Patrick A. Naylor

Curriculum Vitae, February 15, 2012

Imperial College London
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Current position

Reader in Speech and Audio Signal Processing, Imperial College London

Areas of specialization

Signal processing for speech, acoustics and audio

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-	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

Professional membership

Senior Member of IEEE
Fellow of IET
Chartered Engineer

Education

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

Awards

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, 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)

External Committees

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

Conference Chairs

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), 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, London, UK.

Conference Tutorial

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

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.

2005 Special Session Chair, 'Partial Update and Sparse Adaptive Systems', European Signal Processing Conference (EUSIPCO-2005), Antalya, Turkey.

Editorships

2008- Associate Editor IEEE Transactions in Audio Speech and Language Processing

2004-2008 Associate Editor IEEE Signal Processing Letters

2011- 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 and Plenary Talks

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

Expert’s Summary: “Trends in Audio and Acoustic Signal Processing”, at *IEEE Int. Conf. on Acoustics, Speech and Signal Processing*, Prague, May 2011.

Opening Keynote: “Intelligibility Estimation in Law Enforcement Speech Processing”, at *ITG Speech Communication Conference*, University Bochum-Ruhr, Germany, Oct 2010.

Plenary Address: “Speech Dereverberation”, at *International Workshop on Acoustic Echo and Noise Control*, Netherlands, Sep 2005.

Invited Talks

“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

Mar 2011 “Lightweight Noise Protection System”, QinetiQ.

Feb 2010	"SpotForming", Marie Curie Intra-European Fellowships, Dr Emanuel Habets, EURO 182K.
Jan 2009	"Self-Configuring ENvironment-aware Intelligent aCoustic sensing - SCENIC", EC 7th Framework Programme, 3 years, joint with Polytechnic of Milan (IT), Fondatione Bruno Kessler (IT), University of Erlanger-Nurnburg (DE), EURO 1.6M.
Sep 2008	"Speech Analysis for High Performance Phoneme Recognition", Royal Academy of Engineering Global Research Award, Dr Jon Gudnason with Columbia University USA, GBP 39K.
Jan 2008	"Voicemail Conversion System", Royal Academy of Engineering Industrial Secondment, 6 months, GBP 18K.
Oct 2007	"Centre for Law Enforcement Audio Research", Home Office, 5 years, joint with Mike Brookes and Mark Huckvale (UCL), GBP 1.2M.
Oct 2006	"Speech to Text Conversion", SpinVox Ltd, 2 years, GBP 160K.
May 2004	"Enhancement of Reverberant Speech for Telecommunications Application", EPSRC, 3 years, joint with Darren Ward, GBP 220K.
Sep 2003	"Multisensor Acoustic Data Fusion for Enhanced Voice-based Human-Machine Interfaces", DTC Data and Information Fusion, 3 years, GBP 55K.
Nov 2002	"Echo Cancellation for Packet-switched Networks", Trinity Convergence Inc., 1 year, GBP 85K.
Feb 2002	"Identification and Exploitation of Sparseness in Adaptive Systems", EPSRC travel award for visit to Bell Labs, Murray Hill NJ, USA, GBP 5K.
Mar 2001	"Zero IF Architectures for DAB Receivers", Panasonic System LSI Design Europe, 2 years, joint with Mike Brookes, Peter Cheung, Alison Burdett, GBP 240K.
Jun 2000	"Novel Features for Speaker Verification", EPSRC, 3 years, joint with Mike Brookes, GBP 144K.
Jun 1997	"Multi-channel speech enhancement with applications to conferencing and multimedia systems", EPSRC, 3 years, joint with Jonathon Chambers, GBP 181K.
Mar 1997	"Acoustic Echo Control", LG Semicon Ltd, 2 years, joint with Mike Brookes, GBP 110K.
Jan 1992	"Enhancement of Hands-free Communications - FREETEL", EU ESPRIT, 2 years.

PhD Students Graduated

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"
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"
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"

Oct 2002 - Dec 2006 Nikolay D. Gaubitch, "Blind Identification of Acoustic Systems and Enhancement of Reverberant Speech". *First destination:* Postdoc Imperial College.

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"

Oct 1998 - Feb 2002 Amere Oakman, "Dynamic Nonuniform Filter Banks for Subband Adaptive Filtering"

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"

Consultancies

2011 - 2012 Expert witness, London.
2008 - Nuance inc., signal processing for SpinVox voicemail-to-text.
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] P. A. Naylor and N. D. Gaubitch, Eds., *Speech Dereverberation.* Springer, 2010.

Journal Articles

[1] 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 Language Proc.*, vol. 20, no. 3, pp. 888–899, March 2012.

[2] E. A. P. Habets, J. Benesty, and P. A. Naylor, "A speech distortion and interference rejection constraint beamformer," *IEEE Trans. Audio Speech Language Proc.*, vol. 20, no. 3, pp. 854–867, 2012.

[3] 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 Language Proc.*, vol. 20, no. 1, pp. 83–91, Jan 2012.

[4] 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. 199 –211, February 2012.

[5] 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 Language Proc.*, vol. 20, no. 3, pp. 994–1006, March 2012.

[6] M. Slaney and P. A. Naylor, "Audio and acoustic signal processing," *IEEE Signal Processing Magazine*, vol. 28, 2011.

[7] N. D. Gaubitch and P. A. Naylor, "Equalization of multichannel acoustic systems in oversampled subbands," *IEEE Trans. Audio Speech Language Proc.*, vol. 17, no. 6, pp. 1061 – 1070, Aug. 2009.

[8] 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.

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

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

[11] 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.

[12] 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, and Language Processing*, vol. 16, no. 6, pp. 1181–1193, Aug 2008.

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

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

[15] 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.

[16] 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, and Language Processing*, vol. 15, no. 5, pp. 1681–1695, Jul 2007.

[17] 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*, 2007.

[18] 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, and Language Processing*, vol. 15, no. 1, pp. 34–43, Jan 2007.

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

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

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

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

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

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

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

[26] 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.

[27] 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 Processing Letters*, vol. 12, no. 4, pp. 269–272, 2005.

[28] 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.

[29] 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.

[30] 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.

[31] 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.

[32] 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 Processing*, vol. 6, no. 2, pp. 143–155, 1998.

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

[34] 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] 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*, 2011.

[2] 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*, 2011.

[3] 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 Conference*, 2011.

[4] 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 Conference*, 2011.

[5] D. Sharma, P. A. Naylor, N. D. Gaubitch, and Mi, "Short-time objective assessment of speech quality," in *Proc. European Signal Processing Conference*, 2011.

[6] 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 Conference*, 2011.

[7] 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.

[8] 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.

[9] 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 Int. Conf. on Acoustics, Speech and Signal Processing*, May 2011.

[10] 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 Int. Conf. on Acoustics, Speech and Signal Processing*, May 2011.

[11] 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.

[12] 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. International Workshop on Acoustic Echo and Noise Control*, 2010.

[13] J. Filos, E. A. P. Habets, and P. A. Naylor, "A two-step approach to blindly infer room geometries," in *Proc. International Workshop on Acoustic Echo and Noise Control*, 2010.

[14] E. Habets and P. A. Naylor, "An online quasi-newton algorithm for blind SIMO identification," in *Proc. IEEE Int. Conf. on Acoustics, Speech and Signal Processing*, 2010.

[15] 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 Conference*, August 2010.

[16] 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.

[17] P. Loganathan, E. Habets, and P. A. Naylor, "Performance analysis of IPNLMS for identification of time-varying system," in *Proc. IEEE Int. Conf. on Acoustics, Speech and Signal Processing*, 2010.

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

[19] 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.

[20] 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 Conference*, 2010.

[21] 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. International Workshop on Acoustic Echo and Noise Control (IWAENC)*, 2010.

[22] 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 Int. Conf. on Acoustics, Speech and Signal Processing*, 2010.

[23] 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.

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

[25] 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 Int. Conf. on Acoustics, Speech and Signal Processing*, 2010.

[26] 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 Conference*, 2009.

[27] 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.

[28] 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*, Oct. 2009, pp. 141–144.

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

[30] 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 Conference*, 2009.

[31] 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 Conference*, 2009.

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

[33] 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 Conference*, 2009.

[34] 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.

[35] 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 Conference*, 2009.

[36] 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 Conference*, 2009, pp. 1427–1431.

[37] 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 Conference*, 2009.

[38] 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.

[39] 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.

[40] N. D. Gaubitch, X. Lin, and P. A. Naylor, "Scale factor ambiguity correction for subband blind multichannel identification," in *Proc. International Workshop on Acoustic Echo and Noise Control*, 2008.

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