CURRICULUM VITAE

Keiichi Tokuda

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1 PROFESSIONAL EXPERIENCE

April 2004–present
Professor, Department of Computer Science Nagoya Institute of Technology, Japan

April 1996–March 2004
Associate Professor, Department of Computer Science Nagoya Institute of Technology, Japan

April 1989–March 1996
Research Associate, Department of of Electronic and Electric Engineering Tokyo Institute of Technology, Japan

January 2012–Present
Honorary Professor, University of Edinburgh, U.K.

April 2014–March 2015
Visiting Researcher, Google, U.K.

April 2006–March 2013
Visiting Researcher, National Institute of Information and Communications Technology (NiCT), Japan

April 2002–March 2006
Visiting Researcher, ATR Spoken Language Communication Laboratories, Japan

July 2001–April 2002
Visiting Researcher, Language Technologies Institute Carngie Mellon University

April 2001–September 2001
Invited Lecturer, Nagoya University, Japan

August 2000–July 2001
Visiting Researcher, ATR Spoken Language Translation Laboratories, Japan

November 1997–March 1998
Invited Lecturer, Tokyo Institute of Technology, Japan

2 UNIVERSITY DEGREES

1989
Dr.Eng in information processing, Tokyo Institute of Technology, Japan

1986
M.E in information processing, Tokyo Institute of Technology, Japan

1984
B.E in electrical and electronic engineering, Nagoya Institute of Technology, Japan

3 AWARDS

2019 ISCA Medal for Scientific Achievement 2019, September 16, 2019.

2015 Achievement Award from the Institute of Electronics, Information and Communication Engineers

2014 IEEE Fellow 2014 (Institute of Electrical and Electronics Engineers)

2013 ISCA Fellow 2013 (International Speech Communication Association)

2013 IPSJ Kiyasu Special Industrial Achievement Award (Information Processing Society of Japan)

2013 The 2013 EURASIP-ISCA Best Paper Award (Speech Communication Journal)

2012 Prizes for Science and Technology (Research Category), The Commendation for Science and Technology by the Minister of Education, Culture, Sports, Science and Technology

2008 Information and Systems Society Distinguished Achievement Award from the Institute of Electronics, Information and Communication Engineers

2008 Information and Systems Society Excellent Paper Award from the Institute of Electronics, Information and Communication Engineers

2008 TELECOM System Technology Prize from the Telecommunications Advancement Foundation

2001 Best Paper Award from the Institute of Electronics, Information and Communication Engineers

2001 Inose Award (the highest award) from the Institute of Electronics, Information and Communication Engineers

2001 TELECOM System Technology Prize from the Telecommunications Advancement Foundation

4 PUBLICATION

BOOK CHAPTERS

1. Keiichi Tokuda, Akinobu Lee, Yoshihiko Nankaku, Keiichiro Oura, Kei Hashimoto, Daisuke Yamamoto, Ichi Takumi, Takahiro Uchiya, Shuhei Tsutsumi, Steve Renals, and Junichi Yamagishi, "User Generated Dialogue Systems: uDialogue," Human-Harmonized Information Technology, Volume 2, Springer, pp.77–114, May 2017. (ISBN 978-4-431-56533-8)

2. Heiga Zen, Keiichi Tokuda, "7.3 The HMM-based speech synthesis system (HTS)," in Computer processing of Asian spoken languages, Editors: Shuichi Itahashi, Chiu-yu Tseng, Consideration Books, Los Angeles, March 2010. (ISBN 978-0-935047-72-1)

3. Keiichi Tokuda, "4.3 A unified approach to prosody control using HMM-based speech synthesis," Prosody and Spoken Language Processing, Keikichi Hirose (Ed.), Maruzen, pp.118–127, Jan. 2006 (in Japanese). (ISBN 978-4-621-07674-3)

4. Keiichi Tokuda, Heiga Zen, Alan W. Black, "An HMM-Based Approach to Multilingual Speech Synthesis (Chapter 7)," Text-to-Speech Synthesis: New Paradigms and Advances, Shrikanth Narayanan, Abeer Alwan (Eds.), Prentice Hall, pp.135–153, Aug. 2004. (ISBN 978-0131456617)

5. Shin-ichi Kawamoto, Hiroshi Shimodaira, Tsuneo Nitta, Takuya Nishimoto, Satoshi Nakamura, Katsunobu Itou, Shigeo Morishima, Tatsuo Yotsukura, Atsuhiko Kai, Akinobu Lee, Yoichi Yamashita, Takao Kobayashi, Keiichi Tokuda, Keikichi Hirose, Nobuaki Minematsu, Atsushi Yamada, Yasuharu Den, Takehito Utsuro, Shigeki Sagayama, "Galatea: Open-source Software for Developing Anthropomorphic Spoken Dialog Agents," Life-Like Characters: Tools, Affective Functions, and Applications, Series: Cognitive Technologies, Helmut Prendinger, Mitsuru Ishizuka (Eds.), Springer-Verlag, pp.187–211, 2004. (ISBN 978-3540008675)

6. Takao Kobayashi, Keiichi Tokuda, "14.3 Speech Spectral Estimation," in Handbook of Spectral Analyais, Mikio Hino (Ed.), Asakura, pp.481–492, 2004 (in Japanese). (ISBN 978-4-254-20108-6)

INVITED REVIEW AND TUTORIAL

1. Keiichi Tokuda, Yoshihiko Nankaku, Tomoki Toda, Heiga Zen, Junichi Yamagishi, Keiichiro Oura, "Speech Synthesis Based on Hidden Markov Models," Proceedings of the IEEE, vol.101, no.5, pp.1234–1252, May 2013.

2. Keiichi Tokuda, "Recent advances in statistical parametric speech synthesis," the Journal of the Acoustical Society of Japan, vol.67, no.1, pp.17–22, January 2011 (in Japanese).

3. Keiichiro Oura, Heiga Zen, Shinji Sako, Keiichi Tokuda, "Construction of Speech Synthesis Systems using HTS," Journal of Human Interface Society : human interface, vol.12, no.1, pp.35–40, February 2010 (in Japanese).

4. Hisashi Kawai, Keiichi Tokuda, "Multi-language Speech Synthesis," The Journal of The Institute of Electrical Engineers of Japan, vol.130, no.1, pp.16–19, January 2010 (in Japanese).

5. Kiyohiro Shikano, Kazuya Takeda, Tatsuya Kawahara, Hideki Kawahara, Hiroshi Saruwatari, Keiichi Tokuda, Akinobu Lee, Hiromichi Kawanami, Ryuichi Nishimura, Randy GOMEZ, Tomoki Toda, Takanobu Nishiura, Toru Takahashi, Hideki Banno, Heiga Zen, “E-Society Software Development Project for Speech Recognition and Synthesis,” Journal of IEICE, vol.92, no.6, June 2009 (in Japanese).

6. Heiga Zen, Keiichi Tokuda, “TechWare: HMM-Based Speech Synthesis Resources,” IEEE Signal Processing Magazine, July 2009 (in Japanese).

7. Kiyohiro Shikano, Tatsuya Kawahara, Hiroshi Saruwatari, Kazuya Takeda, Hideki Kawahara, Keiichi Tokuda, Takanobu Nishiura, Akinobu Lee, “Advanced development of fundamental software through tight collaboration of academia and industry: Human friendly speech interface,” IPSJ Magazine, vol.49, no.11, pp.1297–1301, November 2008 (in Japanese).

8. Keiichi Tokuda, Alan W. Black, ”Speech synthesis research in a new age of cooperation and competition: The Blizzard Challenge,” the Journal of the Acoustical Society of Japan, vol.62, no.6, pp.466–472, June 2006 (in Japanese).

9. Keiichi Tokuda, ”Speech Recognition and Speech Synthesis based on Hidden Markov Models,” IPSJ Magazine, vol.45, no.10, (in Japanese).

10. Takao Kobayashi, Keiichi Tokuda, ”Technology Trends in Corpus-based Speech Synthesis [IV]: HMM-Based Speech Synthesis,” The Journal of IEICE, vol.87, no.4, April 2004 (in Japanese).

INVITED PAPER

1. Keiichiro Oura, Daisuke Yamamoto, Ichi Takumi, Akinobu Lee, Keiichi Tokuda, “On-Campus, User-Participatable, and Voice-Interactive Digital Signage,” Journal of The Japanese Society for Artificial Intelligence, vol.28, no.1, pp.60–67, January 2013.

2. Keiichi Tokuda, Takashi Masuko, Noboru Miyazaki, Takao Kobayashi, “Multi-space probability distribution HMM,” IEICE Trans. Information and Systems, vol.E85-D, no.3, pp.455–464, Mar. 2002.

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1. Xin Wang, Shinji Takaki, Junichi Yamagishi, Simon King, and Keiichi Tokuda, “A vector quantized variational autoencoder (VQ-VAE) autoregressive neural F0 model for statistical parametric speech synthesis,” IEEE/ACM Transactions on Audio, Speech, and Language Processing, vol.28, pp.157–170, October 2019.

2. Kei Sawada, Kei Hashimoto, Keiichiro Oura, Yoshihiko Nankaku, and Keiichi Tokuda, “A Bayesian framework for image recognition based on hidden Markov eigen-image models,” IEEJ Transactions on Electrical and Electronic Engineering, vol.13, Issue 9, pp.1335–1347, September 2018.

3. Takenori Yoshimura, Kei Hashimoto, Keiichiro Oura, Yoshihiko Nankaku, and Keiichi Tokuda, “Mel-cepstrum-based quantization noise shaping applied to neural-network-based speech waveform synthesis,” IEEE/ACM Transactions on Audio, Speech and Language Processing, vol. 26, Issue 7, pp.1173–1180, July 2018.

4. Kei Sawada, Kei Hashimoto, Keiichiro Oura, Yoshihiko Nankaku, and Keiichi Tokuda, “Constructing text-to-speech systems for languages with unknown pronunciations,” Acoustical Science and Technology, vol.39, Issue 2, pp.119–129, March 2018.

5. Takenori Yoshimura, Kei Hashimoto, Keiichiro Oura, Yoshihiko Nankaku, and Keiichi Tokuda, “Simultaneous optimization of multiple tree-based factor analyzed HMM for speech synthesis,” IEEE/ACM Transactions on Audio, Speech and Language Processing, vol. 25, Issue 9, pp.1532–1541, September 2017.

6. Kei Sawada, Akira Tamamori, Kei Hashimoto, Yoshihiko Nankaku, and Keiichi Tokuda, “A Bayesian approach to image recognition based on separable lattice hidden Markov models,” IEICE Transactions on Information and Systems, vol.E99-D, no.12, pp.3119–3131, December 2016.

7. Kazuhiro Nakamura, Keiichiro Oura, Yoshihiko Nankaku, Keiichi Tokuda, “Hidden Markov model-based English singing voice synthesis,” IEICE Transactions on Information and Systems, vol.J97-D, no.10, pp.1572–1581, October 2014 (in Japanese).

8. Akira Tamamori, Yoshihiko Nankaku, Keiichi Tokuda, “Image recognition based on separable lattice trajectory 2-D HMMs,” IEICE Transactions on Information and Systems, vol.E97-D, no.7, pp.1842–1854, July 2014.

9. Hongwu Yang, Keiichiro Oura, Haiyan Wang, Zhenye Gan, Keiichi Tokuda, “Using speaker adaptive training to realize Mandarin-Tibetan cross-lingual speech synthesis,” Springer, Multimedia Tools and Applications, June 2014.

10. Kazuhiro Nakamura, Kei Hashimoto, Yoshihiko Nankaku, Keiichi Tokuda, “Integration of spectral feature extraction and modeling for HMM-based speech synthesis,” IEICE Transactions on Information and Systems, vol.E97-D, no.6, pp.1438–1448, June 2014.

11. Shinji Takaki, Yoshihiko Nankaku and Keiichi Tokuda, “Contextual additive structure for HMM-based speech synthesis,” IEEE Journal of Selected Topics in Signal Processing, vol.8, issue 2, pp.229–238, April 2014.

12. Sayaka Shiota, Kei Hashimoto, Yoshihiko Nankaku, Keiichi Tokuda, “A Bayesian framework using multiple model structures for speech recognition,” IEICE Transactions on Information and Systems, vol.E96-D, no.4, pp.939–948, April 2013.

13. John Dines, Hui Liang, Lakshmi Saheer, Matthew Gibson, William Byrne, Keiichiro Oura, Keiichi Tokuda, Junichi Yamagishi, Simon King, Mirjam Wester, Teemu Hirsimäki, Reima Karhila, Mikko Kurimo, “Personalising speech-to-speech translation: unsupervised cross-lingual speaker adaptation for HMM-based speech synthesis,” Computer Speech and Language, vol.27, pp.420–437, February 2013.

14. Kei Hashimoto, Junichi Yamagishi, William Byrne, Simon King, Keiichi Tokuda, “Impacts of machine translation and speech synthesis on speech-to-speech translation,” Speech Communication, vol.54, no.7, pp.857–866, September 2012.

15. Akira Tamamori, Yoshihiko Nankuaku, Keiichi Tokuda, “An extension of separable lattice 2-D HMMs for rotational data variations,” IEICE Transactions on Information and Systems, vol.E95-D, no.8, pp.2074–2083, August 2012.

16. Keiichiro Oura, Junichi Yamagishi, Mirjam Wester, Simon King, Keiichi Tokuda, "Analysis of unsupervised cross-lingual speaker adaptation for HMM-based speech synthesis using KLD-based transform mapping," Speech Communication, vol.54, no.6, pp.703–714, July 2012.

17. Sayaka Shiota, Kei Hashimoto, Heiga Zen, Yoshihiko Nankaku, Akinobu Lee, Keiichi Tokuda, "Speech recognition based on statistical models including multiple phonetic decision trees" Acoustical Science and Technology. (accepted)

18. Kei Hashimoto, Heiga Zen, Yoshihiko Nankaku, Akinobu Lee, Keiichi Tokuda, "Bayesian context clustering using cross validation for speech recognition" IEICE Transactions on Information and Systems, vol.E94-D, no.3, pp.668–678, March 2011.

19. Ryuta Terashima, Heiga Zen, Yoshihiko Nankaku, Keiichi Tokuda, "A frame-based context-dependent acoustic modeling for speech recognition," The transactions of the Institute of Electrical Engineers of Japan C (A publication of Electronics, Information and System Society), vol.130, no.4, pp.557–564, April 2010 (in Japanese).

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21. Ryuta Terashima, Takayoshi Yoshimura, Toshihiro Wakita, Keiichi Tokuda, Tadashi Kitamura, "Prediction method of speech recognition performance based on HMM-based speech synthesis technique," The transactions of the Institute of Electrical Engineers of Japan C (A publication of Electronics, Information and System Society), vol.130, no.4, pp.557–564, April 2010 (in Japanese).

22. Keiichiro Oura, Heiga Zen, Yoshihiko Nankaku, Akinobu Lee, Keiichi Tokuda, "A covariance-tying technique for HMM-based speech synthesis," IEICE Transactions on Information and Systems, vol.E93-D, no.3, pp.595–601, March 2010.

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24. Kei Hashimoto, Hirohumi Yamamoto, Hideo Okuma, Eiichiro Sumita, Keiichi Tokuda, "A reordering model using a source-side parse-tree for statistical machine translation," IEICE Transactions on Information and Systems, vol.E92-D, no.12, pp.2386–2393, December 2009.

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29. Tomoki Toda, Alan W. Black, and Keiichi Tokuda, "Statistical mapping between articulatory movements and acoustic spectrum with aGaussian mixture model," Speech Communication, vol.50, no.3, pp.215-227, March 2008.

30. Tomoki Toda, Alan W. Black, and Keiichi Tokuda, "Voice conversion based on maximum likelihood estimation of speech parameter trajectory," IEEE Transactions on Audio, Speech and Language Processing, vol.15, no.8, pp.2222–2235, November 2007.

31. Ranniery Maia, Heiga Zen, Keiichi Tokuda, Tadashi Kitamura, and Fernando G. V. Resende, "An HMM-based Brazilian Portuguese speech synthesizer and its characteristics," Journal of Communication and Information Systems, vol.21, no.2, pp.132–145, Aug. 2006.

32. Heiga Zen, Keiichi Tokuda, Takashi Masuko, Takao Kobayashi, and Tadashi Kitamura, "Hidden Semi-Markov Model Based Speech Synthesis System," IEICE Transactions on Information Systems, vol.E90-D, no.5, pp.825–834, May 2007.

33. Tomoki Toda and Keiichi Tokuda, "Speech Parameter Generation Algorithm Considering Global Variance for HMM-Based Speech Synthesis," IEICE Transactions on Information Systems, vol.E90-D, no.5, pp.816–824, May 2007.

34. Heiga Zen, Tomoki Toda, Masaru Nakamura, and Keiichi Tokuda, "Details of Nitech HMM-based speech synthesis system for the Blizzard Challenge 2005," IEICE Transactions on Information and Systems, vol.E90-D, no.1, pp.325–333, Jan. 2007.

35. Hisashi Kawai, Tomoki Toda, Junichi Yamagishi, Toshio Hirai, Jinfu Ni, Nobuyuki Nishizawa, Minoru Tsuzaki, and Keiichi Tokuda, "XIMERA: a concatenative speech synthesis system with large scale corpora," IEICE Transactions on Information and Systems, J89-D-II, no.12, pp.2688–2698, Dec. 2006 (in Japanese).

36. Heiga Zen, Keiichi Tokuda, Tadashi Kitamura, "Reformulating the HMM as a trajectory model by imposing explicit relationships between static and dynamic feature vector sequences," Computer Speech and Language, vol.21, no.1, pp.153–173, Jan. 2007.

37. Amaro Lima, Heiga Zen, Yoshihiko Nankaku, Keiichi Tokuda, Tadashi Kitamura, and Fernando G. Resende, "Applying sparse KPCA for feature extraction in speech recognition," IEICE Transactions on Information Systems, vol.E88-D, no.3, pp.401–409, March 2005.

38. Hiroyuki Suzuki, Heiga Zen, Yoshihiko Nankaku, Chiyomi Miyajima, Keiichi Tokuda, and Tadashi Kitamura, "Continuous speech recognition based on general factor dependent acoustic models," IEICE Transactions on Information Systems, vol.E88-D, no.3, pp.410–417, March 2005.

39. Hiroyoshi Yamamoto, Yoshihiko Nankaku, Chiyomi Miyajima, Keiichi Tokuda, and Tadashi Kitamura, “Parameter Sharing in Mixture of Factor Analyzers for Speaker Identification,” IEICE Transactions on Information Systems, vol.E88-D, no.3, pp.419–424, March 2005.

40. Yohei Itaya, Heiga Zen, Yoshihiko Nankaku, Chiyomi Miyajima, Keiichi Tokuda, and Tadashi Kitamura, “Deterministic annealing EM algorithm in acoustic modeling for speaker and speech recognition,” IEICE Transactions on Information Systems, vol.E88-D, no.3, pp.425–431, March 2005.

41. Amaro Lima, Heiga Zen, Yoshihiko Nankaku, Chiyomi Miyajima, Keiichi Tokuda, Tadashi Kitamura, “On the use of kernel PCA for feature extraction in speech recognition,” IEICE Transactions on Information Systems, vol.E87-D, no.12, pp.2802–2811, Dec. 2004.

42. Heiga Zen, Keiichi Tokuda, Tadashi Kitamura, ”Decision tree based simultaneous clustering of phonetic contexts, dimensions, and state positions for acoustic modeling,” IEICE Trans. Inf. & Syst., vol.87-D-II, no.8, pp.1593–1602, Aug. 2004 (in Japanese).

43. Takayoshi Yoshimura, Keiichi Tokuda, Takashi Masuko, Takao Kobayashi, Tadashi Kitamura, “Incorporation of mixed excitation model and postfilter into HMM-based text-to-speech synthesis,” IEICE Trans. Inf. & Syst., vol.87-D-II, no.8, pp.1565–1571, Aug. 2004. (in Japanese). Translation: Systems and Computers in Japan, John Wiley & Sons, Inc., vol.36, no.12, pp.43–50, Nov. 15, 2005.

44. Shinji Sako, Chiyomi Miyajima, Keiichi Tokuda, Tadashi Kitamura, “A singing voice synthesis system based on hidden Markov model,” Transactions of Information Processing Society of Japan, vol.45, no.3, pp.719-727, Mar. 2004 (in Japanese).

45. Toru Takahashi, Keiichi Tokuda, Takao Kobayashi, Tadashi Kitamura, “Mixture density models based on mel-cepstral representation of Gaussian process,” IEICE Trans. Fundamentals, vol.E86-A, no.8, pp.1971–1978, Aug. 2003.

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48. Yoshihiko Nankaku, Keiichi Tokuda, Tadashi Kitamura, Takao Kobayashi, “Normalized training for HMM-based visual speech recognition,” IEICE Trans. Inf. & Syst., vol.J86-D-II, no.2, pp.163–172, Feb. 2003 (in Japanese).

49. Takashi Masuko, Keiichi Tokuda, Takao Kobayashi, “Very Low Bit Rate Speech Coding Based on HMM with Speaker Adaptation,” IEICE Trans. Inf. & Syst., vol.J85-D-II, no.12, pp.1749–1759, Dec. 2002 (in Japanese).

50. Takayuki Satoh, Takashi Masuko, Takao Kobayashi, Keiichi Tokuda, “Discrimination of synthetic speech generated by an HMM-based speech synthesis system for speaker verification,” IPSJ Journal, vol.43, no.7, pp.2197-2204, July 2002 (in Japanese).

51. Shinichi Kawamoto, Hiroshi Shimodaira, Tsuneo Nitta, Takuya Nishimoto, Satoshi Nakamura, Kazunori Itou, Shigeo Morishima, Tatsuo Yotsukura, Akihiko Kai, Akinobu Lee, Yohichi Yamashita, Takao Kobayashi, Keiichi Tokuda, Keikichi Hirose, Nobuaki Minematsu, Atsushi Yamada, Yasuharu Den, Takenori Utsuro, Shigeki Sagayama, "Implementation of anthropomorphic spoken dialog agent," Transactions of Information Processing Society of Japan, vol.43, no.7, pp.2249-2263, July 2002 (in Japanese).

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53. Mitsuru Katsumata, Hisashi Suzuki, Keiichi Tokuda, and Tadashi KITAMURA, "A Hierarchical Algorithm for Monotonic and Continuous Two-Dimensional Dynamic Programming," IEICE Trans. Inf. & Syst., vol.J85-D-II, no.9, pp.1382–1391, Sep. 2002 (in Japanese).

54. Toru Takahashi, Keiichi Tokuda, Takao Kobayashi, Tadashi Kitamura, "Vector quantization of mel-cepstrum coefficients using distortion measure for spectral analysis," IEICE Trans. Inf. & Syst., vol.J85-D-II, no.8, pp.1273–1283, Aug. 2002 (in Japanese).

55. Masatsune Tamura, Takashi Masuko, Keiichi Tokuda, Takao Kobayashi, "Speaker adaptation of pitch and spectrum for HMM-based speech synthesis," IEICE Trans. Inf. & Syst., vol.J85-D-II, no.4, pp.545-553, Apr. 2002 (in Japanese).

56. Kazuhito Koishida, Keiichi Tokuda, Takashi Masuko and Takao Kobayashi, "Vector quantization of speech spectral parameters using statistics of static and dynamic features," IEICE Trans. Inf. & Syst., vol.E84-D, no.10, pp.1427-1434, Oct. 2001.

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59. Takashi Masuko, Masatsune Tamura, Keiichi Tokuda, and Takao Kobayashi, "Voice characteristics conversion for HMM-based speech synthesis system using MAP-VFS," IEICE Trans. Inf. & Syst., vol.J83-D-II, no.12. pp.2509–2516, Dec. 2000 (in Japanese).

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63. Oscar Vanegas, Keiichi Tokuda, Tadashi Kitamura, "Lip location normalized training for visual speech recognition," IEICE Trans. Inf. & Syst., vol.E83-D, no.11, pp.1969–1977, Nov. 2000.

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65. Takashi Masuko, Keiichi Tokuda, Noboru Miyazaki and Takao Kobayashi, "Pitch pattern generation using multi-space probability distribution HMM," IEICE Trans., vol.J83-D-II, no.7, pp.1600–1609, July 2000 (in Japanese).

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173. Shinji Sako, Keiichi Tokuda, Takashi Masuko, Takao Kobayashi, Tadashi Kitamura, "HMM-based text-to-audio-visual speech synthesis," International Conference on Spoken Language Processing (ICSLP2000/INTERSPEECH2000), vol.III, pp.25–28, Beijing, China, Oct. 16–20, 2000 (invited session).

174. Chiyomi Miyajima, Keiichi Tokuda, Tadashi Kitamura, "Audio-visual speech recognition using MCE-based HMMs and model-dependent stream weights," International Conference on Spoken Language Processing (ICSLP2000/INTERSPEECH2000), vol.II, pp.1023–1026, Beijing, China, Oct. 16–20, 2000.

175. Takashi Masuko, Keiichi Tokuda and Takao Kobayashi, "Imposture using synthetic speech against speaker verification based on spectrum and pitch," International Conference on Spoken Language Processing (ICSLP2000/INTERSPEECH2000), vol.III, pp.302–305, Beijing, China, Oct. 16–20, 2000.

176. Toru Takahashi, Keiichi Tokuda, Takao Kobayashi, Tadashi Kitamura, "Vector quantization of mel-cepstral coefficients based on a statistical measure," IEEE International Synposium on Intelligent Signal Processing and Communication Synstems, Honolulu, Hawaii, pp.692–695, Nov. 5-8, 2000.

177. Junibakti Sanubari, Keiichi Tokuda, "Robust estimation of an AR multi-channel model by using t-distribution assumption," Proc. The 10th European Signal Processing Conference (EUSIPCO-2000), vol.3, pp.1783–1786, Tempere, Finland, 4–8, Sep. 2000.

178. Yoshihiko Nankaku, Keiichi Tokuda, Takao Kobayashi, Tadashi Kitamura, "Normalized training for HMM-based visual speech recognition," Proc. of IEEE International Conferece on Image Processing, Vancouver, Canada, vol.3, pp.234–237, Sep. 2000.

179. Keiichi Tokuda, Takayoshi Yoshimura, Takashi Masuko, Takao Kobayashi, Tadashi Kitamura, "Speech parameter generation algorithms for HMM-based speech synthesis," Proceedings of IEEE International Conference on Acoustics, Speech, and Signal Processing, Istanbul, Turkey, vol.3, pp.1315–1318, June 2000.

180. Junibakti Sanubari, Keiichi Tokuda, "Image modeling using two dimentional exponential systems," IEEE International Conference on Image Processing, Kobe, Japan, pp.28AP4.8.1–28AP4.8.4, Oct. 1999.

181. Junibakti Sanubari, Keiichi Tokuda, "Two dimensional adaptive filter based on t-distribution assumption and full-plane support," Proceedings of Thirty-Third Asilomar Conference on Signal, Systems, and Computers, Pcific Groove, California, Oct. 24–27, pp.815–819, 1999.

182. Oscar Vanegas, Keiichi Tokuda, Tadashi Kitamura, "Location normalization of HMM-based lip reading: Experiments for the M2VTS Database," IEEE International Conference on Image Processing, Kobe, Japan, Oct. 1999 (accepted).

183. Takayoshi Yoshimura, Keiichi Tokuda, Takashi Masuko, Takao Kobayashi and Tadashi Kitamura, "Simultaneous modeling of spectrum, pitch and duration in HMM-based speech synthesis," Proceedings of European Conference on Speech Communication and Technology, Budapest, Hungary, vol.5, pp.2347–2350, Sep. 1999.

184. Yoshihiko Nankaku, Keiichi Tokuda and Tadashi Kitamura, "Intensity- and location-normalized training for HMM-based visual speech recognition," Proceedings of European Conference on Speech Communication and Technology, Budapest, Hungary, vol.3, pp.1287–1290, Sep. 1999.

185. Takashi Masuko, Takafumi Hitotsumatsu, Keiichi Tokuda and Takao Kobayashi, "On the security of HMM-based speaker verification systems against imposture using synthetic speech," Proceedings of European Conference on Speech Communication and Technology, Budapest, Hungary, vol.3, pp.1223–1226, Sep. 1999.

186. Junibakti Sanubari, Keiichi Tokuda, “LMS-like two dimensional adaptive filter with t-distribution assumption and non-symmetral half plane support,” Proceedings of IEEE-EURASIP Workshop on Nonlinear and Image Processing, Antalaya, Turkey, pp.419–423, June 1999.

187. Keiichi Tokuda, Takashi Masuko, Noboru Miyazaki and Takao Kobayashi, “Hidden Markov models based on multi-space probability distribution for pitch pattern modeling,” Proceedings of IEEE International Conference on Acoustics, Speech, and Signal Processing, vol.1, pp.229–232, Phoenix, USA, Mar. 1999.

188. Masatsune Tamura, Takashi Masuko, Takao Kobayashi and Keiichi Tokuda, “Visual speech synthesis based on parameter generation from HMM: speech-driven and text-and-speech-driven approach,” Proc. International Conference of Auditory-Visual Speech Processing, pp.219–224, Terrigal, Australia, Dec. 1998.

189. Kazuhito Koishida, Goh Hirabayashi, Keiichi Tokuda, and Takao Kobayashi, “A 16kbit/s wideband CELP coder using mel-generalized cepstral analysis and its subjective evaluation,” Proc. of International Conference on Spoken Language Processing (ICLSP-98), vol.6., pp.2583–2586, Sydney, Australia, Nov.–Dec., 1998.

190. Oscar Vanegas, Akiji Tanaka, Keiichi Tokuda, Tadashi Kitamura, “HMM-based visual speech recognition using intensity and location normalization,” Proc. of International Conference on Spoken Language Processing (ICLSP-98), vol.2, pp.789–792, Sydney, Australia, Nov.–Dec., 1998.

191. Takayoshi Yoshimura, Keiichi Tokuda, Takashi Masuko, Takao Kobayashi and Tadashi Kitamura, ”Duration modeling for HMM-based speech synthesis,” Proc. of International Conference on Spoken Language Processing (ICLSP-98), vol.2, Sydney, Australia, pp.29–32, Nov.–Dec., 1998.

192. Takashi Masuko, Takao Kobayashi and Keiichi Tokuda, ”A very low bit rate speech coder using HMM with speaker adaptation,” Proc. of International Conference on Spoken Language Processing (ICLSP-98), vol.2, pp.507–510, Sydney, Australia, Nov.–Dec., 1998.

193. Masatsune Tamura, Takashi Masuko, Keiichi Tokuda and Takao Kobayashi, ”Speaker adaptation for HMM-based speech synthesis system using MLLR,” Proc. ESCA/COCOSDA Workshop on Speech Synthesis, pp.273–276, Blue Mountains, Australia, Nov. 1998.

194. Oscar Vanegas, Akiji Tanaka, Keiichi Tokuda, Tadashi Kitamura, “Intensity/location normalization for automatice lipreading,” Proc. International Conference on Signal Processing, vol.2, pp.920–923, Oct. 1998.

195. Junibakti Sanubari, Keiichi Tokuda, “Recursive two dimensional spectral estimation based on an AR model excited by a t-distribution process using AR decomposition approach,” Proceedings of IEEE Asia Pacific Conference on Circuits and Systems, Chiang Mai, Thailand, pp.447–450, Nov. 1998.

196. Junibakti Sanubari, Keiichi Tokuda, “Adaptive two dimensional filter based on an AR model excited by a t-distribution process,” Proceedings of IESTED International Conference on Signal and Image Processing, Las Vegas, Nevada-USA, pp.679–683, Oct. 1998.

197. Junibakti Sanubari, Keiichi Tokuda, “Adaptive spectral estimation based on an exponential model,” Proceedings of IEEE International Conference on Circuits and Systems, Monterey, California, pp.TAA13-11.1–TAA13-11.4, May 1998.

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199. Kazuhito Koishida, Goh Hirabayashi, Keiichi Tokuda, and Takao Kobayashi, “A wideband CELP speech coder at 16 kbit/s based on mel-generalized cepstral analysis,” Proceedings of IEEE International Conference on Acoustics, Speech, and Signal Processing, vol.1, pp.161–164, Seattle, USA, May 1998.

200. Takashi Masuko, Takao Kobayashi, Masatsune Tamura, Jun Masubuchi, and Keiichi Tokuda, “Text-to-visual speech synthesis based on parameter generation from HMM,” Proceedings of IEEE International Conference on Acoustics, Speech, and Signal Processing, vol.6, pp.3745–3748, Seattle, USA, May 1998.

201. Junibakti Sanubari and Keiichi Tokuda, “Non stationary spectral estimation based on robust time varying AR model excited by a t-distribution process,” IEEE Region Ten Annual Conference on Speech and Image Technologies for Computing and Telecommunications, Brisbane, Australia, pp.51–54, Dec. 1997.

202. Takao Kobayashi, Takashi Masuko and Keiichi Tokuda, “HMM compensation for noisy speech recognition based on cepstral parameter generation,” Proceedings of European Conference on Speech Communication and Technology, vol.3, pp.1583–1586, Rhodes, Greece, Sep. 1997.

203. Takayoshi Yoshimura, Takashi Masuko, Keiichi Tokuda, Takao Kobayashi and Tadashi Kitamura, “Speaker interpolation in HMM-based speech synthesis system,” Proceedings of European Conference on Speech Communication and Technology, vol.5, pp.2523–2526, Rhodes, Greece, Sep. 1997.

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205. Chiyomi Miyajima, Yoshitsuna Sugiura, Keiichi Tokuda and Tadashi Kitamura, “Descrete or tied-mixture HMM based of sefl-organizing feature map for robust probability estimation,” Proceedings of International Conference on Speech Processing, vol.2, pp.529–532, Seoul, Korea, Aug. 1997.

206. Kazuhito Koishida, Keiichi Tokuda, Takashi Masuko and Takao Kobayashi, “Vector quantization of speech spectral parameters using statistics of dynamic features,” Proceedings of International Conference on Speech Processing, vol.1, pp.247–252, Seoul, Korea, Aug. 1997.

207. Junibakti Sanubari, Keiichi Tokuda, “Robust spectral estimation based on an AR model excited by a t-distribution process by using QR decomposition algorithm,” Proceedings of IEEE International Conference on Circuits and Systems, pp.2497–2500, June 1997.

208. Fernando G. Resende, Keiichi Tokuda and Mineo Kaneko, “Multi-band decomposition of the linear prediction error applied to the least-mean squares method with fixed and variable step-sizes,” Proceedings of IEEE International Conference on Circuits and Systems, pp.2176–2179, June 1997.

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210. Kazuhito Koishida, Keiichi Tokuda, Takao Kobayashi and Satoshi Imai, “Efficient encoding of mel-generalized cepstrum for CELP coders,” Proceedings of IEEE International Conference on Acoustics, Speech, and Signal Processing, vol.2, pp.1355–1358, Munichi, Germany, Apr. 1997.

211. Yasuichi Hamano, Keiichi Tokuda and Mineo Kaneko, “Image restoration based on estimation of fractal structure,” IEEE Region Ten Conference (Digital Signal Processing Applications), pp.311–316, Nov. 1996.

212. Fernando G. Resende, Keiichi Tokuda and Mineo Kaneko, “RLS algorithms for adaptive AR spectrum analysis based on multi-band decomposition of the linear prediction error,” IEEE Region Ten Conference (Digital Signal Processing Applications), pp.541-546, Nov. 1996.

213. Kazuhito Koishida, Keiichi Tokuda, Takao Kobayashi and Satoshi Imai, “CELP coding system based on mel-generalized cepstral analysis,” Proceedings of International Conference on Spoken Language Processing, Philadelphia, USA, vol.1, pp.314–317, Oct. 1996.

214. Junibakti Sanubari, Keiichi Tokuda and Mahoki Onoda, “The application of t-distribution assumption for robust spectral estimation,” Proc. of the IASTED International Conferences on Signal, Image Processing and Application, SIPA-96, Annecy, France, pp.217–220, June 1996.

215. Junibakti Sanubari, Keiichi Tokuda, Mahoki Onoda, “Robust two-dimensional spectral estimation based on an AR model excited by a t-distribution process,” Proceedings of IEEE International Conference on Acoustics, Speech, and Signal Processing, vol.5, pp.2998–3001, Atlanta, USA, May 1996.

216. Takashi Masuko, Keiichi Tokuda, Takao Kobayashi and Satoshi Imai, “Speech synthesis from HMMs using dynamic features,” Proceedings of IEEE International Conference on Acoustics, Speech, and Signal Processing, vol.1, pp.389–392, May 1996.

217. Fernando G. Resende, Keiichi Tokuda and Mineo Kaneko, “A fast algorithm for adaptive AR spectral estimation based on multi-scale decomposition of linear prediction error,” Proceedings of Midwest Symposium on Circuits and Systems, pp.119–122, Aug. 1995.

218. Keiichi Tokuda, Takashi Masuko, Tetsuya Yamada, Takao Kobayashi and Satoshi Imai, “An algorithm for speech parameter generation from continuous mixture HMMs with dynamic features,” Proceedings of European Conference on Speech Communication and Technology, vol.1, pp.757–760, Sep. 1995.

219. Keiichi Tokuda, Takao Kobayashi and Satoshi Imai, "Speech parameter generation from HMM using dynamic features," Proceedings of IEEE International Conference on Acoustics, Speech, and Signal Processing, vol.1, pp.660–663, May 1995.

220. Kazuhito Koishida, Keiichi Tokuda, Takao Kobayashi and Satoshi Imai, "CELP coding based on mel-cepstral analysis," Proceedings of IEEE International Conference on Acoustics, Speech, and Signal Processing, vol.1, pp.33–36, May 1995.

221. Kazuhito Koishida, Keiichi Tokuda, Takao Kobayashi and Satoshi Imai, "Speech coding based on adaptive mel-cepstral analysis for noisy channels," Proceedings of International Conference on Spoken Language Processing, vol.4, pp.2087–2090, Sep. 1994.

222. Keiichi Tokuda, Takao Kobayashi, Takashi Masuko and Satoshi Imai, "Mel-generalized cepstral analysis —a unified approach to speech spectral estimation," Proceedings of International Conference on Spoken Language Processing, vol.3, pp.1043–1046, Sep. 1994.

223. Fernando G. Resende, Keiichi Tokuda and Mineo Kaneko, "AR spectrum estimation based on wavelet representation," Proceedings of IEEE International Conference on Circuits and Systems, vol.2, pp.625–628, June 1994.

224. Junibakti Sanubari, Keiichi Tokuda, Mahoki Onoda, "Robust recursive spectral estimation based on AR model excited by a t-distribution process," Proceedings of IEEE International Conference on Acoustics, Speech, and Signal Processing, vol.3, pp.497–500, Apr. 1994.

225. Keiichi Tokuda, Hidetoshi Matsumura, Takao Kobayashi and Satoshi Imai, "Speech coding based on adaptive mel-cepstral analysis," Proceedings of IEEE International Conference on Acoustics, Speech, and Signal Processing, vol.1, pp.197–200, Apr. 1994.

226. Junibakti Sanubari, Keiichi Tokuda and Mahoki Onoda, "Spectral estimation based on AR model excited by t-distribution process," Proceedings of IEEE International Conference on Acoustics, Speech, and Signal Processing, vol.5, pp.521–524, Mar. 1992.

227. Takao Kobayashi, Kazuyoshi Fukushi, Keiichi Tokuda and Satoshi Imai, "Design of stable two-dimensional IIR digital filters with arbitrary magnitude function," Proceedings of IEEE International Conference on Acoustics, Speech, and Signal Processing, vol.5, pp.93–96, Mar. 1992.

228. Toshiaki Fukada, Keiichi Tokuda, Takao Kobayashi and Satoshi Imai, "An adaptive algorithm for mel-cepstral analysis of speech," Proceedings of IEEE International Conference on Acoustics, Speech, and Signal Processing, vol.1, pp.137–140, Mar. 1992.

229. Keiichi Tokuda, Takao Kobayashi and Satoshi Imai, "Generalized cepstral analysis of speech —unified approach to LPC and cepstral method," Proceedings of International Conference on Spoken Language Processing, pp.37–40, Nov. 1990.

230. Keiichi Tokuda, Takao Kobayashi, Shoji Shiomoto and Satoshi Imai, "Adaptive filtering based on cepstral representation —adaptive cepstral analysis of speech," Proceedings of IEEE International Conference on Acoustics, Speech, and Signal Processing, pp.377–380, Apr. 1990.

UNREFEREED CONFERENCE AND WORKSHOP PAPERS

More than 400 unrefereed papers have been published. For the complete list, see

http://www.sp.nitech.ac.jp/˜tokuda/gyoseki-j/node5.html
http://www.sp.nitech.ac.jp/˜tokuda/gyoseki-j/node6.html
http://www.sp.nitech.ac.jp/˜tokuda/gyoseki-j/node7.html

5 SOFTWARE ARTIFACTS

The HMM-based Speech Synthesis System (HTS) http://hts.sp.nitech.ac.jp/

The Speech Signal Processing Toolkit (SPTK) http://sp-tk.sourceforge.net/

Multimodal Speech Database (M2TINIT) http://m2tinit.sp.nitech.ac.jp/

The Galatea Toolkit http://hil.t.u-tokyo.ac.jp/˜galatea/

hts_engine API http://hts-engine.sourceforge.net/

HMM-based Japanese TTS system (Open JTalk) http://open-jtalk.sourceforge.net/

A toolkit for building voice interaction systems (MMDAgent) http://www.mmdagent.jp/

HMM-based singing voice synthesis system (Sinsy) http://sinsy.sourceforge.net/

6 EXTERNAL TEACHING EXPERIENCE

Nov. 29, 2008
Invited Lecturer, Meijo University, Japan

Aug. 8, 2003
LTI Seminar, Language Technologies Institute, Carnegie Mellon University

Mar. 12–13, 2002
International Lecture Series on Signal Processing, Tamkang University Lecture Series of EECS, Tamkang University, R.O.C.

Feb. 8, 2002
LTI Seminar, Language Technologies Institute, Carnegie Mellon University

Dec. 4, 2001
Speech and Signal Processing Seminar, Department of Electrical Engineering, University of Washington

April 2001–September 2001
Invited Lecturer, Nagoya University, Japan

November 1997–March 1998
Invited Lecturer, Tokyo Institute of Technology, Japan

7 INVITED TALKS

AT INTERNATIONAL CONFERENCES

Keiichi Tokuda, "Statistical Approach to Speech Synthesis: Past, Present and Future," Interspeech 2019, September, 2019.

Keiichi Tokuda, "Flexible speech synthesis based on hidden Markov models," Asia-Pacific Signal and Information Processing Association Annual Summit and Conference 2013 (APSIPA ASC 2013), Kaohsiung, Taiwan, October 29–November 1, 2013.

Keiichi Tokuda, "Speech Synthesis as A Statistical Machine Learning Problem," IEEE 2011 Automatic Speech Recognition and Understanding Workshop (ASRU 2011), Hawaii, USA, December 11–15, 2011.

Keiichi Tokuda, "Speech Synthesis as A Machine Learning Problem," The International Committee for the Co-ordination and Standardization of Speech Databases and Assessment Techniques (Oriental COCOSDA 2010), Kathmandu, Nepal, Nov. 24–25, 2010.

Keiichi Tokuda, Heiga Zen, "Fundamentals and recent advances in HMM-based speech synthesis," Interspeech 2009, Tutorial, Brighton, U.K., September 6, 2009.

Keiichi Tokuda, "An HMM-based approach to flexible speech synthesis," The 5th International Symposuim on Chinese Spoken Language Processing (ISCSLP 2006), Kent Ridge, Singapore, December 13–16, 2006 (tutorial).

Keiichi Tokuda, "Hidden Markov model-based speech synthesis as a tool for constructing comunicative spoken dialog systems," Proc. of The 4th Joint Meeting of The Acoustical Society of America and The Acoustical Society of Japan, Honolulu, Hawaii, 28 November–2 December 2006 (in J. Acoust. Soc. Am., vol.120, no.5, Part.2, p.3006, November 2006) (invited talk).

Keiichi Tokuda, "The trajectory HMM: Reformulating the HMM as a trajectory model," Trajectory Models for Speech Processing, Supported by EPSRC (The Engineering and Physical Sciences Research Council, U.K.), Edinburgh, U.K., August 31, 2005 (keynote).

More than 16 talks were given at domestic conferences

AT COMPANIES, UNIVERSITIES, AND OTHER WORKSHOPS

The University of Cambridge, CUED Speech Group Seminars, Cambridge, U.K., "Human-like singing and talking machines: flexible speech synthesis in karaoke, anime, smart phones, video games, digital signage, TV and radio programs, etc.," February 6, 2015.

The University of Sheffield, Speech and Hearing Seminars, U.K., "Human-like singing and talking machines," January 13, 2015.

Massachusetts Institute of Technology, Human Language Technology Distinguished Lecture Series, MA, U.S.A, "Human-like Singing and Talking Machines: Flexible Speech Synthesis in Karaoke, Anime, Smart Phones, Video Games, Digital Signage, TV and Radio Programs," December 4, 2014.

5th EU-Japan Symposium in ICT Research and Innovation, Brussels, Belgium, “Statistical approach to flexible speech synthesis –toward human-like talking machines,” October 16–17, 2014.

University of Science and Technology of China, Talk, “Fundamentals and recent advances in HMM-based speech synthesis,” October 30, 2008.

University of Science and Technology of China, “HMM-Based Speech Synthesistoward human-like talking machines,” March 6, 2007.

Microsoft Research Asia, Beijing, China, “An HMM-based approach to automatic voice building,” March 28, 2006.

University of Cambridge, Department of Engineering, Speech Vision and Robotics Group, Cambridge, U.K., “HMM-based speech synthesis —toward Human-like Talking Machines,” Apr. 26, 2002.

Carnegie Mellon University, PA, U.S.A., Talk at Sphinx Lunch, “HMM-based speech synthesis —toward Human-like Talking Machines,” Apr. 18, 2002.

Cepstral, PA, U.S.A., , “HMM-based speech synthesis —toward Human-like Talking Machines,” Apr. 10, 2002.

Lucent Technologies, Bell Labs, NJ, U.S.A., “HMM-based speech synthesis —toward Human-like Talking Machines,” Apr. 2, 2002.

AT&T Labs Research, “HMM-based speech synthesis —toward Human-like Talking Machines,” Mar. 26, 2002.

Nuance Communications, CA, U.S.A., “HMM-based speech synthesis —toward human-like talking machines,” Feb. 28, 2002.

Universidade Federal do Rio de Janeiro, Signal Processing Laboratory, Rio de Janeiro, Brazil, “HMM-based speech synthesis —toward human-like talking machines,” Feb. 19, 2002.

IBM Research, Watson Research Center, NY, U.S.A., “HMM-based speech synthesis —toward human-like talking machines,” Feb. 12, 2002.

Oregon Graduate Institute, Center for Spoken Language Understanding, OR, U.S.A., , “HMM-based speech synthesis —toward human-like talking machines,” Dec. 6, 2001.

Microsoft Research, WA, U.S.A., “HMM-based speech synthesis —toward Human-like Talking Machines,” Nov. 30, 2001.

More than 14 talks were given at domestic companies and universities.

8 PROFESSIONAL ACTIVITITIES

CONFERENCE AND WORKSHOP COMMITTEES

The Blizzard Challenge Workshop Co-organizer, Vienna, Austria, September 23, 2019.

Interspeech 2018 satelite workshop "The Blizzard Challenge Workshop 2018," Co-organizer, Hyderabad, India, September 8, 2018.

The Blizzard Challenge Workshop Co-organizer, Stockholm, Sweden, August 25, 2017 (satellite event of Interspeech 2017).

The Blizzard Challenge Workshop Co-organizer, California, USA, September 16, 2016 (satellite event of 9th ISCA Speech Synthesis Workshop and Interspeech 2016).

9th ISCA Speech Synthesis Workshop Program Committee Member, California, USA, September 13–15, 2016.

Interspeech 2015 satellite workshop "The Blizzard Challenge 2015" Co-organizer, Berlin, Germany, September 11, 2015.

Interspeech 2014 satellite workshop "The Blizzard Challenge 2014: Evaluating corpus-based speech synthesis on common databases," Co-organizer, Singapore, Singapore, September 19, 2014.

The Blizzard Challenge Workshop Co-organizer, Barcelona, Spain, September 3, 2013 (satellite event of 8th ISCA Speech Synthesis Workshop).

8th ISCA Speech Synthesis Workshop Advisory Committee Member, Barcelona, Spain, August 31–September 2, 2013.

Interspeech 2012 satellite workshop "The Blizzard Challenge 2008: Evaluating corpus-based speech synthesis on common databases," Co-organizer, Portland, Oregon, USA, September 14, 2012.

Interspeech 2011 satellite workshop "The Blizzard Challenge 2008: Evaluating corpus-based speech synthesis on common databases," Co-organizer, Turin, Italy, September 2, 2011.

Interspeech 2010 Special Session Chair, Makuhari, Japan, September 26–30, 2010.

The Blizzard Challenge Workshop Co-organizer, Kyoto, Japan, September 25, 2010 (satellite event of 7th ISCA Speech Synthesis Workshop).

7th ISCA Speech Synthesis Workshop Co-Chair, Kyoto, Japan, September 22–24, 2010.

Interspeech 2009 satellite workshop "The Blizzard Challenge 2008: Evaluating corpus-based speech synthesis on common databases," Co-organizer, Edinburgh, U.K., September 4, 2009.

ACL-08 Area Chair, Columbus, Ohio, USA, June 15–20, 2008.

Interspeech 2008 satellite workshop "The Blizzard Challenge 2008: Evaluating corpus-based speech synthesis on common databases," Co-organizer, Brisbane, Australia, September 21, 2008.

IEEE ICASSP 2007 "Statistical Parametric Speech Synthesis," Special Session Co-organizer, Hawaii, USA, April 15–20, 2007.

The Blizzard Challenge Workshop Co-organizer, Bonn, Germany, August 25, 2007 (satellite event of 6th ISCA Speech Synthesis Workshop).

6th ISCA Speech Synthesis Workshop Scientific Committee Member, Bonn, Germany, August 22–24, 2007.

Interspeech 2006 satellite workshop "The Blizzard Challenge 2006: Evaluating corpus-based speech synthesis on common databases," Co-organizer, Pittsburgh, PA, U.S.A., September 16, 2006.

Interspeech 2005 - Eurospeech "The Blizzard Challenge 2005: Evaluating corpus-based speech synthesis on common databases," Special Session Co-organizer, Lisbon, Portugal, September 4–8, 2005.

5th ISCA Speech Synthesis Workshop Scientific Committee Member, Pittsburgh, PA, U.S.A., 14–16 June 2004.

IEEE 2002 Speech Coding Workshop Organizing Committee Member, Tsukuba, Ibaraki, Japan, 6–9 Oct, 2002.

Session Chair of numerous international conferences and symposiums, including ICASSP, Interspeech.

Reviewer of numerous journals and international conferences, including IEEE Transactions on Speech and Audio, Speech Communication, ICASSP, Interspeech.

PROFESSIONAL SOCIETIES

Scientific Advisory Board Member of an EPSRC Programme Grant "Natural Speech Technology (NST)," May 2011-April 2016.

IEEE Journal of Selected Topics in Signal Processing, Guest Editor, 2013.

IEEE Signal Processing Magazine, Guest Editor, 2012.

ISCA Advisory Council member, January 1, 2009–December 31, 2012.

Associate Editor of IEEE Transactions on Audio, Speech & Language Processing, Jan 1, 2009 – Jan 1 2012.

European Science Foundation (ESF) peer reviewer, 2007–present.

Minstry of Economy, Trade and Industry (METI), Japan, Project: "Development of technologies for the use of home information appliance sensors and human interface devices," Academic Committee Member, 2007–2009

Acoustic Society of Japan, Member of The Board of Trustees, 2007–2008

Institute of Electronics, Information and Communication Engineers (IEICE), Guest Editor, 2005

The Robotics Society of Japan, Technical Research Groups for Robotic Audition, Committee Member, 2003–present

Acoustic Society of Japan, Associate Editor, 2003–2005

Institute of Electronics, Information and Communication Engineers (IEICE), Associate Editor, 2002–2006

Institute of Electrical and Electronics Engineers (IEEE), Signal Processing Society, Speech Technical Committee Member, 2000–2003

Institute of Electronics, Information and Communication Engineers (IEICE), Speech Technical Committee Member, 1999–2005

Japanese Society for Artificial Intelligence, Member of The Board of Trustees, 1998–2001

9 STUDENTS ADVISING

PHD STUDENTS AS PRINCIPAL SUPERVISOR

Heiga Zen, 2006

Ranniery deSilva Maia, 2006

Keiichiro Oura, 2010

Ryuta Terashima, 2010

Kei Hashimoto, 2011

Sayaka Shiota, 2012

Shinji Takaki, 2014

Akira Tamamori, 2014

Kazuhiro Nakamura, 2014

Kei Sawada, 2018

Takenori Yoshimura, 2018

10 ANY OTHER RELEVANT INFORMATION

VISITING RESEARCHERS AND STUDENTS

Thomas Hain, University of Sheffield, U.K., 2018 (2 months)

Xin Wang, National Institute of Informatics, Japan, 2018 (3 weeks)

Simon King, University of Edinburgh, U.K., 2017 (2 months)

Amelia Gully, University of York, U.K., 2016 (2 months)

Anocha Rugchatjaroen, NECTEC, Thailand, 2016 (6 weeks)

Sittipong Saychum, NECTEC, Thailand, 2016 (6 weeks)

Rasmus Dall, University of Edinburgh, U.K., 2015 (6 months)

Hongwu Yang, Northwest Normal University, China, 2011 (1 year)

Ching-Tang Hsieh, Tamkang University, China, 2012 (3 months)

Alan W. Black, Carnegie Mellon University, U.S.A., 2012 (3 weeks)

Mikko Kurimo, Helsinki University of Technology, Finland, 2010 (2 months)

Shifeng Pan, Chinese Academy of Sciences, China, 2010 (3 months)

Ulpu Remes, Helsinki University of Technology, Finland, 2010 (8 months)

Linghui Chen, University of Science and Technology of China, China, 2010 (5 months)

Alan W. Black, Carnegie Mellon University, U.S.A., 2010 (2 weeks)

Jun Xu, Tsinghua University, China, 2008 (6 months)

Heng Lu, University of Science and Technology of China, China, 2008 (2 months)

Pascual Martínez Gomez, Universitat Politècnica de València, Spain, 2008 (1 year)

Alan W. Black, Carnegie Mellon University, U.S.A., 2008 (1 month)

Simon King, University of Edinburgh, U.K., 2007 (4 months)

Alan W. Black, Carnegie Mellon University, U.S.A., 2007 (2 weeks)

Alan W. Black, Carnegie Mellon University, U.S.A., 2006 (2 weeks)

Maria João Barros, Universidade Do Porto, Portugal, 2005 (3 months)

Christian Ants Weiss, Universität Bonn, Germany, 2005 (6 months)

Alan W. Black, Carnegie Mellon University, U.S.A., 2005 (1 week)

Fernando Gil Vianna Resende Junior, Universidade Federal do Rio de Janeiro, Brazil, 2004 (4 months)

Maria João Barros, Universidade Do Porto, Portugal, 2004 (1 month)

Fernando Gil Vianna Resende Junior, Universidade Federal do Rio de Janeiro, Brazil, 2003 (4 months)

Fernando Gil Vianna Resende Junior, Universidade Federal do Rio de Janeiro, Brazil, 2000 (2 months)

Junibakti Sanubari, Satya Wacana University, Indonesia, 1997 (1 year)