

EUROSPEECH '93

PROCEEDINGS

UB/TIB Hannover 89
113 210 000



VOLUME 2

- 21.23 Time-Varing Manner on Formant Trajectories of Chinese Diphthong
– *J.Cao, Chinese Academy of Social Sciences, Beijing, China* 735

Coffee Break: 11.00 - 11.20 hrs

PROCEEDINGS VOLUME 2

Session 22: Speech Coding IV

Time and Place: 11.20 - 12.40 hrs Room A

Chairperson: *H. Leung, MIT, USA*

- 22.1 High-Quality Speech Coding at 2.4 Kbps Based on Time-Frequency Interpolation
– *Y. Shoham, AT&T Bell Laboratories, USA* 741
- 22.2 Coding of Speech Signal by Fractal Techniques
– *L. Marcato, E. Mumolo, Universita' di Trieste, Italy* 745
- 22.3 A New Reference Signal for Evaluating the Quality of Speech Coded at Low Bit-Rates
– *N. Asanuma, H. Nagabuchi, NTT Human Interface Lab, Telecommunication Networks Laboratories, Japan* 749
- 22.4 A Psychophysical Study of Fourier Phase and Amplitude Coding of Speech
– *C. Ma, D. O'Shaughnessy, INRS-Telecommunications, Canada* 753

Session 23: Phonetics II

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Chairperson: *L. Nord, KTH Stockholm, Sweden*

- 23.1 Data-Driven Identification of Poly- and Mono-Phonemes for Four European Languages
– *O. Andersen(*), P. Dalsgaard(*), W. Barry(**), (*) University of Aalborg, Denmark, (**) Universität des Saarlandes, Germany* 759
- 23.2 Reversible Letter-to-Sound Sound to-Letter Generation Based on Parsing
Word Morphology
– *S. Hunnicutt, H. Meng, S. Seneff, V.W. Zue, MIT, USA* 763
- 23.3 The Role of Context in the Automatic Recognition of Stressed Syllables
– *J. Moore, P. Roach, University of Leeds, UK* 767
- 23.4 Metrical Structure and the Perception of Time-Compressed Speech
– *D. Young(*), G.T.M. Altmann(*), A. Cutler(**), D. Norris(**), (*) Sussex University, (**) MRC Applied Psychology Unit, Cambridge, UK* 771
- 23.5 Are Stress and Phonemic String Processed Separately? Evidence from Speech Illusions
– *V. Pasdeloup, J. Morais, R. Kolinsky, Université Libre de Bruxelles, Belgium* 775

Session 24: Prosody II: Analysis and Modelling of F₀ Contours

Time and Place: 11.20 - 13.00 hrs Room C

Chairperson: *L. Boves, University of Nijmegen, The Netherlands*

- 24.1 On the Automatic Classification of Pitch Movements
– *L. ten Bosch, Institute for Perception Research, The Netherlands* 781

- 24.2 Modelling of Intonation Contours at the Sentence Level Using CHMMs and the 1961 O'Connor and Arnold Scheme
– U. Jensen(*), R. K. Moore(**), P. Dalsgaard(*), B. Lindberg(*),
(*) Aalborg University, Denmark, (**) Defence Research Agency Malvern, UK 785
- 24.3 Automatic Recognition of Intonation from F₀ Contours Using the Rise/Fall/Connection Model
– P. Taylor, ATR Interpreting Telecomm. Labs, Kyoto, Japan 789
- 24.4 A Pitch Contour Analysis Guided by Prosodic Event Detection;
– E. Geoffrois, NTT Human Interface Lab., Telecomm. Networks Lab., Tokyo, Japan 793
- 24.5 Analysis and Synthesis of Pitch Movements in a Read Polish Text
– G. Demenko, I. Nowak, J. Imiolczyk, Polish Academy of Sciences, Poland 797

Session 25: Improved Algorithms for HMMs II

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Chairperson: A. Noll, aspect, Hamburg, Germany

- 25.1 Optimization of an HMM-Based Continuous Speech Recognizer
– F. Class, A. Kaltenmeier, P. Regel-Brietzmann, Daimler-Benz AG, Ulm, Germany 803
- 25.2 Linear and Nonlinear Prediction for Speech Recognition with Hidden Markov Models
– M. Saerens(*), H. Bourlard(**), (*) Université Libre de Bruxelles,
(**) Lernout & Hauspie Speech Products, Belgium 807
- 25.3 Segmental Post-Processing of the N-Best Solutions in a Speech Recognition System
– M.N. Lokbani, D. Jouviet, J. Monné, France Télécom, CNET/LAA/TSS/RCP, France 811
- 25.4 A Study of On-Line Bayesian Adaptation for HMM-Based Speech Recognition
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– B.A. Maxwell, P.C. Woodland, Cambridge University, UK 819

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– J.A. Nolasco Flores, S.J. Young, Cambridge University, UK 829
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– M.J.F. Gales, S.J. Young, University of Cambridge, UK 837
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– L. Buniêt(*), D. Fohr(*), Y. Anglade(*), J-C. Junqua(**), J-M. Pierrel(*),
(*) CRIN-CNRS & INRIA, France, (**) Panasonic, USA 841

Session 27: Speaker Independency**Time and Place:** 11.20 - 12.40 hrs Room F*Chairperson: E. Paulus, Technical University of Braunschweig, Germany*

- 27.1 A Baseline of a Speaker Independent Continuous Speech Recognizer of Italian
– B. Angelini, F. Brugnara D. Falavigna, D. Giuliani, R. Gretter, M. Omologo,
Istituto per la Ricerca Scientifica e Tecnologica, Italy 847
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– L.R. Bahl, P.V. de Souza, P.S. Gopalakrishnan, D. Nahamoo, M. Picheny, IBM,
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– D.B. Grayden, M.S. Scordilis, The University of Melbourne, Australia 855
- 27.4 A Continuous Speech Recognition System Using Phonotactic Constraints
– B. Plannerer, G. Ruske, TU München, Germany 859

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– M. Ouadou(*), A. Rajouani(*), M. Zyoute (*), J. Rosenfeld(**), M. Najim(***),
(* LEESA, Maroc, (***) Université de Bordeaux, (***) ENSERB, France 865
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– E.L. López-Gonzalo(*), G. Olaszy(**), G. Németh(***), (*) Universidad Politécnica
de Madrid, Spain, (***) Hungarian Academy of Science, (***) Technical University of
Budapest, Hungary 869
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– M. Ljungqvist(*), H. Fujisaki(**), (*) Infovox AB, Sweden, (***) Science University
of Tokyo, Japan 873
- 28.4 PHRITTS - A Text-to-Speech Synthesizer for the German Language
– P. Meyer(*), H.W. Rühl(*), R. Krüger(*), M. Kugler(*), L.L.M. Vogten(**),
A. Dirksen (**), K. Belhoula(***), (*) PHILIPS Kommunikations Industrie AG,
Germany, (***) Institute for Perception Research, Eindhoven, The Netherlands,
(***) Ruhr-Universität Bochum, Germany 877
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– K. Belhoula, Ruhr-Universität Bochum, Germany 881
- 28.6 A Prototype Text-to-Speech System for Scottish Gaelic
– I.R. Murray, M.M. Black, University of Dundee, UK 885
- 28.7 A Text-to-Speech System for Polish
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– M. Macchi, M.J. Altom, D. Kahn, S. Singhal, M. Spiegel, Bellcore, USA 893
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– M. Gaved, British Telecom Res. Labs., Ipswich, UK 897

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Chairperson: R. Billi, CSELT Torino, Italy

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