Session 47: Perception
Time and Place: 09.20 - 11.00 hrs  Room E
Chairperson: R. Carlson, KTH, Sweden

47.1 The Perceptual Relevance of CV- and VC- Transitions in Identifying Stop Consonants: Cross-Language Results
- A. van Wieringen(*), J.K. Cullen(**), L.C.W. Pols(*), (*) University of Amsterdam, The Netherlands, (**) Louisiana State University, USA

47.2 Perceptual Effects of Place and Voicing Assimilation in Dutch Consonants
- V.J. van Heuven, W. Jongenburger, Leiden University, Holland Institute of Generative Linguistics, The Netherlands

47.3 Detection of Vowels and Consonants by Human Listeners: Effects of Minimising Auditory Memory Load
- B. van Ooyen, University of Leiden, The Netherlands and MRC Applied Psychology Unit, UK

47.4 Resonances as Possible Representation of Speech in the Auditory-to-Articulatory Transform
- G. Bailly, Université Stendhal, France

47.5 A Perceptual Explanation of the Weightlessness of the Syllable Onset
- R. Goedemans, V.J. van Heuven, Leiden University, Holland Institute of Generative Linguistics, The Netherlands

Session 48: Search Algorithms
Time and Place: 09.20 - 11.00 hrs  Room F
Chairperson: R. Hoffmann, Technical University of Dresden, Germany

48.1 A Study of the Beam-Search Algorithm for Large Vocabulary Continuous Speech Recognition and Methods for Improved Efficiency
- E. Bocchieri, AT&T Bell Laboratories, USA

48.2 Using Grammars in Forward and Backward Search
- L. Fissore(*), E. Giachin(*), P. Laface(**), P. Massafra(**), (*) CSELT, (**) Politecnico di Torino, Italy

48.3 Robust Interpretation of Speech
- G.A. Fink, F. Kummert, G. Sagerer, B. Seestaedt, Universität Bielefeld, Germany

48.4 A* Word Network Search for Continuous Speech Recognition
- I.L. Hetherington, M.S. Phillips, J.R. Glass, V.W. Zue, MIT, USA

48.5 Efficient Lexical Access Strategies
- R. Lacouture, Y. Normandin, McGill College, Canada
Session 49: Speech Recognition, HMMs, NNs
Poster Session 7

Time and Place: 09.20 - 11.00 hrs Room G

Chairperson: M.A. Jack, University of Edinburgh, UK

49.1 Multiple Codebook Spanish Phone Recognition Using Semicontinuous Hidden
Markov Models
- M.I. Torres(*), F. Casacuberta(**), (*) Universidad del País Vasco, (**) Universidad Politécnica de Valencia, Spain

49.2 An Efficient Algorithm to Find the Best State Sequence in HSMM
- A. Bonafonte, X. Ros, J.B. Mariño, Universitat Politècnica de Catalunya, Spain

49.3 Robust HMM-Based Endpoint Detector
- A. Acero, C. Crespo, C. de la Torre, J.C. Torrecilla, Speech Technology Group, Telefonica I+D, Spain

49.4 Experiments on Spanish Phone Recognition Using Automatically Derived Phonemic
Baseforms
- I. Galiano, F. Casacuberta, Universidad Politécnica de Valencia, Spain

49.5 Evaluation of VQ-Distortion Based HMM
- S. Nakagawa, H. Suzuki, L. ZhaoToyoohashi University of Technology, Japan

49.6 Continuous HMM for Word Spotting and Rejection of Non Vocabulary Word in
Speech Recognition over Telephone Networks
- J. Song, University of Sydney, Australia

49.7 Bayesian Learning of the Parameters of Discrete and Tied-Mixture HMMs for Speech
Recognition
- Q. Huo(*), C. Chan(*), C.-H. Lee(**), (*) University of Hong Kong, Hong Kong, (**) AT&T Bell Laboratories, USA

49.8 Speech Recognition Using Semantic Hidden Markov Networks
- G.A. Fink, F. Kummert, G. Sagerer, E.G. Schukat-Talamazzini(*), Universität Bielefeld, (*) Universität Erlangen-Nürnberg, Germany

49.9 Experiments in Vocabulary Independent Speech Recognition Using Phoneme Decision
Trees
- S. Downey(*), M. Russell(*), P. Nowell(*), D. Bijl(**) K. Galloway (*), K. Ponting(*), (*) DRA Malvern, (**) Imperial College, London, UK

49.10 Segmental Hidden Markov Models
- M.J.F. Gales, S.J. Young, Cambridge University, UK

49.11 Impact of Dimensionality and Correlation of Observation Vectors in HMM-Based
Speech Recognition
- X. Wang, L.F.M. ten Bosch, L.C.W. Pols, University of Amsterdam, The Netherlands

49.12 Evaluation of an HMM Speech Recognizer with Various Continuous Speech Databases
- F. Class, A. Kaltenmeier, P. Regel-Brietzmann, Daimler-Benz AG, Germany

49.13 Hidden Markov Models for Noisy Speech Recognition
- A. Wrzoskowicz, Polish Academy of Science, Poland

49.14 Neural Network Speech Enhancer Utilizing Masking Properties
- D.E. Tsoukalas, J. Mourjopolos, G. Kokkinakis, University of Patras, Greece

49.15 Comparison of Geometric, Connectionist and Structural Techniques on a Difficult
Isolated Word Recognition Task
- M.J. Castro, J.C. Perez, Universidad Politécnica de Valencia, Spain
49.16 Prediction and Discrimination in Neural Networks for Continuous Speech Recognition
- A. Mellouk(*), P. Gallinari(**), F. Rauscher(**), (*) LRI, Orsay, (**) LAFORIA, France

49.17 Two Schemes of Phonetic Feature Extraction Using Artificial Neural Networks
- S. Ran, J.B. Millar, Australian National University, Australia

49.18 On Use of Discriminant Analysis in Predictive Connectionist Speech Recognition
- B. Petek, A. Ferligoj, University of Ljubljana, Slovenia

49.19 Non-Linear Time Compression for Lexical Access
- N.H. Russell, F. Fallside, R.W. Prager, Cambridge University, UK

49.20 Talker Enrolment for Speech Recognition by Synthesis
- R. Brierton, N. Sedgwick, Cambridge Algorithmica Ltd., UK

49.21 Improving Robustness of Network Grammar by Using Class HMM
- K. Takeda, N. Inoue, S. Kuroiwa, T. Konuma, S. Yamamoto, KDD R&D Laboratories, Japan

49.22 Parallelising K-Means Clustering on Distributed Memory MIMD Computers
- J.A. Elliott, M.E. Forsyth, F.R. McInnes, N.W. Ramsey, CSTR, Edinburgh, UK

49.23 On the Proper Sub-Word Unit Inventory for CSR
- P. Berényi, K. Viczi, Hungarian Academy of Sciences “György Békésy”, Hungary

49.24 Speech Recognition Using the Atomic Speech Units Constructed from Overlapping Articulatory Features
- L. Deng(*,**), D. Sun(**), (*) MIT, USA, (**) University of Waterloo, Canada

49.25 A Bayesian Approach to Phone Duration Adaptation for Lombard Speech Recognition
- O. Siohan, Y. Gong, J.-P. Haton, CRIN-CNRS/INRIA Lorraine, France

49.26 Multiple Multilabeling to Improve HMM-Based Speech Recognition in Noise
- J. Hernando, J.B. Mariño, C. Nadeu, Polytechnical University of Catalonia, Spain

49.27 Discrimination of Polish Stop Consonants Based on Mapped Techniques
- L. Richter, P. Domagala, Polish Academy of Science, Poland

Coffee Break: 11.00 - 11.20 hrs

Session 50: Spoken Language Dialogue
Time and Place: 11.20 - 13.00 hrs Room A

Chairperson: S. Seneff, MIT, USA

50.1 Managing Spoken Dialogues for Information Services
- W. Eckert(*), S. McGlashan(**), (*) Friedrich-Alexander-Universität Erlangen-Nürnberg, Germany, (**) University of Surrey, UK

50.2 Ambiguity and Uncertainty in Spoken Dialogue
- P. Heisterkamp, Daimler-Benz AG, Germany

50.3 Managing Dialogue in a Continuous Speech Understanding System
- E. Gerbino, M. Danieli, Centro Studi e Laboratori Telecomunicazioni, Italy

50.4 Speaking with Computers: A Multimodal Approach
- P. Lefebvre(*), G. Duncan(*), F. Poirier(**), (*) Siemens-Nixdorf Information Systems, Cergy, (**) Télécom Paris, France

50.5 Habitable Interaction in Goal-Oriented Multimodal Dialogue Systems.
- P. Morin(*,**), J.-C. Junqua(*), (*) Panasonic, USA, (**) CRIN-CNRS/INRIA, France

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53.4 Using LVQ to Enhance Semi-Continuous Hidden Markov Models for Phonemes
   - M. Kurimo, Helsinki University of Technology, Finland

53.5 An Improvement of the Two-Level DP Matching Algorithm Using K-NN Techniques for Acoustic-Phonetic Decoding
   - P. Aibar, F. Casacuberta, Universidad Politécnica de Valencia, Spain

Session 54: Visual Cues

Time and Place: 11.20 - 12.40 hrs Room E
Chairperson: G. Benoit, Université Stendhal, France

54.1 Visual Coarticulation Effects in Syllable Environment
   - H.-H. Bothe, F. Rieger, R. Tackmann, Technical University of Berlin, Germany

54.2 Depth Measurement of Face and Palate by Structured Light
   - C.H. Shadle, J.N. Carter, T.P. Monks, J. Field, University of Southampton, UK

54.3 VISIOLAB: A Multimedia Environment for the Study of Bimodal Speech Perception
   - L.-J. Boe, S. Kandel, A. Chapelet, T. Lallouache, Université Stendhal, France

54.4 Integrating Auditory and Visual Representations for Audiovisual Vowel Recognition
   - J. Robert-Ribes, T. Lallouache, P. Escudier, J.-L. Schwartz, Université Stendhal, Grenoble, France

Session 55: Segmentation and Labelling

Time and Place: 11.20 - 13.00 hrs Room F
Chairperson: R. Gubrinowicz, Polish Academy of Science, Warsaw, Poland

55.1 Iterative Transformation and Alignment for Speech Labelling
   - Y. Gong, J.-P. Haton, CRIN - CNRS/INRIA, Vandoeuvre, France,

55.2 Controlling Search in Segmentation Lattices of Speech Signals
   - K. Hübner, A. Hauenstein, University of Hamburg, Germany

55.3 Accent Phrase Segmentation Using Transition Probabilities between Pitch Pattern Templates
   - H. Shimodaira(*), M. Nakai(**), (*) Japan School of Information Science, (**) Tohoku University, Japan

55.4 Syllable Segmentation of Continuous Speech with Artificial Neural Networks
   - W. Reichl, G. Ruske, TU München, Germany

55.5 Labelling of Speech Given Its Text Representation
   - M. Blomberg, R. Carlson, Department of Speech Communication and Music Acoustics, KTH Stockholm, Sweden

Session 56: Telecomm., Application Aspects

Poster Session 8
Time and Place: 11.20 - 13.00 hrs Room G
Chairperson: H.W. Rühl, PKI Nürnberg, Germany

56.1 Speech Recognition over Packetized Voice Systems
   - B. Baungaard, J.S. Nielsen, Jydsk Telefon, Denmark

56.2 Voice Applications on BT's Derived Services Network
   - I.W.G. Jenkins, BT Laboratories, UK
Session 51: Speech Input/Output Assessment I

Time and Place: 11.20 - 13.00 hrs Room B

Chairperson: A. Fourcin, University College London, UK

51.1 Test of Voice Quality on ATM Based Equipment
- J.S. Nielsen, B. Baungaard, Jydsk Telefon, Denmark

51.2 An Evaluation System for Ascertaining the Quality of Synthetic Speech Based on Subjective Category Rating Tests.
- H. Klaus(*), H. Klix(*), J. Sotscheck(**), K. Fellbaum(*), (*) Technical University of Berlin, (**) German Telekom, Germany

51.3 A Global Framework for the Assessment of Synthetic Speech without Subjects
- A. Mariniak, Ruhr-Universität Bochum, Germany

51.4 Comprehension of KTH Text-to-Speech with “Listening Speed” Paradigm
- L. Neovius, P. Raghavendra, KTH Stockholm, Sweden

51.5 Theoretical Principles Concerning Segmentation, Labelling Strategies and Levels of Categorial Annotation for Spoken Language Database Systems
- H.G. Tillmann, B. Pompeino-Marshall, Ludwig-Maximilians Universität München, Germany

Session 52: Synthesis: Sound Generation

Time and Place: 11.20 - 13.00 hrs Room C

Chairperson: C. Sorin, CNET, Lannion, France

52.1 A Speech Formant Synthesizer Based on Harmonic + Random Formant-Waveforms Representations
- S. Grau, C. d’Alessandro, G. Richard, LIMSI-CNRS, France

52.2 SpeakEZ: A First Experiment in Concatenation Synthesis from a Large Corpus
- A.G. Hauptmann, Carnegie Mellon University, USA

52.3 Designing Control Rules for a Serial Pole-Zero Vocal Tract Model
- J. Kerkhoff, L. Boves, Nijmegen University, The Netherlands

52.4 English Speech Synthesis Based on Multi-Layered Context Oriented Clustering; Towards Multi-Lingual Speech Synthesis
- S. Nakajima, NTT Human Interface Laboratories, Japan

52.5 Speech Synthesis Using Artificial Neural Networks Trained on Cepstral Coefficients
- C. Tuerk, T. Robinson, Cambridge University, UK

Session 53: Hybrid HMM/ANNs for Speech Recognition I

Time and Place: 11.20 - 13.00 hrs Room D

Chairperson: H. Bourlard, L&H Speech Products, Belgium

53.1 Bayesian Regularisation Methods in a Hybrid MLP-HMM System
- S. Renals, D. MacKay, Cambridge University, UK

53.2 Real-Time, Neural Network-Based, French Alphabet Recognition with Telephone Speech
- P. Schmid(*), R. Cole(*), M. Fanty(*), H. Bourlard(**), M. Haessen(**), (*) Oregon Graduate Institute, Portland, USA, (**) L&H Speech Products, Belgium

53.3 Joint Optimization of Multiple Neural Codebooks in a Hybrid Connectionist-HMM Speech Recognition System
- G. Rigoll, University of Duisburg, Germany
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<td>A French Oral Dialogue System for Flight Reservations over the Telephone</td>
<td>J.-Y. Magadur, F. Gavignet, F. Andry, F. Charpentier, CAP Gemini Innovation, France</td>
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<td>56.5</td>
<td>The VOIS Project in Retrospect</td>
<td>W.C.G. Ortel, D. Yashchin, NYNEX Science and Technology Inc., USA</td>
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<td>TELEMACO - A Real-Time Keyword Spotting Application for Voice Dialling</td>
<td>E. Lleida, J.B. Mariño, A. Moreno, U.P.C., Barcelona, Spain</td>
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<td>The Relative Importance of the Factors Affecting Recogniser Performance with Telephone Speech</td>
<td>P. Wyard, BT Laboratories, UK</td>
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<td>A Robust Acoustic Echo Canceller for a Hands-Free Voice-Controlled Telecommunication Terminal</td>
<td>T. Burger(*), U. Schultheiβ(**), TH Darmstadt, Deutsche Telekom, Germany</td>
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<td>Polyphase Allpass IIR Structures for Sub-Band Acoustic Echo Cancellation</td>
<td>J.E. Hart, P.A. Naylor, O. Tanrikulu, Imperial College, London, UK</td>
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<td>Speech Input Systems and Their Effect on Written Language Skills</td>
<td>J. Monaghan, C. Cheepen, University of Hertfordshire, UK</td>
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<td>VOXAID: An Interactive Speaking Communication Aid Software for the Speech Impaired</td>
<td>G. Olasy(*), G. Németh(**), Hungarian Academy of Sciences, TU Budapest, Hungary</td>
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<td>56.12</td>
<td>Feature Extraction for Profoundly Deaf People</td>
<td>U. Hartmann, K. Hermansen, F.K. Fink, Aalborg University, Denmark</td>
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<td>56.13</td>
<td>Architecture of a 10 000 Word Real Time Speech Recognizer</td>
<td>A. Hauenstein, TU München, Germany</td>
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<td>A Noise-Robust Real-Time Word Recognition Hardware Module</td>
<td>T. Hörmann, H. Eckhardt, M. Trompf, H. Hackbarth, Alcatel SEL AG, Germany</td>
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<td>KARS: A Speaker-Independent, Vocabulary-Independent Speech Recognition System</td>
<td>M.-W. Koo, Korea Telecom, Korea</td>
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<td>56.16</td>
<td>A Parallel Processing Keyword Recogniser for Police National Computer Enquiries</td>
<td>F.R. McInnes, J.A. Elliott, N.W. Ramsey, M.E. Forsyth, A.M. Sutherland, M.A. Jack, CSTR, Edinburgh, UK</td>
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<td>56.17</td>
<td>COST232: Speech Recognition over the Telephone Line</td>
<td>A. Paoloni(<em>), T. Svendsen(<strong>), B. Kaspar(</strong>), D. Johnston(</em><em><strong>), G. Hult(</strong></em>**), FUB Italy, NTH, Norway, FTZ, Germany, BT Laboratories, Telia, Sweden</td>
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<td>Individual Variability in the Perception of Synthetic Speech</td>
<td>V. Hazan, B. Shi, University College London, UK</td>
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<td>Speech Recognition System and its Application for Blind PC Users</td>
<td>Y.K. Ludovic, V.V. Pilipenko, G.E. Tseitlin, L.I. Nagornaya, T. Terzian, Kiev, Ukraine</td>
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Break for Lunch: 13.00 - 14.00 hrs
Session 57: Spoken Language Dialogue Application I

Time and Place: 14.00 - 15.40 hrs  
Room A

Chairperson: W. Eckert, University of Erlangen, Germany

57.1 The NLP Module of a Spoken Language Dialogue System for Danish Flight Reservations  
- B. Music, C. Povlsen, Centre for Language Technology, Denmark  
1859

57.2 A Man-Machine Dialogue System for Speech Access to Train Timetable Information  
- D. Clementino, L. Fissore, CSELT, Italy  
1863

57.3 An Experimental Dialogue System: WAXHOLM  
1867

57.4 A Spoken Dialogue System for German Intercity Train Timetable Enquiries  
- W. Eckert, T. Kuhn, H. Niemann, S. Rieck, A. Scheuer, E.G. Schukat-Talamazzini, Universität Erlangen-Nürnberg, Germany  
1871

57.5 A Telephone Banking System Based on HMM Keyword Recognition  
- K. Labropoulou(*), N. Fakotakis(**), (*) Knowledge SA, (**) University of Patras, Greece  
1875

Session 58: Speech Input/Output Assessment II

Time and Place: 14.00 - 15.40 hrs  
Room B

Chairperson: V. Vittorelli, Torino, Italy

58.1 The Comparative Assessment of Commercial Speech Recognisers  
- P. Wyard, BT Laboratories, UK  
1881

58.2 Reliable Assessment of Speech Recognisers for Telephone Environment  
- A. Riccio, F. Ceglie, A. Brancaccio, Alcatel Face, Italy  
1885

58.3 Evaluation of a Rule-Based Text-to-Speech System for French at the Segmental Level  
- M. Garnier-Rizet, LIMSI-CNRS, France  
1889

58.4 Intelligibility of Speech Produced by Text-to-Speech Synthesizers over the Orthophonic and Telephonic Channel  
- C. Delogu(*), A. Poaloni(*), P. Ridolfi(**), K. Vagges(**), (*) Fondazione Ugo Bordoni, (**) Instituto Superiore della Poste e Telecomunicazioni, (***) CNR, Italy  
1893

58.5 Using the Orator® Synthesizer for a Public Reverse-Directory Service: Design, Lessons and Recommendations  
- M.F. Spiegel, Bell Communications Research, USA  
1897

Session 59: Synthesis: Articulatory and Source Modelling

Time and Place: 14.00 - 15.40 hrs  
Room C

Chairperson: C. Jürgens, Technical University of Berlin, Germany

59.1 A Gestural Approach for Controlling an Articulatory Speech Synthesizer  
- B.J. Kröger, Universität zu Köln, Germany  
1903

59.2 An Articulatory Synthesizer for the Simulation of Consonants  
- P. Boersma, University of Amsterdam, The Netherlands  
1907
59.3 Vowel Dynamics in a Text-to-Speech System - Some Considerations
- R. Carlson, L. Nord, KTH Stockholm, Sweden
1911

59.4 Improving the Spectral Balance of Digital Speech Synthesis, Applied to a Female, Synthetic Voice
- I. Frehr, M. Elmlund, H. Nielsen, Telecommunications Research Laboratory, Denmark
1915

59.5 A New Model of Excitation for Text-to-Speech Synthesis
- Y. Ishikawa, T. Ebihara, K. Nakajima, Mitsubishi Electric Corporation, Japan
1919

Session 60: Hybrid HMMs/ANNs for Speech Recognition II
Time and Place: 14.00 - 15.40 hrs Room D

Chairperson: J. Glass, MIT, USA

60.1 Performance Comparison of Hidden Markov Models and Neural Networks for Task Dependent and Independent Isolated Word Recognition
1925

60.2 Connectionist Speech Recognition with a Global MMI Algorithm
- P. Haffner, France Télécom, CNET LAATSS/RCP, France
1929

60.3 Connectionist Segmental Post-Processing of the N-Best Solutions in Isolated and Connected Word Recognition Task
- D. Boiteau, P. Haffner, France Télécom, CNET LAATSS/RCP, France
1933

60.4 A New Dynamic Programming/Multi-Layer Perceptron Hybrid for Continuous Speech Recognition
- J.P. Martens, A. Vorsterrmans, N. Cremelie, University of Gent, Belgium
1937

60.5 A Neural Network Based, Speaker Independent, Large Vocabulary, Continuous Speech Recognition System: The WERNICKE Project
1941

Session 61: Syntactical Constraints
Time and Place: 14.00 - 15.20 hrs Room E

Chairperson: H. Niemann, University of Erlangen, Germany

61.1 A Level-Building Top-Down Parsing Algorithm for Context-Free Grammars in Continuous Speech Recognition
- F. Charpillet, J. Di Martino, CRIN-CNRS & INRIA LORRAINE, France
1947

61.2 Using Anti-Grammar and Semantic Categories for the Recognition of Spontaneous Speech
- R.J. Collingham, R. Garigliano, University of Durham, UK
1951

61.3 Speech Recognition Using Particle N-Grams and Content-Word N-Grams
- R. Isotani(*), S. Sagayama(**), (*) ATR Interpreting Telephony, Research Laboratories, (** NTT Human Interface Laboratories, Tokyo, Japan
1955

61.4 Dynamic Use of Syntactical Knowledge in Continuous Speech Recognition
- P. Dupont, France Télécom - CNET, France
1959

XLII
Session 62: Pathological Voice Analysis

Time and Place: 14.00 - 15.40 hrs  Room F

Chairperson: R. Damper, University of Southampton, UK

62.1 Acoustic Detection of Laryngeal Diseases in Children
- F. Plante, J. Borel, C. Berger-Vachon, I. Kauffmann, URA CNRS, France  
1965

62.2 Acoustic Model and Evaluation of Pathological Voice Production
- D. D. Deliyski, KAY Elemetrics Corp., USA  
1969

62.3 Novel Acoustic Measurements of Jitter and Shimmer Characteristics from Pathological Voice
- H. Kasuya(*), Y. Endo(*), S. Saliiu(**), (*) Utsonomiya University, Japan,  
(**) Polytechnic University of Tirana, Albania  
1973

62.4 An Experiment Involving the Consistency and Reliability of Voice Quality Ratings for Different Types of Speech Fragments
- G. de Krom, University of Utrecht, The Netherlands  
1977

62.5 Laryngectomee Speech in Noise - Voice Effort and Intelligibility
- L. Nord, B. Hammarberg, E. Lundström, KTH Stockholm, Sweden  
1981

Session 63: Speech Analysis: Pitch and Prosody

Poster Session 9

Time and Place: 14.00 - 15.40 hrs  Room G

Chairperson: D. Mehnert, Humboldt University of Berlin, Germany

63.1 Analysing Prosody by Means of a Double Tree Structure
- B. Horvei, G. Ottesen, S. Stensby, SINTEF DELab, Norway  
1987

63.2 Prosody and Discourse Interpretation
- G. Caelen-Haumont, ICPI/INPG et Université Stendhal, France  
1991

63.3 Duration Modelling for the Greek Language
- G. Epitropakis, D. Tambakas, N. Fakotakis, G. Kokkinakis, University of Patras, Greece  
1995

63.4 Prosody Control of TTS-Systems Based on Linguistic Analysis
- G. Epitropakis, N. Yiourgalis, G. Kokkinakis, University of Patras, Greece  
1999

63.5 Prosody Takes Over: A Prosodically Guided Dialog System
- R. Kompe, A. Kiefiling, T. Kuhn, M. Mast, H. Niemann, E. Nöth, K. Ott, A. Batliner(*)  
Universität Erlangen-Nürnberg, (*) Universität München, Germany  
2003

63.6 Integration of a Prosodic Component in an Automatic Speech Recognition System
- P. Langlais, H. Meloni, Université d’Avignon, France  
2007

63.7 Referent Tracking in Restricted Texts Using a Lemmatized Lexicon: Implications for Generation of Intonation
- M. Horne(*), M. Filispson(*), M. Ljungqvist(**), A. Lindström(**), (*) University of Lund, (**) INFOVOX, Sweden  
2011

63.8 Perceptual Significance of Focus Accent in Spoken Swedish
- R. Bannert, University of Umeå, Sweden  
2015

63.9 Pitch Estimation of Speech Signal with the Wavelet Transform
- S. Montresor, M. Baudry, Université du Maine, France  
2017
63.10 A Spectral AMDF Method for Pitch Extraction of Noise-Corrupted Speech  
- J.Y. Rheem(*), M. Bae(**), S. Ann(*), (*) Seoul National University, (**) Soong Sil University, Korea  
2021

63.11 A Reliable Postprocessor for Pitch Determination Algorithms  
- G. Yang(*,**), H. Leich(*), (*) Faculté Polytechnique de Mons, (**) Lernout & Hauspie Speech Products, Belgium  
2025

63.12 Vowel Pitch Period Extraction by Models of Neurones in the Mammalian Brain-Stem  
- G. F. Meyer, W.A. Ainsworth, University of Keele, UK  
2029

63.13 Auto-Regressive Linear Models of Jitter  
- J. Schoentgen, R. De Guchteneere, Université libre de Bruxelles, Belgium  
2033

63.14 Larynx Period Detection Methods in Speech Pattern Hearing Aids  
- J. Wei, D. Howells, A. Faulkner, A. Fourcin, University College London, UK  
2037

63.15 Fundamental Frequency of Dutch Women: An Evaluative Study  
- R. van Bezooijen, Catholic University of Nijmegen, The Netherlands  
2041

Coffee Break: 15.40 - 16.00 hrs

Session 64: Spoken Language Dialogue Appl. II  
Time and Place: 16.00 - 17.40 hrs Room A  
Chairperson: J. Peckham, Logica Cambridge, UK

64.1 A Speech-Based Route Enquiry System Built from General-Purpose Components  
2047

64.2 The INRS ATIS System and Its N-Best Interface  
- C. Yang, D. O'Shaughnessy, INRS-Telecommunications, Canada  
2051

64.3 A Multimodal Directory Guidance System with an Interactive Mechanism  
- T. Nitta, Y. Masai, J. Iwasaki, S. Tanaka, H. Kamio, H. Matsu'ura, Toshiba Corp., Japan  
2055

64.4 A French Version of the MIT-ATIS System: Portability Issues  
2059

64.5 A Bilingual Voyager System  
- J. Glass, D. Goodine, M. Phillips, S. Sakai, S. Seneff, VW. Zue, MIT, USA  
2063

Session 65: Applications  
Time and Place: 16.00 - 17.40 hrs Room B  
Chairperson: H. Mangold, Daimler Benz AG, Ulm, Germany

65.1 Proposal and Implementation of a Spoken Word Recognizer Using Utterance Normalization and Multiple Templates on a Single VLSI Chip  
- H. Fujisaki(*), S. Ohno(*), H. Nasuno(*), K. Hirose(**), (*) Science University of Tokyo, (**) University of Tokyo, Japan  
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65.2 CASPER: A Speech Interface for the Macintosh  
- R. Strong, Apple Computer Inc., USA  
2073
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<td>C. Ellermann, S. van Even, C. Huang, L. Manganaro, Dragon Systems Inc., USA</td>
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**Time and Place:** 16.00 - 17.40 hrs  Room C

**Chairperson:** J. Blauert, Ruhr University of Bochum, Germany

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**Time and Place:** 16.00 - 17.40 hrs  Room D

**Chairperson:** H. Höge, Siemens, Munich, Germany

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Chairperson: G. Kokkinakis, University of Patras, Greece

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   - S. Matsunaga(*), T. Yamada(*), K. Shikano(**), (*) ATR Interpreting
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   - M.-Y. Hwang, F. Alleva, X. Huang, Carnegie Mellon University, USA

68.4 CMU’s Robust Spoken Language Understanding System
   - S. Issar, W. Ward, Carnegie Mellon University, USA

68.5 J-Summit: Japanese Spontaneous Speech Recognition
   - S. Sakai, M. Phillips, MIT, USA

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Chairperson: M. Rossi, CNRS Université de Provence, France

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69.3 Comparing Synthesizers for Name and Address Provision: Field Trial Results
   - S. Basson, D. Yashchin, A. Kalyanswamy, K. Silverman, NYNEX Science and
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69.4 Synthesiser Intelligibility in the Context of a Name-and-Address Information Service
   - K. Silverman, A. Kalyanswamy, J. Silverman, S. Basson, D. Yashchin,
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   - R. Marzi, Technical University of Berlin, Germany

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70.31 The Effect of Utterance Length and Content on Speaker-Verifier Performance
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