Tomohiro NAKATANI

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NTT Communication Science Laboratories
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Visiting Associate Professor (2008.4.1∼)
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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

Academic Activities

Jan. 2011 - Present	Chair of IEEE Kansai Section Technical Program Committee
Apr. 2009 - Present	Member of Audio and Acoustics Technical Committee, IEEE Signal Processing Society
May 2007 - Present	Member of Blind Signal Processing Technical Committee, IEEE Circuits and Systems Society
Jan. 2009 - Dec. 2010	Secretary of IEEE Kansai Section Technical Program Committee
Mar. 2008 - Mar 2010	Associate Editor for IEEE Trans. Audio, Speech, and Language Processing
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) 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.

  2) Katsuhiko Ishiguro, Takeshi Yamada, Shoko Araki, Tomohiro Nakatani, and Hiroshi Sawada, "Probabilistic Speaker Clustering for DOA-based Diarization", *IEEE Trans. Audio, Speech, and Language Processing,* vol. 20, no. 2, pp. 447–460, 2012.

  3) 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.

  4) 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.

  5) 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.

  6) 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.

  7) 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.

8) 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.

9) 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.

10) 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.

11) 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.

12) 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.

13) 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.

14) 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.

15) 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.

16) 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.

17) 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.

18) 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.

19) 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.

20) 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.

21) 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.

22) 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.

23) 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.

24) 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) 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.

  2) Keisuke Kinoshita, Takuya Yoshioka, Tomohiro Nakatani, "Recent advances in blind speech dereverberation", *IEICE Fundamentals Review,* Vol.4 No.4 pp.301-310, Apr. 2011.

  3) 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.

  4) 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.

  5) 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.

  6) 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.

  7) 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.

  8) 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.

  9) 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, 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.

  2) 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.

  3) 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.

  4) 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.

  5) 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.

6) 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.

7) 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.

8) 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.

9) 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 non-stationary 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.

10) 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.

11) 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.

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

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

14) 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.

15) 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.

16) 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.

17) 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.

18) 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.

19) Shoko Araki and Tomohiro Nakatani, “Hybrid approach for multichannel source separation combining time-frequency mask with multi-channel Wiener filter”, *Proc. the 2011 IEEE International Conference on Acoustics, Speech, and Signal Processing (ICASSP 2011),* pp. 225–228, May 2011.

20) Masakiyo Fujimoto, Shinji Watanabe, Tomohiro Nakatani, “Non-stationary noise estimation method based on bias-residual component decomposition for robust speech recognition”, *Proc. the 2011 IEEE International Conference on Acoustics, Speech, and Signal Processing (ICASSP 2011),* pp. 4816–4818, May 2011.

21) Shoko Araki, Takaaki Hori, Masakiyo Fujimoto, Shinji Watanabe, Takuya Yoshioka, and Tomohiro Nakatani, "Online meeting recognizer with multichannel speaker diarization", *Proc. the 2010 Asilomar Conference on Signals, Systems, and Computers,* Nov. 2010.

22) Takaaki Hori, Shoko Araki, Takuya Yoshioka, Masakiyo Fujimoto, Shinji Watanabe, Takanobu Oba, Atsunori Ogawa, Kazuhiro Otsuka, Dan Mikami, Keisuke Kinoshita, Tomohiro Nakatani, Atsushi Nakamura, Junji Yamato, "Real-time meeting recognition and understanding using distant microphones and omni-directional camera", *Proc. the 2010 IEEE Workshop on Spoken Language Technology,* pp. 412–417, Dec. 2010.

23) Tomohiro Nakatani, Shoko Araki, Takuya Yoshioka, and Masakiyo Fujimoto, "Multichannel source separation based on source location cue with log-spectral shaping by hidden Markov source model", *Interspeech 2010,* pp. 2766-2769, Sep. 2010.

24) Masakiyo Fujimoto, Shinji Watanabe, Tomohiro Nakatani, "Voice Activity Detection Using Frame-Wise Model Re-Estimation Method Based on Gaussian Pruning with Weight Normalization", *Interspeech 2010,* pp. 3102-3105, Sep. 2010.

25) Tomohiro Nakatani, Shoko Araki, "Single channel source separation based on sparse source observation model with harmonic constraint", *International Conference on Audio, Speech, and Signal Processing (ICASSP-2010),* pp. 13–16, 2010.

26) Keisuke Kinoshita, Tomohiro Nakatani, Masato Miyoshi, "Blind upmix of stereo music signals using multi-step linear prediction based reverberation extraction", *International Conference on Audio, Speech, and Signal Processing (ICASSP-2010),* pp. 49–52, 2010.

27) Takuya Yoshioka, Tomohiro Nakatani, Hiroshi G. Okuno, "Noisy speech enhancement based on prior knowledge about spectral envelope and harmonic structure", *International Conference on Audio, Speech, and Signal Processing (ICASSP-2010),* pp. 4270–4273, 2010.

28) Shoko Araki, Tomohiro Nakatani, Hiroshi Sawada, "Simultaneous clustering of mixing and spectral model parameters for blind sparse source separation", *International Conference on Audio, Speech, and Signal Processing (ICASSP-2010),* pp. 5–8, 2010.

29) Naoki Yasuraoka, Takuya Yoshioka, Tomohiro Nakatani, Atsushi Nakamura, Hiroshi G. Okuno, "Music dereverberation using harmonic structure source model and Wiener filter", *International Conference on Audio, Speech, and Signal Processing (ICASSP-2010),* pp. 53–56, 2010.

30) Yumi Ansai, Shoko Araki, Shoji Makino, Tomohiro Nakatani, Takeshi Yamada, Atsushi Nakamura, Nobuhiko Kitawaki, "Cepstral smoothing of separated signals for underdetermined speech separation", *International Symposium on Circuits and Systems (ISCAS-2010),* pp. 2506–2509, 2010.

31) Takuya Yoshioka, Hirokazu Kameoka, Tomohiro Nakatani, Hiroshi G. Okuno, "Statistical models for speech dereverberation", *IEEE International Workshop on Applications of Signal Processing to Audio and Acoustics (WASPAA-2009),* pp. 145–148, Oct. 2009.

32) Katsuhiko Ishiguro, Takeshi Yamada, Shoko Araki, Tomohiro Nakatani, "A Probabilistic Speaker Clustering for DOA-based Diarization", *IEEE International Workshop on Applications of Signal Processing to Audio and Acoustics (WASPAA-2009),* pp. 241–244, Oct. 2009.

33) Takuya Yoshioka, Tomohiro Nakatani, Masato Miyoshi, "Fast algorithm for conditional separation and dereverberation", *Proc. the 17th European Signal Processing Conference (EUSIPCO2009),* 2009.

34) Kentaro Ishizuka, Shoko Araki, Kazuhiro Otsuka, Tomohiro Nakatani, Masakiyo Fujimoto, "A Speaker Diarization Method based on the Probabilistic Fusion of Audio-Visual Location Information", *Proc. ICML-MLMI-2009,* 2009.

35) Shoko Araki, Tomohiro Nakatani, Hiroshi Sawada, Shoji Makino, "Stereo source separation and source counting with MAP estimation with Dirichlet prior considering spatial aliasing problem", *Proc. Independent Component Analysis and Signal Separation (ICA-2009),* Mar. 2009.

36) Tomohiro Nakatani, Takuya Yoshioka, Keisuke Kinoshita, Masato Miyoshi, Biing-Hwang Juang, "Real-time speech enhancement in noisy reverberant multi-talker environments using probabilistic model of room acoustics", *Proc. International Conference on Audio, Speech, and Signal Processing (ICASSP-2009),* pp.137-140, 2009.

37) Takuya Yoshioka, Hideyuki Tachibana, Tomohiro Nakatani, Masato Miyoshi, "Adaptive dereverberation of speech signals with speaker-position change detection", *Proc. International Conference on Audio, Speech, and Signal Processing (ICASSP-2009),* pp.3733-3736, 2009.

38) Hirokazu Kameoka, Tomohiro Nakatani, Takuya Yoshioka, "Robust speech dereverberation based on non-negativity and sparse nature of speech spectrograms", *Proc. International Conference on Audio, Speech, and Signal Processing (ICASSP-2009),* pp.45-48, 2009.

39) Shoko Araki, Tomohiro Nakatani, Hiroshi Sawada, Shoji Makino, "Blind sparse source separation for unknown number of sources based on Gaussian mixture model with Dirichlet prior", *Proc. International Conference on Audio, Speech, and Signal Processing (ICASSP-2009),* pp.33-36, 2009.

40) Keisuke Kinoshita, Tomohiro Nakatani, Masato Miyoshi, Toshiyuki Kubota, "A new audio post-production tool for speech dereverberation", *Audio Engineering Society Convention Paper (AES San Francisco),* Oct., 2008.

41) Keisuke Kinoshita, Tomohiro Nakatani, Masato Miyoshi, "Upmixing stereo music signals based on dereverberation mechanism", *Audio Engineering Society Convention Paper (AES Osaka),* Jul., 2008.

42) Tomohiro Nakatani, Takuya Yoshioka, Keisuke Kinoshita, Masato Miyoshi, Biing-Hwang Juang, "Incremental estimation of reverberation with uncertainty using prior knowledge of room acoustics for speech dereverberation", *Proc. International Workshop on Acoustic Echo and Noise Control (IWAENC-2008),* Sep., 2008.

43) Takuya Yoshioka, Tomohiro Nakatani, Masato Miyoshi, "Enhancement of noisy reverberant speech by linear filtering followed by nonlinear noise suppression", *Proc. International Workshop on Acoustic Echo and Noise Control (IWAENC-2008),* Sep., 2008.

44) Tobias Hager, Shoko Araki, Kentaro Ishizuka, Masakiyo Fujimoto, Tomohiro Nakatani, Shoji Makino, "Handling speaker position changes in a meeting diarization system by combining DOA clustering and speaker identification", *Proc. International Workshop on Acoustic Echo and Noise Control (IWAENC-2008),* Sep, 2008.

45) Masakiyo Fujimoto, Kentaro Ishizuka, Tomohiro Nakatani, "Study of Integration of Statistical Model-Based Voice Activity Detection And Noise Suppression", *Proc. Interspeech-2008,* pp. 2008-2011, 2008.

46) Tomohiro Nakatani, Takuya Yoshioka, Keisuke Kinoshita, Masato Miyoshi, Biing-Hwang Juang, "Blind speech dereverberation with multi-channel linear prediction based on short time Fourier transform representation", *IEEE International Conference on Audio, Speech, and Signal Processing (ICASSP-2008),* pp. 85–88, 2008.

47) Takuya Yoshioka, Tomohiro Nakatani, Masato Miyoshi, "Maximum likelihood approach to speech enhancement for noisy reverberant signals", *IEEE International Conference on Audio, Speech, and Signal Processing (ICASSP-2008),* pp. 4585–4588, 2008.

48) Marc Delcroix, Tomohiro Nakatani, Shinji Watanabe, "Combined static and dynamic variance adaptation for efficient interconnection of speech enhancement pre-processor with speech recognizer", *IEEE International Conference on Audio, Speech, and Signal Processing (ICASSP-2008),* pp. 4073–4076, 2008.

49) Masakiyo Fujimoto, Kentaro Ishizuka, Tomohiro Nakatani, "A voice activity detection based on the adaptive integration of multiple speech features and a signal decision scheme", *IEEE International Conference on Audio, Speech, and Signal Processing (ICASSP-2008),* pp. 4441–4444, 2008.

50) Tomohiro Nakatani, Takuya Yoshioka, Keisuke Kinoshita, Masato Miyoshi, Biing-Hwang Juang, "Speech dereverberation in short time Fourier transform domain with crossband effect compensation", *Joint Workshop on Hands-free Speech Communication and Microphone Array (HSCMA-2008),* May 2008.

51) Tomohiro Nakatani, Biing-Hwang Juang, Takuya Yoshioka, Keisuke Kinoshita, and Masato Miyoshi, "Importance of energy and spectral features in Gaussian source model for speech dereverberation", *Proc. IEEE Workshop on Applications of Signal Processing to Audio and Acoustics (WASPAA-2007),* pp. 299–302, Oct. 2007.

52) Takuya Yoshioka, Tomohiro Nakatani, Takafumi Hikichi, and Masato Miyoshi, "Overfitting-Resistant Speech Dereverberation", *IEEE Workshop on Applications of Signal Processing to Audio and Acoustics (WASPAA-2007),* pp. 163–166, Oct. 2007.

53) Tomohiro Nakatani, Takafumi Hikichi, Keisuke Kinoshita, Takuya Yoshioka, Marc Delcroix, Masato Miyoshi, and Biing-Hwang Juang, "Robust blind dereverberation of speech signals based on characteristics of short-time speech segments", *Proc. IEEE International Symposium on Circuits and Systems (ISCAS-2007),* pp. 2986–2989, June 2007.

54) Biing-Hwang Juang and Tomohiro Nakatani, "Joint source-channel modeling and estimation for speech dereverberation", *Proc. IEEE International Symposium on Circuits and Systems (ISCAS-2007),* pp. 2990–2993, June 2007.

55) Keisuke Kinoshita, Marc Delcroix, Tomohiro Nakatani, and Masato Miyoshi, "Dereverberation of real recordings using linear prediction-based microphone array", *Proc. Audio Engineering Society (AES) 13th Regional Convention,* Tokyo, August 2007.

56) Keisuke Kinoshita, Marc Delcroix, Tomohiro Nakatani, and Masato Miyoshi, "Multi-step linear prediction based speech enhancement in noisy reverberant environment", *Proc. Interspeech-2007,* pp. 854–857, August 2007.

57) Kentaro Ishizuka, Tomohiro Nakatani, Masakiyo Fujimoto, and Noboru Miyazaki, "Noise robust front-end with voice activity detection based on periodic to aperiodic component ratio", *Proc. Interspeech-2007,* pp. 230–233, August 2007.

58) Tomohiro Nakatani, Biing-Hwang Juang, Takafumi Hikichi, Takuya Yoshioka, Keisuke Kinoshita, Marc Delcroix, and Masato Miyoshi, "Study on speech dereverberation with autocorrelation codebook", *Proc. IEEE International Conference on Acoustics, Speech, and Signal Processing (ICASSP-2007),* vol. I, pp. 193–196, April 2007.

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