$	probability
$\mathrm{Var}\{\cdot\}$	variance operator
$\mathrm{Re}\{\cdot\}$	real part
$\mathrm{Im}\{\cdot\}$	imaginary part
$\delta(\cdot)$	Dirac delta function
δ_{nm}	Kronecker delta function
i	$\sqrt{-1}$
$\boldsymbol{I}_n$	$n\times n$ identity matrix
$\mathbb{C}^n$	n dimensional complex number space
$\mathbb{R}^n$	n dimensional real number space
$\mathbb{Z}^*$	set of non-negative integers
CDF	cumulative density function
DFT	discrete Fourier transform
DRR	direct-to-reverberant energy ratio
MTF	modulation transfer function
PDF	probability density function
SNR	signal-to-noise ratio
STI	speech transmission index
ULA	uniform linear array
WNG	white noise gain

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