

**11th Annual Conference of the
International Speech
Communication Association 2010**

(INTERSPEECH 2010)

**Makuhari, Chiba, Japan
26-30 September 2010**

Volume 1 of 4

ISBN: 978-1-61782-123-3

Printed from e-media with permission by:

Curran Associates, Inc.
57 Morehouse Lane
Red Hook, NY 12571



Some format issues inherent in the e-media version may also appear in this print version.

Copyright© (2010) by the International Speech Communications Association
All rights reserved.

Printed by Curran Associates, Inc. (2011)

For permission requests, please contact the International Speech Communications Association
at the address below.

International Speech Communications Association
c/o Emmanuelle Foxonet
Lous Tourils
F-66390 Baixas, France

Phone: 33 468 385 827
Fax: 49 228 735 639

secretariat@isca-speech.org

Additional copies of this publication are available from:

Curran Associates, Inc.
57 Morehouse Lane
Red Hook, NY 12571 USA
Phone: 845-758-0400
Fax: 845-758-2634
Email: curran@proceedings.com
Web: www.proceedings.com

TABLE OF CONTENTS

VOLUME 1

KEYNOTE I

Still Talking to Machines (Cognitively Speaking)	1
<i>Steve Young</i>	

KEYNOTE II

Sound-Based Assistive Technology Supporting "Seeing", "Hearing" and "Speaking" for the Disabled and the Elderly	11
<i>Tohru Ifukube</i>	

KEYNOTE III

Beyond Sentence Prosody	20
<i>Chiu-Yu Tseng</i>	

MON-SES2-S1: SPECIAL SESSION: MODELS OF SPEECH – IN SEARCH OF BETTER REPRESENTATIONS

A Procedure for Estimating Gestural Scores from Natural Speech	30
<i>Hosung Nam, Vikramjit Mitra, Mark Tiede, Elliot Saltzman, Louis Goldstein, Carol Espy-Wilson, Mark Hasegawa-Johnson</i>	
On the Interdependencies Between Voice Quality, Glottal Gaps, and Voice-Source Related Acoustic Measures	34
<i>Yen-Liang Shue, Gang Chen, Abeer Alwan</i>	
Simplification and Extension of Non-Periodic Excitation Source Representations for High-Quality Speech Manipulation Systems	38
<i>Hideki Kawahara, Masanori Morise, Toru Takahashi, Hideki Banno, Ryuichi Nisimura, Toshio Irino</i>	
Phase Equalization-Based Autoregressive Model of Speech Signals	42
<i>Sadao Hiroya, Takemi Mochida</i>	
Articulatory-Functional Modeling of Speech Prosody: A Review	46
<i>Yi Xu, Santitham Prom-On</i>	
Two New Estimation Methods for a Superpositional Intonation Model	50
<i>Humberto M. Torres, Hansjorg Mixdorff, Jorge A. Gurlekian, Hartmut R. Pfitzinger</i>	

MON-SES2-O1: ASR: ACOUSTIC MODELS I

A Discriminative Splitting Criterion for Phonetic Decision Trees	54
<i>Simon Wiesler, Georg Heigold, Markus Nussbaum-Thom, Ralf Schluter, Hermann Ney</i>	
Canonical State Models for Automatic Speech Recognition	58
<i>M. J. F. Gales, Kai Yu</i>	
Restructuring Exponential Family Mixture Models	62
<i>Pierre L. Dognin, John R. Hershey, Vaibhava Goel, Peder Olsen</i>	
Unsupervised Discovery and Training of Maximally Dissimilar Cluster Models	66
<i>Francoise Beaufays, Vincent Vanhoucke, Brian Stroppe</i>	
Probabilistic State Clustering Using Conditional Random Field for Context-Dependent Acoustic Modelling	70
<i>Khe Chai Sim</i>	
Integrate Template Matching and Statistical Modeling for Speech Recognition	74
<i>Xie Sun, Yunxin Zhao</i>	

MON-SES2-O2: SPOKEN DIALOGUE SYSTEMS I

Cross-Lingual Spoken Language Understanding from Unaligned Data Using Discriminative Classification Models and Machine Translation	78
<i>Fabrice Lefevre, Francois Mairesse, Steve Young</i>	
Techniques for Topic Detection Based Processing in Spoken Dialog Systems	82
<i>Rajesh Balchandran, Leonid Rachevsky, Bhuvana Ramabhadran, Miroslav Novak</i>	
Optimizing Spoken Dialogue Management with Fitted Value Iteration	86
<i>Senthilkumar Chandramohan, Matthieu Geist, Olivier Pietquin</i>	
Natural Belief-Critic: A Reinforcement Algorithm for Parameter Estimation in Statistical Spoken Dialogue Systems	90
<i>F. Jurcicek, B. Thomson, S. Keizer, Francois Mairesse, M. Gasic, Kai Yu, Steve Young</i>	
Is It Possible to Predict Task Completion in Automated Troubleshooters?	94
<i>Alexander Schmitt, Michael Scholz, Wolfgang Minker, Jackson Liscombe, David Suendermann</i>	

Minimally Invasive Surgery for Spoken Dialog Systems	98
<i>David Suendermann, Jackson Liscombe, Roberto Pieraccini</i>	

MON-SES2-O3: SPEECH PERCEPTION I: FACTORS INFLUENCING PERCEPTION

Detecting Categorical Perception in Continuous Discrimination Data	102
<i>Paul Boersma, Katerina Chladkova</i>	
The Interrelation Between the Stimulus Range and the Number of Response Categories in Vowel Categorization	106
<i>Titia Benders, Paola Escudero</i>	
The Relation Between Pitch Perception Preference and Emotion Identification	110
<i>Marie Nilsonova, Martijn Goudbeek, Luuk Kempen</i>	
Competition in the Perception of Spoken Japanese Words	114
<i>Takashi Otake, James M. McQueen, Anne Cutler</i>	
Influence of Musical Training on Perception of L2 Speech	118
<i>Makiko Sadakata, Lotte Van Der Zanden, Kaoru Sekiyama</i>	
Full Body Aero-Tactile Integration in Speech Perception	122
<i>Donald Derrick, Bryan Gick</i>	

MON-SES2-O4: PROSODY: MODELS

Nucleus Position Within the Intonation Phrase: A Typological Study of English, Czech and Hungarian	126
<i>Tomas Dubeda, Katalin Mady</i>	
Focus-Sensitive Operator or Focus Inducer: Always and Only	130
<i>Yong-Cheol Lee, Satoshi Nambu</i>	
F₀ Declination in English and Mandarin Broadcast News Speech	134
<i>Jiahong Yuan, Mark Liberman</i>	
Frequency of Occurrence Effects on Pitch Accent Realisation	138
<i>Katrin Schweitzer, Michael Walsh, Bernd Mobius, Hinrich Schutze</i>	
On the Automatic ToBI Accent Type Identification from Data	142
<i>Cesar Gonzalez-Ferreras, Carlos Vivaracho-Pascual, David Escudero-Mancebo, Valentin Cardenoso-Payo</i>	
AuToBI --- A Tool for Automatic ToBI Annotation	146
<i>Andrew Rosenberg</i>	

MON-SES2-P1: SPEECH SYNTHESIS I: UNIT SELECTION AND OTHERS

A Classifier-Based Target Cost for Unit Selection Speech Synthesis Trained on Perceptual Data	150
<i>Volker Strom, Simon King</i>	
Applying Scalable Phonetic Context Similarity in Unit Selection of Concatenative Text-to-Speech	154
<i>Wei Zhang, Xiaodong Cui</i>	
Speech Database Reduction Method for Corpus-Based TTS System	158
<i>Mitsuaki Isogai, Hideyuki Mizuno</i>	
Automatic Error Detection for Unit Selection Speech Synthesis Using Log Likelihood Ratio Based SVM Classifier	162
<i>Heng Lu, Zhen-Hua Ling, Si Wei, Lirong Dai, Ren-Hua Wang</i>	
Using Robust Viterbi Algorithm and HMM-Modeling in Unit Selection TTS to Replace Units of Poor Quality	166
<i>Hanna Silen, Elina Helander, Jani Nurminen, Konsta Koppinen, Moncef Gabbouj</i>	
Automatic Detection of Abnormal Stress Patterns in Unit Selection Synthesis	170
<i>Yeon-Jun Kim, Mark C. Beutnagel</i>	
Enhancements of Viterbi Search for Fast Unit Selection Synthesis	174
<i>Daniel Tihelka, Jiri Kala, Jindrich Matousek</i>	
Accurate Pitch Marking for Prosodic Modification of Speech Segments	178
<i>Thomas Ewender, Beat Pfister</i>	
A Novel Hybrid Approach for Mandarin Speech Synthesis	182
<i>Shifeng Pan, Meng Zhang, Jianhua Tao</i>	
Modeling Liaison in French by Using Decision Trees	186
<i>Josafa De Jesus Aguiar Pontes, Sadaoki Furu</i>	
Improvement on Plural Unit Selection and Fusion	190
<i>Jian Luan, Jian Li</i>	
Improving Speech Synthesis of Machine Translation Output	194
<i>Alok Parlikar, Alan W. Black, Stephan Vogel</i>	
Paraphrase Generation to Improve Text-to-Speech Synthesis	198
<i>Ghislain Putois, Jonathan Chevelu, Cedric Boidin</i>	

MON-SES2-P2: ASR: SEARCH, DECODING AND CONFIDENCE MEASURES I

Phone Mismatch Penalty Matrices for Two-Stage Keyword Spotting via Multi-Pass Phone Recognizer	202
<i>Chang Woo Han, Shin Jae Kang, Chul Min Lee, Nam Soo Kim</i>	

English Spoken Term Detection in Multilingual Recordings	206
<i>Petr Motlicek, Fabio Valente, Philip N. Garner</i>	
A Hybrid Approach to Robust Word Lattice Generation via Acoustic-Based Word Detection	210
<i>Icksang Han, Chiyoun Park, Jeongmi Cho, Jeongsu Kim</i>	
Direct Observation of Pruning Errors (DOPE): A Search Analysis Tool	214
<i>V. Steinbiss, Martin Sundermeyer, Hermann Ney</i>	
Direct Construction of Compact Context-Dependency Transducers from Data	218
<i>David Rybach, Michael Riley</i>	
Incremental Composition of Static Decoding Graphs with Label Pushing	222
<i>Miroslav Novak</i>	
A Novel Path Extension Framework Using Steady Segment Detection for Mandarin Speech Recognition	226
<i>Zhanlei Yang, Wenju Liu</i>	
On the Relation of Bayes Risk, Word Error, and Word Posteriors in ASR	230
<i>Ralf Schluter, Markus Nussbaum-Thom, Hermann Ney</i>	
Time Conditioned Search in Automatic Speech Recognition Reconsidered	234
<i>D. Nolden, Hermann Ney, Ralf Schluter</i>	
Efficient Data Selection for Speech Recognition Based on Prior Confidence Estimation Using Speech and Context Independent Models	238
<i>Satoshi Kobashikawa, Taichi Asami, Yoshikazu Yamaguchi, Hirokazu Masataki, Satoshi Takahashi</i>	
A Novel Confidence Measure Based on Marginalization of Jointly Estimated Error Cause Probabilities	242
<i>Atsunori Ogawa, Atsushi Nakamura</i>	

MON-SES2-P3: SPECIAL-PURPOSE SPEECH APPLICATIONS

Evaluation of a Silent Speech Interface Based on Magnetic Sensing	246
<i>Robin Hofe, Stephen R. Ell, Michael J. Fagan, James M. Gilbert, Phil D. Green, Roger K. Moore, Sergey I. Rybchenko</i>	
Advanced Speech Communication System for Deaf People	250
<i>R. San-Segundo, V. Lopez, R. Martin, S. Lufti, J. Ferreiros, R. Cordoba, J. M. Pardo</i>	
Unsupervised Acoustic Model Adaptation for Multi-Origin Non Native ASR	254
<i>Sethserey Sam, Eric Castelli, Laurent Besacier</i>	
Speech-Based Automated Cognitive Status Assessment	258
<i>Dilek Hakkani-Tur, Dimitra Vergyri, Gokhan Tur</i>	
Speech Recognition with a Seamlessly Updated Language Model for Real-Time Closed-Captioning	262
<i>Toru Imai, Shinichi Homma, Akio Kobayashi, Takahiro Oku, Shohei Sato</i>	
The Comparison Between the Deletion-Based Methods and the Mixing-Based Methods for Audio CAPTCHA Systems	266
<i>Takuya Nishimoto, Takayuki Watanabe</i>	
Comparing Mono- & Multilingual Acoustic Seed Models for a Low e-Resourced Language: A Case-Study of Luxembourgish	270
<i>Martine Adda-Decker, Lori Lamel, Natalie D. Snoeren</i>	
Manipulating Treacheoesophageal Speech	274
<i>R. J. J. H. Van Son, Irene Jacobi, Frans Hilgers</i>	
Towards Mixed Language Speech Recognition Systems	278
<i>David Inseng, Herve Bourlard, Mathew Magimai Doss</i>	
Voice Search for Development	282
<i>Etienne Barnard, Johan Schalkwyk, Charl Van Heerden, Pedro J. Moreno</i>	
Cross-Cultural Investigation of Prosody in Verbal Feedback in Interactional Rapport	286
<i>Gina-Anne Levow, Susan Duncan, Edward T. King</i>	
Multimodal Speaker Diarization Using Oriented Optical Flow Histograms	290
<i>Mary Tai Knox, Gerald Friedland</i>	
Towards an ASR-Free Objective Analysis of Pathological Speech	294
<i>Catherine Middag, Yvan Saeys, Jean-Pierre Martens</i>	

MON-SES2-P4: SPEECH ANALYSIS

Session Variability Contrasts in the MARP Corpus	298
<i>Keith W. Godin, John H. L. Hansen</i>	
Estimation of Two-to-One Forced Selection Intelligibility Scores by Speech Recognizers Using Noise-Adapted Models	302
<i>Kazuhiro Kondo, Yusuke Takano</i>	
Analysis of Gender Normalization Using MLP and VTLN Features	306
<i>Thomas Schaaf, Florian Metze</i>	
Discovering an Optimal Set of Minimally Contrasting Acoustic Speech Units: A Point of Focus for Whole-Word Pattern Matching	310
<i>Guillaume Aimetti, Roger K. Moore, L. Ten Bosch</i>	
Improvements to the Equal-Parameter BIC for Speaker Diarization	314
<i>Themis Stafylakis, Xavier Anguera</i>	
A Multistream Multiresolution Framework for Phoneme Recognition	318
<i>Nima Mesgarani, Samuel Thomas, Hynek Hermansky</i>	

Cluster Analysis of Differential Spectral Envelopes on Emotional Speech	322
<i>Giampiero Salvi, Fabio Tesser, Enrico Zovato, Piero Cosi</i>	
Modeling Pronunciation Variation with Context-Dependent Articulatory Feature Decision Trees	326
<i>Sam Bowman, Karen Livescu</i>	
Ungrounded Independent Non-Negative Factor Analysis	330
<i>Bhiksha Raj, Kevin W. Wilson, Alexander Krueger, Reinhold Haeb-Umbach</i>	
Signal Interaction and the Devil Function	334
<i>John R. Hershey, Peder Olsen, Steven J. Rennie</i>	

MON-SES3-01: SYSTEMS FOR LVCSR

Semi-Automated Update of Automatic Transcription System for the Japanese National Congress	338
<i>Yuya Akita, Masato Mimura, Graham Neubig, Tatsuya Kawahara</i>	
Language Model Cross Adaptation for LVCSR System Combination	342
<i>X. Liu, M. J. F. Gales, P. C. Woodland</i>	
Large Vocabulary Continuous Speech Recognition Using WFST-Based Linear Classifier for Structured Data	346
<i>Shinji Watanabe, Takaaki Hori, Atsushi Nakamura</i>	
Accelerating Hierarchical Acoustic Likelihood Computation on Graphics Processors	350
<i>Pavel Kveton, Miroslav Novak</i>	
Search by Voice in Mandarin Chinese	354
<i>Jiulong Shan, Genqing Wu, Zhihong Hu, Xiliu Tang, Martin Jansche, Pedro J. Moreno</i>	
The AMIDA 2009 Meeting Transcription System	358
<i>Thomas Hain, Lukas Burget, John Dines, Philip N. Garner, Asmaa El Hannani, Marijn Huijbregts, Martin Karafiat, Mike Lincoln, Vincent Wan</i>	

MON-SES3-02: SPEAKER CHARACTERIZATION AND RECOGNITION I

Simple and Efficient Speaker Comparison Using Approximate KL Divergence	362
<i>W. M. Campbell, Zahi N. Karam</i>	
The IIR NIST SRE 2008 and 2010 Summed Channel Speaker Recognition Systems	366
<i>Hanwu Sun, Bin Ma, Chien-Lin Huang, Trung Hieu Nguyen, Haizhou Li</i>	
Speaker Characterization Using Long-Term and Temporal Information	370
<i>Chien-Lin Huang, Hanwu Sun, Bin Ma, Haizhou Li</i>	
Score-Level Compensation of Extreme Speech Duration Variability in Speaker Verification	374
<i>Sergio Perez-Gomez, Daniel Ramos, Javier Gonzalez-Dominguez, Joaquin Gonzalez-Rodriguez</i>	
Speaker Recognition Experiments Using Connectionist Transformation Network Features	378
<i>Alberto Abad, Isabel Trancoso</i>	
Speaker Recognition Using Supervised Probabilistic Principal Component Analysis	382
<i>Yun Lei, John H. L. Hansen</i>	

MON-SES3-03: SOURCE SEPARATION

A Factorial Sparse Coder Model for Single Channel Source Separation	386
<i>Robert Peharz, Michael Stark, Franz Pernkopf, Yannis Stylianou</i>	
Oriented PCA Method for Blind Speech Separation of Convolutional Mixtures	390
<i>Yasmina Benabderrahmane, Sid Ahmed Selouani, Douglas O'Shaughnessy</i>	
Online Gaussian Process for Nonstationary Speech Separation	394
<i>Hsin-Lung Hsieh, Jen-Tzung Chien</i>	
Convexity and Fast Speech Extraction by Split Bregman Method	398
<i>Meng Yu, Wenye Ma, Jack Xin, Stanley Osher</i>	
Reducing Musical Noise in Blind Source Separation by Time-Domain Sparse Filters and Split Bregman Method	402
<i>Wenye Ma, Meng Yu, Jack Xin, Stanley Osher</i>	
Combining Monaural and Binaural Evidence for Reverberant Speech Segregation	406
<i>John Woodruff, Rohit Prabhavalkar, Eric Fosler-Lussier, Deliang Wang</i>	

MON-SES3-04: SPEECH SYNTHESIS II: HMM-BASED SPEECH SYNTHESIS

Speaker and Language Adaptive Training for HMM-Based Polyglot Speech Synthesis	410
<i>Heiga Zen</i>	
Context Adaptive Training with Factorized Decision Trees for HMM-Based Speech Synthesis	414
<i>Kai Yu, Heiga Zen, Francois Mairesse, Steve Young</i>	
Roles of the Average Voice in Speaker-Adaptive HMM-Based Speech Synthesis	418
<i>Junichi Yamagishi, Oliver Watts, Simon King, Bela Usabaev</i>	
An HMM Trajectory Tiling (HTT) Approach to High Quality TTS	422
<i>Yao Qian, Zhi-Jie Yan, Yijian Wu, Frank K. Soong, Xin Zhuang, Shengyi Kong</i>	
A Perceptual Study of Acceleration Parameters in HMM-Based TTS	426
<i>Yi-Ning Chen, Zhi-Jie Yan, Frank K. Soong</i>	

Evaluation of Prosodic Contextual Factors for HMM-Based Speech Synthesis	430
<i>Shuji Yokomizo, Takashi Nose, Takao Kobayashi</i>	

MON-SES3-O5: MULTI-MODAL SIGNAL PROCESSING

Learning Words and Speech Units Through Natural Interactions	434
<i>Jonas Hornstein, Jose Santos-Victor</i>	
Bimodal Coherence Based Scale Ambiguity Cancellation for Target Speech Extraction and Enhancement	438
<i>Qingju Liu, Wenwu Wang, Philip Jackson</i>	
Speech Estimation in Non-Stationary Noise Environments Using Timing Structures Between Mouth Movements and Sound Signals	442
<i>Hiroaki Kawashima, Yu Horii, Takashi Matsuyama</i>	
Synthesizing Photo-Real Talking Head via Trajectory-Guided Sample Selection	446
<i>Lijuan Wang, Xiaojun Qian, Wei Han, Frank K. Soong</i>	
Silent vs Vocalized Articulation for a Portable Ultrasound-Based Silent Speech Interface	450
<i>Victoria M. Florescu, Lise Crevier-Buchman, Bruce Denby, Thomas Hueber, Antonia Colazo-Simon, Claire Pillot-Loiseau, Pierre Roussel, Cedric Gendrot, Sophie Quattrocchi</i>	
Comparison of HMM and TMDN Methods for Lip Synchronisation	454
<i>Gregor Hofer, Korin Richmond</i>	

MON-SES3-P1: PARALANGUAGE

Rhythm and Formant Features for Automatic Alcohol Detection	458
<i>Florian Schiel, Christian Heinrich, Veronika Neumeyer</i>	
An Exploration of Voice Source Correlates of Focus	462
<i>Irena Yanushevskaya, Christer Gobl, John Kane, Ailbhe Ni Chasaide</i>	
Modeling Perceived Vocal Age in American English	466
<i>James D. Harnsberger, Rahul Shrivastav, W. S. Brown Jr.</i>	
Multivariate Analysis of Vocal Fatigue in Continuous Reading	470
<i>Marie-Jose Caraty, Claude Montacie</i>	
Frequency-Domain Delexicalization Using Surrogate Vowels	474
<i>Alexander Kain, Jan P. H. Van Santen</i>	
Emotion Recognition Using Imperfect Speech Recognition	478
<i>Florian Metze, Anton Batliner, Florian Eyben, Tim Polzehl, Bjorn Schuller, Stefan Steidl</i>	
A Novel Feature Extraction Strategy for Multi-Stream Robust Emotion Identification	482
<i>Gang Liu, Yun Lei, John H. L. Hansen</i>	
Setup for Acoustic-Visual Speech Synthesis by Concatenating Bimodal Units	486
<i>Asterios Toutios, Utpala Musti, Slim Ouni, Vincent Colotte, Brigitte Wrobel-Dautcourt, Marie-Odile Berger</i>	
Towards Affective State Modeling in Narrative and Conversational Settings	490
<i>Bart Jochems, Martha Larson, Roeland Ordelman, Ronald Poppe, Khiet P. Truong</i>	
Detection of Anger Emotion in Dialog Speech Using Prosody Feature and Temporal Relation of Utterances	494
<i>Narichika Nomoto, Hirokazu Masataki, Osamu Yoshioka, Satoshi Takahashi</i>	
Gesture and Speech Coordination: The Influence of the Relationship Between Manual Gesture and Speech	498
<i>Benjamin Roustan, Marion Dohen</i>	
Analysis and Detection of Cognitive Load and Frustration in Drivers' Speech	502
<i>Hynek Boril, Seyed Omid Sadjadi, Tristan Kleinschmidt, John H. L. Hansen</i>	
Acoustic-Based Recognition of Head Gestures Accompanying Speech	506
<i>Akira Sasou, Yasuharu Hashimoto, Katsuhiko Sakaue</i>	
Multimodal Dialog in the Car: Combining Speech and Turn-and-Push Dial to Control Comfort Functions	510
<i>Sandro Castronovo, Angela Mahr, Margarita Pentcheva, Christian Muller</i>	
Hands Free Audio Analysis from Home Entertainment	514
<i>Danil Korchagin, Philip N. Garner, Petr Motlicek</i>	
Affective Story Teller: A TTS System for Emotional Expressivity	518
<i>Mostafa Al Masum Shaikh, Antonio Rui Ferreira Rebordao, Keikichi Hirose</i>	

MON-SES3-P2: ASR: SPEAKER ADAPTATION, ROBUSTNESS AGAINST REVERBERATION

Enhancing Children's Speech Recognition Under Mismatched Condition by Explicit Acoustic Normalization	522
<i>Shweta Ghai, Rohit Sinha</i>	
Comparison of Discriminative Input and Output Transformations for Speaker Adaptation in the Hybrid NN/HMM Systems	526
<i>Bo Li, Khe Chai Sim</i>	
Augmentation of Adaptation Data	530
<i>Ravichander Vippera, Steve Renals, Joe Frankel</i>	
Discriminative Adaptation Based on Fast Combination of DMAP and DfMLLR	534
<i>Lukas Machlica, Zbynek Zajic, Ludek Muller</i>	
Revisiting VTLN Using Linear Transformation on Conventional MFCC	538
<i>D. R. Sanand, Ralf Schluter, Hermann Ney</i>	

Speaker Adaptation Based on Nonlinear Spectral Transform for Speech Recognition	542
<i>Toyohiro Hayashi, Yoshihiko Nankaku, Akinobu Lee, Keiichi Tokuda</i>	
Speaker Adaptation Based on System Combination Using Speaker-Class Models	546
<i>Tetsuo Kosaka, Takashi Ito, Masaharu Kato, Masaki Kohda</i>	
Speaker Adaptation in Transformation Space Using Two-Dimensional PCA	550
<i>Yongwon Jeong, Young Rok Song, Hyung Soon Kim</i>	
On Speaker Adaptive Training of Artificial Neural Networks	554
<i>Jan Trmal, Jan Zelinka, Ludek Muller</i>	
Model Synthesis for Band-Limited Speech Recognition	558
<i>Yongjun He, Jiqing Han</i>	
Performance Estimation of Reverberant Speech Recognition Based on Reverberant Criteria RSR-D_n with Acoustic Parameters	562
<i>Takahiro Fukumori, Masanori Morise, Takanobu Nishiura</i>	
A Novel Approach for Matched Reverberant Training of HMMs Using Data Pairs	566
<i>Armin Sehr, Christian Hofmann, Roland Maas, Walter Kellermann</i>	
An Auditory Based Modulation Spectral Feature for Reverberant Speech Recognition	570
<i>Harikrishna Maganti, Marco Matassoni</i>	
On the Potential of Channel Selection for Recognition of Reverberated Speech with Multiple Microphones	574
<i>Martin Wolf, Climent Nadeu</i>	
An Improved Wavelet-Based Dereverberation for Robust Automatic Speech Recognition	578
<i>Randy Gomez, Tatsuya Kawahara</i>	
Methods for Robust Speech Recognition in Reverberant Environments: A Comparison	582
<i>Rico Petrick, Thomas Feher, Masashi Unoki, Rudiger Hoffmann</i>	

MON-SES3-P3: LANGUAGE LEARNING, TTS, AND OTHER APPLICATIONS

Integration of Multilayer Regression Analysis with Structure-Based Pronunciation Assessment	586
<i>Masayuki Suzuki, Yu Qiao, Nobuaki Minematsu, Keiichi Hirose</i>	
Using Non-Native Error Patterns to Improve Pronunciation Verification	590
<i>Joost Van Doremalen, Catia Cucchiarini, Helmer Strik</i>	
Regularized-MLLR Speaker Adaptation for Computer-Assisted Language Learning System	594
<i>Dean Luo, Yu Qiao, Nobuaki Minematsu, Yutaka Yamauchi, Keiichi Hirose</i>	
Automatic Evaluation of English Pronunciation by Japanese Speakers Using Various Acoustic Features and Pattern Recognition Techniques	598
<i>Kuniaki Hirabayashi, Seiichi Nakagawa</i>	
Decision Tree Based Tone Modeling with Corrective Feedbacks for Automatic Mandarin Tone Assessment	602
<i>Hsien-Cheng Liao, Jiang-Chun Chen, Sen-Chia Chang, Ying-Hua Guan, Chin-Hui Lee</i>	
CASTLE: A Computer-Assisted Stress Teaching and Learning Environment for Learners of English as a Second Language	606
<i>Jingli Lu, Ruili Wang, Liyanage C. De Silva, Yang Gao, Jia Liu</i>	
Automatic Reference Independent Evaluation of Prosody Quality Using Multiple Knowledge Fusions	610
<i>Shen Huang, Hongyan Li, Shijin Wang, Jiaen Liang, Bo Xu</i>	
Landmark-Based Automated Pronunciation Error Detection	614
<i>Su-Youn Yoon, Mark Hasegawa-Johnson, Richard Sproat</i>	
HMM Based TTS for Mixed Language Text	618
<i>Zhiwei Shuang, Shiyin Kang, Yong Qin, Lirong Dai, Lianhong Cai</i>	
An Analysis of Language Mismatch in HMM State Mapping-Based Cross-Lingual Speaker Adaptation	622
<i>Hui Liang, John Dines</i>	
Classroom Note-Taking System for Hearing Impaired Students Using Automatic Speech Recognition Adapted to Lectures	626
<i>Tatsuya Kawahara, Norihiro Katsumaru, Yuya Akita, Shinsuke Mori</i>	
Exploring Web-Browser Based Runtimes Engines for Creating Ubiquitous Speech Interfaces	630
<i>Paul R. Dixon, Sadaoki Furui</i>	

MON-SES3-P4: PITCH AND GLOTTAL-WAVEFORM ESTIMATION AND MODELING I

Efficient Three-Stage Pitch Estimation for Packet Loss Concealment	633
<i>Xuejing Sun, Sameer Gadre</i>	
On Evaluation of the F₀ Estimation Based on Time-Varying Complex Speech Analysis	637
<i>Keiichi Funaki</i>	
Pitch Estimation in Noisy Speech Based on Temporal Accumulation of Spectrum Peaks	641
<i>Feng Huang, Tan Lee</i>	
Multi-Pitch Estimation by a Joint 2-D Representation of Pitch and Pitch Dynamics	645
<i>Tianyu T. Wang, Thomas F. Quatieri</i>	
On the Effect of Fundamental Frequency on Amplitude and Frequency Modulation Patterns in Speech Resonances	649
<i>Pirros Tsiakoulis, Alexandros Potamianos</i>	
Pitch Determination Using Autocorrelation Function in Spectral Domain	653
<i>M. Shahidur Rahman, Tatsuya Shimamura</i>	

Chirp Complex Cepstrum-Based Decomposition for Asynchronous Glottal Analysis	657
<i>Thomas Drugman, Thierry Dutoit</i>	
Exploiting Glottal Formant Parameters for Glottal Inverse Filtering and Parameterization	661
<i>Alan O Cinneide, David Dorran, Mikel Gainza, Eugene Coyle</i>	
Glottal Parameters Estimation on Speech Using the Zeros of the Z-Transform	665
<i>Nicolas Sturmel, Christophe D'Alessandro, Boris Doval</i>	
Significance of Pitch Synchronous Analysis for Speaker Recognition Using AANN Models	669
<i>Sri Harish Reddy M., Kishore Prahallad, Suryakanth V. Gangashetty, B. Yegnanarayana</i>	
On Using Voice Source Measures in Automatic Gender Classification of Children's Speech	673
<i>Gang Chen, Xue Feng, Yen-Liang Shue, Abeer Alwan</i>	

TUE-SES1-S1: SPECIAL SESSION: OPEN VOCABULARY SPOKEN DOCUMENT RETRIEVAL

Constructing Japanese Test Collections for Spoken Term Detection	677
<i>Yoshiaki Itoh, Hiromitsu Nishizaki, Xinhui Hu, Hiroaki Nanjo, Tomoyosi Akiba, Tatsuya Kawahara, Seiichi Nakagawa, Tomoko Matsui, Yoichi Yamashita, Kiyooki Aikawa</i>	
Japanese Spoken Term Detection Using Syllable Transition Network Derived from Multiple Speech Recognizers' Outputs	681
<i>Satoshi Natori, Hiromitsu Nishizaki, Yoshihiro Sekiguchi</i>	
Combining Chinese Spoken Term Detection Systems via Side-Information Conditioned Linear Logistic Regression	685
<i>Sha Meng, Wei-Qiang Zhang, Jia Liu</i>	
Metric Subspace Indexing for Fast Spoken Term Detection	689
<i>Taisuke Kaneko, Tomoyosi Akiba</i>	
Unsupervised Spoken-Term Detection with Spoken Queries Using Segment-Based Dynamic Time Warping	693
<i>Chun-An Chan, Lin-Shan Lee</i>	
Contextual Verification for Open Vocabulary Spoken Term Detection	697
<i>Daniel Schneider, Timo Mertens, Martha Larson, Joachim Kohler</i>	
Augmented Set of Features for Confidence Estimation in Spoken Term Detection	701
<i>Javier Tejedor, Doroteo T. Toledano, Miguel Bautista, Simon King, Dong Wang, Jose Colas</i>	
Cluster-Based Language Model for Spoken Document Retrieval Using NMF-Based Document Clustering	705
<i>Xinhui Hu, Ryosuke Isotani, Hisashi Kawai, Satoshi Nakamura</i>	

TUE-SES1-O1: ROBUST ASR

Asymptotically Exact Noise-Corrupted Speech Likelihoods	709
<i>R. C. Van Dalen, M. J. F. Gales</i>	
A MMSE Estimator in Mel-Cepstral Domain for Robust Large Vocabulary Automatic Speech Recognition Using Uncertainty Propagation	713
<i>Ramon Fernandez Astudillo, Reinhold Orglmeister</i>	
Non-Negative Matrix Factorization Based Compensation of Music for Automatic Speech Recognition	717
<i>Bhiksha Raj, Tuomas Virtanen, Sourish Chaudhuri, Rita Singh</i>	
Feature versus Model Based Noise Robustness	721
<i>Kris Demuyne, Xueru Zhang, Dirk Van Compernelle, Hugo Van Hamme</i>	
SNR-Based Mask Compensation for Computational Auditory Scene Analysis Applied to Speech Recognition in a Car Environment	725
<i>Ji Hun Park, Seon Man Kim, Jae Sam Yoon, Hong Kook Kim, Sung Joo Lee, Yunkeun Lee</i>	
Automatic Selection of Thresholds for Signal Separation Algorithms Based on Interaural Delay	729
<i>Chanwoo Kim, Richard M. Stern, Kiwan Eom, Jaewon Lee</i>	

TUE-SES1-O2: LANGUAGE AND DIALECT IDENTIFICATION

Channel Detectors for System Fusion in the Context of NIST LRE 2009	733
<i>Florian Verdet, Driss Matrouf, Jean-Francois Bonastre, Jean Hennebert</i>	
Selecting Phonotactic Features for Language Recognition	737
<i>Rong Tong, Bin Ma, Haizhou Li, Eng Siong Chng</i>	
Improved Language Recognition Using Mixture Components Statistics	741
<i>Abualsoud Hanani, Michael Carey, Martin J. Russell</i>	
Using Cross-Decoder Co-Occurrences of Phone N-Grams in SVM-Based Phonotactic Language Recognition	745
<i>Mikel Penagarikano, Amparo Varona, Luis Javier Rodriguez-Fuentes, German Borden</i>	
Exploiting Variety-Dependent Phones in Portuguese Variety Identification Applied to Broadcast News Transcription	749
<i>Oscar Koller, Alberto Abad, Isabel Trancoso, Ceu Viana</i>	
Dialect Recognition Using a Phone-GMM-Supervector-Based SVM Kernel	753
<i>Fadi Biadry, Julia Hirschberg, Michael Collins</i>	

TUE-SES1-O3: TECHNOLOGIES FOR LEARNING AND EDUCATION

Discriminative Acoustic Model for Improving Mispronunciation Detection and Diagnosis in Computer-Aided Pronunciation Training (CAPT)	757
<i>Xiaojun Qian, Frank K. Soong, Helen Meng</i>	
Automatic Pronunciation Scoring Using Learning to Rank and DP-Based Score Segmentation	761
<i>Liang-Yu Chen, Jyh-Shing Roger Jang</i>	
Automatic Derivation of Phonological Rules for Mispronunciation Detection in a Computer-Assisted Pronunciation Training System	765
<i>Wai-Kit Lo, Shuang Zhang, Helen Meng</i>	

VOLUME 2

Adapting a Duration Synthesis Model to Rate Children's Oral Reading Prosody	769
<i>Minh Duong, Jack Mostow</i>	
Predicting Word Accuracy for the Automatic Speech Recognition of Non-Native Speech	773
<i>Su-Youn Yoon, Lei Chen, Klaus Zechner</i>	
A New Approach for Automatic Tone Error Detection in Strong Accented Mandarin Based on Dominant Set	777
<i>Taotao Zhu, Dengfeng Ke, Zhenbiao Chen, Bo Xu</i>	

TUE-SES1-O4: EMOTIONAL SPEECH

Analysis of Excitation Source Information in Emotional Speech	781
<i>S. R. M. Prasanna, D. Govind</i>	
Acoustic Feature Analysis in Speech Emotion Primitives Estimation	785
<i>Dongrui Wu, Thomas D. Parsons, Shrikanth S. Narayanan</i>	
Spectro-Temporal Modulations for Robust Speech Emotion Recognition	789
<i>Lan-Ying Yeh, Tai-Shih Chi</i>	
Quantification of Prosodic Entrainment in Affective Spontaneous Spoken Interactions of Married Couples	793
<i>Chi-Chun Lee, Matthew Black, Athanasios Katsamanis, Adam C. Lammert, Brian R. Baucom, Andrew Christensen, Panayiotis G. Georgiou, Shrikanth S. Narayanan</i>	
A Cluster-Profile Representation of Emotion Using Agglomerative Hierarchical Clustering	797
<i>Emily Mower, Kyu J. Han, Sungbok Lee, Shrikanth S. Narayanan</i>	
Incremental Acoustic Valence Recognition: An Inter-Corpus Perspective on Features, Matching, and Performance in a Gating Paradigm	801
<i>Bjorn Schuller, Laurence Devillers</i>	

TUE-SES1-P1: SPEECH SYNTHESIS III: HMM-BASED SPEECH SYNTHESIS

Sinusoidal Model Parameterization for HMM-Based TTS System	805
<i>Slava Shechtman, Alex Sorin</i>	
Improved Training of Excitation for HMM-Based Parametric Speech Synthesis	809
<i>Yoshinori Shiga, Tomoki Toda, Shinsuke Sakai, Hisashi Kawai</i>	
Excitation Modeling Based on Waveform Interpolation for HMM-Based Speech Synthesis	813
<i>June Sig Sung, Doo Hwa Hong, Kyung Hwan Oh, Nam Soo Kim</i>	
Formant-Based Frequency Warping for Improving Speaker Adaptation in HMM TTS	817
<i>Xin Zhuang, Yao Qian, Frank K. Soong, Yijian Wu, Bo Zhang</i>	
Improved Modelling of Speech Dynamics Using Non-Linear Formant Trajectories for HMM-Based Speech Synthesis	821
<i>Hongwei Hu, Martin J. Russell</i>	
Global Variance Modeling on the Log Power Spectrum of LSPs for HMM-Based Speech Synthesis	825
<i>Zhen-Hua Ling, Yu Hu, Lirong Dai</i>	
Autoregressive Clustering for HMM Speech Synthesis	829
<i>Matt Shannon, William Byrne</i>	
An Implementation of Decision Tree-Based Context Clustering on Graphics Processing Units	833
<i>Nicholas Pilkington, Heiga Zen</i>	
Quantized HMMs for Low Footprint Text-to-Speech Synthesis	837
<i>Alexander Gutkin, Xavi Gonzalvo, Stefan Breuer, Paul Taylor</i>	
The Role of Higher-Level Linguistic Features in HMM-Based Speech Synthesis	841
<i>Oliver Watts, Junichi Yamagishi, Simon King</i>	
HMM-Based Singing Voice Synthesis System Using Pitch-Shifted Pseudo Training Data	845
<i>Ayami Mase, Keiichiro Oura, Yoshihiko Nankaku, Keiichi Tokuda</i>	
An Unsupervised Approach to Creating Web Audio Contents-Based HMM Voices	849
<i>Jinju Ni, Hisashi Kawai</i>	
Conversational Spontaneous Speech Synthesis Using Average Voice Model	853
<i>Tomoki Koriyama, Takashi Nose, Takao Kobayashi</i>	

TUE-SES1-P2: NEW PARADIGMS IN ASR I

Mandarin Digit Recognition Assisted by Selective Tone Distinction	857
<i>Xiao-Dong Wang, Kunihiko Owa, Makoto Shozakai</i>	
Brazilian Portuguese Acoustic Model Training Based on Data Borrowing from Other Language	861
<i>Kazuhiro Abe, Sakriani Sakti, Ryosuke Isotani, Hisashi Kawai, Satoshi Nakamura</i>	
Rapid Bootstrapping of Five Eastern European Languages Using the Rapid Language Adaptation Toolkit	865
<i>Ngoc Thang Vu, Tim Schlippe, Franziska Kraus, Tanja Schultz</i>	
Cross-Lingual Speaker Adaptation via Gaussian Component Mapping	869
<i>Houwei Cao, Tan Lee, P. C. Ching</i>	
Cross-Lingual Acoustic Modeling for Dialectal Arabic Speech Recognition	873
<i>Mohamed Elmahdy, Rainer Gruhn, Wolfgang Minker, Slim Abdennadher</i>	
Cross-Lingual and Multi-Stream Posterior Features for Low Resource LVCSR Systems	877
<i>Samuel Thomas, Sriram Ganapathy, Hynek Hermansky</i>	
Latent Perceptual Mapping: A New Acoustic Modeling Framework for Speech Recognition	881
<i>Shiva Sundaram, Jerome R. Bellegarda</i>	
Unsupervised Model Adaptation on Targeted Speech Segments for LVCSR System Combination	885
<i>Richard Dufour, Fethi Bougares, Yannick Esteve, Paul Deleglise</i>	
Incremental Word Learning Using Large-Margin Discriminative Training and Variance Floor Estimation	889
<i>Irene Ayllon Clemente, Martin Heckmann, Alexander Denecke, Britta Wrede, Christian Goerick</i>	
State-Based Labelling for a Sparse Representation of Speech and Its Application to Robust Speech Recognition	893
<i>Tuomas Virtanen, Jort F. Gemmeke, Antti Hurmalainen</i>	
Similarity Scoring for Recognizing Repeated Out-of-Vocabulary Words	897
<i>Mirko Hannemann, Stefan Kombrink, Martin Karafiat, Lukas Burget</i>	
Data Pruning for Template-Based Automatic Speech Recognition	901
<i>Dino Seppi, Dirk Van Compernelle</i>	

TUE-SES1-P3: SPEECH PRODUCTION I: VARIOUS APPROACHES

Speaking Style Dependency of Formant Targets	905
<i>Akiko Amano-Kusumoto, John-Paul Hosom, Alexander Kain</i>	
Similarity of Effects of Emotions on the Speech Organ Configuration with and without Speaking	909
<i>Tatsuya Kitamura</i>	
A Study of Intra-Speaker and Inter-Speaker Affective Variability Using Electroglottograph and Inverse Filtered Glottal Waveforms	913
<i>Daniel Bone, Samuel Kim, Sungbok Lee, Shrikanth S. Narayanan</i>	
Modal Analysis of Vocal Fold Vibrations Using Laryngotopography	917
<i>Ken-Ichi Sakakibara, Hiroshi Imagawa, Miwako Kimura, Hisayuki Yokonishi, Niro Tayama</i>	
Laryngeal Voice Quality in the Expression of Focus	921
<i>Martti Vainio, Matti Airas, Juhani Jarvikivi, Paavo Alku</i>	
Laryngeal Characteristics During the Production of Geminate Consonants	925
<i>Masako Fujimoto, Kikuo Maekawa, Seiya Funatsu</i>	
Numerical Study of Turbulent Flow-Induced Sound Production in Presence of a Tooth-Shaped Obstacle: Towards Sibilant [s] Physical Modeling	929
<i>Julien Cisonni, Kazunori Nozaki, Annemie Van Hirtum, Shigeo Wada</i>	
Morphological and Predictability Effects on Schwa Reduction: The Case of Dutch Word-Initial Syllables	933
<i>Iris Hanique, Barbara Schuppler, Mirjam Ernestus</i>	
Acoustic-to-Articulatory Inversion Based on Local Regression	937
<i>Samer Al Moubayed, G. Ananthakrishnan</i>	
Korean Lenis, Fortis, and Aspirated Stops: Effect of Place of Articulation on Acoustic Realization	941
<i>Mirjam Broersma</i>	
Speech Synthesis by Modeling Harmonics Structure with Multiple Function	945
<i>Toru Nakashika, Ryuki Tachibana, Masafumi Nishimura, Tetsuya Takiguchi, Yasuo Arika</i>	
Physics of Body-Conducted Silent Speech --- Production, Propagation and Representation of Non-Audible Murmur	949
<i>Makoto Otani, Tatsuya Hirahara</i>	

TUE-SES1-P4: SPEECH ENHANCEMENT

Multichannel Noise Reduction Using Low Order RTF Estimate	953
<i>Subhojit Chakladar, Nam Soo Kim, Yu Gwang Jin, Tae Gyoong Kang</i>	
Reinforced Blocking Matrix with Cross Channel Projection for Speech Enhancement	957
<i>Inho Lee, Jongsung Yoon, Yoonjae Lee, Hanseok Ko</i>	
Masking Property Based Microphone Array Post-Filter Design	961
<i>Ning Cheng, Wenju Liu, Lan Wang</i>	
Reduction of Broadband Noise in Speech Signals by Multilinear Subspace Analysis	965
<i>Yusuke Sato, Tetsuya Hoya, Hovagim Bakardjian, Andrzej Cichocki</i>	

Novel Probabilistic Control of Noise Reduction for Improved Microphone Array Beamforming	969
<i>Jungpyo Hong, Seunggho Han, Sangbae Jeong, Minsoo Hahn</i>	
Speech Enhancement Using Improved Generalized Sidelobe Canceller in Frequency Domain with Multi-Channel Postfiltering	973
<i>Kai Li, Qiang Fu, Yonghong Yan</i>	
Close Speaker Cancellation for Suppression of Non-Stationary Background Noise for Hands-Free Speech Interface	977
<i>Jani Even, Carlos Ishi, Hiroshi Saruwatari, Norihiro Hagita</i>	
Multi-Channel Iterative Dereverberation Based on Codebook Constrained Iterative Multi-Channel Wiener Filter	981
<i>Ajay S., T. V. Sreenivas</i>	
Speaker-Dependent Mapping of Source and System Features for Enhancement of Throat Microphone Speech	985
<i>J. M. Anand, R. M. Sri Harish, B. Yegnanarayana</i>	
An Analytic Modeling Approach to Enhancing Throat Microphone Speech Commands for Keyword Spotting	989
<i>Jun Cai, Stefano Marini, Pierre Malarme, Francis Grenez, Jean Schoentgen</i>	
Single-Channel Speech Enhancement Using Kalman Filtering in the Modulation Domain	993
<i>Stephen So, Kamil K. Wojcicki, Kuldip K. Paliwal</i>	
Integrated Feedback and Noise Reduction Algorithm in Digital Hearing Aids via Oscillation Detection	997
<i>Miao Yao, Weiqian Liang</i>	
A Blind Signal-to-Noise Ratio Estimator for High Noise Speech Recordings	1001
<i>Charles Mercier, Roch Lefebvre</i>	

TUE-SES2-S1: SPECIAL SESSION: FACT AND REPLICA OF SPEECH PRODUCTION

Estimation of Glottal Area Function Using Stereo-Endoscopic High-Speed Digital Imaging	1005
<i>Hiroshi Imagawa, Ken-Ichi Sakakibara, Isao T. Tokuda, Mamiko Otsuka, Niro Tayama</i>	
Toward Aero-Acoustical Analysis of the Sibilant /s/: An Oral Cavity Modeling	1009
<i>Kazunori Nozaki, Youhei Ohnishi, Takeshi Suda, Shigeo Wada, Shinji Shimojo</i>	
Effects of Wall Impedance on Transmission and Attenuation of Higher-Order Modes in Vocal-Tract Model	1013
<i>Kunitoshi Motoki</i>	
Articulatory Synthesis and Perception of Plosive-Vowel Syllables with Virtual Consonant Targets	1017
<i>Peter Birkholz, Bernd J. Kroger, Christiane Neuschaefer-Rube</i>	
Speech Robot Mimicking Human Articulatory Motion	1021
<i>Kotaro Fukui, Toshihiro Kusano, Yoshikazu Mukaeda, Yuto Suzuki, Atsuo Takanishi, Masaaki Honda</i>	
Mechanical Vocal-Tract Models for Speech Dynamics	1025
<i>Takayuki Arai</i>	
Prosodic Timing Analysis for Articulatory Re-Synthesis Using a Bank of Resonators with an Adaptive Oscillator	1029
<i>Michael C. Brady</i>	

TUE-SES2-O1: ASR: LANGUAGE MODELING

Decoding with Shrinkage-Based Language Models	1033
<i>Ahmad Emami, Stanley F. Chen, Abraham Ittycheriah, Hagen Soltau, Bing Zhao</i>	
Enhanced Word Classing for Model M	1037
<i>Stanley F. Chen, Stephen M. Chu</i>	
Improved Neural Network Based Language Modelling and Adaptation	1041
<i>J. Park, X. Liu, M. J. F. Gales, P. C. Woodland</i>	
Recurrent Neural Network Based Language Model	1045
<i>Tomas Mikolov, Martin Karafiat, Lukas Burget, Jan Cernocky, Sanjeev Khudanpur</i>	
Discriminative Language Modeling Using Simulated ASR Errors	1049
<i>Preethi Jyothi, Eric Fosler-Lussier</i>	
Learning a Language Model from Continuous Speech	1053
<i>Graham Neubig, Masato Mimura, Shinsuke Mori, Tatsuya Kawahara</i>	

TUE-SES2-O2: SPEAKER CHARACTERIZATION AND RECOGNITION II

Looking for Relevant Features for Speaker Role Recognition	1057
<i>Benjamin Bigot, Julien Pinquier, Isabelle Ferrane, Regine Andre-Obrecht</i>	
Prosodic Speaker Verification Using Subspace Multinomial Models with Intersession Compensation	1061
<i>Marcel Kockmann, Lukas Burget, Ondrej Glembek, Luciana Ferrer, Jan Cernocky</i>	
The Estimation and Kernel Metric of Spectral Correlation for Text-Independent Speaker Verification	1065
<i>Eryu Wang, Kong Aik Lee, Bin Ma, Haizhou Li, Wu Guo, Lirong Dai</i>	
Improving Monaural Speaker Identification by Double-Talk Detection	1069
<i>Rahim Saeidi, Pejman Mowlae, Tomi Kinnunen, Zheng-Hua Tan, Mads Graesboll Christensen, Soren Holdt Jensen, Pasi Franti</i>	
Exploring Subsegmental and Suprasegmental Features for a Text-Dependent Speaker Verification in Distant Speech Signals	1073
<i>B. Avinash, S. Guruprasad, B. Yegnanarayana</i>	
A Fast Implementation of Factor Analysis for Speaker Verification	1077
<i>Qingsong Liu, Wei Huang, Dongxing Xu, Hongbin Cai, Beiqian Dai</i>	

TUE-SES2-O3: SINGLE-CHANNEL SPEECH ENHANCEMENT

Fast Converging Iterative Kalman Filtering for Speech Enhancement Using Long and Overlapped Tapered Windows with Large Side Lobe Attenuation	1081
<i>Stephen So, Kuldip K. Paliwal</i>	
Robust Noise Estimation Using Minimum Correction with Harmonicity Control	1085
<i>Xuejing Sun, Kuan-Chieh Yen, Rogerio Alves</i>	
New Insights into Subspace Noise Tracking	1089
<i>Mahdi Triki</i>	
Bias Considerations for Minimum Subspace Noise Tracking	1093
<i>Mahdi Triki, Kees Jansse</i>	
A Corpus-Based Approach to Speech Enhancement from Nonstationary Noise	1097
<i>Ji Ming, Ramji Srinivasan, Danny Crookes</i>	
Bandwidth Expansion of Speech Based on Wavelet Transform Modulus Maxima Vector Mapping	1101
<i>Zhe Chen, You-Chi Cheng, Fuliang Yin, Chin-Hui Lee</i>	

TUE-SES2-O4: SPEECH SYNTHESIS IV: MISCELLANEOUS TOPICS

Hidden Markov Models with Context-Sensitive Observations for Grapheme-to-Phoneme Conversion	1105
<i>Kalu U. Ogbureke, Peter Cahill, Julie Carson-Berndsen</i>	
Evaluating a Dialog Language Generation System: Comparing the Mountain System to Other NLG Approaches	1109
<i>Brian Langner, Stephan Vogel, Alan W. Black</i>	
Active Appearance Models for Photorealistic Visual Speech Synthesis	1113
<i>Wesley Mattheyses, Lukas Latacz, Werner Verhelst</i>	
Latent Affective Mapping: A Novel Framework for the Data-Driven Analysis of Emotion in Text	1117
<i>Jerome R. Bellegarda</i>	
Native and Non-Native Speaker Judgements on the Quality of Synthesized Speech	1121
<i>Anna C. Janska, Robert A. J. Clark</i>	
Machine Learning for Text Selection with Expressive Unit-Selection Voices	1125
<i>Dominic Espinosa, Michael White, Eric Foster-Lussier, Chris Brew</i>	

TUE-SES2-P1: PROSODY: BASICS & APPLICATIONS

Acoustic Correlates of Meaning Structure in Conversational Speech	1129
<i>Alexei V. Ivanov, Giuseppe Riccardi, S. Ghosh, S. Tonelli, E. A. Stepanov</i>	
HMM-Based Prosodic Structure Model Using Rich Linguistic Context	1133
<i>Nicolas Obin, Xavier Rodet, Anne Lacheret</i>	
Audiovisual Congruence and Pragmatic Focus Marking	1137
<i>Charlotte Wollermann, Bernhard Schroder, Ulrich Schade</i>	
Redescribing Intonational Categories with Functional Data Analysis	1141
<i>Margaret Zellers, Michele Gubian, Brechtje Post</i>	
Exploring Goodness of Prosody by Diverse Matching Templates	1145
<i>Shen Huang, Hongyan Li, Shijin Wang, Jiaen Liang, Bo Xu</i>	
A Language-Identification Inspired Method for Spontaneous Speech Detection	1149
<i>Mickael Rouvier, Richard Dufour, Georges Linares, Yannick Esteve</i>	
Speech Dominoes and Phonetic Convergence	1153
<i>Gerard Bailly, Amelie Lelong</i>	
A Quick Sequential Forward Floating Feature Selection Algorithm for Emotion Detection from Speech	1157
<i>Matyas Brendel, Riccardo Zaccarelli, Laurence Devillers</i>	
Automated Vocal Emotion Recognition Using Phoneme Class Specific Features	1161
<i>Geza Kiss, Jan P. H. Van Santen</i>	
Feature Selection for Pose Invariant Lip Biometrics	1165
<i>Adrian Pass, Jianguo Zhang, Darryl Stewart</i>	
Signal-Based Accent and Phrase Marking Using the Fujisaki Model	1169
<i>Hussein Hussein, Rudiger Hoffmann</i>	
A Study of Interplay Between Articulatory Movement and Prosodic Characteristics in Emotional Speech Production	1173
<i>Jangwon Kim, Sungbok Lee, Shrikanth S. Narayanan</i>	

TUE-SES2-P2: ASR: FEATURE EXTRACTION I

Improved Phoneme Recognition by Integrating Evidence from Spectro-Temporal and Cepstral Features	1177
<i>Shang-Wen Li, Liang-Che Sun, Lin-Shan Lee</i>	
Using Spectro-Temporal Features to Improve AFE Feature Extraction for ASR	1181
<i>Suman V. Ravuri, Nelson Morgan</i>	
Using Harmonic Phase Information to Improve ASR Rate	1185
<i>Ibon Saratxaga, Inma Hernaez, Igor Odriozola, Eva Navas, Iker Luengo, Daniel Erra</i>	

Speech Recognition Using Long-Term Phase Information	1189
<i>Kazumasa Yamamoto, Eiichi Sueyoshi, Seiichi Nakagawa</i>	
Low-Dimensional Space Transforms of Posteriors in Speech Recognition	1193
<i>Jan Zelinka, Jan Trmal, Ludek Muller</i>	
Hierarchical Bottle Neck Features for LVCSR	1197
<i>Christian Plahl, Ralf Schluter, Hermann Ney</i>	
Hierarchical Neural Net Architectures for Feature Extraction in ASR	1201
<i>Frantisek Grezl, Martin Karaftat</i>	
Mutual Information Analysis for Feature and Sensor Subset Selection in Surface Electromyography Based Speech Recognition	1205
<i>Vivek Kumar Rangarajan Sridhar, Rohit Prasad, Prem Natarajan</i>	
Learning from Human Errors: Prediction of Phoneme Confusions Based on Modified ASR Training	1209
<i>Bernd T. Meyer, Birger Kollmeier</i>	

TUE-SES2-P3: SPEECH PERCEPTION II: CROSS LANGUAGE AND AGE

Speech Intelligibility of Diagonally Localized Speech with Competing Noise Using Bone-Conduction Headphones	1213
<i>Kazuhiro Kondo, Takayuki Kanda, Yosuke Kobayashi, Hiroyuki Yagyu</i>	
Masking of Vowel-Analog Transitions by Vowel-Analog Distracters	1217
<i>Pierre L. Divenyi</i>	
2010, a Speech Oddity: Phonetic Transcription of Reversed Speech	1221
<i>Francois Pellegrino, Emmanuel Ferragne, Fanny Meunier</i>	
Perception on Pitch Reset at Discourse Boundaries	1225
<i>Hsin-Yi Lin, Janice Fon</i>	
Effect of Spatial Separation on Speech-in-Noise Comprehension in Dyslexic Adults	1229
<i>Marjorie Dole, Michel Hoen, Fanny Meunier</i>	
Speech Categorization Context Effects in Seven- to Nine-Month-Old Infants	1233
<i>Ellen Marklund, Francisco Lacerda, Anna Ericsson</i>	
Changes in Temporal Processing of Speech Across the Adult Lifespan	1237
<i>Diane Kewley-Port, Larry E. Humes, Daniel Fogerty</i>	
Fluency and Structural Complexity as Predictors of L2 Oral Proficiency	1241
<i>Jared Bernstein, Jian Cheng, Masanori Suzuki</i>	
Semantic Facilitation in Bilingual Everyday Speech Comprehension	1245
<i>Marco Van De Ven, Benjamin V. Tucker, Mirjam Ernestus</i>	
L2 Experience and Non-Native Vowel Categorization of L1-Mandarin Speakers	1249
<i>Bo-Ren Hsieh, Ho-Hsien Pan</i>	
Cross-Lingual Talker Discrimination	1253
<i>Mirjam Wester</i>	
Dajare is Not the Lowest Form of Wit	1257
<i>Takashi Otake</i>	

TUE-SES2-P4: SLP SYSTEMS

Comparison of Methods for Topic Classification in a Speech-Oriented Guidance System	1261
<i>Rafael Torres, Shota Takeuchi, Hiromichi Kawanami, Tomoko Matsui, Hiroshi Saruwatari, Kiyohiro Shikano</i>	
Using Dependency Parsing and Machine Learning for Factoid Question Answering on Spoken Documents	1265
<i>Pere R. Comas, Jordi Turmo, Lluís Marquez</i>	
A Spoken Term Detection Framework for Recovering Out-of-Vocabulary Words Using the Web	1269
<i>Carolina Parada, Abhinav Sethy, Mark Dredze, Frederick Jelinek</i>	
Improved Spoken Term Detection by Discriminative Training of Acoustic Models Based on User Relevance Feedback	1273
<i>Hung-Yi Lee, Chia-Ping Chen, Ching-Feng Yeh, Lin-Shan Lee</i>	
A Lightweight Keyword and Tag-Cloud Retrieval Algorithm for Automatic Speech Recognition Transcripts	1277
<i>Sebastian Tschopel, Daniel Schneider</i>	
Lecture Subtopic Retrieval by Retrieval Keyword Expansion Using Subordinate Concept	1281
<i>Noboru Kanedera, Tetsuo Funada, Seiichi Nakagawa</i>	
Spoken Document Retrieval for Oral Presentations Integrating Global Document Similarities into Local Document Similarities	1285
<i>Hiroaki Nanjo, Yusuke Iyonaga, Takehiko Yoshimi</i>	
Combining Word-Based Features, Statistical Language Models, and Parsing for Named Entity Recognition	1289
<i>Joseph Polifroni, Stephanie Seneff</i>	
Efficient Combined Approach for Named Entity Recognition in Spoken Language	1293
<i>Azeddine Zidouni, Sophie Rosset, Herve Glotin</i>	
Prominence Based Scoring of Speech Segments for Automatic Speech-to-Speech Summarization	1297
<i>Sree Harsha Yella, Vasudeva Varma, Kishore Prahallad</i>	
Maximum Lexical Cohesion for Fine-Grained News Story Segmentation	1301
<i>Zihan Liu, Lei Xie, Wei Feng</i>	
Phoneme Lattice Based TextTiling Towards Multilingual Story Segmentation	1305
<i>Xiaoxuan Wang, Lei Xie, Bin Ma, Eng Siong Chng, Haizhou Li</i>	

TUE-SES3-S1: SPECIAL SESSION: QUALITY OF EXPERIENCING SPEECH SERVICES

The Characterization of the Relative Information Content by Spectral Features for the Objective Intelligibility Assessment of Nonlinearly Processed Speech	1309
<i>Anton Schlesinger, Marinus M. Boone</i>	
Analytical Assessment and Distance Modeling of Speech Transmission Quality	1313
<i>Marcel Waltermann, Alexander Raake, Sebastian Moller</i>	
An Intrusive Super-Wideband Speech Quality Model: DIAL	1317
<i>Nicolas Cote, Vincent Koehl, Valerie Gautier-Turbin, Alexander Raake, Sebastian Moller</i>	
It Takes Two to Tango --- Assessing the Impact of Delay on Conversational Interactivity on Perceived Speech Quality	1321
<i>Sebastian Egger, Raimund Schatz, Stefan Scherer</i>	
Comparison of Approaches for Instrumentally Predicting the Quality of Text-to-Speech Systems	1325
<i>Sebastian Moller, Florian Hinterleitner, Tiago H. Falk, Tim Polzehl</i>	
A Hybrid Architecture for Mobile Voice User Interfaces	1329
<i>Imre Kiss, Joseph Polifroni, Chao Wang, Ghinwa Choueiter, Mike Phillips</i>	
Assessment of Spoken and Multimodal Applications: Lessons Learned from Laboratory and Field Studies	1333
<i>Markku Turunen, Jaakko Hakulinen, Tomi Heimonen</i>	
Improving Cross Database Prediction of Dialogue Quality Using Mixture of Experts	1337
<i>Klaus-Peter Engelbrecht, Hamed Ketabdar, Sebastian Moller</i>	

TUE-SES3-O1: ASR: ACOUSTIC MODELS II

Boosting Systems for LVCSR	1341
<i>George Saon, Hagen Soltau</i>	
Incorporating Sparse Representation Phone Identification Features in Automatic Speech Recognition Using Exponential Families	1345
<i>Vaibhava Goel, Tara N. Sainath, Bhuvana Ramabhadran, Peder Olsen, David Nahamoo, Dimitri Kanevsky</i>	
Integrating MLP Features and Discriminative Training in Data Sampling Based Ensemble Acoustic Modeling	1349
<i>Xin Chen, Yunxin Zhao</i>	
Semi-Supervised Training of Gaussian Mixture Models by Conditional Entropy Minimization	1353
<i>Jui-Ting Huang, Mark Hasegawa-Johnson</i>	
A Study of Irrelevant Variability Normalization Based Training and Unsupervised Online Adaptation for LVCSR	1357
<i>Guangchuan Shi, Yu Shi, Qiang Huo</i>	
Improvements to Generalized Discriminative Feature Transformation for Speech Recognition	1361
<i>Roger Hsiao, Florian Metze, Tanja Schultz</i>	

TUE-SES3-O2: LANGUAGE PROCESSING

Improving ASR-Based Topic Segmentation of TV Programs with Confidence Measures and Semantic Relations	1365
<i>Camille Guinaudeau, Guillaume Gravier, Pascale Sebillot</i>	
The Relevance of Timing, Pauses and Overlaps in Dialogues: Detecting Topic Changes in Scenario Based Meetings	1369
<i>Saturnino Luz, Jing Su</i>	
Semi-Supervised Part-of-Speech Tagging in Speech Applications	1373
<i>Richard Dufour, Benoit Favre</i>	
Memory-Based Active Learning for French Broadcast News	1377
<i>Frederic Tantini, Christophe Cerisara, Claire Gardent</i>	
Can Conversational Word Usage Be Used to Predict Speaker Demographics?	1381
<i>Dan Gillick</i>	
Prosodic Word-Based Error Correction in Speech Recognition Using Prosodic Word Expansion and Contextual Information	1385
<i>Chao-Hong Liu, Chung-Hsien Wu</i>	

TUE-SES3-O3: SPEECH AND AUDIO SEGMENTATION

Fully Automatic Segmentation for Prosodic Speech Corpora	1389
<i>Sarah Hoffmann, Beat Pfister</i>	
A Novel Text-Independent Phonetic Segmentation Algorithm Based on the Microcanonical Multiscale Formalism	1393
<i>Vahid Khanagha, Khalid Daoudi, Oriol Pont, Hussein Yahia</i>	
Phone Boundary Detection Using Sample-Based Acoustic Parameters	1397
<i>You-Yu Lin, Yih-Ru Wang, Yuan-Fu Liao</i>	
HMM-Based Automatic Visual Speech Segmentation Using Facial Data	1401
<i>Utpala Musti, Asterios Toutios, Slim Ouni, Vincent Colotte, Brigitte Wrobel-Dautcourt, Marie-Odile Berger</i>	
Bayes Factor Based Speaker Segmentation for Speaker Diarization	1405
<i>D. Wang, Robert Vogt, Sridha Sridharan</i>	
Using High-Level Information to Detect Key Audio Events in a Tennis Game	1409
<i>Qiang Huang, Stephen Cox</i>	

TUE-SES3-O4: PROSODY: ANALYSIS

What Do You Mean, You're Uncertain?: The Interpretation of Cue Words and Rising Intonation in Dialogue	1413
<i>Catherine Lai</i>	
Coping Imbalanced Prosodic Unit Boundary Detection with Linguistically-Motivated Prosodic Features	1417
<i>Yi-Fen Liu, Shu-Chuan Tseng, Jyh-Shing Roger Jang, C.-H. Alvin Chen</i>	
Improving Prosodic Phrase Prediction by Unsupervised Adaptation and Syntactic Features Extraction	1421
<i>Zhigang Chen, Guoping Hu, Wei Jiang</i>	
Perception-Based Automatic Approximation of F0 Contours in Cantonese Speech	1425
<i>Yujia Li, Tan Lee</i>	
Discriminative Training and Unsupervised Adaptation for Labeling Prosodic Events with Limited Training Data	1429
<i>Raul Fernandez, Bhuvana Ramabhadran</i>	
Prosody for the Eyes: Quantifying Visual Prosody Using Guided Principal Component Analysis	1433
<i>Erin Cvejic, Jeesun Kim, Chris Davis, Guillaume Gibert</i>	

TUE-SES3-P1: SPEAKER CHARACTERIZATION AND RECOGNITION III

An Investigation into Direct Scoring Methods Without SVM Training in Speaker Verification	1437
<i>Ce Zhang, Rong Zheng, Bo Xu</i>	
Large Margin Gaussian Mixture Models for Speaker Identification	1441
<i>Reda Jourani, Khalid Daoudi, Regine Andre-Obrecht, Driss Aboutajdine</i>	
On the Use of Gaussian Component Information in the Generative Likelihood Ratio Estimation for Speaker Verification	1445
<i>Rong Zheng, Bo Xu</i>	
Acoustic Vector Resampling for GMMSVM-Based Speaker Verification	1449
<i>Man-Wai Mak, Wei Rao</i>	
A Fast Speaker Indexing Using Vector Quantization and Second Order Statistics with Adaptive Threshold Computation	1453
<i>Konstantin Biatov</i>	
Using Phoneme Recognition and Text-Dependent Speaker Verification to Improve Speaker Segmentation for Chinese Speech	1457
<i>Gang Wang, Xiaojun Wu, Thomas Fang Zheng</i>	
On Enhancing Feature Sequence Filtering with Filter-Bank Energy Transformation in Speaker Verification with Telephone Speech	1461
<i>Claudio Garreton, Nestor Becerra Yoma</i>	
MAP Estimation of Subspace Transform for Speaker Recognition	1465
<i>Donglai Zhu, Bin Ma, Kong Aik Lee, Cheung-Chi Leung, Haizhou Li</i>	
A Longest Matching Segment Approach for Text-Independent Speaker Recognition	1469
<i>Ayeh Jafari, Ramji Srinivasan, Danny Crookes, Ji Ming</i>	
Approaching Human Listener Accuracy with Modern Speaker Verification	1473
<i>Ville Hautamaki, Tomi Kinnunen, Mohaddeseh Nosratighods, Kong Aik Lee, Bin Ma, Haizhou Li</i>	
Extended Weighted Linear Prediction (XLP) Analysis of Speech and Its Application to Speaker Verification in Adverse Conditions	1477
<i>Jouni Pohjalainen, Rahim Saeidi, Tomi Kinnunen, Paavo Alku</i>	
The Use of Subvector Quantization and Discrete Densities for Fast GMM Computation for Speaker Verification	1481
<i>Guoli Ye, Brian Mak</i>	

TUE-SES3-P2: SYSTEMS FOR LVCSR AND RICH TRANSCRIPTION

Parallel Lexical-Tree Based LVCSR on Multi-Core Processors	1485
<i>Naveen Parihar, Ralf Schluter, David Rybach, Eric A. Hansen</i>	
Exploring Recognition Network Representations for Efficient Speech Inference on Highly Parallel Platforms	1489
<i>Jike Chong, Ekaterina Gonina, Kisun You, Kurt Keutzer</i>	
WFST Compression for Automatic Speech Recognition	1493
<i>Diamantino Caseiro</i>	
Speech Recognizer Optimization Under Speed Constraints	1497
<i>Ivan Bulyko</i>	
The 2010 CMU GALE Speech-to-Text System	1501
<i>Florian Metzke, Roger Hsiao, Qin Jin, Udhyakumar Nallasamy, Tanja Schultz</i>	
Speaker Diarization in Meeting Audio for Single Distant Microphone	1505
<i>Tin Lay Nwe, Hanwu Sun, Bin Ma, Haizhou Li</i>	
Extending the Punctuation Module for European Portuguese	1509
<i>Fernando Batista, Helena Moniz, Isabel Trancoso, Hugo Meinedo, Ana Isabel Mata, Nuno Mamede</i>	
Utilizing a Noisy-Channel Approach for Korean LVCSR	1513
<i>Sakriani Sakti, Ryosuke Isotani, Hisashi Kawai, Satoshi Nakamura</i>	
The RWTH 2009 Quaero ASR Evaluation System for English and German	1517
<i>Markus Nussbaum-Thom, Simon Wiesler, Martin Sundermeyer, Christian Plahl, Stefan Hahn, Ralf Schluter, Hermann Ney</i>	

TUE-SES3-P3: PHONETICS

When is Indexical Information About Speech Activated? Evidence from a Cross-Modal Priming Experiment	1521
<i>Benjamin Munson, Renata Solum</i>	
The Influence of Actual and Perceived Sexual Orientation on Diadochokinetic Rate in Women and Men	1525
<i>Benjamin Munson</i>	
Laryngealization and Features for Chinese Tonal Recognition	1529
<i>Kristine M. Yu</i>	
Production and Perception of Vietnamese Short Vowels in V1V2 Context	1533
<i>Viet Son Nguyen, Eric Castelli, Rene Carre</i>	
Measuring Basic Tempo Across Languages and Some Implications for Speech Rhythm	1537
<i>Gertraud Fenk-Oczlon, August Fenk</i>	
Durational Structure of Japanese Single/Geminate Stops in Three- and Four-Mora Words Spoken at Varied Rates	1541
<i>Yukari Hirata, Shigeaki Amano</i>	
Distribution and Trichotomic Realization of Voiced Velars in Japanese --- An Experimental Study	1545
<i>Shin-Ichiro Sano, Tomohiko Ooigawa</i>	
Specification in Context --- Devoicing Processes in Polish, French, American English and German Sonorants	1549
<i>Jagoda Sieczkowska, Bernd Mobius, Grzegorz Dogil</i>	
Phonetic Imitation of Japanese Vowel Devoicing	1553
<i>Kuniko Nielsen</i>	

VOLUME 3

Post-Aspiration in Standard Italian: Some First Cross-Regional Acoustic Evidence	1557
<i>Mary Stevens, John Hajek</i>	
Articulatory Grounding of Southern Salentino Harmony Processes	1561
<i>Mirko Grimaldi, Andrea Calabrese, Francesco Sigona, Luigina Garrapa, Bianca Sisinni</i>	
Effects of Accent Typicality and Phonotactic Frequency on Nonword Immediate Serial Recall Performance in Japanese	1565
<i>Yuuki Tanida, Taiji Ueno, Satoru Saito, Matthew A. Lambon Ralph</i>	
How Abstract Is Phonetics?	1568
<i>Osamu Fujimura</i>	

TUE-SES3-P4: SPEECH PRODUCTION II: VOCAL TRACT MODELING AND IMAGING

Data-Driven Analysis of Realtime Vocal Tract MRI Using Correlated Image Regions	1572
<i>Adam C. Lammert, Michael I. Proctor, Shrikanth S. Narayanan</i>	
Rapid Semi-Automatic Segmentation of Real-Time Magnetic Resonance Images for Parametric Vocal Tract Analysis	1576
<i>Michael I. Proctor, Daniel Bone, Athanasios Katsamanis, Shrikanth S. Narayanan</i>	
Improved Real-Time MRI of Oral-Velar Coordination Using a Golden-Ratio Spiral View Order	1580
<i>Yoon-Chul Kim, Shrikanth S. Narayanan, Krishna S. Nayak</i>	
Statistical Multi-Stream Modeling of Real-Time MRI Articulatory Speech Data	1584
<i>Erik Bresch, Athanasios Katsamanis, Louis Goldstein, Shrikanth S. Narayanan</i>	
Predicting Unseen Articulations from Multi-Speaker Articulatory Models	1588
<i>G. Ananthakrishnan, Pierre Badin, Julian Andres Valdes Vargas, Olov Engwall</i>	
Estimating Missing Data Sequences in X-Ray Microbeam Recordings	1592
<i>Chao Qin, Miguel A. Carreira-Perpinan</i>	
Adaptation of a Tongue Shape Model by Local Feature Transformations	1596
<i>Chao Qin, Miguel A. Carreira-Perpinan, Mohsen Farhadloo</i>	
Vocal Tract Contour Analysis of Emotional Speech by the Functional Data Curve Representation	1600
<i>Sungbok Lee, Shrikanth S. Narayanan</i>	
Locally-Weighted Regression for Estimating the Forward Kinematics of a Geometric Vocal Tract Model	1604
<i>Adam C. Lammert, Louis Goldstein, Khalil Iskarous</i>	
Identifying Articulatory Goals from Kinematic Data Using Principal Differential Analysis	1608
<i>Michael Reimer, Frank Rudzicz</i>	
Estimation of Speech Lip Features from Discrete Cosinus Transform	1612
<i>Zuheng Ming, Denis Beautemps, Gang Feng, Sebastien Schmerber</i>	
Autoregressive Modelling for Linear Prediction of Ultrasonic Speech	1616
<i>Farzaneh Ahmadi, Ian V. McLoughlin, Hamid R. Sharifzadeh</i>	

WED-SES1-S1: SPECIAL SESSION: SPEECH INTELLIGIBILITY ENHANCEMENT FOR ALL AGES, HEALTH CONDITIONS AND ENVIRONMENTS

Enhanced Speech Yielding Higher Intelligibility for All Listeners and Environments	1620
<i>Takayuki Arai, Nao Hodoshima</i>	

Quality Conversion of Non-Acoustic Signals for Facilitating Human-to-Human Speech Communication Under Harsh Acoustic Conditions	1624
<i>Seyed Omid Sadjadi, Sanjay A. Patil, John H. L. Hansen</i>	
The Use of Air-Pressure Sensor in Electrolaryngeal Speech Enhancement Based on Statistical Voice Conversion	1628
<i>Keigo Nakamura, Tomoki Toda, Hiroshi Saruwatari, Kiyohiro Shikano</i>	
A New Binary Mask Based on Noise Constraints for Improved Speech Intelligibility	1632
<i>Gibak Kim, Philipos C. Loizou</i>	
Energy Reallocation Strategies for Speech Enhancement in Known Noise Conditions	1636
<i>Yan Tang, Martin Cooke</i>	
Effects of Enhancement of Spectral Changes on Speech Quality and Subjective Speech Intelligibility	1640
<i>Jing Chen, Thomas Baer, Brian C. J. Moore</i>	

WED-SES1-O1: ASR: ACOUSTIC MODEL ADAPTATION

Prior Information for Rapid Speaker Adaptation	1644
<i>C. Breslin, K. K. Chin, M. J. F. Gales, Kate Knill, H. Xu</i>	
Discriminative Adaptation for Log-Linear Acoustic Models	1648
<i>Jonas Loof, Ralf Schluter, Hermann Ney</i>	
Automatic Speech Recognition of Multiple Accented English Data	1652
<i>Dimitra Vergyri, Lori Lamel, Jean-Luc Gauvain</i>	
Shrinkage Model Adaptation in Automatic Speech Recognition	1656
<i>Jinyu Li, Yu Tsao, Chin-Hui Lee</i>	
Unscented Transform with Online Distortion Estimation for HMM Adaptation	1660
<i>Jinyu Li, Dong Yu, Yifan Gong, L. Deng</i>	
HMM Adaptation Using Linear Spline Interpolation with Integrated Spline Parameter Training for Robust Speech Recognition	1664
<i>Michael L. Seltzer, Alex Acero</i>	

WED-SES1-O2: SLP SYSTEMS FOR INFORMATION EXTRACTION/RETRIEVAL

CRF-Based Stochastic Pronunciation Modeling for Out-of-Vocabulary Spoken Term Detection	1668
<i>Dong Wang, Simon King, Nicholas Evans, Raphael Troncy</i>	
Improved Spoken Term Detection by Feature Space Pseudo-Relevance Feedback	1672
<i>Chia-Ping Chen, Hung-Yi Lee, Ching-Feng Yeh, Lin-Shan Lee</i>	
Towards Spoken Term Discovery at Scale with Zero Resources	1676
<i>Aren Jansen, Kenneth Church, Hynek Hermansky</i>	
Vocabulary Independent Spoken Query: A Case for Subword Units	1680
<i>Evandro Gouvea, Tony Ezzat</i>	
Extractive Speech Summarization --- From the View of Decision Theory	1684
<i>Shih-Hsiang Lin, Yao-Ming Yeh, Berlin Chen</i>	
The Impact of ASR on Abstractive vs. Extractive Meeting Summaries	1688
<i>Gabriel Murray, Giuseppe Carenini, Raymond Ng</i>	

WED-SES1-O3: SPEECH REPRESENTATION

Binary Coding of Speech Spectrograms Using a Deep Auto-Encoder	1692
<i>L. Deng, Michael L. Seltzer, Dong Yu, Alex Acero, Abdel-Rahman Mohamed, G. Hinton</i>	
A Super-Resolution Spectrogram Using Coupled PLCA	1696
<i>Juhan Nam, Gautham J. Mysore, Joachim Ganseman, Kyogu Lee, Jonathan S. Abel</i>	
Fast Least-Squares Solution for Sinusoidal, Harmonic and Quasi-Harmonic Models	1700
<i>Georgios Tzedakis, Yannis Pantazis, Olivier Rosec, Yannis Stylianou</i>	
Sparse Component Analysis for Speech Recognition in Multi-Speaker Environment	1704
<i>Afsaneh Asaei, Herve Bourlard, Philip N. Garner</i>	
Intra-Frame Variability as a Predictor of Frame Classifiability	1708
<i>Trond Skogstad, Torbjorn Svendsen</i>	
Autocorrelation and Double Autocorrelation Based Spectral Representations for a Noisy Word Recognition System	1712
<i>Tetsuya Shimamura, Ngoc Dinh Nguyen</i>	

WED-SES1-O4: VOICE CONVERSION

Maximum a posteriori Voice Conversion Using Sequential Monte Carlo Methods	1716
<i>Elina Helander, Hanna Silen, Joaquin Miguez, Moncef Gabbouj</i>	
Dynamic Model Selection for Spectral Voice Conversion	1720
<i>Pierre Lanchantin, Xavier Rodet</i>	
Speaker-Independent HMM-Based Voice Conversion Using Quantized Fundamental Frequency	1724
<i>Takashi Nose, Takao Kobayashi</i>	

Probabilistic Integration of Joint Density Model and Speaker Model for Voice Conversion	1728
<i>Daisuke Saito, Shinji Watanabe, Atsushi Nakamura, Nobuaki Minematsu</i>	
Text-Independent F0 Transformation with Non-Parallel Data for Voice Conversion	1732
<i>Zhi-Zheng Wu, Tomi Kinnunen, Eng Siong Chng, Haizhou Li</i>	
A Minimum Converted Trajectory Error (MCTE) Approach to High Quality Speech-to-Lips Conversion	1736
<i>Xiaodan Zhuang, Lijuan Wang, Frank K. Soong, Mark Hasegawa-Johnson</i>	

WED-SES1-P1: PROSODY: LANGUAGE-SPECIFIC MODELS

Influence of Lexical Tones on Intonation in Kammu	1740
<i>Anastasia Karlsson, David House, Jan-Olof Svantesson, Damrong Tayanin</i>	
Phonetic Realization of Second Occurrence Focus in Japanese	1744
<i>Satoshi Nambu, Yong-Cheol Lee</i>	
Prosodic Grouping and Relative Clause Disambiguation in Mandarin	1748
<i>Jianjing Kuang</i>	
Text-Based Unstressed Syllable Prediction in Mandarin	1752
<i>Ya Li, Jianhua Tao, Meng Zhang, Shifeng Pan, Xiaoying Xu</i>	
"Flat Pitch Accents" in Czech	1756
<i>Tomas Dubeda</i>	
Positional Variability of Pitch Accents in Czech	1760
<i>Tomas Dubeda</i>	
Modeling of Sentence-Medial Pauses in Bangla Readout Speech: Occurrence and Duration	1764
<i>Shyamal Das Mandal, Arup Saha, Tulika Basu, Keikichi Hirose, Hiroya Fujisaki</i>	
Declarative Sentence Intonation Patterns in 8 Swiss German Dialects	1768
<i>Adrian Leemann, Lucy Zuberbuhler</i>	
Syllable-Level Prominence Detection with Acoustic Evidence	1772
<i>Je Hun Jeon, Yang Liu</i>	
Prosody Cues for Classification of the Discourse Particle "ha" in Hindi	1776
<i>Sankalan Prasad, Kalika Bali</i>	
Interaction of Syntax-Marked Focus and Wh-Question Induced Focus in Standard Chinese	1780
<i>Yuan Jia, Aijun Li</i>	
Prominence Detection in Swedish Using Syllable Correlates	1784
<i>Samer Al Moubayed, Jonas Beskow</i>	
Automatic Analysis of the Intonation of a Tone Language. Applying the Momel Algorithm to Spontaneous Standard Chinese (Beijing)	1788
<i>Na Zhi, Daniel Hirst, Pier Marco Bertinotto</i>	
Towards Long-Range Prosodic Attribute Modeling for Language Recognition	1792
<i>Raymond W. M. Ng, Cheung-Chi Leung, Ville Hautamaki, Tan Lee, Bin Ma, Haizhou Li</i>	
A Modified Parameterization of the Fujisaki Model	1796
<i>Robert Schubert, Oliver Jokisch, Diane Hirschfeld</i>	

WED-SES1-P2: ASR: LANGUAGE MODELING AND SPEECH UNDERSTANDING I

Within and Across Sentence Boundary Language Model	1800
<i>Saeedeh Momtazi, Friedrich Faubel, Dietrich Klakow</i>	
Impact of Word Classing on Shrinkage-Based Language Models	1804
<i>Ruhi Sarikaya, Stanley F. Chen, Abhinav Sethy, Bhuvana Ramabhadran</i>	
Combination of Probabilistic and Possibilistic Language Models	1808
<i>Stanislas Oger, Vladimir Popescu, Georges Linares</i>	
On-Demand Language Model Interpolation for Mobile Speech Input	1812
<i>Brandon Ballinger, Cyril Allauzen, Alexander Gruenstein, Johan Schalkwyk</i>	
Text Normalization Based on Statistical Machine Translation and Internet User Support	1816
<i>Tim Schlippe, Chenfei Zhu, Jan Gebhardt, Tanja Schultz</i>	
Efficient Estimation of Maximum Entropy Language Models with N-Gram Features: An SRILM Extension	1820
<i>Tanel Alumae, Mikko Kurimo</i>	
Similar N-Gram Language Model	1824
<i>Christian Gillot, Christophe Cerisara, David Langlois, Jean-Paul Haton</i>	
Topic and Style-Adapted Language Modeling for Thai Broadcast News ASR	1828
<i>Markpong Jongtaveesataporn, Sadaoki Furui</i>	
Augmented Context Features for Arabic Speech Recognition	1832
<i>Ahmad Emami, Hong-Kwang J. Kuo, Imed Zitouni, Lidia Mangu</i>	
A Statistical Segment-Based Approach for Spoken Language Understanding	1836
<i>Lucia Ortega, Isabel Galiano, Lluís-F. Hurtado, Emilio Sanchis, Encarna Segarra</i>	

WED-SES1-P3: FIRST AND SECOND LANGUAGE ACQUISITION

Cantonese Tone Word Learning by Tone and Non-Tone Language Speakers	1840
<i>Angela Cooper, Yue Wang</i>	

Validation of a Training Method for L2 Continuous-Speech Segmentation	1844
<i>Anne Cutler, Janise Shanley</i>	
Linguistic Rhythm in Foreign Accent	1848
<i>Jiahong Yuan</i>	
The Effect of a Word Embedded in a Sentence and Speaking Rate Variation on the Perceptual Training of Geminate and Singleton Consonant Distinction	1850
<i>Mee Sonu, Keiichi Tajima, Hiroaki Kato, Yoshinori Sagisaka</i>	
Foreign Accent Matters Most When Timing is Wrong	1854
<i>Chiharu Tsurutani</i>	
Effects of Korean Learners' Consonant Cluster Reduction Strategies on English Speech Recognition Performance	1858
<i>Hyejin Hong, Jina Kim, Minhwa Chung</i>	
The Effects of EMA-Based Augmented Visual Feedback on the English Speakers' Acquisition of the Japanese Flap: A Perceptual Study	1862
<i>June S. Levitt, William F. Katz</i>	
Perception of Voiceless Fricatives by Japanese Listeners of Advanced and Intermediate Level English Proficiency	1866
<i>Hinako Masuda, Takayuki Arai</i>	
Perception of Estonian Vowel Categories by Native and Non-Native Speakers	1870
<i>Lya Meister, Einar Meister</i>	
Spoken English Assessment System for Non-Native Speakers Using Acoustic and Prosodic Features	1874
<i>Qin Shi, Kun Li, Shilei Zhang, Stephen M. Chu, Ji Xiao, Zhijian Ou</i>	
Russian Infants and Children's Sounds and Speech Corpora for Language Acquisition Studies	1878
<i>Elena E. Lyakso, Olga V. Frolova, Anna V. Kurazhova, Julia S. Gaikova</i>	
Language-Specific Influence on Phoneme Development: French and Drehu Data	1882
<i>Julia Monnin, Helene Loevenbruck</i>	
Did you Say Susi or Shushi? Measuring the Emergence of Robust Fricative Contrasts in English- and Japanese-Acquiring Children	1886
<i>Jeffrey J. Holliday, Mary E. Beckman, Chantelle Mays</i>	

WED-SES1-P4: SPOKEN LANGUAGE RESOURCES, SYSTEMS AND EVALUATION I

An Empirical Comparison of the T³, Juicer, HDecode and Sphinx3 Decoders	1890
<i>Josef R. Novak, Paul R. Dixon, Sadaoki Furui</i>	
Tracter: A Lightweight Dataflow Framework	1894
<i>Philip N. Garner, John Dines</i>	
Verifying Pronunciation Dictionaries Using Conflict Analysis	1898
<i>Marelie H. Davel, Febe De Wet</i>	
Automatic Estimation of Transcription Accuracy and Difficulty	1902
<i>Brandon C. Roy, Soroush Vosoughi, Deb Roy</i>	
Creating a Linguistic Plausibility Dataset with Non-Expert Annotators	1906
<i>Benjamin Lambert, Rita Singh, Bhiksha Raj</i>	
Construction and Evaluations of an Annotated Chinese Conversational Corpus in Travel Domain for the Language Model of Speech Recognition	1910
<i>Xinhui Hu, Ryosuke Isotani, Hisashi Kawai, Satoshi Nakamura</i>	
Building Transcribed Speech Corpora Quickly and Cheaply for Many Languages	1914
<i>Thad Hughes, Kaisuke Nakajima, Linne Ha, Atul Vasu, Pedro J. Moreno, Mike Lebeau</i>	
The CHiME Corpus: A Resource and a Challenge for Computational Hearing in Multisource Environments	1918
<i>Heidi Christensen, Jon Barker, Ning Ma, Phil D. Green</i>	
Developing a Chinese L2 Speech Database of Japanese Learners with Narrow-Phonetic Labels for Computer Assisted Pronunciation Training	1922
<i>Wen Cao, Dongning Wang, Jinsong Zhang, Ziyu Xiong</i>	
How Children Acquire Situation Understanding Skills?: A Developmental Analysis Utilizing Multimodal Speech Behavior Corpus	1926
<i>Shogo Ishikawa, Shinya Kiriya, Yoichi Takebayashi, Shigeyoshi Kitazawa</i>	
The Influence of Expertise and Efficiency on Modality Selection Strategies and Perceived Mental Effort	1930
<i>Ina Wechsung, Stefan Schaffer, Robert Schleicher, Anja Naumann, Sebastian Moller</i>	
Parameters Describing Multimodal Interaction --- Definitions and Three Usage Scenarios	1934
<i>Christine Kuhnle, Benjamin Weiss, Sebastian Moller</i>	
Repair Strategies on Trial: Which Error Recovery Do Users Like Best?	1938
<i>Alexander Zgorzelski, Alexander Schmitt, Tobias Heinroth, Wolfgang Minker</i>	

WED-SES2-O1: ASR: SEARCH, DECODING AND CONFIDENCE MEASURES II

CRF-Based Combination of Contextual Features to Improve a posteriori Word-Level Confidence Measures	1942
<i>Julien Fayolle, Fabienne Moreau, Christian Raymond, Guillaume Gravier, Patrick Gros</i>	
Recognition of Spontaneous Conversational Speech Using Long Short-Term Memory Phoneme Predictions	1946
<i>Martin Wollmer, Florian Eyben, Bjorn Schuller, Gerhard Rigoll</i>	
Improving ASR Error Detection with Non-Decoder Based Features	1950
<i>Thomas Pellegrini, Isabel Trancoso</i>	

Phoneme Classification and Lattice Rescoring Based on a k-NN Approach	1954
<i>Ladan Golipour, Douglas O'Shaughnessy</i>	
Online Adaptive Learning for Speech Recognition Decoding	1958
<i>Jeff Bilmes, Hui Lin</i>	
Improvements of Search Error Risk Minimization in Viterbi Beam Search for Speech Recognition	1962
<i>Takaaki Hori, Shinji Watanabe, Atsushi Nakamura</i>	

WED-SES2-O2: SPOKEN LANGUAGE RESOURCES, SYSTEMS AND EVALUATION II

Say What? Why Users Choose to Speak Their Web Queries	1966
<i>Maryam Kamvar, Doug Beeferman</i>	
The Effect of Audience Familiarity on the Perception of Modified Accent	1970
<i>Jonathan Teutenberg, Catherine I. Watson</i>	
On Generating Complex Pronunciations via Morphological Analysis	1974
<i>Korin Richmond, Robert A. J. Clark, Sue Fitt</i>	
Say It As You Mean It --- Analyzing Free User Comments in the VOICE Awards Corpus	1978
<i>Florian Godde, Sebastian Moller</i>	
A New Multichannel Multi Modal Dyadic Interaction Database	1982
<i>Viktor Rozgic, Bo Xiao, Athanasios Katsamanis, Brian R. Baucom, Panayiotis G. Georgiou, Shrikanth S. Narayanan</i>	
SEAME: A Mandarin-English Code-Switching Speech Corpus in South-East Asia	1986
<i>Dau-Cheng Lyu, Tien-Ping Tan, Eng Siong Chng, Haizhou Li</i>	

WED-SES2-O3: SPEECH PRODUCTION III: ANALYSIS

Relying on Critical Articulators to Estimate Vocal Tract Spectra in an Articulatory-Acoustic Database	1990
<i>Daniel Felps, Christian Geng, Michael Berger, Korin Richmond, Ricardo Gutierrez-Osuna</i>	
Investigating Articulatory Setting --- Pauses, Ready Position, and Rest --- Using Real-Time MRI	1994
<i>Vikram Ramanarayanan, Dani Byrd, Louis Goldstein, Shrikanth S. Narayanan</i>	
Articulatory Inversion of American English /r/ by Conditional Density Modes	1998
<i>Chao Qin, Miguel A. Carreira-Perpinan</i>	
Can Tongue Be Recovered from Face? The Answer of Data-Driven Statistical Models	2002
<i>Atef Ben Youssef, Pierre Badin, Gerard Bailly</i>	
Phrase-Medial Vowel Devoicing in Spontaneous French	2006
<i>Francisco Torreira, Mirjam Ernestus</i>	
Exploring the Mechanism of Tonal Contraction in Taiwan Mandarin	2010
<i>Chierh Cheng, Yi Xu, Michele Gubian</i>	

WED-SES2-O4: PARALANGUAGE & COGNITION

Voice Attributes Affecting Likability Perception	2014
<i>Benjamin Weiss, Felix Burkhardt</i>	
Turn-Alignment Using Eye-Gaze and Speech in Conversational Interaction	2018
<i>Kristiina Jokinen, Kazuaki Harada, Masafumi Nishida, Seiichi Yamamoto</i>	
An Investigation of Formant Frequencies for Cognitive Load Classification	2022
<i>Tet Fei Yap, Julien Epps, Eliathamby Ambikairajah, Eric H. C. Choi</i>	
Language Specific Effects of Emotion on Phoneme Duration	2026
<i>Martijn Goudbeek, Mirjam Broersma</i>	
Automatic Classification of Married Couples' Behavior Using Audio Features	2030
<i>Matthew Black, Athanasios Katsamanis, Chi-Chun Lee, Adam C. Lammert, Brian R. Baucom, Andrew Christensen, Panayiotis G. Georgiou, Shrikanth S. Narayanan</i>	
Influence of Gestural Salience on the Interpretation of Spoken Requests	2034
<i>Gideon Kowadlo, Patrick Ye, Ingrid Zukerman</i>	

WED-SES2-P1: ROBUST ASR AGAINST NOISE

Robust Word Recognition Using Articulatory Trajectories and Gestures	2038
<i>Vikramjit Mitra, Hosung Nam, Carol Espy-Wilson, Elliot Saltzman, Louis Goldstein</i>	
Performance Estimation of Noisy Speech Recognition Considering Recognition Task Complexity	2042
<i>Takeshi Yamada, Tomohiro Nakajima, Nobuhiko Kitawaki, Shoji Makino</i>	
Estimating Noise from Noisy Speech Features with a Monte Carlo Variant of the Expectation Maximization Algorithm	2046
<i>Friedrich Faubel, Dietrich Klakow</i>	
Template-Based Spectral Estimation Using Microphone Array for Speech Recognition	2050
<i>Satoshi Tamura, Eriko Hishikawa, Wataru Taguchi, Satoru Hayamizu</i>	
A Particle Filter Feature Compensation Approach to Robust Speech Recognition	2054
<i>Aleem Mushtaq, Yu Tsao, Chin Hui-Lee</i>	
Nonlinear Enhancement of Onset for Robust Speech Recognition	2058
<i>Chanwoo Kim, Richard M. Stern</i>	

Mask Estimation in Non-Stationary Noise Environments for Missing Feature Based Robust Speech Recognition	2062
<i>Shirin Badieezadegan, Richard C. Rose</i>	
Robust Automatic Speech Recognition with Decoder Oriented Ideal Binary Mask Estimation	2066
<i>Lae-Hoon Kim, Kyung-Tae Kim, Mark Hasegawa-Johnson</i>	
A Robust Speech Recognition System Against the Ego Noise of a Robot	2070
<i>Gokhan Ince, Kazuhiro Nakadai, Tobias Rodemann, Hiroshi Tsujino, Jun-Ichi Imura</i>	
Empirical Mode Decomposition for Noise-Robust Automatic Speech Recognition	2074
<i>Kuo-Hao Wu, Chia-Ping Chen</i>	
An Effective Feature Compensation Scheme Tightly Matched with Speech Recognizer Employing SVM-Based GMM Generation	2078
<i>Wooil Kim, Jun-Won Suh, John H. L. Hansen</i>	
Artificial and Online Acquired Noise Dictionaries for Noise Robust ASR	2082
<i>Jort F. Gemmeke, Tuomas Virtanen</i>	
Voice Activity Detection Based on Conditional Random Fields Using Multiple Features	2086
<i>Akira Saito, Yoshihiko Nankaku, Akinobu Lee, Keiichi Tokuda</i>	
A Comparative Study of Noise Estimation Algorithms for VTS-Based Robust Speech Recognition	2090
<i>Yong Zhao, Bing-Hwang Juang</i>	
On Using Missing-Feature Theory with Cepstral Features --- Approximations to the Multivariate Integral	2094
<i>Frank Seide, Pei Zhao</i>	
Using a DBN to Integrate Sparse Classification and GMM-Based ASR	2098
<i>Yang Sun, Jort F. Gemmeke, Bert Cranen, L. Ten Bosch, Lou Boves</i>	

WED-SES2-P2: SPEAKER CHARACTERIZATION AND RECOGNITION IV

Transcript-Dependent Speaker Recognition Using Mixer 1 and 2	2102
<i>Fred S. Richardson, Joseph P. Campbell</i>	
On the Potential of Glottal Signatures for Speaker Recognition	2106
<i>Thomas Drugman, Thierry Dutoit</i>	
Acoustic Feature Diversity and Speaker Verification	2110
<i>R. Padmanabhan, Hema A. Murthy</i>	
A Discriminative Performance Metric for GMM-UBM Speaker Identification	2114
<i>Omid Dehzangi, Bin Ma, Eng Siong Chng, Haizhou Li</i>	
A Novel Speaker Binary Key Derived from Anchor Models	2118
<i>Xavier Anguera, Jean-Francois Bonastre</i>	
Variant Time-Frequency Cepstral Features for Speaker Recognition	2122
<i>Wei-Qiang Zhang, Yan Deng, Liang He, Jia Liu</i>	
Exploitation of Phase Information for Speaker Recognition	2126
<i>Ning Wang, P. C. Ching, Tan Lee</i>	
Effects of the Phonological Relevance in Speaker Verification	2130
<i>Yanhua Long, Lirong Dai, Bin Ma, Wu Guo</i>	
Topological Representation of Speech for Speaker Recognition	2134
<i>Gabriel H. Sierra, Jean-Francois Bonastre, Driss Matrouf, Jose R. Calvo</i>	
Assessment of Single-Channel Speech Enhancement Techniques for Speaker Identification Under Mismatched Conditions	2138
<i>Seyed Omid Sadjadi, John H. L. Hansen</i>	
Speaker Recognition Using the Resynthesized Speech via Spectrum Modeling	2142
<i>Xiang Zhang, Chuan Cao, Lin Yang, Hongbin Suo, Jianping Zhang, Yonghong Yan</i>	

WED-SES2-P3: VOICE CONVERSION AND SPEECH SYNTHESIS

Shape-Invariant Speech Transformation with the Phase Vocoder	2146
<i>Axel Robel</i>	
A Phonetic Alternative to Cross-Language Voice Conversion in a Text-Dependent Context: Evaluation of Speaker Identity	2150
<i>Kayoko Yanagisawa, Mark Huckvale</i>	
Evaluation of Speaker Mimic Technology for Personalizing SGD Voices	2154
<i>Esther Klabbbers, Alexander Kain, Jan P. H. Van Santen</i>	
Adaptive Voice-Quality Control Based on One-to-Many Eigenvoice Conversion	2158
<i>Kumi Ohta, Tomoki Toda, Yamato Ohtani, Hiroshi Saruwatari, Kiyohiro Shikano</i>	
Applying Voice Conversion to Concatenative Singing-Voice Synthesis	2162
<i>Fernando Villavicencio, Jordi Bonada</i>	
Improved Generation of Fundamental Frequency in HMM-Based Speech Synthesis Using Generation Process Model	2166
<i>Miaomiao Wang, Miaomiao Wen, Keikichi Hirose, Nobuaki Minematsu</i>	
A Hierarchical F0 Modeling Method for HMM-Based Speech Synthesis	2170
<i>Ming Lei, Yijian Wu, Frank K. Soong, Zhen-Hua Ling, Lirong Dai</i>	
Training a Parametric-Based LogF0 Model with the Minimum Generation Error Criterion	2174
<i>Javier Latorre, M. J. F. Gales, Heiga Zen</i>	

Improving Mandarin Segmental Duration Prediction with Automatically Extracted Syntax Features	2178
<i>Miaomiao Wen, Miaomiao Wang, Keikichi Hirose, Nobuaki Minematsu</i>	
An Intonation Model for TTS in Sepedi	2182
<i>Daniel R. Van Niekerk, Etienne Barnard</i>	
Synthesis of Fast Speech with Interpolation of Adapted HSMs and Its Evaluation by Blind and Sighted Listeners	2186
<i>Michael Pucher, Dietmar Schabus, Junichi Yamagishi</i>	
A Comparison of Pronunciation Modeling Approaches for HMM-TTS	2190
<i>Gabriel Webster, Sacha Krstulovic, Kate Knill</i>	
HMM-Based Text-to-Articulatory-Movement Prediction and Analysis of Critical Articulators	2194
<i>Zhen-Hua Ling, Korin Richmond, Junichi Yamagishi</i>	

WED-SES2-P4: DETECTION, CLASSIFICATION, AND SEGMENTATION

Audio-Based Sports Highlight Detection by Fourier Local Auto-Correlations	2198
<i>Jiaying Ye, Takumi Kobayashi, Tetsuya Higuchi</i>	
Automatic Excitement-Level Detection for Sports Highlights Generation	2202
<i>Hynek Boril, Abhijeet Sangwan, Taufiq Hasan, John H. L. Hansen</i>	
Detecting Novel Objects in Acoustic Scenes Through Classifier Incongruence	2206
<i>Jorg-Hendrik Bach, Jorn Anemuller</i>	
A Multidomain Approach for Automatic Home Environmental Sound Classification	2210
<i>Stavros Ntalampiras, Ilyas Potamitis, Nikos Fakotakis</i>	
Content-Based Advertisement Detection	2214
<i>Patrick Cardinal, Vishwa Gupta, Gilles Boulianne</i>	
Identification of Abnormal Audio Events Based on Probabilistic Novelty Detection	2218
<i>Stavros Ntalampiras, Ilyas Potamitis, Nikos Fakotakis</i>	
Lightly Supervised Recognition for Automatic Alignment of Large Coherent Speech Recordings	2222
<i>Norbert Braunschweiler, M. J. F. Gales, Sabine Buchholz</i>	
Incremental Diarization of Telephone Conversations	2226
<i>Oshry Ben-Harush, Itshak Lapidot, Hugo Guterman</i>	
Audio Analytics by Template Modeling and 1-Pass DP Based Decoding	2230
<i>Srikanth Cherla, V. Ramasubramanian</i>	
Perceptual Wavelet Decomposition for Speech Segmentation	2234
<i>Mariusz Ziolk, Jakub Galka, Bartosz Ziolk, Tomasz Drwiega</i>	
A Comparative Study of Constrained and Unconstrained Approaches for Segmentation of Speech Signal	2238
<i>Venkatesh Keri, Kishore Prahallad</i>	
Automatic Discriminative Measurement of Voice Onset Time	2242
<i>Morgan Sonderegger, Joseph Keshet</i>	
Selective Gammatone Filterbank Feature for Robust Sound Event Recognition	2246
<i>Yi Ren Leng, Huy Dat Tran, Norihide Kitaoka, Haizhou Li</i>	

WED-SES3-S1: SPECIAL SESSION: COMPRESSIVE SENSING FOR SPEECH AND LANGUAGE PROCESSING

Towards a Robust Face Recognition System Using Compressive Sensing	2250
<i>Allen Y. Yang, Zihan Zhou, Yi Ma, S. Shankar Sastry</i>	
Sparse Representation Features for Speech Recognition	2254
<i>Tara N. Sainath, Bhuvana Ramabhadran, David Nahamoo, Dimitri Kanevsky, Abhinav Sethy</i>	
Data Selection for Language Modeling Using Sparse Representations	2258
<i>Abhinav Sethy, Tara N. Sainath, Bhuvana Ramabhadran, Dimitri Kanevsky</i>	
Observation Uncertainty Measures for Sparse Imputation	2262
<i>Jort F. Gemmeke, Ulpu Remes, Kalle J. Palomaki</i>	
Sparse Representations for Text Categorization	2266
<i>Tara N. Sainath, Sameer R. Maskey, Dimitri Kanevsky, Bhuvana Ramabhadran, David Nahamoo, Julia Hirschberg</i>	
Sparse Auto-Associative Neural Networks: Theory and Application to Speech Recognition	2270
<i>G. S. V. S. Sivaram, Sriram Ganapathy, Hynek Hermansky</i>	

WED-SES3-01: ASR: LEXICAL AND PRONUNCIATION MODELING

FSM-Based Pronunciation Modeling Using Articulatory Phonological Code	2274
<i>Chi Hu, Xiaodan Zhuang, Mark Hasegawa-Johnson</i>	
Detailed Pronunciation Variant Modeling for Speech Transcription	2278
<i>Denis Jouviet, Dominique Fohr, Irina Illina</i>	
A Minimum Classification Error Approach to Pronunciation Variation Modeling of Non-Native Proper Names	2282
<i>Line Adde, Bert Reveil, Jean-Pierre Martens, Torbjorn Svendsen</i>	
Acoustics-Based Phonetic Transcription Method for Proper Nouns	2286
<i>Antoine Laurent, Sylvain Meignier, Teva Merlin, Paul Deleglise</i>	
Wiktionary as a Source for Automatic Pronunciation Extraction	2290
<i>Tim Schlippe, Sebastian Ochs, Tanja Schultz</i>	

Learning New Word Pronunciations from Spoken Examples	2294
<i>Ibrahim Badr, Ian McGraw, James Glass</i>	

WED-SES3-O2: SPEAKER RECOGNITION AND DIARIZATION

Phonetic Subspace Mixture Model for Speaker Diarization	2298
<i>I-Fan Chen, Shih-Sian Cheng, Hsin-Min Wang</i>	
Overlap Detection for Speaker Diarization by Fusing Spectral and Spatial Features	2302
<i>Martin Zelenak, Carlos Segura, Javier Hernandez</i>	
Floor Holder Detection and End of Speaker Turn Prediction in Meetings	2306
<i>Alfred Dielmann, Giulia Garau, Herve Bourlard</i>	
Confidence Measures for Speaker Segmentation and Their Relation to Speaker Verification	2310
<i>Carlos Vaquero, Alfonso Ortega, Jesus Villalba, Antonio Miguel, Eduardo Lleida</i>	
Decoupling Session Variability Modelling and Speaker Characterisation	2314
<i>Anthony Larcher, Christophe Levy, Driss Matrouf, Jean-Francois Bonastre</i>	
Incorporating MAP Estimation and Covariance Transform for SVM Based Speaker Recognition	2318
<i>Cheung-Chi Leung, Donglai Zhu, Kong Aik Lee, Bin Ma, Haizhou Li</i>	

WED-SES3-O3: SPEECH AND AUDIO CLASSIFICATION

Single-Speaker/Multi-Speaker Co-Channel Speech Classification	2322
<i>Stephane Rossignol, Olivier Pietquin</i>	
Discriminative Training for Hierarchical Clustering in Speaker Diarization	2326
<i>Oriol Vinyals, Gerald Friedland, Nelson Morgan</i>	
GMM-UBM Based Open-Set Online Speaker Diarization	2330
<i>Jurgen Geiger, Frank Wallhoff, Gerhard Rigoll</i>	
A Segment-Based Non-Parametric Approach for Monophone Recognition	2334
<i>Ladan Golipour, Douglas O'Shaughnessy</i>	
A Fast One-Pass-Training Feature Selection Technique for GMM-Based Acoustic Event Detection with Audio-Visual Data	2338
<i>Taras Butko, Climent Nadeu</i>	
Effects of Modelling Within- and Between-Frame Temporal Variations in Power Spectra on Non-Verbal Sound Recognition	2342
<i>Nobuhide Yamakawa, Tetsuro Kitahara, Toru Takahashi, Kazunori Komatani, Tetsuya Ogata, Hiroshi G. Okuno</i>	

VOLUME 4

WED-SES3-O4: EMOTION RECOGNITION

On the Importance of Glottal Flow Spectral Energy for the Recognition of Emotions in Speech	2346
<i>Ling He, Margaret Lech, Nicholas Allen</i>	
Real-Life Emotion-Related States Detection in Call Centers: A Cross-Corpora Study	2350
<i>Laurence Devillers, Christophe Vaudable, Clement Chastagnol</i>	
Multi-Class and Hierarchical SVMs for Emotion Recognition	2354
<i>Ali Hassan, Robert I. Dampier</i>	
Determining Optimal Features for Emotion Recognition from Speech by Applying an Evolutionary Algorithm	2358
<i>David Hubner, Bogdan Vlasenko, Tobias Grosser, Andreas Wendemuth</i>	
Context-Sensitive Multimodal Emotion Recognition from Speech and Facial Expression Using Bidirectional LSTM Modeling	2362
<i>Martin Wollmer, Angeliki Metallinou, Florian Eyben, Bjorn Schuller, Shrikanth S. Narayanan</i>	
Data-Dependent Evaluator Modeling and Its Application to Emotional Valence Classification from Speech	2366
<i>Kartik Audhkhasi, Shrikanth S. Narayanan</i>	

WED-SES3-P1: SPEECH CODING, MODELING, AND TRANSMISSION

Modelling Speech Line Spectral Frequencies with Dirichlet Mixture Models	2370
<i>Zhanyu Ma, Arne Leijon</i>	
PDF-Optimized LSF Vector Quantization Based on Beta Mixture Models	2374
<i>Zhanyu Ma, Arne Leijon</i>	
Non-Linear Predictive Vector Quantization of Feature Vectors for Distributed Speech Recognition	2378
<i>Jose Enrique Garcia, Alfonso Ortega, Antonio Miguel, Eduardo Lleida</i>	
Superwideband Extension of G.718 and G.729.1 Speech Coders	2382
<i>Lasse Laaksonen, Mikko Tammi, Vladimir Malenovsky, Tommy Vaillancourt, Mi Suk Lee, Tomofumi Yamanashi, Masahiro Oshikiri, Claude Lamblin, Balazs Kovesi, Lei Miao, Deming Zhang, Jon Gibbs, Holly Francois</i>	
A Multipulse FEC Scheme Based on Amplitude Estimation for CELP Coders Over Packet Networks	2386
<i>Jose L. Carmona, Angel M. Gomez, Antonio M. Peinado, Jose L. Perez-Cordoba, Jose A. Gonzalez</i>	

Voice Quality Evaluation of Recent Open Source Codecs	2390
<i>Anssi Ramo, Henri Toukoma</i>	
Efficient HMM-Based Estimation of Missing Features, with Applications to Packet Loss Concealment	2394
<i>Bengt J. Borgstrom, Per H. Borgstrom, Abeer Alwan</i>	
Speech Inventory Based Discriminative Training for Joint Speech Enhancement and Low-Rate Speech Coding	2398
<i>Xiaoqiang Xiao, Robert M. Nickel</i>	
Quality-Based Playout Buffering with FEC for Conversational VoIP	2402
<i>Qipeng Gong, Peter Kabal</i>	
Sub-Band Basis Spectrum Model for Pitch-Synchronous Log-Spectrum and Phase Based on Approximation of Sparse Coding	2406
<i>Masatsune Tamura, Takehiko Kagoshima, Masami Akamine</i>	
A Multimodal Density Function Estimation Approach to Formant Tracking	2410
<i>Harshavardhan S., Chandra Sekhar Seelamantula, T. V. Sreenivas</i>	
Estimation Studies of Vocal Tract Shape Trajectory Using a Variable Length and Lossy Kelly-Lochbaum Model	2414
<i>Heikki Rasilo, Unto K. Laine, Okko Rasanen</i>	

WED-SES3-P2: ASR: LANGUAGE MODELING AND SPEECH UNDERSTANDING II

Improving Back-Off Models with Bag of Words and Hollow-Grams	2418
<i>Benjamin Lecouteux, Raphael Rubino, Georges Linares</i>	
Study on Interaction Between Entropy Pruning and Kneser-Ney Smoothing	2422
<i>Ciprian Chelba, Thorsten Brants, Will Neveitt, Peng Xu</i>	
Dynamic Language Model Adaptation Using Keyword Category Classification	2426
<i>Hitoshi Yamamoto, Ken Hanazawa, Kiyokazu Miki, Koichi Shinoda</i>	
Integration of Cache-Based Model and Topic Dependent Class Model with Soft Clustering and Soft Voting	2430
<i>Welly Naptali, Masatoshi Tsuchiya, Seiichi Nakagawa</i>	
Conditional Models for Detecting Lambda-Functions in a Spoken Language Understanding System	2434
<i>Frederic Duvert, Renato De Mori</i>	
Novel Weighting Scheme for Unsupervised Language Model Adaptation Using Latent Dirichlet Allocation	2438
<i>Akmal Haidar, Douglas O'Shaughnessy</i>	
Automatic Speech Recognition System Channel Modeling	2442
<i>Qun Feng Tan, Kartik Audhkhasi, Panayiotis G. Georgiou, Emil Ettelaie, Shrikanth S. Narayanan</i>	
Round-Robin Discrimination Model for Reranking ASR Hypotheses	2446
<i>Takanobu Oba, Takaaki Hori, Atsushi Nakamura</i>	
On-the-Fly Lattice Rescoring for Real-Time Automatic Speech Recognition	2450
<i>Hasim Sak, Murat Saraclar, Tunga Gungor</i>	

WED-SES3-P3: SPEECH PERCEPTION III: PROCESSING AND INTELLIGIBILITY

A Feature Extraction Method for Automatic Speech Recognition Based on the Cochlear Nucleus	2454
<i>Serajul Haque, Roberto Togneri</i>	
A Phoneme Recognition Framework Based on Auditory Spectro-Temporal Receptive Fields	2458
<i>Samuel Thomas, Kailash Patil, Sriram Ganapathy, Nima Mesgarani, Hynek Hermansky</i>	
Perceptual Compensation for Effects of Reverberation in Speech Identification: A Computer Model Based on Auditory Efferent Processing	2462
<i>Amy V. Beeston, Guy J. Brown</i>	
Predicting Human Perception and ASR Classification of Word-Final [t] by Its Acoustic Sub-Segmental Properties	2466
<i>Barbara Schuppler, Mirjam Ernestus, Wim Van Dommelen, Jacques Koreman</i>	
Multichannel Source Separation Based on Source Location Cue with Log-Spectral Shaping by Hidden Markov Source Model	2470
<i>Tomohiro Nakatani, Shoko Araki, Takuya Yoshioka, Masakiyo Fujimoto</i>	
A Speech-in-Noise Test Based on Spoken Digits: Comparison of Normal and Impaired Listeners Using a Computer Model	2474
<i>Matthew Robertson, Guy J. Brown, Wendy Lecluyse, Manasa Panda, Christine M. Tan</i>	
Evaluation of Bone-Conducted Ultrasonic Hearing-Aid Regarding Transmission of Paralinguistic Information: A Comparison with Cochlear Implant Simulator	2478
<i>Takayuki Kagomiya, Seiji Nakagawa</i>	
Challenging the Speech Intelligibility Index: Macroscopic vs. Microscopic Prediction of Sentence Recognition in Normal and Hearing-Impaired Listeners	2482
<i>Tim Jurgens, Stefan Fredelake, Ralf M. Meyer, Birger Kollmeier, Thomas Brand</i>	
Does Sentence Complexity Interfere with Intelligibility in Noise? Evaluation of the Oldenburg Linguistically and Audiologically Controlled Sentence Test (OLACS)	2486
<i>Verena N. Uslar, Thomas Brand, Mirko Hanke, Rebecca Carroll, Esther Ruigendijk, Cornelia Hamann, Birger Kollmeier</i>	
Intelligibility Predictions for Speech Against Fluctuating Masker	2490
<i>Juan-Pablo Ramirez, Hamed Ketabdar, Alexander Raake</i>	
An Effect of Formant Amplitude in Vowel Perception	2494
<i>Masashi Ito, Keiji Ohara, Akinori Ito, Masafumi Yano</i>	

WED-SES3-P4: SPOKEN LANGUAGE UNDERSTANDING AND SPOKEN LANGUAGE TRANSLATION I

Functional Imaging of Brain Regions Sensitive to Communication Sounds in Primates	2498
<i>Christopher I. Petkov, Benjamin Wilson</i>	
Strategies for Statistical Spoken Language Understanding with Small Amount of Data --- An Empirical Study	2502
<i>Ye-Yi Wang</i>	
Investigating Multiple Approaches for SLU Portability to a New Language	2506
<i>Bassam Jabaian, Laurent Besacier, Fabrice Lefevre</i>	
Learning Naturally Spoken Commands for a Robot	2510
<i>Anja Austermann, Seiji Yamada, Kotaro Funakoshi, Mikio Nakano</i>	
A Semi-Supervised Cluster-and-Label Approach for Utterance Classification	2514
<i>Amparo Albalade, Aparna Suchindranath, David Suendermann, Wolfgang Minker</i>	
Classifying Dialog Acts in Human-Human and Human-Machine Spoken Conversations	2518
<i>Silvia Quarteroni, Giuseppe Riccardi</i>	
Exploring Speaker Characteristics for Meeting Summarization	2522
<i>Fei Liu, Yang Liu</i>	
Semi-Supervised Extractive Speech Summarization via Co-Training Algorithm	2526
<i>Shasha Xie, Hui Lin, Yang Liu</i>	
Extractive Summarization Using a Latent Variable Model	2530
<i>Asli Celikyilmaz, Dilek Hakkani-Tur</i>	
Hierarchical Classification for Speech-to-Speech Translation	2534
<i>Emil Ettelaie, Panayiotis G. Georgiou, Shrikanth S. Narayanan</i>	
Rapid Development of Speech Translation Using Consecutive Interpretation	2538
<i>Matthias Paulik, Alex Waibel</i>	

THU-SES1-S1: SPECIAL SESSION: SOCIAL SIGNALS IN SPEECH

Combining Many Alignments for Speech to Speech Translation	2542
<i>Sameer R. Maskey, Steven J. Rennie, Bowen Zhou</i>	
Detecting Politeness and Efficiency in a Cooperative Social Interaction	2546
<i>Paul M. Brunet, Marcela Charfuelan, Roderick Cowie, Marc Schroder, Hastings Donnan, Ellen Douglas-Cowie</i>	
Comparing Measures of Synchrony and Alignment in Dialogue Speech Timing with Respect to Turn-Taking Activity	2550
<i>Nick Campbell, Stefan Scherer</i>	
Resources for Turn Competition in Overlap in Multi-Party Conversations: Speech Rate, Pausing and Duration	2554
<i>Emina Kurtic, Guy J. Brown, Bill Wells</i>	
Disambiguating the Functions of Conversational Sounds with Prosody: The Case of 'Yeah'	2558
<i>Khiet P. Truong, Dirk Heylen</i>	
Prosody and Voice Quality of Vocal Social Signals: The Case of Dominance in Scenario Meetings	2562
<i>Marcela Charfuelan, Marc Schroder, Ingmar Steiner</i>	

THU-SES1-O2: PHYSIOLOGY AND PATHOLOGY OF SPOKEN LANGUAGE

Online SLU Model Adaptation with a Partial Oracle	2566
<i>Pierre Gotab, Geraldine Damnati, Frederic Bechet, Lionel Delphin-Poulat</i>	
The Prosody of Swedish Conversational Grunts	2570
<i>D. Neiberg, J. Gustafson</i>	
Reliable Tracking Based on Speech Sample Saliency of Vocal Cycle Length Perturbations	2574
<i>C. Mertens, Francis Grenéz, Lise Crevier-Buchman, Jean Schoentgen</i>	
Longitudinal Changes of Selected Voice Source Parameters	2578
<i>Hideki Kasuya, Hajime Yoshida, Satoshi Ebihara, Hiroki Mori</i>	
Automatic Perceptual Categorization of Disordered Connected Speech	2582
<i>A. Alpan, Jean Schoentgen, Y. Maryn, Francis Grenéz</i>	
Kinematic Analysis of Tongue Movement Control in Spastic Dysarthria	2586
<i>Heejin Kim, Panying Rong, Torrey M. Loucks, Mark Hasegawa-Johnson</i>	

THU-SES1-O3: PITCH AND GLOTTAL-WAVEFORM ESTIMATION AND MODELING II

Pre- and Short-Term Posttreatment Vocal Functioning in Patients with Advanced Head and Neck Cancer Treated with Concomitant Chemoradiotherapy	2590
<i>Irene Jacobi, Lisette Van Der Molen, Maya Van Rossum, Frans Hilgers</i>	
Acoustic Analysis of Intonation in Parkinson's Disease	2594
<i>Joan K. Y. Ma, Rudiger Hoffmann</i>	
SAFE: A Statistical Algorithm for F0 Estimation for Both Clean and Noisy Speech	2598
<i>Wei Chu, Abeer Alwan</i>	
Robust and Efficient Pitch Estimation Using an Iterative ARMA Technique	2602
<i>Jung Ook Hong, Patrick J. Wolfe</i>	

Statistical Modeling of F0 Dynamics in Singing Voices Based on Gaussian Processes with Multiple Oscillation Bases	2606
<i>Yasunori Ohishi, Hirokazu Kameoka, Daichi Mochihashi, Hidehisa Nagano, Kunio Kashino</i>	
Applying Geometric Source Separation for Improved Pitch Extraction in Human-Robot Interaction	2610
<i>Martin Heckmann, Claudius Glaser, Frank Joublin, Kazuhiro Nakadai</i>	

THU-SES1-O4: ASR: FEATURE EXTRACTION II

A Spectral LF Model Based Approach to Voice Source Parameterisation	2614
<i>John Kane, Mark Kane, Christer Gobl</i>	
Glottal-Based Analysis of the Lombard Effect	2618
<i>Thomas Drugman, Thierry Dutoit</i>	
Hidden Logistic Linear Regression for Support Vector Machine Based Phone Verification	2622
<i>Bo Li, Khe Chai Sim</i>	
Jointly Optimized Discriminative Features for Speech Recognition	2626
<i>Tim Ng, Bing Zhang, Long Nguyen</i>	
Invariant Integration Features Combined with Speaker-Adaptation Methods	2630
<i>Florian Muller, Alfred Mertins</i>	
Multi Resolution Discriminative Models for Subvocalic Speech Recognition	2634
<i>Mark Raugas, Vivek Kumar Rangarajan Sridhar, Rohit Prasad, Prem Natarajan</i>	

THU-SES1-P1: SPEAKER DIARIZATION

A Comparative Large Scale Study of MLP Features for Mandarin ASR	2638
<i>Fabio Valente, Mathew Magimai Doss, Christian Plahl, Suman V. Ravuri, Wen Wang</i>	
Recognizing Cochlear Implant-Like Spectrally Reduced Speech with HMM-Based ASR: Experiments with MFCCs and PLP Coefficients	2642
<i>Cong-Thanh Do, Dominique Pastor, Gael Le Lan, Andre Goalic</i>	
A Hybrid Approach to Online Speaker Diarization	2646
<i>Carlos Vaquero, Oriol Vinyals, Gerald Friedland</i>	
System Output Combination for Improved Speaker Diarization	2650
<i>Simon Bozonnet, Nicholas Evans, Xavier Anguera, Oriol Vinyals, Gerald Friedland, Corinne Fredouille</i>	
An Integrated Top-Down/Bottom-Up Approach to Speaker Diarization	2654
<i>Simon Bozonnet, Nicholas Evans, Corinne Fredouille, Dong Wang, Raphael Troncy</i>	
Advances in Fast Multistream Diarization Based on the Information Bottleneck Framework	2658
<i>Deepu Vijayaseenan, Fabio Valente, Herve Bourlard</i>	
Audio-Visual Synchronisation for Speaker Diarisation	2662
<i>Giulia Garau, Alfred Dielmann, Herve Bourlard</i>	
An Improved Cluster Model Selection Method for Agglomerative Hierarchical Speaker Clustering Using Incremental Gaussian Mixture Models	2666
<i>Kyu J. Han, Shrikanth S. Narayanan</i>	
Dialog Prediction for a General Model of Turn-Taking	2670
<i>Nigel G. Ward, Olac Fuentes, Alejandro Vega</i>	

THU-SES1-P2: MULTI-MODAL ASR, INCLUDING AUDIO-VISUAL ASR

Speaker Tracking in an Unsupervised Speech Controlled System	2674
<i>Tobias Herbig, Franz Gerl, Wolfgang Minker</i>	
MultiBIC: An Improved Speaker Segmentation Technique for TV Shows	2678
<i>Paula Lopez-Otero, Laura Docio-Fernandez, Carmen Garcia-Mateo</i>	
Automatic Speech Recognition for Assistive Writing in Speech Supplemented Word Prediction	2682
<i>John-Paul Hosom, Tom Jakobs, Allen Baker, Susan Fager</i>	
Viseme-Dependent Weight Optimization for CHMM-Based Audio-Visual Speech Recognition	2686
<i>Alexey Karpov, Andrey Ronzhin, Konstantin Markov, Milos Zelezny</i>	
Audio-Visual Anticipatory Coarticulation Modeling by Human and Machine	2690
<i>Louis H. Terry, Karen Livescu, Janet B. Pierrehumbert, Aggelos K. Katsaggelos</i>	
Impact of Lack of Acoustic Feedback in EMG-Based Silent Speech Recognition	2694
<i>Mathias Janke, Michael Wand, Tanja Schultz</i>	
Using Prosody to Improve Mandarin Automatic Speech Recognition	2698
<i>Chong-Jia Ni, Wenju Liu, Bo Xu</i>	
A Robust Audio-Visual Speech Recognition Using Audio-Visual Voice Activity Detection	2702
<i>Satoshi Tamura, Masato Ishikawa, Takashi Hashiba, Shin'Ichi Takeuchi, Satoru Hayamizu</i>	
Efficient Manycore CHMM Speech Recognition for Audiovisual and Multistream Data	2706
<i>Dorothea Kolossa, Jike Chong, Steffen Zeiler, Kurt Keutzer</i>	

THU-SES1-P3: SPEAKER AND LANGUAGE RECOGNITION

Two-Layered Audio-Visual Integration in Voice Activity Detection and Automatic Speech Recognition for Robots	2710
<i>Takami Yoshida, Kazuhiro Nakadai</i>	
Non-Audible Murmur Recognition Based on Fusion of Audio and Visual Streams	2714
<i>Panikos Heracleous, Norihiro Hagita</i>	
Improved N-Gram Phonotactic Models for Language Recognition	2718
<i>Mohamed Faouzi Benzeghiba, Jean-Luc Gauvain, Lori Lamel</i>	
A Study of Term Weighting in Phonotactic Approach to Spoken Language Recognition	2722
<i>Sirinoot Boonsuk, Donglai Zhu, Bin Ma, Atiwong Suchato, Proadpran Punyabukkana, Nattanun Thatphithakkul, Chai Wutiwivatchai</i>	
Exploiting Context-Dependency and Acoustic Resolution of Universal Speech Attribute Models in Spoken Language Recognition	2726
<i>Sabato Marco Siniscalchi, Jeremy Reed, Torbjorn Svendsen, Chin-Hui Lee</i>	
Hierarchical Multilayer Perceptron Based Language Identification	2730
<i>David Imseng, Mathew Magimai Doss, Herve Bourlard</i>	
The NIST 2010 Speaker Recognition Evaluation	2734
<i>Alvin F. Martin, Craig S. Greenberg</i>	
Bayesian Speaker Recognition Using Gaussian Mixture Model and Laplace Approximation	2738
<i>Shih-Sian Cheng, I-Fan Chen, Hsin-Min Wang</i>	
What Else is New Than the Hamming Window? Robust MFCCs for Speaker Recognition via Multitapering	2742
<i>Tomi Kinnunen, Rahim Saeidi, Johan Sandberg, Maria Hansson-Sandsten</i>	
Fast Computation of Speaker Characterization Vector Using MLLR and Sufficient Statistics in Anchor Model Framework	2746
<i>A. K. Sarkar, S. Umesh</i>	
Graph-Embedding for Speaker Recognition	2750
<i>Zahi N. Karam, W. M. Campbell</i>	
A Hybrid Modeling Strategy for GMM-SVM Speaker Recognition with Adaptive Relevance Factor	2754
<i>Chang Huai You, Haizhou Li, Kong Aik Lee</i>	

THU-SES1-P4: SOURCE LOCALIZATION AND SEPARATION

Robust Mixture Modeling Using T-Distribution: Application to Speaker ID	2758
<i>Harshavardhan S., T. V. Sreenivas</i>	
A Variable Frame Length and Rate Algorithm Based on the Spectral Kurtosis Measure for Speaker Verification	2762
<i>Chi-Sang Jung, Kyu J. Han, Hyunson Seo, Shrikanth S. Narayanan, Hong-Goo Kang</i>	
Near Field Sound Source Localization Based on Cross-Power Spectrum Phase Analysis with Multiple Microphones	2766
<i>Kohei Hayashida, Masanori Morise, Takanobu Nishiura</i>	
A Maximum a Posteriori Sound Source Localization in Reverberant and Noisy Conditions	2770
<i>Jinho Choi, Chang D. Yoo</i>	
A DOA Estimation Algorithm Based on Equalization-Cancellation Theory	2774
<i>Duc Thanh Chau, Junfeng Li, Masato Akagi</i>	
Concurrent Speaker Localization Using Multi-Band Position-Pitch (M-PoPi) Algorithm with Spectro-Temporal Pre-Processing	2778
<i>Tania Habib, Harald Romsdorfer</i>	
On Using Gaussian Mixture Model for Double-Talk Detection in Acoustic Echo Suppression	2782
<i>Ji-Hyun Song, Kyu-Ho Lee, Yun-Sik Park, Sang-Ick Kang, Joon-Hyuk Chang</i>	
Catalog-Based Single-Channel Speech-Music Separation	2786
<i>Cemil Demir, A. Taylan Cemgil, Murat Saraclar</i>	
Unvoiced Speech Segregation Based on CASA and Spectral Subtraction	2790
<i>Ke Hu, Deliang Wang</i>	

THU-SES2-S1: SPECIAL SESSION: INTERSPEECH 2010 PARALINGUISTIC CHALLENGE

Unsupervised Sequential Organization for Cochannel Speech Separation	2794
<i>Ke Hu, Deliang Wang</i>	
The INTERSPEECH 2010 Paralinguistic Challenge	2798
<i>Bjorn Schuller, Stefan Steidl, Anton Batliner, Felix Burkhardt, Laurence Devillers, Christian Muller, Shrikanth S. Narayanan</i>	
Age and Gender Classification from Speech Using Decision Level Fusion and Ensemble Based Techniques	2802
<i>Florian Lingenfelser, Johannes Wagner, Thuriid Vogt, Jonghwa Kim, Elisabeth Andre</i>	
Level of Interest Sensing in Spoken Dialog Using Multi-Level Fusion of Acoustic and Lexical Evidence	2806
<i>Je Hun Jeon, Rui Xia, Yang Liu</i>	
Fuzzy Support Vector Machines for Age and Gender Classification	2810
<i>Phuoc Nguyen, Trung Le, Dat Tran, Xu Huang, Dharmendra Sharma</i>	
Gender and Affect Recognition Based on GMM and GMM-UBM Modeling with Relevance MAP Estimation	2814
<i>Rok Gajsek, Janez Zibert, Tadej Justin, Vitomir Struc, Bostjan Vesnicer, France Mihelic</i>	
Age Recognition Based on Speech Signals Using Weights Supervector	2818
<i>Royl Porat, Dan Lange, Yaniv Zigel</i>	

Age and Gender Classification Using Fusion of Acoustic and Prosodic Features	2822
<i>Hugo Meinedo, Isabel Trancoso</i>	
Brno University of Technology System for Interspeech 2010 Paralinguistic Challenge	2826
<i>Marcel Kockmann, Lukas Burget, Jan Cernocky</i>	
Combining Five Acoustic Level Modeling Methods for Automatic Speaker Age and Gender Recognition	2830
<i>Ming Li, Chi-Sang Jung, Kyu J. Han</i>	
Age and Gender Recognition Based on Multiple Systems --- Early vs. Late Fusion	2834
<i>Tobias Bocklet, Georg Stemmer, Viktor Zeissler, Elmar Noth</i>	

THU-SES2-O1: NEW PARADIGMS IN ASR II

Automatic Speaker Age and Gender Recognition in the Car for Tailoring Dialog and Mobile Services	2838
<i>Michael Feld, Felix Burkhardt, Christian Muller</i>	
Improved Topic Classification and Keyword Discovery Using an HMM-Based Speech Recognizer Trained Without Supervision	2842
<i>Man-Hung Siu, Herbert Gish, Arthur Chan, William Belfield</i>	
An Analysis of Sparseness and Regularization in Exemplar-Based Methods for Speech Classification	2846
<i>Dimitri Kanevsky, Tara N. Sainath, Bhuvana Ramabhadran, David Nahamoo</i>	
Investigation of Full-Sequence Training of Deep Belief Networks for Speech Recognition	2850
<i>Abdel-Rahman Mohamed, Dong Yu, L. Deng</i>	
Mandarin Tone Recognition Using Affine-Invariant Prosodic Features and Tone Posteriorgram	2854
<i>Yow-Bang Wang, Lin-Shan Lee</i>	
Continuous Speech Recognition with a TF-IDF Acoustic Model	2858
<i>Geoffrey Zweig, Patrick Nguyen, Jasha Droppo, Alex Acero</i>	

THU-SES2-O2: SPOKEN LANGUAGE UNDERSTANDING AND SPOKEN LANGUAGE TRANSLATION II

SCARF: A Segmental Conditional Random Field Toolkit for Speech Recognition	2862
<i>Geoffrey Zweig, Patrick Nguyen</i>	
Role of Language Models in Spoken Fluency Evaluation	2866
<i>Om D. Deshmukh, Harish Doddala, Ashish Verma, Karthik Visweswariah</i>	
Social Role Discovery from Spoken Language Using Dynamic Bayesian Networks	2870
<i>Sibel Yaman, Dilek Hakkani-Tur, Gokhan Tur</i>	
Domain Adaptation and Compensation for Emotion Detection	2874
<i>Michelle Hewlett Sanchez, Gokhan Tur, Luciana Ferrer, Dilek Hakkani-Tur</i>	
Phrase Alignment Confidence for Statistical Machine Translation	2878
<i>Sankaranarayanan Ananthkrishnan, Rohit Prasad, Prem Natarajan</i>	
Named-Entity Projection and Data-Driven Morphological Decomposition for Field Maintainable Speech-to-Speech Translation Systems	2882
<i>Ian R. Lane, Alex Waibel</i>	

THU-SES2-O3: SIGNAL PROCESSING FOR MUSIC AND SONG

Acoustic Correlates of Voice Quality Improvement by Voice Training	2886
<i>Kiyoaki Aikawa, Junko Uenuma, Tomoko Akitake</i>	
Phonetic Segmentation of Singing Voice Using MIDI and Parallel Speech	2890
<i>Minghui Dong, Paul Chan, Ling Cen, Haizhou Li, Jason Teo, Ping Jen Kua</i>	
A Singing Style Modeling System for Singing Voice Synthesizers	2894
<i>Keijiro Saino, Makoto Tachibana, Hideki Kenmochi</i>	
A Fast Query by Humming System Based on Notes	2898
<i>Jingzhou Yang, Jia Liu, Wei-Qiang Zhang</i>	
Melody Pitch Estimation Based on Range Estimation and Candidate Extraction Using Harmonic Structure Model	2902
<i>Seokhwan Jo, Sihyun Joo, Chang D. Yoo</i>	
Modified Spatial Audio Object Coding Scheme with Harmonic Extraction and Elimination Structure for Interactive Audio Service	2906
<i>Jihoon Park, Kwangki Kim, Jeongil Seo, Minsoo Hahn</i>	

THU-SES2-O4: MODELING FIRST LANGUAGE ACQUISITION

Modelling the Effect of Speaker Familiarity and Noise on Infant Word Recognition	2910
<i>Christina Bergmann, Michele Gubian, Lou Boves</i>	
Unsupervised Learning of Vowels from Continuous Speech Based on Self-Organized Phoneme Acquisition Model	2914
<i>Kouki Miyazawa, Hideaki Kikuchi, Reiko Mazuka</i>	
Learning Speaker Normalization Using Semisupervised Manifold Alignment	2918
<i>Andrew R. Plummer, Mary E. Beckman, Mikhail Belkin, Eric Fosler-Lussier, Benjamin Munson</i>	

Fully Unsupervised Word Learning from Continuous Speech Using Transitional Probabilities of Atomic Acoustic Events	2922
<i>Okko Johannes Rasanen</i>	
Language Acquisition and Cross-Modal Associations: Computational Simulation of the Result of Infant Studies	2926
<i>L. Ten Bosch, Lou Boves</i>	
Active Word Learning Under Uncertain Input Conditions	2930
<i>Maarten Versteegh, L. Ten Bosch, Lou Boves</i>	

THU-SES2-P1: ASR: ACOUSTIC MODELS III

Parallel Training of Neural Networks for Speech Recognition	2934
<i>Karel Vesely, Lukas Burget, Frantisek Grezl</i>	
The Use of Sense in Unsupervised Training of Acoustic Models for ASR Systems	2938
<i>Rita Singh, Benjamin Lambert, Bhiksha Raj</i>	
Boosted Mixture Learning of Gaussian Mixture HMMs for Speech Recognition	2942
<i>Jun Du, Yu Hu, Hui Jiang</i>	
On the Exploitation of Hidden Markov Models and Linear Dynamic Models in a Hybrid Decoder Architecture for Continuous Speech Recognition	2946
<i>Volker Leutnant, Reinhold Haeb-Umbach</i>	
Context Dependent Modelling Approaches for Hybrid Speech Recognizers	2950
<i>Alberto Abad, Thomas Pellegrini, Isabel Trancoso, Joao Neto</i>	
A Regularized Discriminative Training Method of Acoustic Models Derived by Minimum Relative Entropy Discrimination	2954
<i>Yotaro Kubo, Shinji Watanabe, Atsushi Nakamura, Tetsunori Kobayashi</i>	
Decision Tree State Clustering with Word and Syllable Features	2958
<i>Hank Liao, Chris Alberti, Michiel Bacchiani, Olivier Siohan</i>	
A Duration Modeling Technique with Incremental Speech Rate Normalization	2962
<i>Hiroshi Fujimura, Takashi Masuko, Mitsuyoshi Tachimori</i>	
Long Short-Term Memory Networks for Noise Robust Speech Recognition	2966
<i>Martin Wollmer, Yang Sun, Florian Eyben, Bjorn Schuller</i>	
One-Model Speech Recognition and Synthesis Based on Articulatory Movement HMMs	2970
<i>Tsuneo Nitta, Takayuki Onoda, Masashi Kimura, Yurie Iribe, Kouichi Katsurada</i>	
Acoustic Modeling with Bootstrap and Restructuring for Low-Resourced Languages	2974
<i>Xiaodong Cui, Jian Xue, Pierre L. Dognin, Upendra V. Chaudhari, Bowen Zhou</i>	
Lecture Speech Recognition by Combining Word Graphs of Various Acoustic Models	2978
<i>Tetsuo Kosaka, Keisuke Goto, Takashi Ito, Masaharu Kato</i>	
Semi-Parametric Trajectory Modelling Using Temporally Varying Feature Mapping for Speech Recognition	2982
<i>Khe Chai Sim, Shilin Liu</i>	
Deep-Structured Hidden Conditional Random Fields for Phonetic Recognition	2986
<i>Dong Yu, L. Deng</i>	
Semi-Supervised Learning for Improved Expression of Uncertainty in Discriminative Classifiers	2990
<i>Jonathan Malkin, Jeff Bilmes</i>	
Modeling Posterior Probabilities Using the Linear Exponential Family	2994
<i>Peder Olsen, Vaibhava Goel, Charles Micchelli, John R. Hershey</i>	

THU-SES2-P2: SPOKEN DIALOGUE SYSTEMS II

New Technique to Enhance the Performance of Spoken Dialogue Systems Based on Dialogue States-Dependent Language Models and Grammatical Rules	2998
<i>Ramon Lopez-Cozar, David Griol</i>	
A Stochastic Finite-State Transducer Approach to Spoken Dialog Management	3002
<i>Lluis-F. Hurtado, Joaquin Planells, Encarna Segarra, Emilio Sanchis, David Griol</i>	
Enhanced Monitoring Tools and Online Dialogue Optimisation Merged into a New Spoken Dialogue System Design Experience	3006
<i>Romain Laroche, Philippe Bretier, Ghislain Putois</i>	
Optimising a Handcrafted Dialogue System Design	3010
<i>Romain Laroche, Ghislain Putois, Philippe Bretier</i>	
Utterance Selection for Speech Acts in a Cognitive Tourguide Scenario	3014
<i>Felix Putze, Tanja Schultz</i>	
Lexical Entrainment of Real Users in the Let's Go Spoken Dialog System	3018
<i>Gabriel Parent, Maxine Eskenazi</i>	
Combining User Intention and Error Modeling for Statistical Dialog Simulators	3022
<i>Silvia Quarteroni, Meritxell Gonzalez, Giuseppe Riccardi, Sebastian Varges</i>	
Parallel Processing of Interruptions and Feedback in Companions Affective Dialogue System	3026
<i>Jaakko Hakulinen, Markku Turunen, Raul Santos De La Camara, Nigel Crook</i>	
Dynamic Language Modeling Using Bayesian Networks for Spoken Dialog Systems	3030
<i>Antoine Raux, Neville Mehta, Deepak Ramachandran, Rakesh Gupta</i>	
Automatic Detection of Task-Incompleted Dialog for Spoken Dialog System Based on Dialog Act N-Gram	3034
<i>Sunao Hara, Norihide Kitaoka, Kazuya Takeda</i>	

Dialogue Act Detection in Error-Prone Spoken Dialogue Systems Using Partial Sentence Tree and Latent Dialogue Act Matrix	3038
<i>Wei-Bin Liang, Chung-Hsien Wu, Yu-Cheng Hsiao</i>	
Detection of Hot Spots in Poster Conversations Based on Reactive Tokens of Audience	3042
<i>Tatsuya Kawahara, Kouhei Sumi, Zhi-Qiang Chang, Katsuya Takanashi</i>	
Psychological Evaluation of a Group Communication Activation Robot in a Party Game	3046
<i>Yoichi Matsuyama, Shinya Fujie, Hikaru Taniyama, Tetsunori Kobayashi</i>	
Analyzing User Utterances in Barge-in-Able Spoken Dialogue System for Improving Identification Accuracy	3050
<i>Kyoko Matsuyama, Kazunori Komatani, Ryu Takeda, Toru Takahashi, Tetsuya Ogata, Hiroshi G. Okuno</i>	
Pitch Similarity in the Vicinity of Backchannels	3054
<i>Mattias Heldner, Jens Edlund, Julia Hirschberg</i>	
A Rule-Based Backchannel Prediction Model Using Pitch and Pause Information	3058
<i>Khiet P. Truong, Ronald Poppe, Dirk Heylen</i>	

THU-SES2-P3: DISCOURSE AND DIALOGUE

Combining Text Categorization and Dialog Modeling for Speaker Role Identification on Call Center Conversations	3062
<i>Remi Lavalley, Chloe Clavel, Patrice Bellot, Marc El-Beze</i>	
Topic-Dependent N-Gram Models Based on Optimization of Context Lengths in LDA	3066
<i>Akira Nakamura, Satoru Hayamizu</i>	
Expectations for Discourse Genre Identification: A Prosodic Study	3070
<i>Nicolas Obin, Volker Dellwo, Anne Lacheret, Xavier Rodet</i>	
Dialogue Act Tagging and Segmentation with a Single Perceptron	3074
<i>Ramon Granell, Stephen Pulman, Carlos-D. Martinez-Hinarejos, Jose Miguel Benedi</i>	
Improving the Readability of Class Lecture ASR Results Using a Confusion Network	3078
<i>Yasuhisa Fujii, Kazumasa Yamamoto, Seiichi Nakagawa</i>	

THU-SES2-P4: VOICE ACTIVITY AND TURN DETECTION

Toward Detecting Voice Activity Employing Soft Decision in Second-Order Conditional MAP	3082
<i>Sang-Kyun Kim, Jae-Hun Choi, Sang-Ick Kang, Ji-Hyun Song, Joon-Hyuk Chang</i>	
Voice Activity Detection in a Regularized Reproducing Kernel Hilbert Space	3086
<i>Xugang Lu, Masashi Unoki, Ryosuke Isotani, Hisashi Kawai, Satoshi Nakamura</i>	
A New VAD Framework Using Statistical Model and Human Knowledge Based Empirical Rule	3090
<i>Ji Wu, Xiao-Lei Zhang, Wei Li</i>	
Adaptive High Accuracy Approaches to Speech Activity Detection in Noisy and Hostile Audio Environments	3094
<i>Mark Huggins, Brett Smolenski, Aaron Lawson</i>	
Robust Voice Activity Detection in Stereo Recording with Crosstalk	3098
<i>Prasanta Kumar Ghosh, Andreas Tsiartas, Panayiotis G. Georgiou, Shrikanth S. Narayanan</i>	
Voice Activity Detection Using Frame-Wise Model Re-Estimation Method Based on Gaussian Pruning with Weight Normalization	3102
<i>Masakiyo Fujimoto, Shinji Watanabe, Tomohiro Nakatani</i>	
Spectral Entropy-Based Voice Activity Detector for Videoconferencing Systems	3106
<i>Bowon Lee, Debargha Mukherjee</i>	
The QUT-NOISE-TIMIT Corpus for the Evaluation of Voice Activity Detection Algorithms	3110
<i>David Dean, Sridha Sridharan, Robert Vogt, Michael Mason</i>	
A Bayesian Approach to Voice Activity Detection Using Multiple Statistical Models and Discriminative Training	3114
<i>Tao Yu, John H. L. Hansen</i>	
Noise Robust Voice Activity Detection Using Features Extracted from the Time-Domain Autocorrelation Function	3118
<i>Houman Ghaemmaghami, Brendan Baker, Robert Vogt, Sridha Sridharan</i>	
VAD-Measure-Embedded Decoder with Online Model Adaptation	3122
<i>Tasuku Oonishi, Koji Iwano, Sadaoki Furui</i>	
Robust Statistical Voice Activity Detection Using a Likelihood Ratio Sign Test	3126
<i>Shiwen Deng, Jiqing Han</i>	
Automatic Turn Segmentation in Spoken Conversations	3130
<i>Alexei V. Ivanov, Giuseppe Riccardi</i>	
Turn Taking-Based Conversation Detection by Using DOA Estimation	3134
<i>Yohei Kawaguchi, Masahito Togami, Yasunari Obuchi</i>	
Author Index	