Patrick A. Naylor

CURRICULUM VITAE, January 16, 2019

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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

2018-	Digital Filters and Signal Processing. MSc and MEng (part IV)
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

2019-2021	President, European Association for Signal Processing
2018	(President Elect, European Association for Signal Processing)
2015-2016	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 Review Panel

2011 Government Chief Scientific Advisor's Committee on Speech Technology

2011-2014 Chair IEEE AASP Technical Subcommittee for 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 Tutorials

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 International 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 multimedia 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 - Patrick A. Naylor

January 16, 2019

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.
.

Book Chapters

[1] E. A. P. Habets and P. A. Naylor, ”Dereverberation,” in *Audio Source Separation and Speech Enhancement,* E. Vincent, T. Virtanen, S. Gannot, Eds. Wiley, 2018, ch. 15.

[2] 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.

[3] 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.

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

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

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

Journal Articles

[1] A. H. Moore, W. Xue, P. A. Naylor, and M. Brookes, “Noise covariance matrix estimation for rotating microphone arrays,” *IEEE Trans. Audio, Speech, Lang. Process.*, vol. 27, no. 3, pp. 519–530, Mar. 2018.

[2] W. Xue, A. H. Moore, M. Brookes, and P. A. Naylor, “Modulation-domain multichannel Kalman filtering for speech enhancement,” *IEEE Trans. Audio, Speech, Lang. Process.*, vol. 26, no. 10, pp. 1833–1847, 2018.

[3] 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.

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

[5] 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.

[6] 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.

[7] 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.

[8] 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.

[9] 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.

[10] 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.

[11] 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*, 2017.

[12] 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.

[13] 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.

[14] 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.

[15] 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.

[16] 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.

[17] 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.

[18] 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.

[19] 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.

[20] 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.

[21] 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.

[22] 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.

[23] 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.

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