Tomohiro NAKATANI

Senior Research Scientist (Supervisor)
NTT Communication Science Laboratories
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Visiting Associate Professor (2008.4.1∼)
Graduate School of Information Science
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Education

- B.E., Department of Precision Engineering, Kyoto University, Japan, 1989, thesis title: parameter adaptation methods for a learning control system, Supervisor: Professor Yutaka Yamamoto
- M.E., Division of Applied System Science, Kyoto University, Japan, 1991, thesis title: composite neural networks, Supervisor: Professor Yutaka Yamamoto
- PhD., Department of Applied Analysis and Complex Dynamical Systems, Kyoto University, Japan, March, 2002, dissertation title: computational auditory scene analysis based on residue-driven architecture and its application to mixed speech recognition, Supervisor: Professor Yutaka Yamamoto

Professional Experience

2009 - Present	Senior Research Scientist (Supervisor), NTT Communication Science Laboratories, NTT Corporation
2008 - Present	Visiting Associate Professor, Nagoya University
2001 - 2009	Senior Research Scientist, NTT Communication Science Laboratories, NTT Corporation
2005 - 2006	Visiting Scholar, Georgia Institute of Technology
1998 - 2001	Technical staff member, Corporate Business Headquarters, NTT-east Corporation
1998 - 1998	Technical staff member, Multimedia Business Department, NTT Corporation
1991 - 1998	Researcher, NTT Basic Research Laboratory, NTT Corporation

Research Interests

1. Speech enhancement and computational auditory scene analysis
   - Dereverberation, denoising, and source separation with statistical signal processing approach
2. Robust automatic speech recognition (ASR)
   - Integration of speech enhancement, ASR, and speech synthesis
3. Other robust speech feature extraction
   - Fundamental frequency and voiced/unvoiced segments estimation

Awards

- 1997 JSAI Conference Best Paper Award
- 2002 ASJ Poster Award
- 2005 IEICE Paper Awards
- 2011 ASJ Technical Development Award
- 2012 Japan Audio Society Award

Academic Activities

Apr. 2009 - Present	Member of Audio and Acoustics Technical Committee, IEEE Signal Processing Society (May 2013 - Present: Chair of Review Subcommittee)
Jan. 2009 - Present	Secretary (Jan. 2009-Dec. 2010), Chair (Jan. 2011-Dec. 2012), and Past-Chair (Jan. 2013-Present) of IEEE Kansai Section Technical Program Committee
Nov. 2013 - Present	Chair of REVERB (REverberant Voice Enhancement and Recognition Benchmark) Challenge Workshop
Dec. 2011 - Present	Member of REVERB Challenge Organizing Committee
Mar. 2008 - Mar. 2010	Associate Editor for IEEE Trans. Audio, Speech, and Language Processing
May 2007 - May 2009	Member of Blind Signal Processing Technical Committee, IEEE Circuits and Systems Society
Mar. 2012	Presenter of a tutorial "Reverberant Speech Processing for Human Communication and Automatic Speech Recognition" for IEEE International Conference on Audio, Speech, and Signal Processing (ICASSP 2012)
Sep. 2010	Organizer of a Special Issue "Processing reverberant speech: methodologies and applications" for IEEE Trans. Audio, Speech, and Language Processing
Apr. 2009	Organizer of a Special Session "Handling reverberant speech: methodologies and applications" for IEEE International Conference on Audio, Speech, and Signal Processing (ICASSP 2009)
May 2008	Member of Organizing Committee, IEEE workshop on Computer Based Signal Processing
Oct. 2007	Technical Program Co-Chair of IEEE Workshop on Applications of Signal Processing to Audio and Acoustics (WASPAA 2007)
June 2007	Special Session Organizer for IEEE International Symposium on Circuits and Systems (ISCAS 2007)

- Senior member of the Institute of Electrical and Electronics Engineers, Inc (IEEE)
- Member of the Institute of Electronics, Information and Communication Engineers (IEICE)
- Member of the Acoustical Society of Japan (ASJ)

Publications

- **Journal Papers (in English)**

1) Mehrez Souden, Kinoshita Kinoshita, Marc Delcroix and Tomohiro Nakatani, "Location feature integration for clustering-based speech separation in distributed microphone arrays", *IEEE Trans. on Speech, Audio and Language Processing,* vol. 22, no. 2, pp. 354–367, Feb., 2014.

2) Takuya Yoshioka and Tomohiro Nakatani, "Noise model transfer: novel approach to robustness against nonstationary noise", *IEEE Transactions on Audio, Speech, and Language Processing,* vol. 21, no. 10, pp. 2182–2192, Oct., 2013.

3) Tomohiro Nakatani, Shoko Araki, Takuya Yoshioka, Marc Delcroix, and Masakiyo Fujimoto, "Dominance based integration of spatial and spectral features for speech enhancement", *IEEE Transactions on Audio, Speech, and Language Processing,* vol. 21, no. 12, pp. 2516–2531, Dec. 2013.

4) Mehrez Souden, Shoko Araki, Keisuke Kinoshita, Tomohiro Nakatani, and Hiroshi Sawada, "A Multichannel MMSE-based framework for speech source separation and noise reduction", *IEEE Transactions on Audio Speech and Language Processing,* vol. 21, no. 9, pp. 1913–1928, Sep, 2013.

5) Mehrez Souden, Kinoshita Kinoshita and Tomohiro Nakatani, "Towards online maximum likelihood speech clustering and separation", *Journal of Acoust. Soc. America (JASA) Express letter,* vol. 133, no. 5, pp. EL339-EL345, 2013.

6) Marc Delcroix, Shinji Watanabe, Tomohiro Nakatani, and Atsushi Nakamura, "Cluster-based dynamic variance adaptation for interconnecting speech enhancement pre-processor and speech recognizer", *Computer Speech and Language, Elsevier,* vol. 27, no. 3, pp. 851–873, 2013.

7) Marc Delcroix, Keisuke Kinoshita, Tomohiro Nakatani, Shoko Araki, Atsunori Ogawa, Takaaki Hori, Shinji Watanabe, Masakiyo Fujimoto, Takuya Yoshioka, Takanobu Oba, Yotaro Kubo, Mehrez Souden, Seong-Jun Hahm, and Atsushi Nakamura, "Speech recognition in living rooms: Integrated speech enhancement and recognition system based on spatial, spectral & temporal modeling of sounds", *Computer Speech and Language, Elsevier,* vol. 27, no. 3, pp. 851–873, 2013.

8) Takuya Yoshioka, Armin Sehr, Marc Delcroix, Keisuke Kinoshita, Roland Maas, Tomohiro Nakatani, and Walter Kellermann, "Making machines understand us in reverberant rooms: robustness against reverberation for automatic speech recognition", *IEEE Signal Processing Magazine,* vol. 29, no. 6, pp. 114–126, Nov., 2012.

9) Takuya Yoshioka and Tomohiro Nakatani, "Generalization of multi-channel linear prediction methods for blind MIMO impulse response shortening", *IEEE Transactions on Audio, Speech, and Language Processing,* vol. 20, no. 10, pp. 2707–2720, 2012.

10) Mehrez Souden, Marc Delcroix, Keisuke Kinoshita, Takuya Yoshioka, and Tomohiro Nakatani, "Noise power spectral density tracking: a maximum likelihood perspective", *IEEE Signal Processing Letters,* vol. 19, no. 8, pp. 495-498, Aug., 2012.

11) Takaaki Hori, Shoko Araki, Takuya Yoshioka, Masakiyo Fujimoto, Shinji Watanabe, Takanobu Oba, Atsunori Ogawa, Kazuhiro Otsuka, Dan Mikami, Keisuke Kinoshita, Tomohiro Nakatani, Atsushi Nakamura, and Junji Yamato, "Low-latency Real-time Meeting Recognition and Understanding Using Distant Microphones and Omni-directional Camera", *IEEE Trans. Audio, Speech, and Language Processing,* vol. 20, no. 2, pp. 499–513, 2012.

12) Katsuhiko Ishiguro, Takeshi Yamada, Shoko Araki, Tomohiro Nakatani, and Hiroshi Sawada, "Probabilistic Speaker Diarization with Bag-of-words Representations of Speaker Angle Information", *IEEE Trans. Audio, Speech, and Language Processing,* vol. 20, no. 2, pp. 447–460, 2012.

13) Masakiyo Fujimoto, Shinji Watanabe, Tomohiro Nakatani, "Frame-wise model re-estimation method based on Gaussian pruning with weight normalization for noise robust voice activity detection", *Speech Communication,* vol. 54, pp. 229–244, 2012.

14) Shoko Araki, Tomohiro Nakatani, and Hiroshi Sawada, "Sparse source separation based on simultaneous clustering of source locational and spectral features", *Acoustical Science and Technology, Acoustic Letter,* vol. 32 (2011), no. 4, pp. 161–164, 2011.

15) Takuya Yoshioka, Tomohiro Nakatani, Masato Miyoshi, Hiroshi Okuno, "Blind separation and dereverberation of speech mixtures by joint optimization", *IEEE Trans. Audio, Speech, and Language Processing,* vol. 19, no. 1, pp. 69–84, Jan. 2011.

16) Tomohiro Nakatani, Takuya Yoshioka, Keisuke Kinoshita, Masato Miyoshi, Biing-Hwang Juang, "Speech dereverberation based on variance-normalized delayed linear prediction", *IEEE Trans. Audio, Speech, and Language Processing,* vo. 17, no. 7, pp. 1717–1731, Sep. 2010.

17) Kentaro Ishizuka, Tomohiro Nakatani, Masakiyo Fujimoto, Noboru Miyazaki, "Noise robust voice activity detection based on periodic to aperiodic component ratio", *Speech communication,* vol.52, no.1, pp.41-60, 2010.

18) Shigeaki Amano, Tadahisa Kondo, Kazumi Kato, Tomohiro Nakatani, "Development of Japanese infant speech database from longitudinal recordings", *Speech Communication,* vol 51, no. 6, pp. 510–520, Jun. 2009.

19) Keisuke Kinoshita, Marc Delcroix, Tomohiro Nakatani, Masato Miyoshi, "Suppression of late reverberation effect on speech signal using long-term multiple-step linear prediction", *IEEE Trans. Audio, Speech, and Language Processing,* vol.17, no.4, pp.534-545, 2009.

20) Marc Delcroix, Tomohiro Nakatani, Shinji Watanabe, "Static and dynamic variance compensation for recognition of reverberant speech with dereverberation pre-processing", *IEEE Trans. Audio, Speech, and Language Processing,* vol. 17, no. 2, pp. 324–334, 2009.

21) Takuya Yoshioka, Tomohiro Nakatani, Masato Miyoshi, "Integrated Speech Enhancement Method using Noise Suppression and Dereverberation", *IEEE Trans. Audio, Speech, and Language Processing,* vol. 17, no. 2, pp. 231–246, 2009.

22) Tomohiro Nakatani, Biing-Hwang Juang, Takuya Yoshioka, Keisuke Kinoshita, Marc Delcroix, Masato Miyoshi, "Speech dereverberation based on maximum likelihood estimation with time-varying Gaussian source model", *IEEE Trans. Audio, Speech, and Language Processing,* vol. 16, no. 8, pp. 1512–1527, 2008.

23) Tomohiro Nakatani, Shigeaki Amano, Toshio Irino, Kentaro Ishizuka and Tadahisa Kondo, "A method for fundamental frequency estimation and voicing decision: application to infant utterances recorded in real acoustical environments", *Speech Communication,* vol. 50, no. 3, pp. 203–214, 2008.

24) Hiroko Kato, M. Taniguchi, Tomohiro Nakatani, and Shigeaki Amano, "Classification and similarity analysis of fundamental frequency patterns in infant spoken language acquisition", *Journal of Statistical Methodology,* vol. 5, no. 3, pp. 187–208, May 2008.

25) Keisuke Kinoshita, Tomohiro Nakatani, and Masato Miyoshi, "Fast estimation of a precise dereverberation filter based on the harmonic structure of speech", *Acoustical Science and Technology,* vol. 28, no. 2, pp. 105–114, 2007.

26) Tomohiro Nakatani, Keisuke Kinoshita, and Masato Miyoshi, "Harmonicity based blind dereverberation for single channel speech signals", *IEEE Trans. Audio, Speech, and Language Processing,* vol. 15, no. 1, pp. 80–95, 2007.

27) Kentaro Ishizuka, Tomohiro Nakatani, Yasuhiro Minami, and Noboru Miyazaki, "Speech feature extraction method using subband-based periodicity and non-periodicity decomposition", *Journal of the Acoustical Society of America,* vol. 120, Issue 1, pp. 443–452, 2006.

28) Kentaro Ishizuka and Tomohiro Nakatani, "A feature extraction method using subband based periodicity and aperiodicity decomposition with noise robust frontend processing for automatic speech recognition", *Speech Communication,* vol. 48, no. 11, pp. 1447–1457, 2006.

29) Tomohiro Nakatani, Keisuke Kinoshita, and Masato Miyoshi, "Blind dereverberation of monaural speech signals based on harmonic structure", *Systems and Computers in Japan,* vol. 37, Issue 6, pp. 1–12, June 2006.

30) Shigeaki Amano, Tomohiro Nakatani, and Tadahisa Kondo, "Fundamental frequency of infants' and parents' utterances in longitudinal recordings", *Journal of the Acoustical Society of America (JASA),* 119(3), pp. 1636–1647, Mar. 2006.

31) Keisuke Kinoshita, Tomohiro Nakatani, and Masato Miyoshi, "Harmonicity based dereverberation for improving automatic speech recognition performance and speech intelligibility", *IEICE Trans. on Fundamentals of Electronics, Communications and Computer Sciences,* E88-A, no. 7, pp. 1724–1731, 2005.

32) Tomohiro Nakatani and Toshio Irino, "Robust and accurate fundamental frequency estimation based on dominant harmonic components", *Journal of the Acoustical Society of America (JASA),* vol. 116, Issue 6, pp. 3690–3700, Dec., 2004.

33) Tomohiro Nakatani and Hiroshi G. Okuno, "Harmonic Sound Stream Segregation Using Localization and Its Application to Speech Stream Segregation", *Speech Communication,* vol. 27, nos.3-4, pp. 209–222, Elsevier, Apr., 1999.

34) Hiroshi G. Okuno, Tomohiro Nakatani, and Takeshi Kawabata, "Listening to Two Simultaneous Speeches", *Speech Communication,* vol. 27, nos.3-4, pp. 299–310, Elsevier, 1999.

- **Journal papers (in Japanese)**

1) Nobutaka Ito, Shoko Araki, Keisuke Kinoshita, and Tomohiro Nakatani, “Permutation-free clutering method for underdetermined blind source separation based on source location information”, *IEICE Trans. Fundamentals,* vol. J97-A, no. 4, pp. 1–13, Apr., 2014.

2) Yumi Ansai, Shook Araki, Shoji Makino, Tomohiro Nakatani, Takeshi Yamada, Atsushi Nakamura, Nobuhiko Kitawaki, “Cepstral smoothing of separated signals for underdetermined speech separation”, *Acoustical Science and Technology,* vol. 68, no. 2, pp. 74–85, 2011.

3) Keisuke Kinoshita, Takuya Yoshioka, Tomohiro Nakatani, “Recent advances in blind speech dereverberation”, *IEICE Fundamentals Review,* Vol. 4 No. 4 pp. 301–310, Apr. 2011.

4) Tomohiro Nakatani, Takuya Yoshioka, Keisuke Kinoshita, Masato Miyoshi, “Audio signal dereverberation based on time-varying Gaussian source model and multi-channel autoregressive observation model (in Japanese)”, *IEICE Trans. Fundamentals (Invited paper),* vol. J92-A, no. 5, 2009.

5) Tomohiro Nakatani, Masato Miyoshi, Keisuke Kinoshita, “Blind dereverberation of monaural speech signals based on harmonic structure (in Japanese)”, *IEICE Trans. Information and Systems,* vol. J88-D-II, no. 3, pp. 509–520, Mar., 2005.

6) Tomohiro Nakatani and Hiroshi G. Okuno, “Sound ontology based integration of computational auditory scene analysis systems (in Japanese)”, *Journal of Japanese Society for Artificial Intelligence,* Vol. 14, No. 6, pp. 1072–1079, 1999.

7) Tomohiro Nakatani, Masataka Goto, Takeshi Kawabata, and Hiroshi G. Okuno, “Residue-Driven Architecture and Its Application to Sound Stream Segregation (in Japanese)”, *Journal of Japanese Society for Artificial Intelligence,* vol. 12, no. 1, pp. 111–120, Jan. 1997.

8) Hiroshi G. Okuno, Tomohiro Nakatani, and Takeshi Kawabata, “Speech stream segregation and preliminary results on listening to several things simultaneously (in Japanese)”, *Transactions of Information Processing Society of Japan,* vol. 38, no. 3, pp. 510–523, Mar. 1997.

9) Tomohiro Nakatani, Hiroshi G. Okuno, and Takeshi Kawabata, “Multi-Agent Based Sound Stream Segregation for Auditory Scene Analysis (in Japanese)”, *Journal of Japanese Society for Artificial Intelligence,* vol. 10, no. 2, pp. 232–241, Mar. 1995.

10) Tomohiro Nakatani, Yutaka Yamamoto, and Yutaka Matsumoto, “On composite neural networks (in Japanese)”, *Transactions of the Institute of System, Control and Information Engineers (ISCIE),* vol. 5, no. 9, pp. 349–356, 1992.

- **International Conferences (peer reviewed)**

1) Marc Delcroix, Takuya Yoshioka, Atsunori Ogawa, Yotaro Kubo, Masakiyo Fujimoto, Nobutaka Ito, Keisuke Kinoshita, Miquel Espi, Takaaki Hori, Tomohiro Nakatani, and Atsushi Nakamura, “Linear Prediction-Based Dereverberation with Advanced Speech Enhancement and Recognition Technologies for the REVERB Challenge”, *Proc. REVERB challenge workshop,* May 2014.

2) Miquel Espi, Masakiyo Fujimoto, Yotaro Kubo, and Tomohiro Nakatani, “Spectrogram patch based acoustic event detection and classification in speech overlapping conditions”, *Proc. of Hands-free speech communication and microphone array (HSCMA),* May, 2014.

3) Masakiyo Fujimoto, Yotaro Kubo, and Tomohiro Nakatani, “Unsupervised non-parametric Bayesian modeling of non-stationary noise for model-based noise suppression”, *Proc. of International Conference on Acoustics, Speech, and Signal Processing (IEEE ICASSP),* May, 2014.

4) Atsunorir Ogawa, Keisuke Kinoshita, Takaaki Hori, Tomohiro Nakatani and Atsushi Nakamura, “Fast segment search for corpus-based speech enhancement based on speech recognition technology”, *Proc. IEEE International Conference on Acoustics, Speech, and Signal Processing (IEEE ICASSP),* May, 2014.

5) Nobutaka Ito, Shoko Araki, and Tomohiro Nakatani, “Probabilistic Integration of Diffuse Noise Suppression and Dereverberation”, *Proc. IEEE International Conference on Acoustics, Speech, and Signal Processing (IEEE ICASSP),* May 2014.

6) Keisuke Kinoshita, Tomohiro Nakatani, “Microphone-location dependent mask estimation for BSS using spatially distributed asynchronous microphones”, *2013 International Symposium on Intelligent*

Signal Processing and Communications Systems (ISPACS), pp. 326-331, Nov., 2013.

7) Keisuke Kinoshita, Marc Delcroix, Takuya Yoshioka, Tomohiro Nakatani, Emanuel Habets, Reinhold Haeb-Umbach, Volker. Leutnant, Armin Sehr, Walter Kellermann, Roland Maas, Sharon Gannot, Bhiksha. Raj, "The REVERB Challenge: A Common Evaluation Framework for Dereverberation and Recognition of Reverberant Speech", *2013 Workshop on Applications of Signal Processing to Audio and Acoustics (WASPAA),* 2013.

8) Tomohiro Nakatani, Marc Delcroix and Masakiyo Fujimoto, "Speech enhancement in a car using spatial and spectral models for speaker and noise", *Proc. of The 6th Biennial Workshop on Digital Signal Processing for In-Vehicle Systems,* Sep., 2013.

9) Ingrid Jafari, Nobutaka Ito, Mehrez Souden, Shoko Araki, and Tomohiro Nakatani, "Source Number Estimation Based on Clustering of Speech Activity Sequences for Microphone Array Processing", *Proc. IEEE International Workshop on Machine Learning for Signal Processing (IEEE MLSP),* pp. 1–6, Southampton, UK, Sep. 2013.

10) Takuya Yoshioka and Tomohiro Nakatani, "Dereverberation for reverberation-robust microphone arrays", *Proc. 21th European Signal Processing Conference (EUSIPCO 2013),* Sep. 2013.

11) Yasufumi Uezu, Keisuke Kinoshita, Mehrez Souden and Tomohiro Nakatani, "On the robustness of distributed EM based BSS in asynchronous distributed microphone array scenarios", *Proc. Interspeech,* pp. 3298–3302, Aug., 2013.

12) Keisuke Kinoshita, Mehrez Souden, Tomohiro Nakatani, "Blind source separation using spatially distributed microphones based on microphone-location dependent source activities", *Proc. Interspeech,* pp. 822–826, Aug., 2013.

13) Armin Sehr, Takuya Yoshioka, Marc Delcroix, Keisuke Kinoshita, Tomohiro Nakatani, Roland Maas and Walter Kellermann, "Conditional emission densities for interconnecting speech enhancement and recognition systems", *Proc. Interspeech,* pp. 3502–3506, Aug., 2013.

14) Keisuke Kinoshita, Mehrez Souden, Tomohiro Nakatani, "Blind source separation using spatially distributed microphones based on microphone-location dependent source activities", *Proc. Interspeech,* pp. 822–826, Aug., 2013.

15) Marc Delcroix, Yotaro Kubo, Tomoiro Nakatani and Atsushi Nakamura, "Is speech enhancement pre-processing still relevant when using deep neural networks for acoustic modeling?", *Proc. Interspeech,* pp. 2992–2996, Aug., 2013.

16) Roland Maas, Walter Kellermann, Armin Sehr, Takuya Yoshioka, Marc Delcroix, Keisuke Kinoshita, Tomohiro Nakatani, "Formulation of the REMOS concept from an uncertainty decoding perspective", *Proc. International conference on digital signal processing,* pp. 1–6, July, 2013.

17) Tomohiro Nakatani, Mehrez Souden, Shoko Araki, Takuya Yoshioka, Takaaki Hori, and Atsuhori Ogawa, "Coupling beamforming with spatial and spectral feature based spectral enhancement and its application to meeting recognition", *Proc. the 2013 IEEE International Conference on Acoustics, Speech, and Signal Processing (ICASSP 2013),* pp. 7249–7253, May 2013.

18) Marc Delcroix, Atsunori Ogawa, Seong-Jun Hahm, Tomohiro Nakatani and Atsushi Nakamura, "Unsupervised discriminative adaptation using differenced maximum mutual information based linear regression", *Proc. the 2013 IEEE International Conference on Acoustics, Speech, and Signal Processing (ICASSP'13),* pp. 7888–7892, May, 2013.

19) Nobutaka Ito, Shoko Araki, and Tomohiro Nakatani, "Permutation-free convolutive blind source separation via full-band clustering based on frequency-independent source presence priors", *Proc. the 2013 IEEE International Conference on Acoustics, Speech, and Signal Processing (ICASSP),* pp. 3238–3242, May 2013.

20) Takuya Yoshioka and Tomohiro Nakatani, "Noise model transfer using affine transformation with application to large vocabulary reverberant speech recognition", *Proc. the 2013 IEEE International Conference on Acoustics, Speech, and Signal Processing (ICASSP),* pp. 7058–7062, May 2013.

21) Mehrez Souden, Keisuke Kinoshita and Tomohiro Nakatani, "An integration of source location cues for speech clustering in distributed microphone arrays", *Proc. the 2013 IEEE International Conference on Acoustics, Speech, and Signal Processing (ICASSP),* pp. 111-115, May 2013.

22) Mehrez Souden, Keisuke Kinoshita, Marc Delcroix and Tomohiro Nakatani, “Distributed microphone array processing for speech source separation with classifier fusion”, *Proc. the 2012 IEEE International Workshop on Machine Learning for Signal Processing (MLSP),* Sept. 2012.

23) Keisuke Kinoshita, Marc Delcroix, Mehrez Souden, and Tomohiro Nakatani, “Example-based speech enhancement with joint utilization of spatial, spectral & temporal cues of speech and noise”, *Proc. Interspeech,* 2012.

24) Marc Delcroix, Atsunori Ogawa, Tomohiro Nakatani and Atsushi Nakamura, “Dynamic variance adaptation using differenced maximum mutual information”, *Proc. Symposium on Machine Learning in Speech and Language Processing (MLSLP),* 2012.

25) Takuya Yoshioka, Armin Sehr, Marc Delcroix, Keisuke Kinoshita, Roland Maas, Tomohiro Nakatani and Walter Kellermann, “Survey on approaches to speech recognition in reverberant environments”, *invited in Proc. the Conference of the Asia-Pacific Signal and Information Processing Association,* 2012.

26) Masakiyo Fujimoto and Tomohiro Nakatani, “A reliable data selection for model-based noise suppression using unsupervised joint speaker adaptation and noise model estimation”, *invited in Proc. IEEE International Conference on Signal Processing Communications and Computing (ICSPCC),* pp. 4713-4716, Aug. 2012.

27) Marc Delcroix, Atsunori Ogawa, Shinji Watanabe, Tomohiro Nakatani, and Atsushi Nakamura, “Discriminative feature transforms using differenced maximum mutual information”, *Proc. the 2012 IEEE International Conference on Acoustics, Speech, and Signal Processing (ICASSP),* pp. 4753–4756, 2012.

28) Mehrez Souden, Shoko Araki, Keisuke Kinoshita, Tomohiro Nakatani, and Hiroshi Sawada, “A multichannel MMSE-based framework for joint blind source separation and noise reduction”, *Proc. the 2012 IEEE International Conference on Acoustics, Speech, and Signal Processing (ICASSP),* pp. 109–112, 2012.

29) Shoko Araki and Tomohiro Nakatani, “Sparse vector factorization for underdetermined BSS using wrapped-phase GMM and source log-spectral prior”, *Proc. the 2012 IEEE International Conference on Acoustics, Speech, and Signal Processing (ICASSP),* pp. 265–268, 2012.

30) Takuro Maruyama, Shoko Araki, Tomohiro Nakatani, Shigeki Miyabe, Takeshi Yamada, Shoji Makino, and Atsushi Nakamura, “New analytical update rule for TDOA inference for underdetermined BSS in noisy environments”, *Proc. the 2012 IEEE International Conference on Acoustics, Speech, and Signal Processing (ICASSP),* pp. 269–272, 2012.

31) Tomohiro Nakatani, Takuya Yoshioka, Shoko Araki, Marc Delcroix, and Masakiyo Fujimoto, “Log-max observation model with MFCC-based spectral prior for reduction of highly nonstationary ambient noise”, *Proc. the 2012 IEEE International Conference on Acoustics, Speech, and Signal Processing (ICASSP),* pp. 4029–4033, 2012.

32) Takuya Yoshioka, and Tomohiro Nakatani, “Time-varying residual noise feature model estimation for multi-microphone speech recognition”, *Proc. the 2012 IEEE International Conference on Acoustics, Speech, and Signal Processing (ICASSP),* pp. 4913–4916, 2012.

33) Masakiyo Fujimoto, Shinji Watanabe, and Tomohiro Nakatani, “Noise suppression with unsupervised joint speaker adaptation and noise mixture model estimation”, *Proc. the 2012 IEEE International Conference on Acoustics, Speech, and Signal Processing (ICASSP),* pp. 4713–4716, Mar., 2012.

34) Yasuaki Iwata and Tomohiro Nakatani, “Introduction of speech log-spectral priors into dereverberation based on Itakura-Saito distance minimization”, *Proc. the 2012 IEEE International Conference on Acoustics, Speech, and Signal Processing (ICASSP),* pp. 245–248, 2012.

35) Marc Delcroix, Keisuke Kinoshita, Tomohiro Nakatani, Shoko Araki, Atsunori Ogawa, Takaaki Hori, Shinji Watanabe, Masakiyo Fujimoto, Takuya Yoshioka, Takanobu Oba, Yotaro Kubo, Mehrez Souden, Seong-Jun Hahm, Atsushi Nakamura, “Speech recognition in the presence of highly nonstationary noise based on spatial, spectral and temporal speech/noise modeling combined with dynamic variance adaptation”, *Proc. of International Workshop on Machine Listening in Multisource Environments (CHiME-2011),* pp. 12–17, Sep. 2011.

36) Tomohiro Nakatani, Shoko Araki, Marc Delcroix, Takuya Yoshioka, and Masakiyo Fujimoto, "Reduction of highly nonstationary ambient noise by integrating spectral and locational characteristics of speech and noise for robust ASR", *Interspeech-2011,* pp. 1785–1788, 2011.

37) Keisuke Kinoshita, Mehrez Souden, Marc Delcroix, and Tomohiro Nakatani, "Single channel dereverberation using example-based speech enhancement with uncertainty decoding technique", *Interspeech-2011,* pp. 197–200, 2011.

38) Mehrez Souden, Marc Delcroix, Keisuke Kinoshita, and Tomohiro Nakatani, "A multichannel feature-based processing for robust speech recognition", *Interspeech-2011,* pp. 689–692, 2011.

39) Masakiyo Fujimoto, Shinji Watanabe, and Tomohiro Nakatani, "A robust estimation method of noise mixture model for noise suppression", *Interspeech-2011,* pp. 697–700, 2011.

40) Tomohiro Nakatani, Takuya Yoshioka, and Keisuke Kinoshita, "Mathematical analysis of speech dereverberation based on time-varying Gaussian source model: Its Solution and Convergence Characteristics", *invited in Proc. IEEE International Conference on Signal Processing, Communications and Computing (ICSPCC-2011),* 2011.

41) Mehrez Souden, Shoko Araki, Keisuke Kinoshita, Tomohiro Nakatani, and Hiroshi Sawada, "Simultaneous speech source separation and noise reduction via clustering and MMSE-based filtering", *invited in Proc. IEEE International Conference on Signal Processing, Communications and Computing (ICSPCC-2011),* 2011.

42) Marc Delcroix, Shinji Watanabe, Tomohiro Nakatani, and Atsushi Nakamura, "Discriminative approach to dynamic variance adaptation for noisy speech recognition", *Proc. the 3rd Joint Workshop on Hands-free Speech Communication and Microphone Arrays (HSCMA 2011),* pp. 7–12, May 2011.

43) Takuya Yoshioka and Tomohiro Nakatani, "A microphone array system integrating beamforming, feature enhancement, and spectral mask-based noise estimation", *Proc. the 3rd Joint Workshop on Hands-free Speech Communication and Microphone Arrays (HSCMA 2011),* pp. 219-224, May 2011.

44) Yoshiki Iso, Shoko Araki, Shoji Makino, Tomohiro Nakatani, Hiroshi Sawada, Takeshi Yamada, Atsushi Nakamura, "Blind source separation of mixed speech in a high reverberant environment", *Proc. the 3rd Joint Workshop on Hands-free Speech Communication and Microphone Arrays (HSCMA 2011),* pp. 36-39, May 2011.

45) Takuya Yoshioka and Tomohiro Nakatani, "Speech enhancement based on log spectral envelope model and harmonicity-derived spectral mask, and its coupling with feature compensation", *Proc. the 2011 IEEE International Conference on Acoustics, Speech, and Signal Processing (ICASSP 2011),* pp. 5064–5067, May 2011.

46) Tomohiro Nakatani, Shoko Araki, Takuya Yoshioka, and Masakiyo Fujimoto, "Joint unsupervised learning of hidden Markov source models and source location models for multichannel source separation", *Proc. the 2011 IEEE International Conference on Acoustics, Speech, and Signal Processing (ICASSP 2011),* pp. 237–240, May 2011.

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- Tomohiro Nakatani, "Blind speech de-reverberation based on statistical signal processing," Winter School on Neuroinfomatics (WSNI), Sogang University, Seoul, Korea, January 29-30, 2009.
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