Patrick A. Naylor

CURRICULUM VITAE, September 13, 2018

Imperial College London
Department of Electrical and Electronic Engineering
London, SW7 2AZ, UK

Phone: +44 (0) 20 7594 6235
email: p.naylor@imperial.ac.uk
URL: http://www.ee.ic.ac.uk/naylor

Current position

Reader in Speech and Audio Signal Processing, Imperial College London

Areas of specialization

Acoustic signal processing. Speech signal processing.

Appointments held

1989-1999	Lecturer, Department of Electrical and Electronic Engineering, Imperial College London
1999-2007	Senior Lecturer, Department of Electrical and Electronic Engineering, Imperial College London
1998-2002	Course Director, Information Systems Engineering Degree Programme, Imperial College London
2004-2012	Director of Postgraduate Studies, Department of Electrical and Electronic Engineering, Imperial College London
2007-	Reader in Speech and Audio Signal Processing, Imperial College London
2008-	co-Director, Centre for Law Enforcement Audio Research
2019-2020	President, European Association for Signal Processing

Professional membership

Senior Member of IEEE
Fellow of IET
Chartered Engineer

Education

1982-1986 BEng (1st Class) in Electronic and Electrical Engineering, University of Sheffield
1986-1989 PhD in Speech Signal Processing, Imperial College London

Awards

2017 Best Paper Award: I. D. Gebru, C. Evers, P. A. Naylor, R. Horaud, "Audio-visual tracking by density approximation in a sequential Bayesian filtering framework", *Hands-free Speech Communication and Microphone Arrays.* San Francisco, March 2017.

2017 Best Paper Award Marie Skłodowska-Curie Research and Innovation, Innovative Training Network 'DREAMS': P. Peso-Parada, D. Sharma, J. Lainez, D. Barreda, T. van Waterschoot, and P. A. Naylor, "A single-channel non-intrusive C50 estimator correlated with speech recognition performance," *IEEE Trans. Audio, Speech, Lang. Process.*, vol. 24, no. 4, pp. 719 – 732, Jan. 2016.

2014 Best Paper Award: R. Stanton, N. D. Gaubitch, P. A. Naylor, and M. Brookes, "A differentiable approximation to speech intelligibility index with applications to listening enhancement," in *Audio Engineering Society 54th International Conference: Audio Forensics*, June 2014.

2008 Royal Academy of Engineering Industrial Secondment Award

2005 Best Student Paper Award, Andy W. H. Khong, International Workshop on Acoustic Echo and Noise Control

1995 Rector's Award for Excellence in Teaching - leadership and innovation in the Information Systems Engineering (now EIE) Programme, Imperial College London

1994 Best Student Paper Award, M. A. Sayid, IEE International Conference on Control

Courses

2003- Speech Processing. MSc and MEng (part IV)
1997- Digital Signal Processing. MEng/BEng (part III)
2005 Communications. MEng/BEng (part II)
1990-2001 Digital Electronics. MEng/BEng (part I)
1991-1993 Digital System Design. MEng/BEng (part IV)
1994-1996 Computing. MEng/BEng (part I)

National and International Committees

2018 President Elect, European Association for Signal Processing
2015-2016 Chair of IEEE Technical Committee on Audio and Acoustic Signal Processing
2014 Vice Chair of IEEE Technical Committee on Audio and Acoustic Signal Processing
2012- Board of Directors, European Association for Signal Processing
2012-2014 Audio Engineering Society UK Committee
2012 Academy of Finland, 'Image, Data and Signal Processing', Research Funding Re-

view Panel
2011 Government Chief Scientific Advisor's Committee on Speech Technology
2011-2014 Chair IEEE AASP Technical Subcommittee for AASP Technology Challenges
2003- Member and Associate Member of IEEE Technical Committee on Audio and Acoustic Signal Processing
1997- Member of Technical Committee of the International Workshop on Acoustic Echo and Noise Control

Public Engagement

2015 "Interaction with Sound in a 3D World", Royal Society Summer Science Exhibition, London, UK. June 28th to July 5th, 2015. Lead exhibitor role. 15,000 visitors plus website, press and video channels.

Conference Chairs

2017 General Chair, IEEE Workshop on Applications of Signal Processing to Audio and Acoustics (WASPAA-2017), New York, USA.
2015 Area Chair, Audio and Acoustic Signal Processing, IEEE International Conference on Acoustics Speech and Signal Processing (ICASSP-2015), Brisbane, Australia.
2015 Tutorials Chair, European Signal Processing Conference EUSIPCO-2015, Nice, France.
2012 Area Coordinator for Speech Production and Enhancement, INTERSPEECH-2012, Portland, Oregon, USA.
2011 Publicity Chair of IEEE International Conference on Acoustics Speech and Signal Processing (ICASSP-2011), Prague, Czech Republic.
2009 Technical Chair of European Signal Processing Conference EUSIPCO-2009, Glasgow, UK.
2006 Track Chair Adaptive Systems and Processing, Asilomar Conference on Signals, Systems and Computers, Pacific Grove, USA.
1997 General Chair of International Workshop on Acoustic Echo and Noise Control IWAENC-1997, London, UK.

Conference Tutorial

2017 "Reverberation and Dereverberation of Audio Music and Speech", P. A. Naylor, E. De Sena, T. van Waterschoot, Tutorial presented at European Signal Processing Conference, 2017.
2010 "Speech Dereverberation", P. A. Naylor, C. Evers, and E. A. P. Habets, Tutorial presented at European Signal Processing Conference, 2010.

Recent Conference Session Chairs

2018 Session Chair, 'Source Localization and Array Calibration', IEEE International Conference on Acoustics Speech and Signal Processing (`ICASSP-2018`), Calgary, Canada
2017 Session Chair, 'Signal Analysis and Processing for Hearing Devices', IEEE Interna-

tional Conference on Acoustics Speech and Signal Processing (ICASSP-2017), New Orleans, USA

2016 Session Chair, 'Microphone Arrays and Spatial Acoustic Processing', IEEE International Conference on Acoustics Speech and Signal Processing (ICASSP-2016), Shanghai, China

2015 Session Chair, 'Reverberation and Source Separation', IEEE Workshop on Applications of Signal Processing to Audio and Acoustics (WASPAA-2015), New York, USA

2015 Session Chair, 'Reverberant Signal Analysis and Decomposition for Audio and Speech Processing', IEEE International Conference on Acoustics Speech and Signal Processing (ICASSP-2015), Brisbane, Australia.

2013 Session Chair, 'Speech Processing', European Signal Processing Conference (EUSIPCO-2013), Marrakech, Morocco.

2013 Session Chair, 'Source Separation and Biomedical Applications', IEEE International Conference on Acoustics Speech and Signal Processing (ICASSP-2013), Vancouver, Canada.

2012 Session Chair, 'Adaptive Beamforming', European Signal Processing Conference (EUSIPCO-2012), Bucharest, Romania.

2011 Session Chair, 'Source Separation and Localization', IEEE Workshop on Applications of Signal Processing to Audio and Acoustics (WASPAA-2011), New Paltz, NY, USA.

2011 Session Chair, 'Acoustic scene reconstruction and environment-aware space-time processing', European Signal Processing Conference (EUSIPCO-2011), Barcelona, Spain.

2011 Session Chair, 'Echo Cancellation', IEEE International Conference on Acoustics Speech and Signal Processing (ICASSP-2011), Prague, Czech Republic.

2010 Plenary Chair, 'Noise and Echo Control for Immersive VoiceCommunication in Spacesuits', International Workshop on Acoustic Echo and Noise Control IWAENC-2010, Tel Aviv, Israel.

2010 Session Chair, 'Speech Analysis', European Signal Processing Conference (EUSIPCO-2010), Aalborg, Denmark.

2010 Session Chair, 'Room Acoustics and Hearing Aids', IEEE International Conference on Acoustics Speech and Signal Processing (ICASSP-2010), Dallas, USA.

2009 Session Chair, 'Acoustics, Active noise control, and Sound Reproduction', IEEE International Conference on Acoustics Speech and Signal Processing (ICASSP-2009), Taipei, Taiwan.

2008 Special Session Chair, 'Blind System Identification, Multi-channel System Inversion and Speech Dereverberation', Asilomar Conference on Signals, Systems and Computers, Pacific Grove, USA.

2008 Session Chair, 'Source Separation I', IEEE International Conference on Acoustics Speech and Signal Processing (ICASSP-2008), Las Vegas, USA.

2008 Session Chair, 'Speech Analysis', European Signal Processing Conference (EUSIPCO-2008), Lausanne, Switzerland.

Editorships

2017- Senior Area Editor, IEEE Transactions in Audio Speech and Language Processing

2012-	Section Editor for Speech and Audio Processing, Academic Press Signal Processing Library
2008-2013	Associate Editor IEEE Transactions in Audio Speech and Language Processing
2004-2008	Associate Editor IEEE Signal Processing Letters
2011-2013	Series Editor, International Journal of Adaptive Control and Signal Processing
2010	Guest Editor, Special Issue of IEEE Transactions in Audio Speech and Language Processing, "Handling Reverberant Speech"
2010	Guest Editor, Special Issue of EURASIP Journal on Advances in Signal Processing, "Microphone Array Speech Processing"
2006	Guest Editor, Special Issue of EURASIP Journal on Audio Speech and Music Processing, "Sparse Adaptive Systems"

Keynote Talks

"Modulation-domain Multichannel Kalman Filtering for Speech Enhancement", at *Symposium on Information Science and Technology* (SSIST), Shanghai, China, Jul. 2018.

"Enhancement of Ambient Speech for Robot Audition", at *8th Speech in Noise Workshop*, Groningen, Netherlands, Jan. 2016.

"What's Happening in Speech Enhancement and Acoustic Signal Processing?", P. A. Naylor, K. Baykaner, A. Hines, A. H. Moore, at *UK-Speech Conference*, Cambridge, UK, Sep. 2013.

"Acoustic Signal Processing in Noise: It's Not Getting Any Quieter", at *International Workshop on Acoustic Echo and Noise Control*, Aachen, Germany, Sep. 2012.

"Space-time Audio Processing: An Overview", at *New Trends in Audio Processing and Rendering*, FAST, Milan, 2011.

"Trends in Audio and Acoustic Signal Processing", Expert's Summary at *IEEE Int. Conf. on Acoustics, Speech and Signal Processing*, Prague, May 2011.

"Intelligibility Estimation in Law Enforcement Speech Processing", at *ITG Speech Communication Conference*, University Bochum-Ruhr, Germany, Oct 2010.

"Speech Dereverberation", at *International Workshop on Acoustic Echo and Noise Control*, Netherlands, Sep 2005.

Invited Talks

"Parametric Multichannel Kalman Filtering with Application to Speech Enhancement", Massachusetts Institute of Technology (MIT), Cambridge MA, USA, Sep.

2018.
“Multichannel Kalman Filtering for Speech Enhancement Applications”, Toshiba Cambridge Research Laboratory, Cambridge, UK, May 2018.
“Multichannel Kalman Filtering for Speech Enhancement Applications”, Cirrus Logic, London, UK, March 2018.
“Measurement and Exploitation of Reverberation in Speech Signals”, Institute of Sound and Vibration Research, University of Southampton, UK, June 2017.
“Measurement and Exploitation of Reverberation in Speech Signals”, Aalborg University, Denmark, Feb 2017.
“Measurement and Exploitation of Reverberation in Speech Signals”, Oticon A/S, Denmark, Jan 2017.
“Multichannel Blind Acoustic System Identification with Under-modelling”, Aachen University, Germany, May 2016.
“Signal Processing for Robot Audition”, University of York, UK, Jan 2015.
“Audition in Robots”, Université LILLE1, France, May 2015.
“Signal Processing Techniques for Acoustic Inference and Dereverberation”, University of Kiel, Germany, April. 2014.
“Acoustic Signal Processing and Applications to Speech Dereverberation”, University of Sheffield, Nov. 2013.
“Audio Signal Processing and Applications to Speech Dereverberation”, Institute of Sound and Vibration Research (ISVR), University of Southampton, May 2013.
“Speech Processing in Law Enforcement Applications”, ITG Acoustic Signal Processing Symposium, Oldengurg University, Oct 2011.
“Speech and Audio Processing with Applications to Speech Dereverberation”, Institute for Communication Systems and Data Processing, RWTH Aachen University, Oct 2011.
“Multichannel Acoustic System Identification and Inversion for Dereverberation”, HP Labs, Palo Alto, USA, Nov 2010.
“Speech Dereverberation”, Universite de Rennes 1, Sep 2010.
“Speech Dereverberation using Blind System Identification and Inversion”, University of York, UK, Jun 2010.
“Recent Advances and Future Strategies in Speech Processing”, COST 2103: 4th Advanced Voice Function Workshop, York, UK, May 2010.
“Speech Dereverberation for Telecommunication Applications”, University of Missouri-Rolla, USA, Apr 2007.
“Speech Dereverberation for Telecommunication Applications”, Kings College London, UK, Nov 2006.
“Adaptive Filters and SIMO System Identification”, National Physical Laboratory, UK, Jun 2006.
“Speech Dereverberation”, IEEE Signal Processing Chapter, hosted at De Montfort University, UK, Nov 2005.
“Aspects of Speech Processing and Adaptive Filters”, Bell Labs, Murray Hill NJ, USA, Feb 2002.

Research Grants and Contracts

Jan 2017	"Acoustic Signal Processing and Scene Analysis for Socially Assistive Robots", EPSRC Fellowship - Christine Evers, 3 years, £330K.
April 2016	"A Novel Technique For Sensing Underwater Objects Using Coherence Modulation Acoustic Speckle Interferometry", DSTL, 1 year, £91K.
Jan 2016	"EPSRC Doctoral Prize fellowship - James Eaton", EPSRC, 1 year, £51K.
Sept 2015	"Environment-aware Listener-Optimized Binaural Enhancement of Speech (E-LOBES)", Co-I, EPSRC, 3 years joint with Mike Brookes and UCL, £600K (total), £471K (Imperial).
July 2015	"Personalized fitting and evaluation of hearing aids with EEG responses", Co-I, EPSRC, 3 years joint with Tobias Reichenbach, University of Southampton and University of Manchester, £917K (total), £297K (Imperial)
Aug 2014	"3-Dimensional Acoustic Imaging of the Dynamic Osteoarthritic Knee", PI, Wellcome Trust Facility Network of Excellence, £22K.
Jan 2014	"Embodied Audition for RobotS - EARS", PI, EC 7th Framework Programme, 3 years, joint with University of Erlangen-Nurnberg (DE), Ben-Gurion University of the Negev, IL, Humboldt University of Berlin, DE, INRIA, FR, Alderbaran Robotics SA, FR, €3.5M (total), €535K (Imperial).
July 2013	"Environment Aware Dereverberation for VoIP Applications", PI, Google, £118K.
Jan 2013	"Dereverberation and Reverberation of Audio, Music and Speech - DREAMS", PI, Marie Curie Initial Training Network, joint with KU Leuven (BE), Uni. Aalborg (DK), Uni. Oldenburg (DE), €4.1M (total), €751K (Imperial).
Mar 2011	"Lightweight Noise Protection System", QinetiQ, Co-I, £112K.
Feb 2010	"SpotForming", Marie Curie Intra-European Fellowships, Dr Emanuel Habets, €182K.
Jan 2009	"Self-Configuring ENvironment-aware Intelligent aCoustic sensing - SCENIC", PI, EC 7th Framework Programme, 3 years, joint with Polytechnic of Milan (IT), Fondatione Bruno Kessler (IT), University of Erlangen-Nurnberg (DE), €1.6M (total), €335K (Imperial).
Sep 2008	"Speech Analysis for High Performance Phoneme Recognition", PI, Royal Academy of Engineering Global Research Award, Dr Jon Gudnason with Columbia University USA, £39K.
Jan 2008	"Voicemail Conversion System", PI, Royal Academy of Engineering Industrial Secondment, 6 months, £18K.
Oct 2007	"Centre for Law Enforcement Audio Research", Co-I, Home Office, 5 years, joint with Mike Brookes and Mark Huckvale (UCL), £1.2M.
Oct 2006	"Speech to Text Conversion", PI, SpinVox Ltd, 2 years, £160K.
May 2004	"Enhancement of Reverberant Speech for Telecommunications Applications", PI, EPSRC, 3 years, £220K.
Sep 2003	"Multisensor Acoustic Data Fusion for Enhanced Voice-based Human-Machine Interfaces", PI, DTC Data and Information Fusion, 3 years, £55K.
Nov 2002	"Echo Cancellation for Packet-switched Networks", PI, Trinity Convergence Inc., 1 year, £85K.
Feb 2002	"Identification and Exploitation of Sparseness in Adaptive Systems", EPSRC travel award for visit to Bell Labs, Murray Hill NJ, USA, £5K.

Mar 2001 “Zero IF Architectures for DAB Receivers”, Co-I, Panasonic System LSI Design Europe, 2 years, joint with Mike Brookes, Peter Cheung, Alison Burdett, £240K.

Jun 2000 “Novel Features for Speaker Verification”, Co-I, EPSRC, 3 years, joint with Mike Brookes, £144K.

Jun 1997 “Multi-channel speech enhancement with applications to conferencing and multi-media systems”, Co-I, EPSRC, 3 years, joint with Jonathon Chambers, £181K.

Mar 1997 “Acoustic Echo Control”, Co-I, LG Semicon Ltd, 2 years, joint with Mike Brookes, £110K.

Jan 1992 “Enhancement of Hands-free Communications - FREETEL”, PI, EU ESPRIT III, £200K

PhD Students Graduated

Oct 2013 - June 2017 Mathieu Hu, “Cross-Relation Based Blind Identification of Acoustic SIMO Systems and Applications”. *First destination:* INRIA, Nancy, France.

Oct 2012 - Sep 2016 Hamza Javed, “Perceptual Modelling and Processing of Reverberant Speech”. *First destination:* Oxford University, UK (post-doc).

Apr 2013 - July 2016 Pablo Peso Parada, “Spatial Features of Reverberant Speech: Estimation and Application to Recognition and Diarization”. *First destination:* Cirrus Logic.

Oct 2012 - Apr 2016 James Eaton, “Non-intrusive Estimation of Acoustic Parameters from Degraded Speech”. *First destination*: Imperial College London (post-doc).

Oct 2011 - Jan 2016 Felicia Lim, “Robust Multichannel Equalization for Blind Speech Dereverberation”. *First destination:* Google Inc..

Oct 2010 - Oct 2013 Daniel Jarrett, “Spherical Microphone Array Processing for Acoustic Parameter Estimation and Signal Enhancement”. *First destination:* Kilburn and Strode LLP.

Oct 2008 - Aug 2012 Jason Filos, “Inferring Room Geometries”. *First destination:* Qualcomm, San Diego, USA.

Oct 2008 - July 2012 Dushyant Sharma, “Speech Assessment and Characterization for Law Enforcement Applications”. *First destination:* Nuance Communications, Inc.

Oct 2008 - Sep 2011 Pradeep Loganathan, “Sparseness-controlled Adaptive Algorithms for Supervised and Unsupervised System Identification”. *First destination:* Schlumberger.

Mar 2007 - Jul 2010 Wancheng Zhang, “Robust Equalization of Multichannel Acoustic Systems”. *First destination:* Beijing Institute of Technology, China.

Oct 2006 - Mar 2010 Mark R. P. Thomas, “Glottal-Synchronous Speech Processing”. *First destination:* Postdoc Imperial College. *Now with:* Microsoft Research, Redmond, USA.

Oct 2005 - Jul 2009 Jimi Y-C Wen, “Reverberation: Models, Estimation and Applications”. *First destination:* Institute for Information Industry, Taiwan.

Oct 2005 - Jul 2009 Xiang Lin, “System Identification with Applications in Speech Enhancement”. *First destination:* Credit-Suisse.

Oct 2002 - Nov 2006 Uttachai Manmontri, “A Gradient-based Approach to Unsupervised Signal Separation Using Signal Properties”. *First destination:* Thailand Institute of Scientific and Technological Research.

Oct 2002 - Dec 2006 Nikolay D. Gaubitch, “Blind Identification of Acoustic Systems and Enhancement of Reverberant Speech”. *First destination:* Postdoc Imperial College. *Now with:* Delft University of Technology / Google, Netherlands.

Oct 2002 - Feb 2006 Andy W. H. Khong, “Adaptive Algorithms Employing Tap Selection for Single Channel and Stereophonic Acoustic Echo Cancellation”. *First destination:* Assistant Prof., Nanyang Technological University, Singapore.

Oct 2001 - Nov 2005 Alexander Wright, “Adaptive Compensation Techniques in COFDM Direct Conversion Receivers”. *First destination:* PA Consulting Group.

Oct 1998 - Feb 2002 Amere Oakman, “Dynamic Nonuniform Filter Banks for Subband Adaptive Filtering”. *First destination:* Unilever Corporation.

Sep 1997 - Oct 2000 Neil Forsyth, “A Subband and Noise Robust Approach to Stereophonic Acoustic Echo Cancellation”. *First destination:* Astrium.

Oct 1997 - Mar 2002 Tassos Kounoudes, “Automatic Segmentation of the Larynx Cycle with Applications to Speaker Verification”. *Now with:* SignalGeneriX Ltd, Cyprus.

Oct 1995 - Mar 1998 Nikos Doukas , “Voice Activity Detection Using Energy Based Measures and Source Separation”

Oct 1997 - Nov 2000 Warren Sherliker , “Acoustic Echo Cancellation Algorithms with Tap Selection for Non-stationary Environments”

Oct 1996 - Jun 2000 Andreas Neocleous, “Speaker Verification using Voice Source Parameters”

Sep 1993 - Jul 1996 Joanna Hart, “Multirate Subband Structures with Application to Adaptive Acoustic Echo Cancellation”. *First destination:* NEC.

Oct 1992 - Feb 1995 — Mohammed Sayid, "Nonlinear Adaptive Filtering: System Identification, Nonlinearity Characterisation and Order Estimation"

Recent External PhD Examining / PhD Review Committees

Feb 2018 — Sebastian Braun, "Speech Dereverberation In Noisy Environments Using Time-Frequency Domain Signal Models". *Advisor:* Emanuel Habets, Friedrich-Alexander-Universitat Erlangen-Nurnberg, Germany.

Oct 2017 — Amin Hassani, "Distributed Signal Processing Algorithms for Multi-Task Wireless Acoustic Sensor Networks". *Advisor:* Marc Moonen, KU Leuven, Belgium.

June 2017 — Mojtaba Farmani, "Informed Sound Source Localization for Hearing Aid Applications". *Advisor:* Jesper Jensen, Aalborg University, Denmark.

Apr 2016 — Christoph Nelke, "Wind Noise Reduction – Signal Processing Concepts". *Advisor:* Peter Vary, RWTH Aachen University, Germany.

Dec 2015 — Ina Kodrasi, "Dereverberation and Noise Reduction Techniques based on Acoustic Multichannel Equalization". *Advisor:* Simon Doclo, Oldenburg University, Germany.

May 2014 — Edwin Mabande, "Robust Time-Invariant Broadband Beamforming as a Convex Optimization Problem". *Advisor:* Walter Kellermann, Oldenburg University, Germany.

May 2013 — Christelle Yemdji, "Acoustic Echo Cancellation for Single and Dual Microphone Devices". *Advisor:* Nicholas Evans, Ecole Nationale Supérieure des Télécommunications, France.

Jan 2013 — Cees Taal, "Prediction and Optimization of Speech Intelligibility in Adverse Conditions ". *Advisor:* Richard Heusdens, Technische Universiteit Delft, The Netherlands.

Nov 2012 — Dejan Markovic, "Plenacoustic processing in the ray space: applications to acoustic scene modeling and analysis". *Advisor:* Augusto Sarti, Politecnico di Milano, Italy.

Consultancies

2014 - 2015 Expert witness, London.
2011 - 2012 Expert witness, London.
2008 - Nuance Communications inc., Scientific Advisor.

2010 - 2011	Expert witness, London.
2003 - 2012	Total, expert witness.
2007 - 2008	SpinVox Ltd, consultant in speech processing.
2008	Expert witness, London.
2001	Trinity Covergence inc., consultant in acoustic echo cancellation.
1999 - 2001	Nokia, training consultant.
2000	Hitachi, training consultant.
1996	Texas Instruments Ltd, technical author and training consultant.
1987 - 1990	KEF Electronics Ltd, consultant in loudspeaker acoustics and measurement.

List of Publications

Books

[1] D. Jarrett, E. A. P. Habets, and P. A. Naylor, *Theory and Applications of Spherical Microphone Array Processing.* Springer, 2016.

[2] P. A. Naylor and N. D. Gaubitch, Eds., *Speech Dereverberation.* Springer, 2010.

Journal Articles

[1] W. Xue, A. H. Moore, M. Brookes, and P. A. Naylor, "Modulation-domain parametric multichannel Kalman filtering for speech enhancement," in *Proc. European Signal Processing Conf. (EUSIPCO)*, 2018.

[2] C. Evers, E. A. P. Habets, S. Gannot, and P. A. Naylor, "DoA reliability for distributed acoustic tracking," *IEEE Signal Process. Lett.*, vol. 25, no. 9, pp. 1320–1324, 2018.

[3] C. Evers and P. A. Naylor, "Acoustic SLAM," *IEEE Trans. Audio, Speech, Lang. Process.*, vol. 26, no. 9, pp. 1484–1498, 2018.

[4] C. Evers and P. A. Naylor, "Optimized self-localization for SLAM in dynamic scenes using probability hypothesis density filters," *IEEE Trans. Signal Process.*, vol. 66, no. 4, pp. 863 – 878, 2018.

[5] E. De Sena, M. Brookes, P. A. Naylor, and T. van Waterschoot, "Localization experiments with reporting by head orientation: statistical framework and case study," *Journal of the Audio Engineering Society*, vol. 65, no. 12, pp. 982–996, 2017.

[6] S. Hafezi, A. H. Moore, and P. A. Naylor, "Augmented intensity vectors for direction of arrival estimation in the spherical harmonic domain," *IEEE Trans. Audio, Speech, Lang. Process.*, vol. 25, no. 10, pp. 1956 – 1968, 2017.

[7] N. Antonello, E. De Sena, M. Moonen, P. A. Naylor, and T. van Waterschoot, "Room impulse response interpolation using a sparse spatio-temporal representation of the sound field," *IEEE Trans. Audio, Speech, Lang. Process.*, vol. 25, no. 10, pp. 1929 – 1941, 2017.

[8] C. S. J. Doire, M. Brookes, and P. A. Naylor, "Robust and efficient bayesian adaptive psychometric function estimation," *J. Acoust. Soc. Am.*, vol. 141, no. 4, pp. 2501–2512, Apr. 2017.

[9] C. S. J. Doire, M. Brookes, P. A. Naylor, C. M. Hicks, D. Betts, M. A. Dmour, and S. H. Jensen, "Single-channel online enhancement of speech corrupted by reverberation and noise," *IEEE Trans. Audio, Speech, Lang. Process.*, vol. 25, no. 3, pp. 572–587, Mar. 2017.

[10] A. H. Moore, P. P. Parada, and P. A. Naylor, "Speech enhancement for robust automatic speech recognition: Evaluation using a baseline system and instrumental measures," *Computer Speech & Language*, 2016.

[11] P. Peso-Parada, D. Sharma, T. van Waterschoot, and P. A. Naylor, "Confidence measures for non-intrusive estimation of speech clarity index," *Journal of the Audio Engineering Society*, vol. 65, no. 1/2, pp. 90–98, 2017.

[12] A. H. Moore, C. Evers, and P. A. Naylor, "Direction of arrival estimation in the spherical harmonic domain using subspace pseudo-intensity vectors," *IEEE Trans. Audio, Speech, Lang. Process.*, vol. 25, no. 1, pp. 178 – 192, 2017.

[13] J. Eaton, N. D. Gaubitch, A. H. Moore, and P. A. Naylor, "Estimation of room acoustic parameters: The ACE challenge," *IEEE Trans. Audio, Speech, Lang. Process.*, vol. 24, no. 10, pp. 1681 – 1693, Jun. 2016.

[14] A. Zahedi, J. Ostergaard, P. A. Naylor, and S. Bech, "Source coding in networks with covariance distortion constraints," *IEEE Trans. Signal Process.*, vol. 64, no. 22, pp. 5943–5958, 2016.

[15] P. Peso-Parada, D. Sharma, J. Lainez, D. Barreda, T. van Waterschoot, and P. A. Naylor, "A single-channel non-intrusive C50 estimator correlated with speech recognition performance," *IEEE Trans. Audio, Speech, Lang. Process.*, vol. 24, no. 4, pp. 719 – 732, Jan. 2016.

[16] D. Sharma, Y. Wang, P. A. Naylor, and M. Brookes, "A data-driven non-intrusive measure of speech quality and intelligibility," *Speech Communication*, vol. 80, pp. 84 – 94, Jun. 2016.

[17] P. Peso-Parada, D. Sharma, P. A. Naylor, and T. van Waterschoot, "Reverberant speech recognition exploiting clarity index estimation," *EURASIP Journal on Advances in Signal Processing*, 2015.

[18] A. Zahedi, J. Ostergaard, S. H. Jensen, S. Bech, and P. A. Naylor, "Audio coding in wireless acoustic sensor networks," *Signal Processing*, vol. 107, pp. 141–152, Feb. 2015.

[19] F. Lim, W. Zhang, E. A. P. Habets, and P. A. Naylor, "Robust multichannel dereverberation using relaxed multichannel least squares," *IEEE Trans. Audio, Speech, Lang. Process.*, vol. 22, no. 9, pp. 1379–1390, Sep. 2014.

[20] D. Jarrett, M. Taseska, E. A. P. Habets, and P. A. Naylor, "Noise reduction in the spherical harmonic domain using a tradeoff beamformer and narrowband DOA estimates," *IEEE Trans. Audio, Speech, Lang. Process.*, vol. 22, no. 5, pp. 965–976, May 2014.

[21] N. D. Gaubitch, M. Brookes, and P. A. Naylor, "Blind channel magnitude response estimation in speech using spectrum classification," *IEEE Trans. Audio, Speech, Lang. Process.*, vol. 21, no. 10, pp. 2162–2171, Oct. 2013.

[22] K. Kowalczyk, E. A. Habets, P. A. Naylor, and W. Kellermann, “Blind system identification using sparse learning for TDOA estimation of room reflections,” *IEEE Signal Process. Lett.*, vol. 20, no. 7, pp. 653 – 656, Jul. 2013.

[23] P. Annibale, J. Filos, P. A. Naylor, and R. Rabenstein, “TDOA-based speed of sound estimation for air temperature and room geometry inference,” *IEEE Trans. Audio, Speech, Lang. Process.*, vol. 21, no. 2, pp. 234–246, Feb. 2013.

[24] F. Antonacci, J. Filos, M. Thomas, E. A. P. Habets, A. Sarti, P. Naylor, and S. Tubaro, “Inference of room geometry from acoustic impulse responses,” *IEEE Trans. Audio, Speech, Lang. Process.*, vol. 20, no. 10, pp. 2683–2695, Dec. 2012.

[25] T. Drugman, M. R. P. Thomas, J. Gudnason, P. A. Naylor, and T. Dutoit, “Detection of glottal closure instants from speech signals: a quantitative review,” *IEEE Trans. Audio, Speech, Lang. Process.*, vol. 20, no. 3, pp. 994–1006, Mar. 2012.

[26] J. Gudnason, M. R. P. Thomas, D. P. W. Ellis, and P. A. Naylor, “Data-driven voice source waveform analysis and synthesis,” *Speech Communication*, vol. 54, no. 2, pp. 199ñ–211, Feb. 2012.

[27] E. A. P. Habets, J. Benesty, and P. A. Naylor, “A speech distortion and interference rejection constraint beamformer,” *IEEE Trans. Audio, Speech, Lang. Process.*, vol. 20, no. 3, pp. 854–867, Mar. 2012.

[28] D. Jarrett, E. Habets, M. Thomas, and P. A. Naylor, “Rigid sphere room impulse response simulation: algorithm and applications,” *J. Acoust. Soc. Am.*, vol. 132, no. 3, Sep. 2012.

[29] X. S. Lin, A. W. H. Khong, and P. A. Naylor, “A forced spectral diversity algorithm for speech dereverberation in the presence of near-common zeros,” *IEEE Trans. Audio, Speech, Lang. Process.*, vol. 20, no. 3, pp. 888–899, Mar. 2012.

[30] M. R. P. Thomas, J. Gudnason, and P. A. Naylor, “Estimation of glottal closing and opening instants in voiced speech using the YAGA algorithm,” *IEEE Trans. Audio, Speech, Lang. Process.*, vol. 20, no. 1, pp. 83–91, Jan. 2012.

[31] M. Slaney and P. A. Naylor, “Audio and acoustic signal processing,” *IEEE Signal Process. Mag.*, vol. 28, 2011.

[32] N. D. Gaubitch, E. A. P. Habets, and P. A. Naylor, “Signal-based performance evaluation of dereverberation algorithms,” *Journal of Electrical and Computer Engineering*, Oct. 2009.

[33] N. D. Gaubitch and P. A. Naylor, “Equalization of multichannel acoustic systems in oversampled subbands,” *IEEE Trans. Audio, Speech, Lang. Process.*, vol. 17, no. 6, pp. 1061 – 1070, Aug. 2009.

[34] P. Loganathan, A. W. H. Khong, and P. A. Naylor, "A class of sparseness-controlled algorithms for echo cancellation," *IEEE Trans. Audio, Speech, Lang. Process.*, vol. 17, no. 8, pp. 1591–1601, Nov. 2009.

[35] M. R. P. Thomas and P. A. Naylor, "The SIGMA algorithm: A glottal activity detector for electroglottographic signals," *IEEE Trans. Audio, Speech, Lang. Process.*, vol. 17, no. 8, pp. 1557–1566, Nov. 2009.

[36] X. Lin, A. W. H. Khong, M. Doroslovaki, and P. A. Naylor, "Frequency-domain adaptive algorithm for network echo cancellation in VoIP," *EURASIP Journal on Audio, Speech, and Music Processing*, 2008.

[37] U. Manmontri and P. A. Naylor, "A class of frobenius norm-based algorithms using penalty term and natural gradient for blind signal separation," *IEEE Trans. Audio, Speech, Lang. Process.*, vol. 16, no. 6, pp. 1181–1193, Aug. 2008.

[38] W. Zhang and P. A. Naylor, "An algorithm to generate representations of system identification errors," *Research Letters in Signal Processing*, vol. 2008, Article ID 529291, 2008.

[39] K. Dogancay and P. A. Naylor, "Adaptive partial-update and sparse system identification," *EURASIP Journal on Audio, Speech, and Music Processing*, vol. 2007, Article ID 12046, 2007.

[40] M. A. Haque, M. S. Bashar, P. A. Naylor, K. Hirose, and M. K. Hasan, "Energy constrained frequency-domain normalized LMS algorithm for blind channel identification," *Signal Image and Video Processing*, vol. 1, no. 3, pp. 203–213, 2007.

[41] A. W. H. Khong and P. A. Naylor, "Selective-tap adaptive filtering with performance analysis for identification of time-varying systems," *IEEE Trans. Audio, Speech, Lang. Process.*, vol. 15, no. 5, pp. 1681–1695, Jul. 2007.

[42] A. W. H. Khong, P. A. Naylor, and J. Benesty, "A low delay and fast converging improved proportionate algorithm for sparse system identification," *EURASIP Journal on Audio, Speech, and Music Processing*, vol. 2007, Article ID 084376, 2007.

[43] P. A. Naylor, A. Kounoudes, J. Gudnason, and D. M. Brookes, "Estimation of glottal closure instants in voiced speech using the DYPSA algorithm," *IEEE Trans. Audio, Speech, Lang. Process.*, vol. 15, no. 1, pp. 34–43, Jan. 2007.

[44] M. Brookes, P. A. Naylor, and J. Gudnason, "A quantitative assessment of group delay methods for identifying glottal closures in voiced speech," *IEEE Trans. Audio, Speech, Lang. Process.*, vol. 14, no. 2, pp. 456–466, Mar. 2006.

[45] N. D. Gaubitch, M. K. Hasan, and P. A. Naylor, "Generalized optimal step-size for blind multichannel LMS system identification," *IEEE Signal Process. Lett.*, vol. 13, no. 10, pp. 624–627, Oct. 2006.

[46] N. D. Gaubitch, P. A. Naylor, and D. B. Ward, "Statistical analysis of the autoregressive modeling of reverberant speech," *J. Acoust. Soc. Am.*, vol. 120, no. 6, pp. 4031–4039, 2006.

[47] A. W. H. Khong, J. Benesty, and P. A. Naylor, "Stereophonic acoustic echo cancellation: Analysis of the misalignment in the frequency domain," *IEEE Signal Process. Lett.*, vol. 13, no. 1, pp. 33–36, Jan. 2006.

[48] A. W. H. Khong and P. A. Naylor, "Stereophonic acoustic echo cancellation employing selective-tap adaptive algorithms," *IEEE Trans. Audio, Speech, Lang. Process.*, vol. 14, no. 3, pp. 785–796, May 2006.

[49] P. A. Naylor, J. Cui, and M. Brookes, "Adaptive algorithms for sparse echo cancellation," *Signal Processing*, vol. 86, no. 6, pp. 1182–1192, 2006.

[50] F. Talantzis, D. B. Ward, and P. A. Naylor, "Performance analysis of dynamic acoustic source separation in reverberant rooms," *IEEE Trans. Audio, Speech, Lang. Process.*, vol. 14, no. 4, pp. 1378–1390, Jul. 2006.

[51] M. K. Hasan, N. M. Hossain, and P. A. Naylor, "Autocorrelation model-based identification method for ARMA systems in noise," *IEE Proc. Vision, Image and Signal Processing*, vol. 152, no. 5, pp. 520–526, 2005.

[52] A. W. H. Khong and P. A. Naylor, "Selective-tap adaptive algorithms in the solution of the nonuniqueness problem for stereophonic acoustic echo cancellation," *IEEE Signal Process. Lett.*, vol. 12, no. 4, pp. 269–272, 2005.

[53] N. T. Forsyth, J. A. Chambers, and P. A. Naylor, "An alternating fixed-point algorithm for stereophonic acoustic echo cancellation," *IEE Proc. Vision, Image and Signal Processing*, vol. 149, no. 1, pp. 1–9, 2002.

[54] T. Hoya, N. Forsyth, J. A. Chambers, and P. A. Naylor, "A study of the epsilon-NLMS algorithm with application to stereophonic acoustic echo cancellation," *Int. Journal of Adaptive Control and Signal Processing*, vol. 14, pp. 609–621, October 2000.

[55] N. T. Forsyth, J. A. Chambers, and P. A. Naylor, "Noise robust alternating fixed-point algorithm for stereophonic acoustic echo cancellation," *IEE Electronics Letters*, vol. 35, no. 21, pp. 1812–1813, 1999.

[56] T. Hoya, Y. Loke, J. A. Chambers, and P. A. Naylor, "Application of the leaky extended LMS algorithm in stereophonic acoustic echo cancellation," *Signal Processing*, vol. 64, pp. 87–91, 1998.

[57] P. A. Naylor, O. Tanrikulu, and A. G. Constantinides, "Sub-band adaptive filtering for acoustic echo control using allpass polyphase IIR filter banks," *IEEE Trans. Speech Audio Process.*, vol. 6, no. 2, pp. 143–155, 1998.

[58] P. A. Naylor, J.Alcazar, J.Boudy, and Y.Grenier, "Enhancement of hands-free telecommunications," *Annales des Telecommunications*, vol. 49, pp. 373–379, 1994.

[59] A. Strong, P. J. Kirkpatrick, R. M. Bucknall, P. A. Naylor, and S. Rudman, "Measurement of superficial cerebral cortical blood flow by imaging umbelliferone clearance," *Journal of the Physiological Society*, vol. 446, 1991.

Conference Papers

[1] D. Sharma, A. Nour-Eldin, P. Harding, S. Karimian-Azari, and P. A. Naylor, "Robust feature extraction from ad-hoc microphones for meeting diarization," in *Proc. Intl. Workshop Acoust. Signal Enhancement (IWAENC)*, 2018.

[2] A. H. Moore, L. Lightburn, W. Xue, P. A. Naylor, and M. Brookes, "Binaural mask-informed speech enhancement for hearing aids with head tracking," in *Proc. Intl. Workshop Acoust. Signal Enhancement (IWAENC)*, 2018.

[3] S. Karimian-Azari, D. Sharma, A. Nour-Eldin, and P. A. Naylor, "Robust speaker localization in noisy environments using speech modulation properties," in *Proc. Intl. Workshop Acoust. Signal Enhancement (IWAENC)*, 2018.

[4] W. Xue, A. H. Moore, M. Brookes, and P. A. Naylor, "Modulation-domain parametric multichannel Kalman filtering for speech enhancement," in *Proc. European Signal Processing Conf. (EUSIPCO)*, 2018.

[5] P. J. Dawson, E. DeSena, and P. A. Naylor, "An acoustic image-source characterisation of surface profiles," in *Proc. European Signal Processing Conf. (EUSIPCO)*, 2018.

[6] W. Xue, A. H. Moore, M. Brookes, and P. A. Naylor, "Multichannel kalman filtering for speech enhancement," in *Proc. IEEE Intl. Conf. on Acoustics, Speech and Signal Processing (ICASSP)*, 2018.

[7] A. H. Moore, P. A. Naylor, and M. Brookes, "Room identification using frequency dependence of spectral decay statistics," in *Proc. IEEE Intl. Conf. on Acoustics, Speech and Signal Processing (ICASSP)*, 2018.

[8] N. Antonello, E. D. Sena, M. Moonen, P. A. Naylor, and T. van Waterschoot, "Joint source localization and dereverberation by sound field interpolation using sparse regularization," in *Proc. IEEE Intl. Conf. on Acoustics, Speech and Signal Processing (ICASSP)*, 2018.

[9] C. Yiallourides, A. H. Moore, E. Auvinet, C. V. D. Straeten, and P. A. Naylor, "Acoustic analysis and assessment of the knee in osteoarthritis during walking," in *Proc. IEEE Intl. Conf. on Acoustics, Speech and Signal Processing (ICASSP)*, 2018.

[10] D. Sharma, U. Jost, and P. A. Naylor, “Non-intrusive bit-rate detection of coded speech,” in *Proc. European Signal Processing Conf. (EUSIPCO)*, 2017.

[11] S. Hafezi, A. H. Moore, and P. A. Naylor, “Multiple doa estimation based on estimation consistency and spherical harmonic multiple signal classification,” in *Proc. European Signal Processing Conf. (EUSIPCO)*, 2017.

[12] P. Peso-Parada, D. Sharma, U. Jost, T. van Waterschoot, and P. A. Naylor, “Robust statistical processing of TDOA estimates for distant speaker diarization,” in *Proc. European Signal Processing Conf. (EUSIPCO)*, Kos, Greece, 2017.

[13] C. Papayiannis, C. Evers, and P. A. Naylor, “Sparse parametric modeling of the early part of acoustic impulse responses,” in *Proc. European Signal Processing Conf. (EUSIPCO)*, 2017.

[14] A. H. Moore, M. Brookes, and P. A. Naylor, “Robust spherical harmonic domain interpolation of spatially sampled array manifolds,” in *Proc. IEEE Intl. Conf. on Acoustics, Speech and Signal Processing (ICASSP)*, 2017.

[15] C. Yiallourides, V. Manning-Eid, A. H. Moore, and P. A. Naylor, “A dynamic programming approach for automatic stride detection and segmentation in acoustic emission from the knee,” in *Proc. IEEE Intl. Conf. on Acoustics, Speech and Signal Processing (ICASSP)*, 2017.

[16] S. Hafezi, A. H. Moore, and P. A. Naylor, “Multiple source localization using estimation consistency in the time-frequency domain,” in *Proc. IEEE Intl. Conf. on Acoustics, Speech and Signal Processing (ICASSP)*, 2017.

[17] L. Lightburn, E. De Sena, A. H. Moore, P. A. Naylor, and M. Brookes, “Improving the perceptual quality of ideal binary masked speech,” in *Proc. IEEE Intl. Conf. on Acoustics, Speech and Signal Processing (ICASSP)*, 2017.

[18] W. Xue, M. Brookes, and P. A. Naylor, “Frequency-domain under-modelled blind system identification based on cross power spectrum and sparsity regularization,” in *Proc. IEEE Intl. Conf. on Acoustics, Speech and Signal Processing (ICASSP)*, 2017.

[19] G. Pinero and P. A. Naylor, “Channel estimation for crosstalk cancellation in wireless acoustic networks,” in *Proc. IEEE Intl. Conf. on Acoustics, Speech and Signal Processing (ICASSP)*, 2017.

[20] C. Evers, Y. Dorfan, S. Gannot, and P. A. Naylor, “Source tracking using moving microphone arrays for robot audition,” in *Proc. IEEE Intl. Conf. on Acoustics, Speech and Signal Processing (ICASSP)*, 2017.

[21] H. Javed, B. Cauchi, S. Doclo, P. A. Naylor, and S. Goetze, “Measuring, modelling and predicting perceived reverberation,” in *Proc. IEEE Intl. Conf. on Acoustics, Speech and Signal Processing (ICASSP)*, 2017.

[22] C. Papayiannis, C. Evers, and P. A. Naylor, "Discriminative feature domains for reverberant acoustic environments," in *Proc. IEEE Intl. Conf. on Acoustics, Speech and Signal Processing (ICASSP)*, 2017.

[23] P. Peso-Parada, D. Sharma, U. Jost, T. van Waterschoot, and P. A. Naylor, "Robust statistical processing of TDOA estimates for distant speaker diarization," in *Proc. European Signal Processing Conf. (EUSIPCO)*, Kos, Greece, 2017.

[24] C. Evers, B. Rafaely, and P. A. Naylor, "Speaker tracking in reverberant environments using multiple directions of arrival," in *Proc. Hands-Free Speech Communication and Microphone Arrays*, San Francisco, USA, 2017.

[25] I. D. Gebru, C. Evers, P. A. Naylor, and R. Horaud, "Audio-visual tracking by density approximation in a sequential bayesian filtering framework," in *Proc. Hands-Free Speech Communication and Microphone Arrays*, San Francisco, USA, 2017.

[26] H. Loellmann, A. H. Moore, P. A. Naylor, B. Rafaely, R. Horaud, A. Mazel, and W. Kellermann, "Microphone array signal processing for robot audition," in *Proc. Hands-Free Speech Communication and Microphone Arrays*, San Francisco, USA, 2017.

[27] S. Hafezi, A. H. Moore, and P. A. Naylor, "Multi-source estimation consistency for improved multiple direction-of-arrival estimation," in *Proc. Hands-Free Speech Communication and Microphone Arrays*, San Francisco, USA, 2017.

[28] J. Eaton, H. Javed, and P. A. Naylor, "Estimation of the perceived level of reverberation using non-intrusive single-channel variance of decay rates," in *Proc. Hands-Free Speech Communication and Microphone Arrays*, San Francisco, USA, 2017.

[29] W. Xue, M. Brookes, and P. A. Naylor, "Under-modelled blind system identification for time delay estimation in reverberant environments," in *Proc. Intl. Workshop Acoust. Signal Enhancement (IWAENC)*, 2016.

[30] A. H. Moore and P. A.Naylor, "Linear prediction based dereverberation for spherical microphone arrays," in *Proc. Intl. Workshop Acoust. Signal Enhancement (IWAENC)*, 2016.

[31] H. Javed, A. H. Moore, and P. A. Naylor, "Spherical harmonic rake receivers for dereverberation," in *Proc. Intl. Workshop Acoust. Signal Enhancement (IWAENC)*, 2016.

[32] W. Zhang, P. A. Naylor, Z. He, and Y. Zhang, "On the evaluation of multichannel blind system identification from the viewpoint of system equalization," in *Proc. Intl. Workshop Acoust. Signal Enhancement (IWAENC)*, 2016.

[33] W. Zhang and P. A. Naylor, "An iterative method for equalization of multichannel acoustic systems robust to system identification errors," in *Proc. Intl. Workshop Acoust. Signal Enhancement (IWAENC)*, 2016.

[34] A. H. Moore, C. Evers, and P. A. Naylor, "2D direction of arrival estimation of multiple moving sources using a spherical microphone array," in *Proc. European Signal Processing Conf. (EUSIPCO)*, 2016.

[35] C. Evers, A. H. Moore, and P. A. Naylor, "Localization of moving microphone arrays from moving sound sources for robot audition," in *Proc. European Signal Processing Conf. (EUSIPCO)*, 2016.

[36] Y. Dorfan, C. Evers, S. Gannot, and P. A. Naylor, "Speaker localization with moving microphone arrays," in *Proc. European Signal Processing Conf. (EUSIPCO)*, 2016.

[37] W. Xue, M. Brookes, and P. A. Naylor, "Cross-correlation based undermodelled multichannel blind acoustic system identification with sparsity regularization," in *Proc. European Signal Processing Conf. (EUSIPCO)*, 2016.

[38] S. Hafezi, A. H. Moore, and P. A. Naylor, "Multiple source localization in the spherical harmonic domain using augmented intensity vectors based on grid search," in *Proc. European Signal Processing Conf. (EUSIPCO)*, 2016.

[39] B. Cauchi, H. Javed, T. Gerkmann, S. Doclo, S. Goetze, and P. A. Naylor, "Perceptual and instrumental evaluation of the perceived level of reverberation," in *Proc. IEEE Intl. Conf. on Acoustics, Speech and Signal Processing (ICASSP)*, 2016.

[40] H. Javed, A. H. Moore, and P. A. Naylor, "Spherical microphone array acoustic RAKE receivers," in *Proc. IEEE Intl. Conf. on Acoustics, Speech and Signal Processing (ICASSP)*, 2016.

[41] S. Hafezi, A. H. Moore, and P. A. Naylor, "3D acoustic source localization in the spherical harmonic domain based on optimized grid search," in *Proc. IEEE Intl. Conf. on Acoustics, Speech and Signal Processing (ICASSP)*, 2016.

[42] C. Evers, A. H. Moore, and P. A. Naylor, "Acoustic simultaneous localization and mapping (A-SLAM) of a moving microphone array and its surrounding speakers," in *Proc. IEEE Intl. Conf. on Acoustics, Speech and Signal Processing (ICASSP)*, 2016.

[43] J. Eaton, N. D. Gaubitch, A. H. Moore, and P. A. Naylor, "The ACE challenge ñ corpus description and performance evaluation," in *Proc. IEEE Workshop on Applications of Signal Processing to Audio and Acoustics (WASPAA)*, 2015.

[44] M. Hu, P. Peso-Parada, D. Sharma, S. Doclo, T. van Waterschoot, M. Brookes, and P. A. Naylor, "Single-channel speaker diarization based on spatial features," in *Proc. IEEE Workshop on Applications of Signal Processing to Audio and Acoustics (WASPAA)*, 2015.

[45] F. Lim, M. R. P. Thomas, P. A. Naylor, and I. J. Tashev, "Acoustic blur kernel with sliding window for blind estimation of reverberation time," in *Proc. IEEE Workshop on Applications of Signal Processing to Audio and Acoustics (WASPAA)*, 2015.

[46] B. Cauchi, P. A. Naylor, T. Gerkmann, S. Doclo, and S. Goetze, "Late reverberant spectral variance estimation using acoustic channel equalization," in *Proc. European Signal Processing Conf. (EUSIPCO)*, 2015.

[47] M. Hu, S. Doclo, D. Sharma, M. Brookes, and P. A. Naylor, "Noise robust blind system identification algorithms based on a Rayleigh quotient cost function," in *Proc. European Signal Processing Conf. (EUSIPCO)*, 2015.

[48] C. S. J. Doire, M. Brookes, P. A. Naylor, and S. H. Jensen, "Data-driven statistical modelling of room impulse responses in the power domain," in *Proc. European Signal Processing Conf. (EUSIPCO)*, 2015.

[49] A. H. Moore, C. Evers, P. A. Naylor, D. Alon, and B. Rafaely, "Direction of arrival estimation using pseudo-intensity vectors with direct-path dominance test," in *Proc. European Signal Processing Conf. (EUSIPCO)*, 2015.

[50] D. Sharma, A. Poddar, S. Manna, and P. A. Naylor, "The SAS project: Speech signal processing in high school education," in *Proc. European Signal Processing Conf. (EUSIPCO)*, 2015.

[51] H. Javed and P. A. Naylor, "An extended reverberation decay tail metric as a measure of perceived late reverberation," in *Proc. European Signal Processing Conf. (EUSIPCO)*, 2015.

[52] C. M. Nelke, P. A. Naylor, and P. Vary, "Corpus based reconstruction of speech degraded by wind noise," in *Proc. European Signal Processing Conf. (EUSIPCO)*, 2015.

[53] S. Hafezi, A. H. Moore, and P. A. Naylor, "Modelling source directivity in room impulse response simulation for spherical microphone arrays," in *Proc. European Signal Processing Conf. (EUSIPCO)*, 2015.

[54] C. Evers, A. H. Moore, P. A. Naylor, J. Sheaffer, and B. Rafaely, "Bearing-only acoustic tracking of moving speakers for robot audition," in *Proc. IEEE Intl. Conf. Digital Signal Processing (DSP)*, 2015, pp. 1206–1210.

[55] A. H. Moore, C. Evers, and P. A. Naylor, "Multichannel equalisation for high-order spherical microphone arrays using beamformed channels," in *Proc. IEEE Intl. Conf. Digital Signal Processing (DSP)*, 2015, pp. 1211–1215.

[56] V. Manning, C. Yiallourides, M. Brevadt, A. H. Moore, E. Auvinet, P. A. Naylor, and J. Cobb, "Knee sounds may predict osteoarthritis severity, symptoms and function: Pilot investigation toward a novel dynamic imaging system," in

Arthritis Rheumatol. 2015; 67 (suppl 10). http://acrabstracts.org/abstract/knee-sounds-may-predict-osteoarthritis-severity-symptoms-and-function-pilot-investigation-toward-a-novel-dynamic-imaging-system/. Accessed October 1, 2015.

[57] C. S. J. Doire, M. Brookes, P. A. Naylor, D. Betts, C. M. Hicks, M. A. Dmour, and S. H. Jensen, "Single-channel blind estimation of reverberation parameters," in *Proc. IEEE Intl. Conf. on Acoustics, Speech and Signal Processing (ICASSP)*, Apr. 2015.

[58] J. Eaton, A. H. Moore, P. A. Naylor, and J. Skoglund, "Direct-to-reverberant ratio estimation using a null-steered beamformer," in *Proc. IEEE Intl. Conf. on Acoustics, Speech and Signal Processing (ICASSP)*, Apr. 2015.

[59] M. Hu, D. Sharma, S. Doclo, M. Brookes, and P. A. Naylor, "Speaker change detection and speaker diarization using spatial information," in *Proc. IEEE Intl. Conf. on Acoustics, Speech and Signal Processing (ICASSP)*, Apr. 2015.

[60] N. Antonello, T. van Waterschoot, M. Moonen, and P. A. Naylor, "Evaluation of a numerical method for identifying surface acoustic impedances in a reverberant room," in *EuroNoise*, Maastricht, May 2015.

[61] D. Sharma, L. Meredith, J. Lainez, D. Barreda, and P. A. Naylor, "A non-intrusive PESQ measure," in *Proc. IEEE GlobalSIP*, 2014.

[62] R. Stanton, N. D. Gaubitch, P. A. Naylor, and M. Brookes, "A differentiable approximation to speech intelligibility index with applications to listening enhancement," in *Audio Engineering Society 54th International Conference: Audio Forensics*, June 2014.

[63] P. Peso-Parada, D. Sharma, P. A. Naylor, and T. van Waterschoot, "Single-channel reverberant speech recognition using C50 estimation," in *Proc. IEEE AASP REVERB Challenge*, Florence, Italy, May 2014.

[64] P. Peso-Parada, D. Sharma, P. A. Naylor, and T. van Waterschoot, "Reverberant speech recognition: A phoneme analysis," in *Proc. IEEE GlobalSIP*, 2014.

[65] M. R. P. Thomas, F. Lim, I. J. Tashev, and P. A. Naylor, "Optimal beamforming as a time domain equalization problem with application to room acoustics," in *Proc. Intl. Workshop Acoust. Signal Enhancement (IWAENC)*, 2014.

[66] N. Antonello, T. van Waterschoot, M. Moonen, and P. A. Naylor, "Identification of surface acoustic impedances in a reverberant room using the FDTD method," in *Proc. Intl. Workshop Acoust. Signal Enhancement (IWAENC)*, 2014.

[67] F. Lim and P. A. Naylor, "Statistical modelling of multichannel blind system identification errors," in *Proc. Intl. Workshop Acoust. Signal Enhancement (IWAENC)*, 2014.

[68] C. Evers, A. H. Moore, and P. A. Naylor, "Multiple source localisation in the spherical harmonic domain," in *Proc. Intl. Workshop Acoust. Signal Enhancement (IWAENC)*, 2014.

[69] P. Peso-Parada, D. Sharma, J. Lainez, D. Barreda, P. A. Naylor, and T. van Waterschoot, "A quantitative comparison of blind C50 estimators," in *Proc. Intl. Workshop Acoust. Signal Enhancement (IWAENC)*, 2014.

[70] N. Antonello, T. van Waterschoot, M. Moonen, and P. A. Naylor, "Source localization and signal reconstruction in a reverberant field using the FDTD method," in *Proc. European Signal Processing Conf. (EUSIPCO)*, 2014.

[71] A. H. Moore, P. A. Naylor, and J. Skoglund, "An analysis of the effect of larynx-synchronous averaging on dereverberation of voiced speech," in *Proc. European Signal Processing Conf. (EUSIPCO)*, 2014.

[72] M. Costa and P. A. Naylor, "ILD preservation in the multichannel wiener filter for binaural hearing aid applications," in *Proc. European Signal Processing Conf. (EUSIPCO)*, 2014.

[73] R. Borges, M. Costa, P. A. Naylor, and A. Ferriera, "Impact of the vent size in the feedback-path and occlusion-effect in hearing aids," in *Proc. Biomedical Circuits and Systems Conference*, October 2014.

[74] J. Eaton and P. A. Naylor, "Noise-robust detection of peak-clipping in decoded speech," in *Proc. IEEE Intl. Conf. on Acoustics, Speech and Signal Processing (ICASSP)*, May 2014, pp. 7069–7073.

[75] P. Peso-Parada, D. Sharma, and P. A. Naylor, "Non-intrusive estimation of the level of reverberation in speech," in *Proc. IEEE Intl. Conf. on Acoustics, Speech and Signal Processing (ICASSP)*, May 2014, pp. 4751–4755.

[76] A. Zahedi, J. Ostergaard, S. H. Jensen, P. A. Naylor, and S. Bech, "Distributed remote vector gaussian source coding with covariance distortion constraints," in *Proc. IEEE International Symposium on Information Theory*, 2014.

[77] A. Zahedi, J. Ostergaard, S. H. Jensen, P. A. Naylor, and S. Bech, "Distributed remote vector gaussian source coding for wireless acoustic sensor networks," in *Proc. IEEE Data Compression Conference*, 2014.

[78] F. Lim, M. R. P. Thomas, and P. A. Naylor, "MINTFormer: a spatially aware channel equalizer," in *Proc. IEEE Workshop on Applications of Signal Processing to Audio and Acoustics (WASPAA)*, Oct. 2013.

[79] A. H. Moore, D. M. Brookes, and P. A. Naylor, "Roomprints for forensic audio applications," in *Proc. IEEE Workshop on Applications of Signal Processing to Audio and Acoustics (WASPAA)*, Oct. 2013.

[80] F. Lim and P. A. Naylor, "Robust speech dereverberation using subband multichannel least squares with variable relaxation," in *Proc. European Signal Processing Conf. (EUSIPCO)*, 2013.

[81] J. Eaton, M. Brookes, and P. A. Naylor, "A comparison of non-intrusive SNR estimation algorithms and the use of mapping functions," in *Proc. European Signal Processing Conf. (EUSIPCO)*, 2013.

[82] J. Eaton and P. A. Naylor, "Detection of clipping in coded speech signals," in *Proc. European Signal Processing Conf. (EUSIPCO)*, 2013.

[83] D. Sharma, P. A. Naylor, and M. Brookes, "Non-intrusive speech intelligibility assessment," in *Proc. European Signal Processing Conf. (EUSIPCO)*, 2013.

[84] A. H. Moore, M. Brookes, and P. A. Naylor, "Room geometry estimation from a single channel acoustic impulse response," in *Proc. European Signal Processing Conf. (EUSIPCO)*, 2013.

[85] F. Lim and P. A. Naylor, "Robust low-complexity multichannel equalization for dereverberation," in *Proc. IEEE Intl. Conf. on Acoustics, Speech and Signal Processing (ICASSP)*, 2013.

[86] D. Jarrett, E. A. P. Habets, and P. A. Naylor, "Spherical harmonic domain noise reduction using an MVDR beamformer and DOA-based second-order statistics estimation," in *Proc. IEEE Intl. Conf. on Acoustics, Speech and Signal Processing (ICASSP)*, 2013.

[87] J. Eaton, N. D. Gaubitch, and P. A. Naylor, "Noise-robust reverberation time estimation using spectral decay distributions with reduced computational cost," in *Proc. IEEE Intl. Conf. on Acoustics, Speech and Signal Processing (ICASSP)*, 2013.

[88] P. Annibale, J. Filos, R. Rabenstein, and P. A. Naylor, "Geometric inference of the room geometry under temperature variations," in *Proc. Int. Symposium on Comms., Control and Signal Proc.*, 2012.

[89] N. D. Gaubitch, H. W. Loellmann, M. Jeub, T. H. Falk, P. A. Naylor, P. Vary, and M. Brookes, "Performance comparison of algorithms for blind reverberation time estimation from speech," in *Proc. Intl. Workshop Acoust. Signal Enhancement (IWAENC)*, Sept 2012.

[90] P. A. Naylor and N. D. Gaubitch, "Acoustic signal processing in noise: It's not getting any quieter," in *Proc. Intl. Workshop Acoust. Signal Enhancement (IWAENC)*, Sep 2012.

[91] M. R. P. Thomas, N. D. Gaubitch, E. A. P. Habets, and P. A. Naylor, "An insight into common filtering in noisy SIMO blind system identification," in *Proc. IEEE Intl. Conf. on Acoustics, Speech and Signal Processing (ICASSP)*, 2012, pp. 521–524.

[92] D. Sharma, G. Hilkhuysen, P. A. Naylor, N. D. Gaubitch, M. Huckvale, and M. Brookes, "Descriptive vocabulary development for degraded speech," in *Proc. 13th Annual Conference of the International Speech Communication Association, INTERSPEECH*, 2012.

[93] D. Sharma, P. A. Naylor, N. D. Gaubitch, and D. M. Brookes, "Non-intrusive codec identification algorithm," in *Proc. IEEE Intl. Conf. on Acoustics, Speech and Signal Processing (ICASSP)*, 2012, pp. 4477–4480.

[94] A. Canclini, F. Antonacci, J. Filos, A. Sarti, and P. A. Naylor, "Exact localization of planar acoustic reflectors in three-dimensional geometries," in *Proc. Intl. Workshop Acoust. Signal Enhancement (IWAENC)*, 2012.

[95] D. P. Jarrett, O. Thiergart, E. A. P. Habets, and P. A. Naylor, "Coherence-based diffuseness estimation in the spherical harmonic domain," in *2012 IEEE 27th Convention of Electrical and Electronics Engineers in Israel*, 2012.

[96] D. P. Jarrett, E. A. P. Habets, J. Benesty, and P. A. Naylor, "A tradeoff beamformer for noise reduction in the spherical harmonic domain," in *Proc. Intl. Workshop Acoust. Signal Enhancement (IWAENC)*, 2012.

[97] F. Lim and P. A. Naylor, "Relaxed multichannel least squares with constrained initial taps for multichannel dereverberation," in *Proc. Intl. Workshop Acoust. Signal Enhancement (IWAENC)*, 2012.

[98] J. Filos, A. Canclini, F. Antonacci, A. Sarti, and P. A. Naylor, "Localization of planar acoustic reflectors from the combination of linear estimates," in *Proc. European Signal Processing Conf. (EUSIPCO)*, 2012, pp. 1019–1023.

[99] P. Annibale, F. Antonacci, P. Bestagini, A. Brutti, A. Canclini, L. Cristoforetti, E. Habets, J. Filos, W. Kellermann, K. Kowalczyk, A. Lombard, E. Mabande, D. Markovic, P. Naylor, M. Omologo, R. Rabenstein, A. Sarti, P. Svaizer, and M. Thomas, "The SCENIC project: Space-time audio processing for environment-aware acoustic sensing and rendering," in *131st AES Convention*, October 2011.

[100] A. Canclini, F. Antonacci, M. R. P. Thomas, J. Filos, A. Sarti, P. A. Naylor, and S. Tubaro, "Exact localization of acoustic reflectors from quadratic constraints," in *Proc. IEEE Workshop on Applications of Signal Processing to Audio and Acoustics (WASPAA)*, 2011.

[101] J. Filos, A. Canclini, M. R. P. Thomas, F. Antonacci, A. Sarti, and P. A. Naylor, "Robust inference of room geometry from acoustic measurements using the Hough transform," in *Proc. European Signal Processing Conf. (EUSIPCO)*, 2011.

[102] N. D. Gaubitch, M. Brookes, P. A. Naylor, and D. Sharma, "Single-microphone blind channel identification in speech using spectrum classification," in *Proc. European Signal Processing Conf. (EUSIPCO)*, 2011.

[103] E. A. P. Habets, J. Benesty, and P. A. Naylor, “A cross-relation based affine projection algorithm for blind SIMO system identification,” in *Proc. European Signal Processing Conf. (EUSIPCO)*, 2011.

[104] D. P. Jarrett, E. A. P. Habets, M. R. P. Thomas, N. D. Gaubitch, and P. A. Naylor, “Dereverberation performance of rigid and open spherical microphone arrays: theory & simulation,” in *Proc. Hands-Free Speech Communication and Microphone Arrays*, 2011.

[105] D. P. Jarrett, M. R. Thomas, E. A. P. Habets, and P. A. Naylor, “Simulating room impulse responses for spherical microphone arrays,” in *Proc. IEEE Intl. Conf. on Acoustics, Speech and Signal Processing (ICASSP)*, May 2011.

[106] P. Loganathan, E. A. P. Habets, and P. A. Naylor, “A proportionate adaptive algorithm with variable partitioned block length for acoustic echo cancellation,” in *Proc. IEEE Intl. Conf. on Acoustics, Speech and Signal Processing (ICASSP)*, May 2011.

[107] D. Sharma, P. A. Naylor, N. D. Gaubitch, and M. Brookes, “Short-time objective assessment of speech quality,” in *Proc. European Signal Processing Conf. (EUSIPCO)*, 2011.

[108] M. R. Thomas, N. Gaubitch, and P. A. Naylor, “Application of channel shortening to acoustic channel equalization in the presence of noise and estimation error,” in *Proc. IEEE Workshop on Applications of Signal Processing to Audio and Acoustics (WASPAA)*, 2011.

[109] N. D. Gaubitch, M. Brookes, P. A. Naylor, and D. Sharma, “Bayesian adaptive method for estimating speech intelligibility in noise,” in *39th International Conference: Audio Forensics: Practices and Challenges*, 2010, pp. 7–1.

[110] B. Castro, N. D. Gaubitch, E. A. P. Habets, S. Gannot, P. A. Naylor, and S. Grant, “Subband scale factor ambiguity correction using multiple filterbanks,” in *Proc. Intl. Workshop Acoust. Echo Noise Control (IWAENC)*, 2010.

[111] J. Filos, E. A. P. Habets, and P. A. Naylor, “A two-step approach to blindly infer room geometries,” in *Proc. Intl. Workshop Acoust. Echo Noise Control (IWAENC)*, 2010.

[112] E. Habets and P. A. Naylor, “An online quasi-newton algorithm for blind SIMO identification,” in *Proc. IEEE Intl. Conf. on Acoustics, Speech and Signal Processing (ICASSP)*, 2010.

[113] D. P. Jarrett, E. A. P. Habets, and P. A. Naylor, “3D source localization in the spherical harmonic domain using a pseudointensity vector,” in *Proc. European Signal Processing Conf. (EUSIPCO)*, August 2010.

[114] D. P. Jarrett, E. A. P. Habets, and P. A. Naylor, "Eigenbeam-based acoustic source tracking in noisy reverberant environments," in *Proc. Asilomar Conference on Signals, Systems and Computers*, 2010.

[115] P. Loganathan, E. A. P. Habets, and P. A. Naylor, "A partitioned block proportionate adaptive algorithm for acoustic echo cancellation," in *Proc. Asia-Pacific Signal and Information Processing Association Conference*, December 2010.

[116] P. Loganathan, E. A. P. Habets, and P. A. Naylor, "Performance analysis of IPNLMS for identification of time-varying systems," in *Proc. IEEE Intl. Conf. on Acoustics, Speech and Signal Processing (ICASSP)*, 2010.

[117] P. A. Naylor, C. Evers, and E. A. P. Habets, "Speech dereverberation," in *Tutorial presented at European Signal Processing Conference*, 2010.

[118] R. Rashobh, A. W. H. Khong, and P. A. Naylor, "Adaptive blind system identification for speech dereverberation using a priori estimates," in *Proc. IEEE Asia Pacific Conference on Circuits and Systems*, December 2010.

[119] D. Sharma, G. Hilkhuysen, N. D. Gaubitch, and P. A. Naylor, "Data driven method for non-intrusive speech intelligibility estimation," in *Proc. European Signal Processing Conf. (EUSIPCO)*, 2010.

[120] M. R. P. Thomas, N. D. Gaubitch, E. A. P. Habets, and P. A. Naylor, "Supervised identification and removal of common filter components in adaptive blind SIMO system identification," in *Proc. Intl. Workshop Acoust. Echo Noise Control (IWAENC)*, 2010.

[121] M. R. P. Thomas, J. Gudnason, P. A. Naylor, B. Geiser, and P. Vary, "Voice source estimation for artificial bandwidth extension of telephone speech," in *Proc. IEEE Intl. Conf. on Acoustics, Speech and Signal Processing (ICASSP)*, 2010.

[122] M. C. Tsakiris, C. G. Lopes, and P. A. Naylor, "An alternative criterion for regularization in recursive least-squares problems," in *Proc. Seventh International Symposium on Wireless Communication Systems*, York, UK, Sep 2010.

[123] W. Zhang, E. A. P. Habets, and P. A. Naylor, "A system identification-error-robust method for equalization of multichannel acoustic systems," in *Proc. IEEE Intl. Conf. on Acoustics, Speech and Signal Processing (ICASSP)*, 2010.

[124] W. Zhang, E. A. P. Habets, and P. A. Naylor, "On the use of channel shortening in multichannel acoustic system equalization," in *Proc. Intl. Workshop Acoust. Echo Noise Control (IWAENC)*, 2010.

[125] N. D. Gaubitch, M. Brookes, and P. A. Naylor, "Blind channel identification in speech using the long-term average speech spectrum," in *Proc. European Signal Processing Conf. (EUSIPCO)*, 2009.

[126] J. Gudnason, M. R. P. Thomas, P. A. Naylor, and D. P. W. Ellis, "Voice source waveform analysis and synthesis using principal component analysis and gaussian mixture modelling," in *Proc. 10th Annual Conference of the International Speech Communication Association, INTERSPEECH*, 2009, pp. 120–123.

[127] E. A. P. Habets, J. Benesty, S. Gannot, P. A. Naylor, and I. Cohen, "On the application of the LCMV beamformer to speech enhancement," in *Proc. IEEE Workshop on Applications of Signal Processing to Audio and Acoustics (WASPAA)*, Oct. 2009, pp. 141–144.

[128] X. Lin, A. W. H. Khong, and P. A. Naylor, "Blind system identification for speech dereverberation with forced spectral diversity," in *Proc. IEEE Intl. Conf. on Acoustics, Speech and Signal Processing (ICASSP)*, 2009, pp. 3737–3740.

[129] P. Loganathan, X. S. Lin, A. W. Khong, and P. A. Naylor, "Frequency-domain adaptive multidelay algorithm with sparseness control for acoustic echo cancellation," in *Proc. European Signal Processing Conf. (EUSIPCO)*, 2009.

[130] D. Sharma and P. A. Naylor, "Evaluation of pitch estimation in noisy speech for application in non-intrusive speech quality assessment," in *Proc. European Signal Processing Conf. (EUSIPCO)*, 2009.

[131] M. R. P. Thomas, J. Gudnason, and P. A. Naylor, "Data-driven voice soruce waveform modelling," in *Proc. IEEE Intl. Conf. on Acoustics, Speech and Signal Processing (ICASSP)*, 2009, pp. 3965–3968.

[132] M. R. P. Thomas, J. Gudnason, and P. A. Naylor, "Detection of glottal closing and opening instants using an improved DYPSA framework," in *Proc. European Signal Processing Conf. (EUSIPCO)*, 2009.

[133] M. C. Tsakiris and P. A. Naylor, "Fast exact affine projection algorithm using displacement structure theory," in *Proc. 16th International Conference on Digital Signal Processing*, 2009, pp. 1–6.

[134] J. Y. C. Wen, A. Sehr, P. A. Naylor, and W. Kellermann, "Blind estimation of a feature-domain reverberation model in non-diffuse environments with variance adjustment," in *Proc. European Signal Processing Conf. (EUSIPCO)*, 2009.

[135] W. Zhang, A. W. H. Khong, and P. A. Naylor, "Acoustic system equalization using channel shortening techniques for speech dereverberation," in *Proc. European Signal Processing Conf. (EUSIPCO)*, 2009, pp. 1427–1431.

[136] W. Zhang and P. A. Naylor, "An experimental study of the robustness of multichannel inverse filtering systems to near-common zeros," in *Proc. European Signal Processing Conf. (EUSIPCO)*, 2009.

[137] N. D. Gaubitch, E. Habets, and P. A. Naylor, “Multimicrophone speech dereverberation using spatiotemporal and spectral processing,” in *Proc. IEEE Int. Symp. on Circuits and Systems*, Apr 2008, pp. 3222–3225.

[138] N. D. Gaubitch, X. Lin, and P. A. Naylor, “Scale factor ambiguity correction for subband blind multichannel identification,” in *Proc. Intl. Workshop Acoust. Echo Noise Control (IWAENC)*, 2008.

[139] E. Habets, N. D. Gaubitch, and P. A. Naylor, “Temporal selective dereverberation of noisy speech using one microphone,” in *Proc. IEEE Intl. Conf. on Acoustics, Speech and Signal Processing (ICASSP)*, Jan 2008, pp. 4577–4580.

[140] A. W. H. Khong, W.-S. Gan, P. A. Naylor, and D. M. Brookes, “A low complexity fast converging partial update adaptive algorithm employing variable step-size for acoustic echo cancellation,” in *Proc. IEEE Intl. Conf. on Acoustics, Speech and Signal Processing (ICASSP)*, Jan 2008, pp. 237–240.

[141] A. W. H. Khong, X. Lin, M. Doroslovacki, and P. A. Naylor, “Frequency domain selective tap adaptive algorithms for sparse system identification,” in *Proc. IEEE Intl. Conf. on Acoustics, Speech and Signal Processing (ICASSP)*, Jan 2008, pp. 229–232.

[142] A. W. H. Khong, X. Lin, and P. A. Naylor, “Algorithms for identifying clusters of near-common zeros in multichannel blind system identification and equalization,” in *Proc. IEEE Intl. Conf. on Acoustics, Speech and Signal Processing (ICASSP)*, Jan 2008, pp. 389–392.

[143] P. Loganathan, W. H. A. Khong, and P. A. Naylor, “A sparseness controlled proportionate algorithm for acoustic echo cancellation,” in *Proc. European Signal Processing Conf. (EUSIPCO)*, 2008.

[144] P. A. Naylor, X. Lin, and A. W. H. Khong, “Near-common zeros in blind identification of SIMO acoustic systems,” in *Proc. Hands-Free Speech Communication and Microphone Arrays*, Apr 2008, pp. 21–24.

[145] A. Sehr, Y.-C. J. Wen, W. Kellermann, and P. A. Naylor, “A combined approach for estimating a feature-domain reverberation model in non-diffuse environments,” in *Proc. Intl. Workshop Acoust. Echo Noise Control (IWAENC)*, 2008.

[146] M. R. P. Thomas, J. Gudnason, and P. A. Naylor, “Application of the DYPSA algorithm to segmented time scale modification of speech,” in *Proc. European Signal Processing Conf. (EUSIPCO)*, 2008.

[147] M. R. P. Thomas and P. A. Naylor, “The SIGMA algorithm for estimation of reference-quality glottal closure instants from electroglottograph signals,” in *Proc. European Signal Processing Conf. (EUSIPCO)*, 2008.

[148] Y.-C. J. Wen, E. Habets, and P. A. Naylor, "Blind estimation of reverberation time based on the distribution of signal decay rates," in *Proc. IEEE Intl. Conf. on Acoustics, Speech and Signal Processing (ICASSP)*, Jan 2008, pp. 329–332.

[149] W. Zhang, N. D. Gaubitch, and P. A. Naylor, "Computationally efficient equalization of room impulse responses robust to system estimation errors," in *Proc. IEEE Intl. Conf. on Acoustics, Speech and Signal Processing (ICASSP)*, Jan 2008, pp. 4025–4028.

[150] W. Zhang, A. W. H. Khong, and P. A. Naylor, "Adaptive inverse filtering of room acoustics," in *Proc. 42nd Asilomar Conference on Signals, Systems and Computers*, 2008, pp. 26–29.

[151] R. Ahmad, N. D. Gaubitch, and P. A. Naylor, "A noise-robust dual filter approach to multichannel blind system identification," in *Proc. European Signal Processing Conf. (EUSIPCO)*, 2007, pp. 385–388.

[152] R. Ahmad, A. W. H. Khong, and P. A. Naylor, "A practical adaptive blind multichannel estimation algorithm with application to acoustic impulse responses," in *Proc. Int. Conf. on Digital Signal Processing*, Jun 2007, pp. 31–34.

[153] N. D. Gaubitch and P. A. Naylor, "Spatiotemporal averaging method for enhancement of reverberant speech," in *Proc. Int. Conf. on Digital Signal Processing*, Jun 2007, pp. 607–610.

[154] N. D. Gaubitch, M. R. Thomas, and P. A. Naylor, "Subband method for multichannel least squares equalization of room transfer functions," in *Proc. IEEE Workshop on Applications of Signal Processing to Audio and Acoustics (WASPAA)*, Sep 2007, pp. 14–17.

[155] X. Lin, N. D. Gaubitch, and P. A. Naylor, "Blind speech dereverberation in the presence of common acoustical zeros," in *Proc. European Signal Processing Conf. (EUSIPCO)*, 2007, pp. 389–393.

[156] H. Maqsood, J. Gudnason, and P. A. Naylor, "Enhanced robustness to unvoiced speech and noise in the DYPSA algorithm for identification of glottal closure instants," in *Proc. European Signal Processing Conf. (EUSIPCO)*, 2007.

[157] H. Maqsood and P. Naylor, "Improved DYPSA algorithm for noise and unvoiced speech," in *Proc. International Conference on Emerging Technologies*, Oct 2007, pp. 243–248.

[158] P. A. Naylor, A. W. H. Khong, and D. M. Brookes, "Misalignment performance of selective tap adaptive algorithms for system identification of time-varying unknown systems," in *Proc. IEEE Intl. Conf. on Acoustics, Speech and Signal Processing (ICASSP)*, vol. 1, Mar 2007, pp. I–97 – I–100.

[159] M. R. Thomas, N. Gaubitch, J. Gudnason, and P. A. Naylor, "A practical multichannel dereverberation algorithm using multichannel DYPSA and spatiotemporal averaging," in *Proc. IEEE Workshop on Applications of Signal Processing to Audio and Acoustics (WASPAA)*, Sep 2007, pp. 50–53.

[160] M. R. P. Thomas, N. D. Gaubitch, and P. A. Naylor, "Multichannel DYPSA for estimation of glottal closure instants in reverberant speech," in *Proc. European Signal Processing Conf. (EUSIPCO)*, 2007.

[161] J. Wen and P. A. Naylor, "Objective measurement of colouration in reverberation," in *Proc. European Signal Processing Conf. (EUSIPCO)*, 2007.

[162] J. Y. Wen and P. Naylor, "Semantic colouration space investigation: Controlled colouration in the bark-sone domain," in *Proc. IEEE Workshop on Applications of Signal Processing to Audio and Acoustics (WASPAA)*, Sep 2007, pp. 311–314.

[163] R. Ahmad, A. W. Khong, M. K. Hasan, and P. A. Naylor, "An extended normalized multichannel FLMS algorithm for blind channel identification," in *Proc. European Signal Processing Conf. (EUSIPCO)*, 2006.

[164] R. Ahmad, A. W. H. Khong, and P. A. Naylor, "Proportionate frequency domain adaptive algorithms for blind channel identification," in *Proc. IEEE Intl. Conf. on Acoustics, Speech and Signal Processing (ICASSP)*, 2006, pp. V29–V32.

[165] V. Bhunjun, M. Brookes, and P. A. Naylor, "Model-based eigen-spectrum estimation for speech enhancement," in *Proc. 40th Asilomar Conference on Signals, Systems and Computers*, 2006, pp. 1331–1334.

[166] N. D. Gaubitch, M. K. Hasan, and P. A. Naylor, "Noise robust adaptive blind channel identification using spectral constraints," in *Proc. IEEE Intl. Conf. on Acoustics, Speech and Signal Processing (ICASSP)*, 2006, pp. V93 – V96.

[167] N. D. Gaubitch and P. A. Naylor, "The complex multichannel LMS algorithm for adaptive blind system identification," in *Proc. Intl. Workshop Acoust. Echo Noise Control (IWAENC)*, 2006.

[168] M. K. Hasan and P. A. Naylor, "Effect of noise on blind adaptive multichannel identification algorithms: Robustness issue," in *Proc. European Signal Processing Conf. (EUSIPCO)*, 2006.

[169] S. Javidi, N. D. Gaubitch, and P. A. Naylor, "An experimental study of the eigen-decomposition methods for blind SIMO system identification in the presence of noise," in *Proc. Intl. Workshop Acoust. Echo Noise Control (IWAENC)*, 2006.

[170] A. W. H. Khong, J. Benesty, and P. A. Naylor, "Effect of interchannel coherence on conditioning and misalignment performance for stereo acoustic echo

cancellation," in *Proc. IEEE Intl. Conf. on Acoustics, Speech and Signal Processing (ICASSP)*, 2006, pp. V265 – V268.

[171] A. W. H. Khong and P. A. Naylor, "Efficient use of sparse adaptive filters," in *Proc. 40th Asilomar Conference on Signals, Systems and Computers*, 2006, pp. 1375–1379.

[172] X. Lin, N. D. Gaubitch, and P. A. Naylor, "Two-stage blind identification of SIMO systems with common zeros," in *Proc. European Signal Processing Conf. (EUSIPCO)*, 2006.

[173] U. Manmontri and P. A. Naylor, "Blind signal separation using a criterion based on principle of minimal disturbance," in *Proc. IEEE Intl. Conf. on Acoustics, Speech and Signal Processing (ICASSP)*, 2006, pp. V829–V832.

[174] L. Mintandjian and P. A. Naylor, "A study of synchronous convergence in μ-law PNLMS for voice over IP," in *Proc. European Signal Processing Conf. (EUSIPCO)*, 2006.

[175] Y.-C. J. Wen, N. D. Gaubitch, E. Habets, T. Myatt, and P. A. Naylor, "Evaluation of speech dereverberation algorithms using the MARDY database," in *Proc. Intl. Workshop Acoust. Echo Noise Control (IWAENC)*, 2006.

[176] Y.-C. J. Wen and P. A. Naylor, "An evaluation measure for reverberant speech using tail decay modelling," in *Proc. European Signal Processing Conf. (EUSIPCO)*, 2006.

[177] K. Dogancay and P. A. Naylor, "Recent advances in partial update and sparse adaptive filters," in *Proc. European Signal Processing Conf. (EUSIPCO)*, 2005.

[178] N. D. Gaubitch, J. Benesty, and P. A. Naylor, "Adaptive common root estimation and the common zeros problem in blind channel identification," in *Proc. European Signal Processing Conf. (EUSIPCO)*, 2005.

[179] N. D. Gaubitch and P. A. Naylor, "Analysis of the dereverberation performance of microphone arrays," in *Proc. Intl. Workshop Acoust. Echo Noise Control (IWAENC)*, 2005.

[180] M. K. Hasan, J. Benesty, P. A. Naylor, and D. B. Ward, "Improving robustness of blind adaptive multichannel identification algorithms using constraints," in *Proc. European Signal Processing Conf. (EUSIPCO)*, 2005.

[181] A. W. H. Khong, J. Benesty, and P. A. Naylor, "An improved proportionate multi-delay block adaptive filter for packet-switched network echo cancellation," in *Proc. European Signal Processing Conf. (EUSIPCO)*, 2005.

[182] A. W. H. Khong and P. A. Naylor, "Frequency domain adaptive algorithms for stereophonic acoustic echo cancellation employing tap selection," in *Proc. Intl. Workshop Acoust. Echo Noise Control (IWAENC)*, 2005.

[183] A. W. H. Khong and P. A. Naylor, “A family of selective-tap algorithms for stereo acoustic echo cancellation,” in *Proc. IEEE Intl. Conf. on Acoustics, Speech and Signal Processing (ICASSP)*, 2005, pp. iii/133 – iii/136.

[184] U. Manmontri and P. A. Naylor, “Blind identification using second-order statistics: a nonstationarity and nonwhiteness approach,” in *Proc. IEEE Intl. Conf. on Acoustics, Speech and Signal Processing (ICASSP)*, 2005, pp. v/305 – v/308.

[185] P. A. Naylor and N. D. Gaubitch, “Speech dereverberation,” in *Proc. Intl. Workshop Acoust. Echo Noise Control (IWAENC)*, 2005.

[186] A. R. Wright and P. A. Naylor, “Blind IQ mismatch compensation in OFDM direct conversion receivers,” in *Proc. IEE/EURASIP Conf. on DSP Enabled Radio*, 2005.

[187] J. Cui, P. A. Naylor, and D. T. Brown, “An improved IPNLMS algorithm for echo cancellation in packet-switched networks,” in *Proc. IEEE Intl. Conf. on Acoustics, Speech and Signal Processing (ICASSP)*, vol. 4, 2004, pp. iv–141 – iv–144.

[188] N. D. Gaubitch, P. A. Naylor, and D. B. Ward, “Multi-microphone speech dereverberation via spatio-temporal averaging,” in *Proc. European Signal Processing Conf. (EUSIPCO)*, 2004, pp. 809–812.

[189] A. W. H. Khong and P. A. Naylor, “Reducing inter-channel coherence in stereophonic acoustic echo cancellation using partial update adaptive filters,” in *Proc. European Signal Processing Conf. (EUSIPCO)*, 2004, pp. 405–408.

[190] A. W. H. Khong and P. A. Naylor, “Affine projection and recursive least squares adaptive filters employing partial updates,” in *Proc. Asilomar Conference on Signals, Systems and Computers*, 2004, pp. 950–954.

[191] F. Talantzis, D. B. Ward, and P. A. Naylor, “Expected performance of a family of blind source separation algorithms in a reverberant room,” in *Proc. IEEE Intl. Conf. on Acoustics, Speech and Signal Processing (ICASSP)*, vol. 4, May 2004, pp. VI 61–64.

[192] N. Gaubitch, P. A. Naylor, and D. B. Ward, “On the use of linear prediction for dereverberation of speech,” in *Proc. Intl. Workshop Acoust. Echo Noise Control (IWAENC)*, 2003, pp. 99–102.

[193] A. W. H. Khong and P. A. Naylor, “The use of partial update schemes to reduce inter-channel coherence in adaptive stereophonic acoustic echo cancellation,” in *Proc. Intl. Workshop Acoust. Echo Noise Control (IWAENC)*, 2003, pp. 59–62.

[194] P. A. Naylor and W. Sherliker, “A short-sort M-Max NLMS partial-update adaptive filter with applications to echo cancellation,” in *Proc. IEEE Intl.*

Conf. on Acoustics, Speech and Signal Processing (ICASSP), 2003, pp. 373–376.

[195] A. R. Wright and P. A. Naylor, "I/Q mismatch compensation in zero-IF OFDM receivers with applications to DAB," in *Proc. IEEE Intl. Conf. on Acoustics, Speech and Signal Processing (ICASSP)*, 2003, pp. II–329–332.

[196] A. Kounoudes, P. A. Naylor, and M. Brookes, "The DYPSA algorithm for estimation of glottal closure instants in voiced speech," in *Proc. IEEE Intl. Conf. on Acoustics, Speech and Signal Processing (ICASSP)*, vol. 1, Orlando, 2002, pp. 349–352.

[197] A. Kounoudes, P. A. Naylor, and M. Brookes, "Automatic epoch extraction for closed-phase analysis of speech," in *Proc 14th International Conference on Digital Signal Processing*, vol. 2, Santorini, 2002, pp. 979–983.

[198] A. Oakman and P. A. Naylor, "Dynamic structures for non-uniform subband adaptive filters," in *Proc. IEEE Intl. Conf. on Acoustics, Speech and Signal Processing (ICASSP)*, 2001, pp. 3717–3720.

[199] N. Forsyth, N. Tangsanguimvisai, P. A. Naylor, and J. A. Chambers, "A subband signal decorrelation scheme for stereophonic acoustic echo cancellation using higher-order time-varying allpass filters," in *Proc. Intl. Workshop Acoust. Echo Noise Control (IWAENC)*, 1999.

[200] T. Hoya, J. A. Chambers, and P. A. Naylor, "Low complexity epsilon-NLMS algorithm and subband structures for stereophonic acoustic echo control," in *Proc. Intl. Workshop Acoust. Echo Noise Control (IWAENC)*, 1999.

[201] T. Hoya, J. A. Chambers, N. Forsyth, and P. A. Naylor, "Steady-state solutions of the extended LMS algorithm for stereophonic acoustic echo cancellation," in *Proc. European Signal Processing Conf. (EUSIPCO)*, 1998, pp. 977–980.

[202] A. Neocleous and P. A. Naylor, "Voice source parameters for speaker verification," in *Proc. European Signal Processing Conf. (EUSIPCO)*, 1998, pp. 697–700.

[203] N. Doukas, P. A. Naylor, and T. Stathaki, "Voice activity detection using source separation techniques," in *Proc. European Signal Processing Conf. (EUSIPCO)*, 1997.

[204] N. Doukas, T. Stathaki, and P. A. Naylor, "Stability of a voice activity detector based on source separation," in *Proc. IEEE Conference on DSP*, 1997, pp. 749–754.

[205] R. J. Wilson, P. A. Naylor, and D. M. Brookes, "Performance limitations in subband acoustic echo controllers," in *Proc. Intl. Workshop Acoust. Echo Noise Control (IWAENC)*, 1997, pp. 176–179.

[206] N. Doukas, P. A. Naylor, and T. Stathaki, "A single sensor source separation approach to noise reduction," in *Proc. Computational Engineering in System Applications Conference*, 1996.

[207] N. Doukas, T. Stathaki, and P. A. Naylor, "Speech enhancement through nonlinear adaptive source separation methods," in *Proc. IEEE Workshop Statistical Signal and Array Processing*, 1996, pp. 297–282.

[208] S. Heidar-Zadeh and P. A. Naylor, "F0 downtrends," in *Proc. ICSP*, 1996.

[209] P. A. Naylor and J. E. Hart, "Subband acoustic echo control using non-critical frequency sampling," in *Proc. European Signal Processing Conf. (EUSIPCO)*, 1996.

[210] O. Tanrikulu, B. Baykal, J. A. Chambers, A. G. Constantinides, and P. A. Naylor, "Finite-precision design and implementation of all-pass polyphase networks for echo cancellation in subbands," in *Proc. IEEE Intl. Conf. on Acoustics, Speech and Signal Processing (ICASSP)*, 1995, pp. 3039–3042.

[211] P. A. Naylor, J. Alcazar, J. Boudy, and Y. Grenier, "Enhancement of handsfree telecommunications," in *Proc. Int. Conf. on Sig. Proc. Applications and Techonology*, vol. 49, no. 7-8, Dallas, Tx, 1994, pp. 142–147.

[212] M. A. Sayid and P. A. Naylor, "A novel nonlinearity measure," in *Proc. IEE Int. Conf. on Control*, 1994, pp. 1368–1373.

[213] J. E. Hart, P. A.Naylor, and O.Tanrikulu, "Polyphase allpass IIR structures for sub-band acoustic echo cancellation," in *Proc. European Signal Processing Conf. (EUSIPCO)*, 1993, pp. 1813–1816.

[214] P. A. Naylor, "Enhancement of hands-free telecommunications - activities of the FREETEL consortium," in *Proc. 3rd Int. Workshop on Acoustic Echo Control*, Plestin les GrËves, France, 1993.

[215] D. M. Brookes and P. A. Naylor, "Speech production modelling with variable glottal reflection coefficient," in *Proc. IEEE Intl. Conf. on Acoustics, Speech and Signal Processing (ICASSP)*, 1988, pp. 671–674.

Book Chapters

[1] J. Benesty, Y. Huang, J. Chen, and P. A. Naylor, "Adaptive Algorithms for the Identification of Sparse Impulse Responses," in *Selected Methods for Acoustic Echo and Noise Control,* E. Hansler and G. Schmidt, Eds. Elsevier, 2006, ch. 5.

[2] P. A. Naylor and A. W. H. Khong, "Selective-Tap Adaptive Algorithms for Echo Cancellation," in *Selected Methods for Acoustic Echo and Noise Control,* E. Hansler and G. Schmidt, Eds. Elsevier, 2006, ch. 6.

[3] P. A. Naylor, ”Introduction to Audio Signal Processing,” in *Library in Signal Processing*. Academic Press, 2013, ch. 26.

[4] P. A. Naylor, ”Introduction to Acoustic Signal Processing,” in *Library in Signal Processing*. Academic Press, 2013, ch. 29.

[5] P. A. Naylor, ”Dereverberation,” in *Library in Signal Processing*. Academic Press, 2013, ch. 31.

PATENTS

[1] D. Sharma, P. A. Naylor, and P. Peso-Parada, “Method for non-intrusive acoustic parameter estimation,” International Patent WO2 015 034 633, 2014.

[2] D. J. Eaton, A. H. Moore, P. A. Naylor, and J. Skoglund, “Reverberation estimator,” US Patent US14 521 104, Oct, 2014.

[3] J. Lainez, D. A. Barreda, D. Sharma, P. A. Naylor, and S. Pilli, “System and method for compressed domain estimation of the signal to noise ratio of a coded speech signal,” US Patent US14 322 369, Jul, 2014.

[4] S. Pilli, J. Lainez, D. Sharma, D. A. Barreda, P. A. Naylor, and M. Godavarti, “System and method to reduce transmission bandwidth via improved discontinuous transmission,” US Patent US14 447 773, 2014.

[5] D. A. Barreda, J. E. Lainez, D. Sharma, and P. A. Naylor, “Voice activity detection (VAD) for a coded speech bitstream without decoding,” US Patent US14 094 025, Dec, 2013.

[6] D. Sharma and P. A. Naylor, “Method for voicemail quality detection,” US Patent US14 019 860, Sep, 2013.

[7] P. A. Naylor, “Providing a plurality of audio files with consistent loudness levels but different audio characteristics,” International Patent WO/1910/005 823, 2010. [Online]. Available: http://www.wipo.int/pctdb/en/wo.jsp?WO=2010005823