

CURRICULUM VITAE

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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

April 2006–present

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 Carnegie 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

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

PUBLICATION

BOOK CHAPTERS

1. 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)
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3. 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).
4. 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).
5. Heiga Zen, Keiichi Tokuda, "TechWare: HMM-Based Speech Synthesis Resources," *IEEE Signal Processing Magazine*, July 2009 (in Japanese).
6. 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).
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43. Chiyomi Miyajima, Hideyuki Watanabe, Keiichi Tokuda, Tadashi Kitamura, Shigeru Katagiri, "A New Approach to Designing a Feature Extractor in Speaker Identification Based on Discriminative Feature Extraction," *Speech Communication*, vol.35, no.3-4, pp.203–218, Oct. 2001.
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UNREFEREED CONFERENCE AND WORKSHOP PAPERS

More than 300 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>

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/>

AWARDS

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

EXTERNAL TEACHING EXPERIENCE

Dec. 4, 2001

Speech and Signal Processing Seminar, Department of Electrical Engineering, University of Washington

Feb. 8, 2002

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.

April 2001–September 2001

Invited Lecturer, Nagoya University, Japan

November 1997–March 1998

Invited Lecturer, Tokyo Institute of Technology, Japan

INVITED TALKS

AT CONFERENCES

Keiichi Tokuda, “Speech Synthesis as A Statistical Machine Learning Problem,” IEEE 2011 Automatic Speech Recognition and Understanding Workshop (ASRU 2011),

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

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, “An HMM-based approach to flexible speech synthesis,” The 5th International Symposium 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 communicative 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).

AT COMPANIES AND UNIVERSITIES

University of Science and Technology of China, “HMM-Based Speech Synthesis toward 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.

Other 17 talks were given at domestic conferences and companies.

PROFESSIONAL ACTIVITIES

CONFERENCE AND WORKSHOP COMMITTEES

ACL-08 Area Chair, Columbus, Ohio, USA, June 15–20, 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 5th 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

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.

Ministry 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