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TR-I-0373

Supplement to Collected Papers of the Speech Processing Department

1993

ATR Interpreting Telephony Research Laboratories

March, 1993

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This volume is the supplement to the series of the complete collection of technical papers from the Speech Processing Department, ATR Interpreting Telephony Research Laboratories:

volume	period	id	#papers	#pages
I	from April 1986 through December 1988	TR-I-0010	32	181
II	from November 1987 through December 1988	TR-I-0065	88	449
III	from January through October 1989	TR-I-0115	69	282
IV	from November 1989 through December 1990	TR-I-0230	113	574
V	from January through December 1991	TR-I-0261	122	633
VI	from January 1992 through December 1992	TR-I-0313	140	629
VII	from January 1993 through March 1993	TR-I-0372	27	126
Suppl	submitted in 1993	TR-I-0373	23	285

ATR Interpreting Telephony Research Laboratories finishes its seven-year research project in interpreting telephony at the end of March 1993. This volume is the supplement of collected papers which have been submitted during the current project and are still in the reviewing process.

The research areas of the Speech Processing Department include:

1. Large Vocabulary Continuous Speech Recognition
   (a) Hidden Markov Models
   (b) Neural Network Approaches
   (c) Feature-Based Approaches
2. Speaker Adaptation and Noise-Robust Speech Recognition
3. Language Source Modeling
4. Speech Synthesis by Rule
5. Voice Conversion
6. Speech Database

The following pages are technical publications, to be published or coauthored by the researchers of the Speech Processing Department, ATR Interpreting Telephony Research Laboratories, and still in the reviewing process.

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1 Technical Publications from January through March 1993

The following pages are technical publications, to be published or coauthored by the researchers of the Speech Processing Department, ATR Interpreting Telephony Research Laboratories, and still in the reviewing process.

Abbreviations used in the list are as follows:

- ASJ: the Acoustical Society of Japan
- IEICE: the Institute of Electronics, Information and Communications Engineers (Japan)
- ICASSP: IEEE International Conference on Acoustics, Speech, and Signal Processing
- Eurospeech: Europian Conference on Speech Communications

List of Publications, 1993

◆

W. N. Campbell: "Predicting Segmental Distribution for Accommodation within a Syllable-level Timing Framework," Proc. of Eurospeech93

◆

福沢 圭二, 加藤 喜永, 杉山 雅英: "FPM-LR による不特定話者連続音声認識," 電子情報通信学会 - 論文誌, D-II, 投稿中 (1993 年 2 月 原稿提出).
Keiji Fukuzawa, Yoshinaga Kato, Masahide Sugiyama: "Speaker-independent Continous Speech Recognition Using FPM-LRs," Trans of IEICE, D-II, contributed (1993).

◆

磯谷 亮輔, 嵯峨山 茂樹: "Speech Recognition Using Particle N-grams and Content-Word N-grams," Proc. of Eurospeech93

◆

岩橋 直人, 匂坂 芳典: "Duration Modeling with Multiple Spilt Regression," Proc. of Eurospeech93

N. Iwahashi, S. Sagisaka: "Speech Segment Network Approach for an Optimal Synthesis Unit Set," Proc. of Speech Communication

岩橋 直人, 海木 延佳, 匂坂 芳典: "Concatenative Speech Synthesis with Minimum Distortion Criteria," 電子情報通信学会英文論文誌

◆

Kosaka92SP09 60

小坂 哲夫, 鷹見 淳一, 嵯峨山 茂樹: “話者混合逐次状態分割法による不特定話者音声認識と話者適応,” 電子情報通信学会論文誌 D-II, 投稿中.

Kosaka93Eurpspeech 73

Tetsuo Kosaka, Edward Willems, Jun-Ichi Takami, Shigeki Sagayama: “A Dynamic Approach to Speaker Adaptation of Hidden markov Networks for Speech Recognition,” Eurospeech93 proposal. 投稿中.

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Lucke93Eurospeech 74

H. Lucke: “Inference of Stochastic Context-free Grammar Rules from Example Data Using the Theory of Bayesian Belief Propagation,” Eurospeech93 proposal. 投稿中.

◆

Miyazawa93FULLPAPER 75

宮沢 康永, 大倉 計美, 嵯峨山 茂樹 : “全音素エルゴディック HMM を用いた教師なし話者適応,” 電子情報通信学会論文誌 D-II, 投稿中.

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Muarakami93Eurospeech 82

Jin'ichi Murakami, Hiroaki Yamamoto, Shigeki Sagayama: “ On the Automatic Acquisition of Stochastic Network Grammar using Ergodic HMM,” (1993)

Murakami93IEICEtransa 83

村上 仁一, 荒木 哲郎, 池原 悟, 嵯峨山 茂樹: “音声におけるアクセント情報の持つ情報量の考察,” 電子情報通信学会論文誌

Murakami93IEICEtransb 121

村上仁一, 嵯峨山茂樹: “自由発話音声認識における音響的および言語的な問題点の検討,” 電子情報通信学会論文誌

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