LARGE VOCABULARY SPEECH RECOGNITION

Chair: S. Furui, NTT Human Interface Labs. (Japan)

New Uses for the N-Best Sentence Hypotheses
Within the Byblos Speech Recognition System
R. Schwartz, S. Austin, F. Kubala, J. Makhoul, L. Nguyen, P. Placeway, BBN Systems and Technologies (USA);
G. Zavaliagkos, Northeastern Univ. (USA)

Exploiting Correlations Among Competing Models with Applications to Large Vocabulary Speech Recognition
R. Rosenfeld, X. Huang, M. Furst, Carnegie Mellon Univ. (USA)

Improvements in Beam Search for 10,000-Word Continuous Speech Recognition
H. Ney, R. Haeb-Umbach, B. Tran, M. Oerder, Philips Res. Labs. (Germany)

Linear Discriminant Analysis for Improved Large Vocabulary Continuous Speech Recognition
R. Haeb-Umbach, H. Ney, Philips Res. Labs. (Germany)

A Fast Match for Continuous Speech Recognition Using Allophonic Models
L. Bahl, P. DeSouza, P. Gopalakrishnan, D. Nahamoo, M. Picheny, IBM T.J. Watson Res. Ctr. (USA)

Continuous Speech Recognition by Context-Dependent Phonetic HMM and an Efficient Algorithm for Finding N-Best Sentence Hypotheses
K. Itou, Tokyo Inst. of Tech. (Japan); S. Hayamizu, Electrotechnical Lab. (Japan); H. Tanaka, Tokyo Inst. of Tech. (Japan)

An Efficient A* Stack Decoder Algorithm for Continuous Speech Recognition with a Stochastic Language Model
D. Paul, MIT Lincoln Lab. (USA)

Dividing the Distributions of HMM and Linear Interpolation in Speech Recognition
K. Asai, S. Hayamizu, K. Handa, Electrotechnical Lab. (Japan)

Subphonicet Modeling with Markov States
M. Hwang, X. Huang, Carnegie Mellon Univ. (USA)

Japanese Dictation System Using Character Source Modeling
T. Yamada, S. Matsunaga, K. Shikano, NTT Human Interface Labs. (Japan)

ANALYSIS BY SYNTHESIS CODING

Chair: J. Campbell, Dept. of Defense (USA)

Techniques for Improving the Quality of LD-CELP Coders at 8 KB/s
R. Soheili, A. Kondoz, B. Evans, Univ. of Surrey (UK)

Low-Delay VXC at 8 kb/s with Interframe Coding
J. Yao, J. Shynk, A. Gersho, Univ. of California, Santa Barbara (USA)

On Reducing the Bit Rate of a CELP Based Speech Coder
Y. Liu, ITT Aerospace/Communications Div. (USA)

Reduced Complexity CELP Coder
M. Mauc, G. Baudoin, ESIEE (France)

A Multi-Stage Perspective on CELP Speech Coding
P. Hedelin, Chalmers Univ. of Tech. (Sweden)

Successive Orthogonalizations in the Multistage CELP Coder
N. Moreau, Telecom Paris (France); P. Dymarski, Tech. Univ. of Warsaw (Poland)

A New Excitation Model for LPC Vocoder at 2.4 Kb/s
X.W. Zhang, X.Z. Chen, Nanjing Inst. of Comm. Eng. (China)

Improving the Performance of the 16 Kb/s LD-CELP Speech Coder
J. Chen, N. Jayant, R. Cox, AT&T Bell Labs. (USA)

Pole-Zero Code Excited Linear Prediction Using a Perceptually Weighted Error Criterion
M. Dunn, B. Murray, A. Fagan, Univ. Col. Dublin (Ireland)
On the Effectiveness of Parameter Reoptimization in Multipulse Based Coders
M. Fratti, Teletlra S.p.A. (Italy); A. Mian, Univ. of Padova (Italy); G. Riccardi, CEFRIEL (Italy)

TOPICS IN SPEECH CODING
Chair: P. Kroon, AT&T Bell Labs. (USA)

Kalman Filtering Techniques in Speech Coding
S. Crisafulli, Australian Nat'l Univ. (Australia); J. Mills, Tellabs Res. Ctr. (USA); R. Bitmead, Australian Nat'l Univ. (Australia)

Time-Scale Modification of Speech Using an Incremental Time-Frequency Approach with Waveform Structure Compensation
B. Sylvestre, P. Kabal, McGill Univ. (Canada)

Signal Reconstruction from Modified Wavelet Transform—An Application to Auditory Signal Processing
T. Irino, H. Kawahara, NTT Basic Res. Labs. (Japan)

An Efficient Approximation-Elimination Algorithm for Fast Nearest-Neighbour Search
V. Ramasubramaniam, K. Paliwal, Tata Inst. of Fundamental Res. (India)

Speech Coding by the Efficient Transformation of the Spectral Envelope of Subwords
V. Algazi, D. Irvine, C. Caldwell, M. Ready, K. Brown, S. Chung, Univ. of California, Davis (USA)

Low Bit-Rate Quantization of LSP Parameters Using Two-Dimensional Differential Coding
C.C. Kuo, F.R. Jean, H.C. Wang, Nat'l Tsing Hua Univ. (China)

Adaptive Vector Quantization for Waveform Coding
T. Wang, J. Foster, S. Ardalan, North Carolina A&T State Univ. (USA)

Tree Searched Multi-Stage Vector Quantization for 4kb/s Speech Coding
B. Bhattacharya, W. LeBlanc, S. Mahmoud, V. Cuperman, Simon Fraser Univ. (Canada)

A Fast VQ Codebook Design Algorithm for a Large Number of Data
M. Nakai, H. Shimodaira, Tohoku Univ. (Japan); M. Kimura, Japan Adv. Inst. of Sci. and Tech. (Japan)

Synthesis/Coding of Audio Signals Using Optimized Wavelets
D. Sinha, A. Tewfik, Univ. of Minnesota, Minneapolis (USA)

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Thinned Lattice Filter for LPC Analysis
C.F. Chan, K.W. Law, City Polytechnic of Hong Kong (Hong Kong)

RASTA-PLP Speech Analysis Technique
H. Hermansky, U.S. WEST Adv. Tech. (USA); N. Morgan, ICSI (USA); A. Bayya, US WEST Adv. Tech. (USA); P. Kohn, ICSI (USA)

Exploiting Recursive Parameter Trajectories in Speech Analysis
N. Hubing, K. Yoo, Univ. of Missouri, Rolla (USA)

Real-Time Robust Pitch Detector
M. Doğan, J. Mendel, Univ. of Southern California (USA)

Pitch Determination of Noisy Speech Using Higher Order Statistics
A. Moreno, J. Fonollosa, ETSIT, Barcelona (Spain)

An Adaptive Algorithm for Mel-Cepstral Analysis of Speech
T. Fukada, Canon Inc. (Japan); K. Tokuda, T. Kobayashi, S. Imai, Tokyo Inst. of Tech. (Japan)

Signal Approximation via Data-Adaptive Normalized Guassian Functions and Its Applications for Speech Processing
S. Qian, National Instruments (USA); K. Chen, Univ. of Maryland (USA); D. Chen, National Instruments (USA)

Voice Transformation Using PSOLA Technique
H. Valbret, E. Moulines, J. Tubach, Telecom Paris (France)

A Scheme for Pitch Extraction of Speech Using Autocorrelation Function with Frame Length Proportional to the Time Lag
K. Hirose, Univ. of Tokyo (Japan); H. Fujisaki, Sci. Univ. of Tokyo (Japan); S. Seto, Toshiba Corp. (Japan)

Cinematic Techniques for Speech Processing: Temporal Decomposition and Multivariate Linear Prediction
C. Montacie, M. Caraty, Univ. P & M Curie (France); P. Deleglise, F. Bimbot, Telecom Paris (France)

LANGUAGE MODELING & SPEECH UNDERSTANDING
Chair: L. Hirschman, MIT (USA)

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S. Lerner, B. Mazor, GTE Labs. (USA)

Non-Linear Spectral Subtraction (NSS) and Hidden Markov Models for Robust Speech Recognition in Car Noise Environments
P. Lockwood, J. Boudy, M. Blanchet, MATRA Comm. (France)

A Robust Speech/Non-Speech Detection Algorithm Using Time and Frequency-Based Features
B. Mak, J. Junqua, B. Reaves, Speech Tech. Lab. (USA)

SPEECH ENHANCEMENT & NOISE REDUCTION

Chair: Y. Ephraim, AT&T Bell Labs. (USA)

Pitch Dependent Phone Modelling for HMM Based Speech Recognition
H. Singer, S. Sagayama, ATR Interpreting Telephony Res. Labs. (Japan)

Single Sensor Active Noise Cancellation Based on the EM Algorithm
A. Oppenheim, MIT (USA); E. Weinstein, Tel-Aviv Univ. (Israel); K. Zangi, MIT (USA); M. Feder, Tel-Aviv Univ. (Israel); D. Gauger, BOSE Corp. (USA)

Hands-Free Voice Communication in an Automobile with a Microphone Array
S. Oh, V. Viswanathan, P. Papamichalis, Texas Instruments (USA)

Beamforming Microphone Arrays for Speech Enhancement
K. Farrell, R. Mammore, J. Flanagan, Rutgers Univ. (USA)

Speech Enhancement Using State Dependent Dynamical System Model
Y. Ephraim, AT&T Bell Labs. (USA)

A Wideband Blind Identification Approach to Speech Data Acquisition Using a Microphone Array
V. Soon, Univ. of Notre Dame (USA); L. Tong, Stanford Univ. (USA); Y. Huang, R. Liu, Univ. of Notre Dame (USA)

Dual-Channel Speech Enhancement with Auditory Spectrum Based Constraints
S. Nandkumar, J. Hansen, Duke Univ. (USA)

Vector Equalization in Hidden Markov Models for Noisy Speech Recognition
B. Juang, K. Paliwal, AT&T Bell Labs. (USA)

A Microphone Array for Car Environments
Y. Grenier, ENST (France)

Robust Estimation of AR Parameters and Its Application for Speech Enhancement
K. Lee, B. Lee, Seoul Nat'l Univ. (Korea); I. Song, KAIST (Korea); S. Ann, Seoul Nat'l Univ. (Korea)

IMPROVED EXCITATION IN CELP CODERS

Chair: V. Cuperman, Simon Fraser Univ. (Canada)

CELP Coding at 4.0 Kb/sec and Below: Improvements to FS-1016
R. Zinser, S. Koch, General Electric (USA)

Ultra-Fast CELP Coding Using Deterministic Multi-Codebook Innovations
D. Lin, International Mobile Machines (USA)

Improved 4.8kb/s CELP Coding Using Two-Stage Vector Quantization with Multiple Candidates (LCELP)
T. Taniguchi, Y. Tanaka, Y. Ohta, Fujitsu Labs. Ltd. (Japan)

Fractional Excitation and Other Efficient Transformed Codebooks for CELP Coding of Speech
M. Delprat, C. Gruet, F. Dervaux, MATRA Comm. (France)

Generalized Analysis-by-Synthesis Coding and Its Application to Pitch Prediction
W. Kleijn, R. Ramachandran, P. Kroon, AT&T Bell Labs. (USA)

A Deconvolution-based Efficient Method for Generating the Excitation in Linear Predictive Speech Coding
D. Docampo, V. Abreu, F. Perez, F. Gonzalez, ETSIT, Vigo (Spain)

Mixture Excitations and Finite-State CELP Speech Coders
A. Benyassine, New Jersey Inst. of Tech. (USA); H. Abut, San Diego State Univ. (USA)

Improved Phonetically-Segmented Vector Excitation Coding at 3.4 kb/s
S. Wang, A. Gersho, Univ. of California, Santa Barbara (USA)

ALGORITHMS FOR RECOGNITION

Chair: J. Tubach, Telecom Paris (France)

Predictor Codebooks for Speaker-Independent Speech Recognition
T. Kawabata, NTT Basic Res. Labs. (Japan)
An Improved VQ Codebook Design Algorithm for HMM
J. Koo, H.S. Lee, C. Un, KAIST (Korea)

Mixture Density Estimators in Viterbi Training
C. Wellekens, L & H Speech Products (Belgium)

HMM Based on Pair-Wise Bayes Classifiers
T. Kawahara, S. Doshita, Kyoto Univ. (Japan)

Temporal Decomposition for the Initialisation of a HMM Isolated Word-Recogniser
M. Taylor, F. Bimbot, Telecom Paris (France)

Modeling Improvement of the Continuous Hidden Markov Model for Speech Recognition
Z. Hu, S. Imai, Tokyo Inst, of Tech. (Japan)

A Family of Parallel Hidden Markov Models
F. Brugnara, IRST (Italy); R. DeMori, McGill Univ. (Canada); D. Giuliani, M. Omologo, IRST (Italy)

Modeling State Durations in Hidden Markov Models for Automatic Speech Recognition
P. Ramesh, J. Wilpon, AT&T Bell Labs. (USA)

Representing Dynamic Features of Phonetic Segment in an Orthogonalized Codebook of HMM Based Speech Recognition System
T. Nitta, J. Iwasaki, Y. Masai, H. Matsu’ura, Toshiba Corp. (Japan)

Context Modeling with the Stochastic Segment Model
M. Ostendorf, I. Bechwati, O. Kimball, Boston Univ. (USA)

NEURAL NETS FOR SPEECH RECOGNITION
Chair: H. Leung, MIT (USA)

Unsupervised Information Theory-Based Training Algorithms for Multilayer Neural Networks
G. Rigoll, NTT Human Interface Labs. (Japan)

Context-Dependent Hidden Control Neural Network Architecture for Continuous Speech Recognition
B. Petek, J. Tebelskis, Carnegie Mellon Univ. (USA)

A Fast Neural Network Training Algorithm for Speech Recognition
E. Buhrke, J. LoCicero, Illinois Inst. of Tech. (USA)

A Neural Fuzzy Training Approach for Continuous Speech Recognition Improvement
Y. Komori, ATR Interpreting Telephony Res. Labs. (Japan)

Speaker-Independent Phoneme Recognition Using Large-Scale Neural Networks
S. Nakamura, Keio Univ. (Japan); H. Sawai, Ricoh Co. Ltd. (Japan); M. Sugiyama, ATR Interpreting Telephony Res. Labs. (Japan)

A Multi-task Neural Network Approach to Speech Recognition
E. Richards, Univ. of Colorado (USA)

Prototype-Based Discriminative Training for Various Speech Units
E. McDermott, S. Katagiri, ATR Auditory & Visual Perception Res. Labs. (Japan)

Incorporating Acoustic-Phonetic Knowledge in Hybrid TDNN/Viterbi Framework
L. Devillers, LISI (France); C. Dugast, Philips Res. Labs. (Germany)

Error-Correcting Training for Phoneme Spotting
L. Niles, L. Wilcox, M. Bush, Xerox PARC (USA)

Parallel Sequential Running Neural Network and Its Application to Automatic Speech Recognition
H. Zeng, T. Yu, Academia Sinica (China)

SPEAKER ADAPTATION
Chair: C. Lee, AT&T Bell Labs. (USA)

A Segment-based Speaker Adaptation Neural Network Applied to Continuous Speech Recognition
K. Fukuzawa, Y. Komori, ATR Interpreting Telephony Res. Labs. (Japan); H. Sawai, Ricoh Co. Ltd. (Japan); M. Sugiyama, ATR Interpreting Telephony Res. Labs. (Japan)

A Bayesian Approach to Speaker Adaptation for the Stochastic Segment Model
B. Necioglu, M. Ostendorf, Boston Univ. (USA); R. Rohlicek, BBN Systems and Technologies (USA)

An LVQ Based Reference Model for Speaker Adaptive Speech Recognition
O. Schmidbauer, Siemens AG (Germany); J. Tebelskis, Carnegie Mellon Univ. (USA)

Robust Speaker Adaptation Using a Piecewise Linear Acoustic Mapping

A Piecewise Linear Spectral Mapping for Supervised Speaker Adaptation
H. Matsukoto, H. Inoue, Shinshu Univ. (Japan)

Rapid Connectionist Speaker Adaptation
M. Witbrock, Carnegie Mellon Univ. (USA); P. Haffner, CNET (France)

Speaker Adaptive Phoneme Recognition Based on Feature Mapping from Spectral Domain to Probabilistic Domain
T. Kobayashi, Waseda Univ. (Japan); Y. Uchiyama, J. Osada, Hosei Univ. (Japan); K. Shirai, Waseda Univ. (Japan)
Fast Speaker Adaption Combined with SOFT Vector Quantization in a HMM Speech Recognition System  
F. Class, A. Kaltenmeier, P. Regel-Brietzmann, Daimler-Benz (Germany); K. Trottler, Deutsche Aerospace (Germany)

Speaker Normalization for Speech Recognition  
X. Huang, Carnegie Mellon Univ. (USA)

Speaker-Independent Speech Recognition Method Using Training Speech from a Small Number of Speakers  
M. Hoshimi, M. Miyata, S. Hiraoka, K. Niyada, Matsushita Research Inst. Tokyo Inc. (Japan)

LEARNING & DISCRIMINATIVE TRAINING  
Chair: M. Picheny, IBM T.J. Watson Res. Ctr. (USA)

Segmental GPD Training of HMM Based Speech Recognizer  
W. Chou, B. Juang, C. Lee, AT&T Bell Labs. (USA)

Adaptation of Large Vocabulary Recognition System Parameters  

Improved Acoustic Modeling with Bayesian Learning  
J. Gauvain, C. Lee, AT&T Bell Labs. (USA)

Vocabulary Learning and Normalization Environment Adaptation in Vocabulary-Independent Speech Recognition  
H. Hon, Carnegie Mellon Univ. (USA); K. Lee, Apple Computer (USA)

Adaptation of the HMM Distributions: Application to a VQ Codebook and to a Noisy Environment  
E. Frangoulis, D. Gaganelis, Logica Cambridge (UK)

Discriminative Template Training for Dynamic Programming Speech Recognition  
P.C. Chang, Ministry of Comm. (China); B.H. Juang, AT&T Bell Labs. (USA)

Application of a Generalized Probabilistic Descent Method to Dynamic Time Warping-Based Speech Recognition  
T. Komori, S. Katagiri, ATR Auditory & Visual Perception Res. Labs. (Japan)

Discriminative Analysis for Feature Reduction in Automatic Speech Recognition  
E. Bocchien, J. Wilpon, AT&T Bell Labs. (USA)

High Performance Connected Digit Recognition Using Codebook Exponents  
R. Cardin, Y. Normandin, CRIM (Canada); R. DeMori, McGill Univ. (Canada)

Hidden Markov Models Using Vector Linear Prediction and Discriminative Output Distributions  
P. Woodland, Cambridge Univ. (UK)

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Chair: F. Soong, AT&T Bell Labs. (USA)

On the Performance of Polynomial and HMM Whole-Word Classifiers for Digit Recognition Over Telephone  
H. Katterfeldt, P. Regel-Brietzmann, B. Vater, Daimler-Benz (Germany)

SWITCHBOARD: Telephone Speech Corpus for Research and Development  
J. Godfrey, E. Holliman, J. McDaniel, Texas Instruments (USA)

Automatic Recognition of Hesitations in Spontaneous Speech  
D. O'Shaughnessy, INRS Telecomm. (Canada)

An Improved Speech Detection Algorithm for Isolated Korean Utterances  
M. Hahn, C. Park, Elec. and Tel. Res. Inst. (Korea)

Static Representation of Speech Dynamics for Isolated Word Recognition  
C. Chan, J. Wu, Univ. of Hong Kong (Hong Kong)

Robust Automatic Time Alignment of Orthographic Transcriptions with Unconstrained Speech  

On Increasing Structural Complexity of Finite State Speech Models  
S. Vaseghi, P. Conner, Univ. of East Anglia (UK)

An Application of the Modulation Model to Speech Recognition  
A. Fineberg, R. Mammeone, J. Fianagan, Rutgers Univ. (USA)

HMM Representation of Quantized Articulatory Features for Recognition of Highly Confusible Words  
K. Erler, L. Deng, Univ. of Waterloo (Canada)

On the Use of Acoustic-Phonetic Features in Interactive Labelling of Multi-Lingual Speech Corpora  
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Chair: M. Ostendorf, Boston Univ. (USA)

Phonemic HMM Constrained by Statistical VQ-Code Transition
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Experiments on Speaker-Independent Phone Recognition Using BREF
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Experiments on Stress-Dependent Phone Modelling for Continuous Speech Recognition
M. Adda-Decker, G. Adda, LIMSI (France)

Use of Semi-Markov Models for Speaker-Independent Phoneme Recognition
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The General Use of Tying in Phoneme-Based HMM Speech Recognisers
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A Successive State Splitting Algorithm for Efficient Allophone Modeling
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Acoustic Modelling of Subword Units in the ISADORA Speech Recognizer
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Recognition of Demisyllable Based Units Using Semicontinuous Hidden Markov Models
B. Plannerer, G. Ruske, Tech. Univ. of Munich (Germany)

The Automatic Recognition of Stop Consonants Using Hidden Markov Models
T. Waardenburg, J. du Preez, Univ. of Stellenbosch (South Africa); W. Coetzer, Datafusion Systems (South Africa)

Relationship among Phoneme/Word Recognition Rate, Perplexity and Sentence Recognition Rate and Comparison of Language Models
S. Nakagawa, Toyohashi Univ. of Tech. (Japan)

NEURAL NET & HYBRID RECOGNITION SYSTEMS

Chair: A. Waibel, Carnegie Mellon Univ. (USA)

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Continuous Speech Recognition with Modified Learning Vector Quantization Algorithm and Two-Level DP-Matching
S. Makino, M. Endo, T. Sone, Tohoku Univ. (Japan); K. Kido, Chiba Inst. of Tech. (Japan)

Connectionist Probability Estimation in the DECIPHER Speech Recognition System
S. Renals, N. Morgan, ICSI (USA); M. Cohen, H. Franco, SRI International (USA)

Expanding the Vocabulary of a Connectionist Recognizer Trained on the Darpa Resource Management Corpus
H. Lucke, F. Fallside, Cambridge Univ. (UK)

A Speech Recognizer Optimally Combining Learning Vector Quantization, Dynamic Programming and Multi Layer Perceptron
X. Driancourt, P. Gallinari, Univ. of Paris, Sud (France)

Speech Recognition Using Stochastic Segmental Neural Networks
H.C. Leung, I. Hetherington, V. Zue, MIT (USA)

A Real-Time Recurrent Error Propagation Network Word Recognition System
T. Robinson, Cambridge Univ. (UK)

Connectionist Word-Level Classification in Speech Recognition
P. Haffner, CNET (France)

Speech Recognition Using Segmental Neural Nets
S. Austin, BBN Systems and Technologies (USA); G. Zavaliagkos, Northeastern Univ. (USA); J. Makhoul, R. Schwartz, BBN Systems and Technologies (USA)

A Speech Recognizer Using Radial Basis Function Neural Networks in an HMM Framework
E. Singer, R. Lippmann, MIT Lincoln Lab. (USA)