个人简历

凌震华，男，生于 1979 年 8 月，安徽省合肥市人。中国科学技术大学信号与信息处理专业博士，现任中国科学技术大学电子工程与信息科学系副教授、博士生导师。主要研究方向为语音信号处理。

教育经历

2005/9 - 2008/6　中国科学技术大学，信号与信息处理，博士，导师：王仁华

2002/9 - 2005/6　中国科学技术大学，信号与信息处理，硕士，导师：王仁华

1996/9 - 2002/6　中国科学技术大学，电子信息工程，学士，导师：王仁华

工作经历

2011/1 - 至今　中国科学技术大学，语音及语言信息处理国家工程实验室/电子工程与信息科学系，副教授

2012/8 - 2013/8　美国华盛顿大学，电子工程系，访问学者

2008/7 – 2010/12　中国科学技术大学/科大讯飞股份有限公司，博士后

2007/10 - 2008/4　英国爱丁堡大学，语音技术研究中心，访问学者

科研项目

项目类别	批准号	名称	起止年月	金额(万元)	项目状态	负责或参加
工信部电子信息产业发展基金	工信部财[2014]425 号	面向少数民族语言的智能语音技术及系统研发	2014/10-2016/10	100	在研	子课题负责人
安徽省科技攻关项目	11010202188	面向高级人工智能领域的类人智能关键技术及系统研发	2014/06-2016/06	200	在研	负责
国家自然科学基金青年-面上连续资助项目	61273032	融合语音产生机理与统计声学建模的层次化语音合成方法	2013/01-2016/12	80	在研	负责
国家自然科学基金委-英国爱丁堡皇家学会国际合作交流项目	61111130120	用于灵活可控语音合成的发音动作参数-声学参数联合建模方法研究	2011/01-2012/12	8.6	结题	负责

国家自然科学基金青年科学基金项目	60905010	结合发音动作参数的统计建模语音合成方法研究	2010/01-2012/12	19	结题	负责
中国博士后科学基金面上项目	20090450823	基于合成语音质量评估的语音合成方法研究	2009/07-2010/10	3	结题	负责
工信部电子信息产业发展基金	工信部财[2013]472 号	基于自然语音人机交互的信息搜索系统研发和产业化	2013/01-2014/12	180	结题	参加
科技部 973 前期研究专项	2012CB326405	声纹识别及声音转换深度学习理论与方法	2012/12-2014/12	61	结题	参加

科研获奖

- 《智能语音交互关键技术及应用开发平台》，国家科技进步奖二等奖（总排名第9，本单位排名第3），2011年
- IEEE信号处理学会最佳青年作者论文奖（IEEE Signal Processing Society Young Author Best Paper Award），2010年
- 《智能语音交互关键技术及应用平台》，安徽省科技进步奖一等奖（排名第9），2008年
- 《面向口语学习的中英文智能语音评测与学习技术》，中国电子学会电子信息科学技术奖二等奖（排名第7），2008年

学术兼职

- IEEE/ACM Transactions on Audio, Speech, and Language Processing(IEEE/ACM TASLP)副编辑(Associate Editor), 2014/01-2017/01
- 国际语音通信协会(ISCA)通讯委员会委员(Communication Committee Member)
- 中国语言学会语音学分会学术委员会委员
- 全国人机语音通讯学术会议常设机构委员会委员
- 2015年口语语音资料库协调暨标准化国际研讨会东方分会和亚太口语语言研究和评价会议（Oriental COCOSDA/CASLRE）科学委员会 (Scientific Committee) 成员
- 2015年IEEE信号与信息处理中国高峰会议（ChinaSIP）技术程序委员会（Technical Program Committee）成员
- 2015年全国人机语音通讯学术会议分会主席
- 2014年IEEE声学、语音和信号处理国际会议(ICASSP)分会主席(Session

Chair)

- 2014年ISCA年会(Interspeech)会议分会主席(Session Chair)
- 2014年IEEE信号与信息处理中国高峰会议（ChinaSIP）技术程序委员会（Technical Program Committee）成员
- 2014年中文口语处理国际研讨会（ISCSLP）程序委员会（Program Committee）成员
- 2013年ISCA 语音合成研讨会(SSW)程序委员会 (Program Committee) 成员
- 2012年语音韵律国际会议(ICSP)科学委员会 (Scientific Committee) 成员
- 2012年ISCA年会(Interspeech)会议分会主席(Session Chair)
- IEEE会员；国际语音通信协会(ISCA)会员
- 任以下期刊审稿人
  - Proceedings of the IEEE
  - The Journal of Selected Topics in Signal Processing
  - IEEE/ACM Transactions on Audio Speech and Language Processing
  - IEEE Transactions on Neural Networks and Learning Systems
  - IEEE Transactions on Fuzzy Systems
  - Computer Speech and Language
  - Speech Communication
  - IEICE Transactions on Information and Systems
  - EURASIP Journal on Audio, Speech, and Music Processing
  - Information Sciences
  - Journal of Signal Processing Systems

特邀报告

邀请单位	**报告名称**	**报告时间及地点**
第 15 届日本口语处理研讨会	Acoustic Modeling Using Restricted Boltzmann Machines and Deep Belief Networks for Statistical Parametric Speech Synthesis and Voice Conversion	2013/12 日本东京
微软研究院	HMM-based Speech Synthesis: Fundamentals and Its Recent Advances	2012/09 美国雷蒙德
英国爱丁堡大学	Automatic Error Detection for Unit Selection Speech Synthesis Using Human Feedbacks	2011/08 英国爱丁堡

发明专利

专利名称	专利号	授权日期
一种基于受限玻尔兹曼机的语音合成方法	201310099895.4	2015.06.17
一种说话人声音转换方法	201210528629.4	2014.12.11
一种语音信号增强系统和方法	201210410212.8	2014.10.14
基于声学统计模型的单元挑选语音合成方法	2007101910786	2012.03.28
语音合成方法及装置	2009102228990	2011.08.03
一种个性化歌唱语音的合成方法	2008101071140	2011.06.29
一种结合自然样本挑选与声学参数建模的语音合成方法	2006100396752	2011.06.29
基于线谱频率及其阶间差分参数的频谱建模与语音增强方法	200610038589X	2010.05.12
一种结合高层描述信息和模型自适应的说话人转换方法	2006100396803	2010.05.12

发表论文

<u>国际期刊论文</u> (*通讯作者)

[1] Ya-Jun Hu, and **Zhen-Hua Ling*,** "DBN-based Spectral Feature Representation for Statistical Parametric Speech Synthesis," IEEE Signal Processing Letters, accepted.

[2] Xin Wang, **Zhen-Hua Ling***, and Li-Rong Dai, "Concept-to-Speech Generation with Knowledge Sharing for Acoustic Modelling and Utterance Filtering," Computer Speech and Language, accepted.

[3] Xiang Yin, Ming Lei, Yao Qian, Frank K. Soong, Lei He, **Zhen-Hua Ling***, and Li-Rong Dai, "Modeling F0 Trajectories in Hierarchically Structured Deep Neural Networks," Speech Communication, vol. 76, pp. 82-92, 2016.

[4] Korin Richmond, **Zhen-Hua Ling**, and Junichi Yamagishi, "The use of articulatory movement data in speech synthesis applications: an overview –Application of articulatory movements using machine learning algorithms–", Acoustical Science and Technology, vol. 36, no. 6, pp. 467-477, 2015.

[5] Ling-Hui Chen, Tuomo Raitio, Cassia Valentini-Botinhao, **Zhen-Hua Ling**, and Junichi Yamagishi, "A Deep Generative Architecture for Postfiltering in Statistical Parametric Speech Synthesis," IEEE/ACM Transactions on Audio, Speech, and Language Processing, vol. 23, no. 11, pp. 2003-2014, 2015.

[6] Ming-Qi Cai, **Zhen-Hua Ling***, Li-Rong Dai, "Statistical Parametric Speech Synthesis Using a Hidden Trajectory Model", Speech Communication, vol. 72, pp. 149-159, 2015.

[7] **Zhen-Hua Ling**, Shi-Yin Kang, Heiga Zen, Andrew Senior, Mike Schuster, Xiao-Jun Qian, Helen Meng, Li Deng, "Deep Learning for Acoustic Modeling in Parametric Speech Generation: A systematic review of existing techniques and future trends," IEEE Signal Processing Magazine, vol. 32, no. 3, pp. 35-52, 2015.

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[9] Jing-Jie Li, Ian V. McLoughlin, Li-Rong Dai, and **Zhen-Hua Ling**, "Whisper-to-speech conversion using restricted Boltzmann machine arrays," Electronics Letters, vol. 50, no. 24, pp. 1781-1782, 2014.

[10] Ling-Hui Chen, **Zhen-Hua Ling***, Li-Juan Liu, and Li-Rong Dai, "Voice Conversion Using Deep Neural Networks with Layer-Wise Generative Training," IEEE/ACM Transactions on Audio, Speech, and Language Processing, vol. 22, no. 12, pp. 1859-1872, 2014.

[11] Xian-Jun Xia, **Zhen-Hua Ling***, Yuan Jiang, and Li-Rong Dai, "HMM-based Unit Selection Speech Synthesis Using Log Likelihood Ratios Derived from Perceptual Data", Speech Communication, vol. 63-64, pp. 27-37, 2014.

[12] Chen-Yu Yang, **Zhen-Hua Ling**, and Li-Rong Dai, "Unsupervised Prosodic Labeling of Speech Synthesis Databases Using Context-Dependent HMMs", IEICE Transactions on Information and Systems, vol.E97-D, no.6, pp. 1449-1460, 2014.

[13] **Zhen-Hua Ling**, Li Deng, and Dong Yu, "Modeling Spectral Envelopes Using Restricted Boltzmann Machines and Deep Belief Networks for Statistical Parametric Speech Synthesis," IEEE Transactions on Audio, Speech, and Language Processing, vol. 21, no. 10, pp. 2129-2139, 2013.

[14] **Zhen-Hua Ling**, Korin Richmond, and Junichi Yamagishi, “Articulatory Control of HMM-based Parametric Speech Synthesis using Feature-Space-Switched Multiple Regression,” IEEE Transactions on Audio, Speech, and Language Processing, vol. 21, no. 1, pp. 207-219, 2013.

[15] **Zhen-Hua Ling**, and Li-Rong Dai, “Minimum Kullback-Leibler Divergence Parameter Generation for HMM-based Speech Synthesis,” IEEE Transactions on Audio, Speech, and Language Processing, vol. 20, no. 5, pp. 1492-1502, 2012.

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[17] Heng Lu, **Zhen-Hua Ling**, Li-Rong Dai, and Ren-Hua Wang, “Cross-Validation and Minimum Generation Error based Decision Tree Pruning for HMM-based Speech Synthesis”, Computational Linguistics and Chinese Language Processing, vol. 15, no. 1, pp. 61-76, March 2010.

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<u>国内期刊论文</u>

[1] 孙晓辉，凌震华，戴礼荣，"融合自动检错的单元挑选语音合成方法研究"，数据采集与处理，已接收.

[2] 凌震华，高丽，戴礼荣，"基于目标逼近特征和双向联想贮存器的情感语音基频转换"，天津大学学报，vol. 48, pp. 670-674, 2015.

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[5] 陈凌辉，凌震华，戴礼荣，"基于话者无关模型的说话人转换方法"，模式识别与人工智能，vol. 26, no. 3, pp. 254-259 2013.

[6] 张元平，凌震华，戴礼荣，刘庆峰，"一种改进的基于决策树的英文韵律短语边界预测方法"，计算机应用研究，vol. 29, no. 8, pp. 2921-2925, 2012.

[7] 刘海波，李辉，凌震华，"用于周期分解语音活动检测的基频提取方法研究"，中国科学技术大学学报，vol. 42, no. 2, pp. 106-111, 2012.

[8] 胡郁，凌震华，王仁华，戴礼荣，"基于声学统计建模的语音合成技术研究"，中文信息学报，vol. 25, no.6, pp. 127-136, 2011.

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[10] 刘航，凌震华，郭武，戴礼荣，"一种改进的跨语种语音合成模型自适应方法"，模式识别与人工智能，vol. 24, no. 4, 2011.

[11] 卢恒，凌震华，雷鸣，戴礼荣，王仁华，"基于最小生成误差的 HMM 模型聚类自动优化"，模式识别与人工智能，vol. 23, no.6, pp. 822-828, 2010.

[12] 赵欢欢，凌震华，王仁华，戴礼荣，"基于最大后验概率的语音合成说话人自适应"，数据采集与处理， vol. 25, no.4, pp.495-499, 2010.

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[14] Ren-Hua Wang, Li-Rong Dai, Zhen-Hua Ling, and Yu Hu, "Trainable unit selection speech synthesis under statistical framework," Chinese Science Bulletin, vol. 54, no. 11, pp. 1963-1969, 2009.

[15] 凌震华, 王仁华, "基于统计声学模型的单元挑选语音合成算法," 模式识别与人工智能, vol. 21, no. 3, 2008, pp. 280-284.

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国际会议论文

[1] Zhen-Hua Ling, Xiao-Hui Sun, Li-Rong Dai, and Yu Hu, "MODULATION SPECTRUM COMPENSATION FOR HMM-BASED SPEECH SYNTHESIS USING LINE SPECTRAL PAIRS," ICASSP 2016, accepted.

[2] Xiang Yin, Zhen-Hua Ling, Ya-Jun Hu, and Li-Rong Dai, "MODELING SPECTRAL ENVELOPES USING DEEP CONDITIONAL RESTRICTED BOLTZMANN MACHINES FOR STATISTICAL PARAMETRIC SPEECH SYNTHESIS," ICASSP 2016, accepted.

[3] Ya-Jun Hu, Zhen-Hua Ling, and Li-Rong Dai, "DEEP BELIEF NETWORK-BASED POST-FILTERING FOR STATISTICAL PARAMETRIC SPEECH SYNTHESIS," ICASSP 2016, accepted.

[4] Xin Wang, Ming-Hui Dong, and Zhen-Hua Ling, "A FULL TRAINING FRAMEWORK OF CROSS-STREAM DEPENDENCE MODELLING FOR HMM-BASED SINGING VOICE SYNTHESIS," ICASSP 2016, accepted.

[5] Qian Chen, Zhen-Hua Ling, Chen-Yu Yang, and Li-Rong Dai, "Automatic Phrase Boundary Labeling of Speech Synthesis Database Using Context-Dependent HMMs and N-Gram Prior Distributions", in Interspeech 2015.

[6] Quan Liu, Hui Jiang, Si Wei, Zhen-Hua Ling, and Yu Hu, "LEARNING SEMANTIC WORD EMBEDDINGS BASED ON ORDINAL KNOWLEDGE CONSTRAINTS", in ACL 2015..

[7] Yu Gu, and Zhen-Hua Ling, "Restoring High Frequency Spectral Envelopes Using Neural Networks for Speech Bandwidth Extension", in IJCNN, IEEE 2015.

[8] Zheng-Chen Liu, Zhen-Hua Ling, and Li-Rong Dai, "LIP MOVEMENT GENERATION USING RESTRICTED BOLTZMANN MACHINES FOR VISUAL SPEECH SYNTHESIS", in ChinaSIP, pp. 606-610, 2015.

[9] Li-Juan Liu, Ling-Hui Chen, Zhen-Hua Ling, Li-Rong Dai, "SPECTRAL CONVERSION USING DEEP NEURAL NETWORKS TRAINED WITH MULTI-SOURCE SPEAKERS", in ICASSP, pp. 4849-4853, 2015.

[10] Ling-Hui Chen, Zhen-Hua Ling, Yi-Qing Zu, Run-Qiang Yan, Yuan Jiang, Xian-Jun Xia, Ying Wang, "The USTC System for Blizzard Challenge 2014", in Blizzard Challenge Workshop, 2014.

[11] Ming-Qi Cai, Zhen-Hua Ling, and Li-Rong Dai, "Formant-Controlled Speech Synthesis Using Hidden Trajectory Model", in Interspeech, pp. 1529-1533, 2014.

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[21] Maria Astrinaki, Alexis Moinet, Junichi Yamagishi, Korin Richmond, Zhen-Hua Ling, Simon King, and Thierry Dutoit, "Mage - Reactive articulatory feature control of HMM-based parametric
speech synthesis", in 8th ISCA Speech Synthesis Workshop, pp. 207-211, 2013.

[22] Ling-Hui Chen, Zhen-Hua Ling, Yan Song, and Li-Rong Dai, “Joint spectral distribution modeling using restricted Boltzmann machines for voice conversion,” in Interspeech, pp. 3052-3056, 2013.

[23] Korin Richmond, Zhen-Hua Ling, Junichi Yamagishi, Benigno Uria, “On the Evaluation of Inversion Mapping Performance in the Acoustic Domain,” in Interspeech, pp. 1012-1016, 2013.

[24] Zhen-Hua Ling, Li Deng, and Dong Yu, "Modeling Spectral Envelopes Using Restricted Boltzmann Machines for Statistical Parametric Speech Synthesis", in ICASSP, pp. 7825-7829, 2013.

[25] Chen-Yu Yang, Zhen-Hua Ling, and Li-Rong Dai, "Unsupervised Prosodic Phrase Boundary Labeling of Mandarin Speech Synthesis Database Using Context-Dependent HMM", in ICASSP, pp. 6875-6879, 2013.

[26] Xin Wang, Zhen-Hua Ling, and Li-Rong Dai, "Cross-Stream Dependency Modeling Using Continuous F0 Model For Hmm-Based Speech Synthesis", In Proc. of ISCSLP, 2012.

[27] Xian-Jun Xia,Zhen-Hua Ling, Chen-Yu Yang, Li-Rong Dai, "Improved Unit Selection Speech Synthesis Method Utilizing Subjective Evaluation Results On Synthetic Speech", in Proc. of ISCSLP, 2012.

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