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Supplement to Collected Papers of
the Speech Processing Department

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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:

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I	from April 1986 through December 1988	TR-I-0010	32	181
II	from November 1987 through December 1988	TR-I-0065	88	449
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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: European Conference on Speech Communications

List of Publications, 1993



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