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# Research Activities of the Speech Processing Department

November 1989 through December 1990

ATR Interpreting Telephony Research Laboratories

October, 1991

ATR Interpreting Telephony Research Laboratories

Abstract

This volume is a complete collection of technical papers from Speech Processing Department, ATR Interpreting Telephony Research laboratories, published from November, 1989 through December, 1990. The research areas of the Speech Processing Department include:

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

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# Preface: The Overview of the Research Activity of Speech Processing Department

## 1 Introduction

At ATR Interpreting Telephony Research Laboratories, speech research is being intensively pursued to realize automatic telephone interpretation which allows speech communication between telephone users speaking different languages. Automatic telephone interpretation, or *speech translation* according to more general terminology, is one of the technologies most eagerly awaited by people all over the world.

The utterance of one party will be translated into another language and heard by another. The basic constituent technologies are speech recognition, language translation, and speech synthesis, which are not only essential for realization of speech translation but also widely applicable in various other areas. Although this concept of automatic telephone interpretation is rather new and research is still in its early stages, it is rapidly attracting the interest of researchers and research organizations throughout the world, and is expected to become an area of common research interest.

## 2 Automatic Telephone Interpretation

### 2.1 History and general concept

ATR Interpreting Telephony Research Laboratories, a subsidiary of Advanced Telecommunications Research International, was established in 1986 to initiate basic research for telephone interpretation.

The basic components of an automatic telephone interpretation ("speech translation" in more general terminology) system are speech recognition, language translation, and speech synthesis. In more realistic applications, speaker adaptation for input, and speaker conversion for output, are also required.

Simple combination of these elements of current technology, however, does not give a satisfactory speech translation system. What is required of these components is somewhat different from a conventional application. The major differences are as follows.

- Particularly high performance both in speech recognition and translation is required. Unlike conventional language translation, there cannot be any "pre/post-editing", since both input and output are speech.
- Real-time operation is a strong requirement.
- Spoken language is quite different from written language. Though sentences are generally short and their structures are not particularly complicated in spoken language, spoken dialogues include elliptic and anaphoric expressions. They may also include many syntactically ill-formed expressions.

Thus, speech translation is a new area which requires significant innovations in these constituent technologies.

Although the ultimate goal of the telephone interpretation system will be universal dialogue in an unlimited domain, at present, the goal should be a more feasible one such as a system which is limited to specific, task oriented areas. ATR Interpreting Telephony Research Laboratories has selected an international conference registration task as a constrained task domain where the dialogue will be goal-directed and the vocabulary is expected to be limited to approximately 1,500 words.

### 2.2 ATR's Experimental Speech Translation System

Fig.1 illustrates the outline of the current experimental Japanese-English speech translation system at ATR Interpreting Telephony Research Laboratories. The input sentence speech uttered phrase by phrase is recognized with speaker adaptation capability under linguistic constraints and converted into several word sequence candidates. The number of candidates is narrowed down by dependency analysis and finally uniquely determined by linguistic analysis, followed by language translation consisting of linguistic analysis, Japanese-to-English language transfer, and English sentence generation. The final stage is to convert English sentences into speech with the original speaker's characteristics although this stage is not integrated in the system. Another direction, English-Japanese conversion, can be considered in the same way. As it seems inefficient to split our research efforts into Japanese and English aspects, our research topics mainly cover Japanese speech recognition, Japanese-English translation, and Japanese speech synthesis. Basically, the same approaches can be applied to the remaining parts.

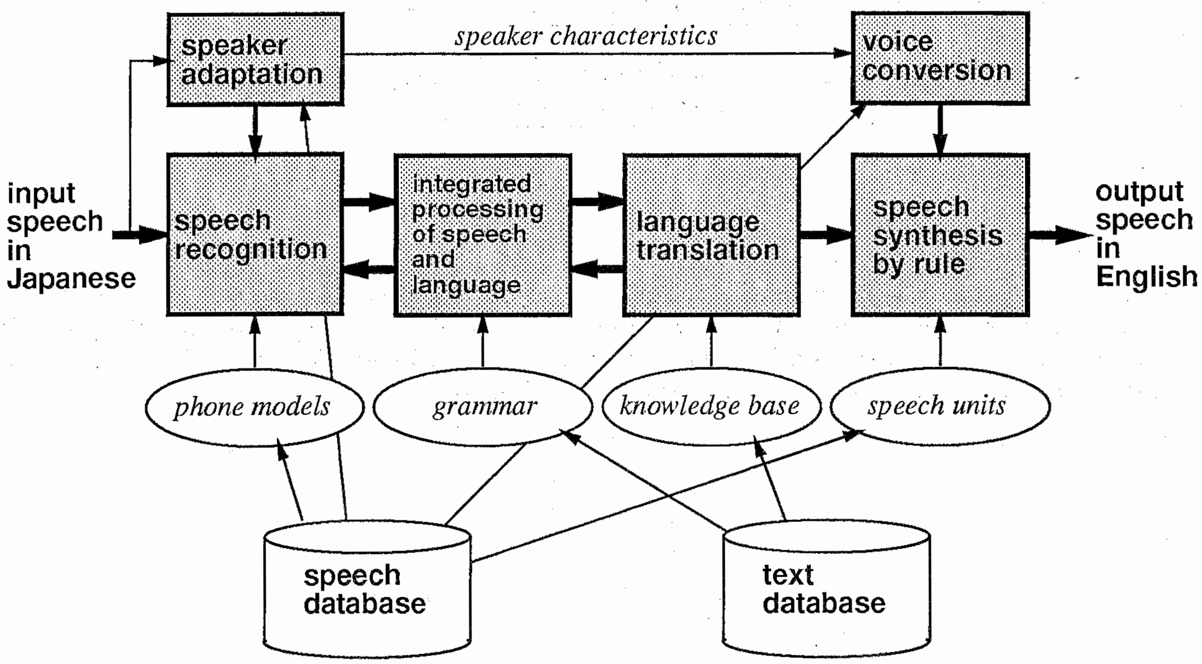



Figure 1: SL-Trans: an experimental speech translation system (Japanese-English)

# 3 Speech Research at ATR Interpreting Telephony Research Laboratories

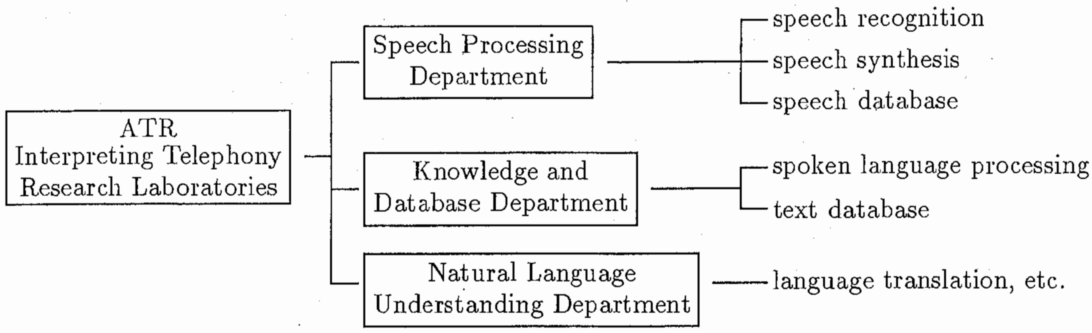
## 3.1 Research project

ATR Interpreting Telephony Research Laboratories (Akira Kurematsu, president) consists of three departments, Natural Language Understanding Department, Knowledge and Database Department, and Speech Processing Department. The laboratory was founded in April of 1986 with the support of the Japan Key Technology Center, ATR International, NTT, KDD, NHK, and other Japanese enterprises. The main target of our institute is fundamental research into speech and language processing, and integration of speech and language processing technologies to demonstrate the feasibility of an automatic telephone interpretation system. As of 1990, the laboratory had a research budget of 2.6 billion yen, a research staff of about 40, and 235 academic publications including 70 papers at international conferences.

## 3.2 Organization

ATR Interpreting Telephony Research Laboratories (Akira Kurematsu, President) consists of three departments, i.e., Speech Processing Department (Shigeki Sagayama, Head), Knowledge and Database Department (Tsuyoshi Morimoto, Head), and Natural Language Understanding Department (Hitoshi Iida, Head).

The outline of the organization is shown below.




## 4 Research Staff

The research staff is mainly composed of members seconded from research institutes, laboratories, and industrial companies which support ATR, and visiting scientists. The following table shows the past and current members of our research staff staying more than 6 months. In addition to the list, we have/had a number of intern students and working students from universities.

Table 1: Research Staff of the Speech Processing Department

| name & title | position | period |
|---|---|---|
| Akira Kurematsu, Dr | President | 1986.4 – |
| Kiyohiro Shikano, Dr | Department Head | 1986.6 –1990.1 |
| Shigeki Sagayama | Department Head | 1990.2– |
| Hisao Kuwabara, Dr | Supervisor | 1986.10 – 1989.7 |
| Yoshinori Sagisaka, Dr | Superviser | 1986.4 – |
| Tetsuo Umeda | Supervisor | 1989.7 – 1990.6 |
| Masahide Sugiyama, Dr | Supervisor | 1990.2– |
| Takeshi Kawabata, Dr | Senior Researcher | 1986.9 –1990.2 |
| Shin-ichi Tamura | Senior Researcher | 1986.9 –1990.2 |
| Hidefumi Sawai, Dr | Senior Researcher | 1988.4 –1991.3 |
| Satoshi Nakamura | Researcher | 1986.9 – 1989.8 |
| Kaichiro Hatazaki | Researcher | 1986.12 – 1989.3 |
| Kazuya Takeda | Researcher | 1986.8 – 1990.2 |
| Masanori Miyatake | Researcher | 1986.9 – 1989.8 |
| Toshiyuki Hanazawa | Researcher | 1987.3 –1990.2 |
| Masanobu Abe | Researcher | 1987.4 – 1991.2 |
| Katsuteru Maruyama | Researcher | 1987.3 –1990.2 |
| Masami Nakamura | Researcher | 1987.9 – 1990.8 |
| Katsuo Abe | Researcher | 1987.3 –1990.2 |
| Yasuhiro Komori | Researcher | 1988.9 – |
| Hiroaki Hattori | Researcher | 1989.5 – |
| Kazumi Ohkura | Researcher | 1989.9 – |
| Nobuyoshi Kaiki | Researcher | 1989.11 – |
| Jun-ichi Takami | Researcher | 1989.11 – |
| Kazuki Katagishi, Dr | Researcher | 1990.2 – |
| Akito Nagai | Researcher | 1990.3 – |
| Keiji Fukuzawa | Researcher | 1990.4 – |
| Katsuhiko Mimura | Researcher | 1990.5 – |
| Shingo Fujiwara | Researcher | 1990.5 – |
| Naoto Iwahashi | Researcher | 1990.10 – |
| Jin-ichi Murakami | Researcher | 1991.3 – |
| Yoshinaga Kato | Researcher | 1991.4 – |
| Kouichi Yamaguchi | Researcher | 1991.5 – |
| Yasunaga Miyazawa | Researcher | 1991.7 – |
| Tetsuo Kosaka | Researcher | 1991.9 – |
| Alex Waibel, Dr | Invited Researcher | 1987.5 – 1988.8, 1989.6 – 1989.9 |
| William Poser, Dr | Invited Researcher | 1988.9 – 1989.2 |
| Alain de Cheveigné, Dr | Invited Researcher | 1989.2 – 1990.2 |
| Wilhelm N. Campbell | Visiting Researcher | 1990.8 – |
| David Rainton, Dr | Visiting Researcher | 1990.4 – |
| Harald Singer | Visiting Researcher | 1991.3 – |
| Kouichi Murayama | Engineer | 1987.9 – 1988.8 |
| Takaharu Tanaka | Engineer | 1988.9 – 1989.8 |
| Kouji Kitagaito | Engineer | 1989.9 – 1990.8 |
| Shinobu Araki | Engineer | 1990.9 – 1991.8 |
| Yoshinori Ono | Engineer | 1991.9 – |
| H. Segot | Student | 1987.3 – 1987.11 |
| Patrick G. Haffner | Student | 1988.3 – 1988.11 |
| Jean-Claude Dang | Student | 1989.3 – 1989.11 |
| Yasuhiro Minami | Student | 1989.4 – 1990.3 |
| Alain Biem | Student | 1990.8 – 1991.1 |
| Romain Brunias | Student | 1991.8 – |
| Richard Lengagne | Student | 1991.8 – |

# 5 Research Facilities

In the year of 1990, we renovated our research facility. Each of research staff is provided a DECstation5000 (speed 24MIPS) or DECstation3100 (speed 16MIPS), with 24 - 32 Mbyte memory and a 600M - 3G byte disk, which is connected to an Ethernet. We introduced 2 DECsystem5000s (currently, 3 systems) with over 30G byte disk memory which are used as file servers containing a large scale speech database and are accessed from workstations through the Ethernet. Some workstations are equiped with speech input/output devices. Older computer systems including VAX8600, a VAX8700, a VAX8800 and an Alliant 9800 (eight-CPU array processor) are also connected to the Ethernet and accessed from workstations.

# 6 Speech Recognition Research

## 6.1 Phone Modeling

Reliable phoneme recognition and segmentation algorithms have been investigated leading to considerable improvements over conventional approaches. We have been pursuing three approaches: feature-based approach, hidden Markov model approach, and neural network approach. These improvements resulted in the successful implementation of a continuous speech recognition system such as HMM-LR, by combining HMM phoneme models with the generalized LR parsing algorithm. Hardware implementation has also been done.

### 6.1.1 Hidden Markov models

Hidden Markov models have been intensively studied for representing phone models in continuous speech recognition.

At the early stage of HMM research, the number of states, tied probabilities of transition and output, probability smoothing techniques, initial probability settings and duration control techniques were studied, based on the task of phoneme recognition using our large vocabulary speech database. Separate VQ (multiple codebook) and fuzzy VQ techniques improved the recognition rate from 86.5% without them, up to 95.7%. Major research topics cover the following items.

- discrete HMM phone models
- HMM phone models based on fuzzy vector quantization
- continuous output probability density HMMs
- allophonic (context dependent) HMM phoneme models
- concatenated training
- segment-based HMM
- HMM - neural network hybrid

HMM phone models are used in HMM-LR where the HMM phoneme models are combined with a generalized LR parser (a language source model) to efficiently recognize a Japanese phrase input, word spotting in continuous speech, phonetic typewriter, and English word recognition.

A recent study on context-dependent phone models represented by single Gaussian output probabilities has attained a speaker-dependent phoneme recognition rate of 98.3% for phonemes in word utterances. For phonemes in phrase utterances, they performed even better than the Gaussian mixture models.

### 6.1.2 Neural network approaches

ATR started neural network research in its early period and intensively investigated its application to speech recognition. The TDNN (time-delay neural network) is one of our developments. Major topics in this area cover:

- time delay neural network (TDNN)
- fast back-propagation algorithm
- neural network work bench
- phoneme spotting
- phoneme filter approach
- robustness across differences of speaking styles

( neighbor integration, KNIT ($k$-nearest neighbor interpolative training), pairwise discriminant TDNNs, fuzzy training )

- new architectures
- Boltzmann machine approach

TDNN attained a phoneme recognition rate of 98.6% for a /b/, /d/, /g/ task, and 96.7% for a full consonant task. Phoneme spotting by TDNN aimed at determining a continuous speech recognition approach by neural networks has been carried out and resulted a phoneme spotting rate of 98%.

Efforts to speed up the back-propagation algorithm resulted in a 1,000-fold speed-up of the TDNN. This speed-up and the use of a mini-supercomputer have made it possible to challenge larger scale neural networks.

Robustness across different speaking styles is one of most important issues in neural networks for speech recognition. Although neural networks show very high phoneme recognition performance for word speech when they have been trained using word utterances, the performance degrades drastically for different speaking styles such as continuous speech. To solve this problem, intensive studies have been made and new schemes of neural network training (e.g., KNIT method, fuzzy training) and new architectures (e.g., pairwise discriminant TDNNs) have been developed, and they perform much better than the original neural network approaches.

Other neural network approaches such as a deterministic Boltzmann machine and a neural prediction model are also being studied. A constructive neural network (CNN) to recognize words in bottom-up fashion using a TDNN phoneme spotting network is also studied.

### 6.1.3 Feature-based approaches

A feature-based approach is also taken for utilizing expert knowledge in spectrogram reading. Major topics in this area are:

- spectrum reading expert system
- hybrid system of knowledge based segmentation and neural network phoneme recognition
- feature based segmentation combined with HMM

Phoneme segmentation knowledge for consonants is described by ART. A phoneme segmentation expert is integrated with a TDNN neural network for consonant discrimination. The system attains a consonant segmentation rate of 94.5% and a consonant recognition rate of 88.8%. Moreover, spotting of vowels, semi-vowels and syllabic nasal has been studied using a vowel spotting TDNN.

## 6.2 Speaker Adaptation and Noise-robust Speech Recognition

Aiming at a general preprocessor for speaker normalization, speaker adaptation research is studied.

- codebook mapping for speaker adaptation
- supplemented codebook mapping
- speaker weighted HMM training
- HMM model modification
- speaker mapping by a neural network
- speaker independent phoneme recognition by neural networks

Codebook mapping is a technique to find a mapping between a pair of vector quantization codebooks of different speakers using fuzzy vector quantization and DTW(dynamic time warping)-based word speech matching. This HMM speaker adaptation algorithm attains a phrase recognition rate of 78.6%, where the phrase recognition rates of speaker independence and speaker dependence are 59.6% and 88.4%, respectively. A supplemented HMM phoneme model training approach has been initiated to cope with speaker coarticulation variations. In the codebook mapping algorithm, a speaker adaptation approach has been undertaken using the concept of fuzzy vector quantization and spectrum mapping.

Recently, instead of such a frame-wise approach, adaptation of continuous output density HMM models has been found to be very successful and has attained a phoneme recognition performance higher than 90% after speaker adaptation.

## 6.3 Language Source Modeling

Phone models are connected under the control of a language model to recognize continuous speech. We are taking both syntactic and stochastic approaches to language modeling. Some of our activities in this area are as follows.

- generalized predictive LR parser
- HMM-LR continuous speech recognition
- phoneme context dependent LR parsing algorithm
- word/syllable/phoneme trigram models
- phonetic typewriter using phoneme sequence statistics
- word spotting based on HMM phone models
- two-stage LR parser for inter-*bunsetsu* syntax
- unknown word detection in HMM-LR
- hardware implementation of the HMM-LR scheme
- the use of pitch patterns for phrase boundary detection
- NETgram: neural network approach for word category prediction
- TDNN-LR continuous speech recognition
- PD-TDNN-LR, fuzzy-TDNN-LR continuous speech recognition schemes

A generalized LR parser has been combined with HMM phone models to search for the optimal grammatical path. Fig.2 shows an outline of the HMM-LR scheme for continuous speech recognition which

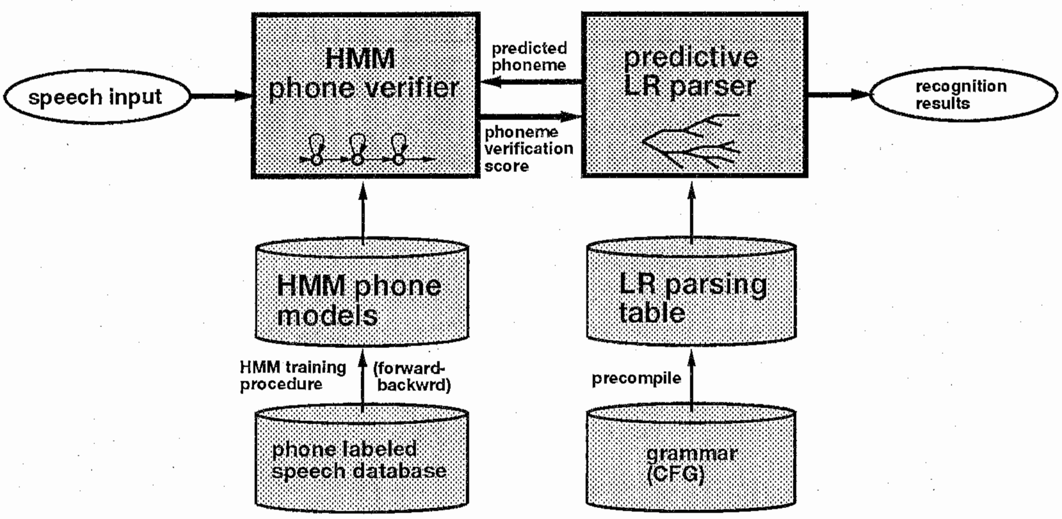



Figure 2: HMM-LR continuous speech recognition system

attained a phrase recognition rate of 88% for a task with a phoneme perplexity of 5.9 with vocabulary of 1,000 words. The cooccurrence of context-free rewriting rules is used to produce a better language source model. The LR parser is also successfully combined with the Sphinx system at Carnegie-Mellon University (CMU) under a research collaboration between ATR and CMU. A recent enhancement of the LR parser has attained a word accuracy of 97.5% and a sentence accuracy of 91.2%.

An experiment to recognize continuous utterances using TDNN phoneme spotting results has been tried by summing up the phoneme spotting output by means of a DTW algorithm with an LR parser. Recently, it is connected to both a PD-TDNN and a fuzzy trained TDNN and is attaining better results.

The NETgram approach for word category prediction using a neural network is also proposed and proved to be efficient when the training data size is relatively small compared with $n$-gram (bigram, trigram, etc.) approaches.

# 7 Speech Synthesis Research

## 7.1 Speech Synthesis by Rule

A high quality speech synthesis system by rule should be realized as an important part of the interpreting telephony system. A synthesis system based on flexible synthesis units has been developed. Moreover, the speech synthesis algorithm is evaluated to realize a better synthesis system. A unit selection algorithm from a large-scale speech database and a unit concatenation algorithm have been studied. The prosody control algorithm has been also studied. Also, rules between speaking styles and prosodic control have been studied to realize various kinds of synthesized voices. Our research interests are:

- non-uniform synthesis units
- optimal selection algorithm of units from a database for speech synthesis
- prosody control models using neural networks
- cepstrum-based speech waveform generation
- optimal segmental duration control using a statistical method
- optimal amplitude control using a statistical method
- analysis of segmental durations in English speech
- automatic phoneme segmentation for speech synthesis using HMM

A prototype speech synthesis-by-rule system has been implemented.

## 7.2 Voice Conversion

Voice conversion from one speaker to another is an important aspect in realizing an automatic telephone interpreting system. Formerly, a voice conversion algorithm based on codebook mapping by vector quantization was developed. Recently, our research interest has focus on two aspects. One is the application to cross language voice conversion, where speech data from speaker $S_1$ of source language $L_1$ and from speaker $S_2$ of destination language $L_2$ are given, and the latter is to be converted into speaker $S_1$'s voice. The other is a segment-based approach to voice conversion, where the input speech is segmented using HMM phone models and segment-wisely converted into anther speaker's voice. Our research subjects cover:

- frame-wise voice conversion using VQ codebook mapping
- cross-language voice conversion
- segment-based voice conversion using HMM

# 8 Speech Database

A large-scale speech database with phonetic transcription, mainly for research in speech recognition and speech synthesis, has been developed with considerable support from other speech research laboratories. This database includes three categories as follows.

(1) set-A: Large-scale database consisting of 5,240 Japanese common words, 216 phonetically-ballanced words, 101 Japanese syllables, 25 numerics, 35 alphabetical letters, 9 foreign words, and 115 sentences in a conference registration task with 3 different speaaking styles, uttered by 20 professional broadcast announcers

(2) set-B: 503 phonetically balanced sentences uttered by 10 professional speakers

(3) set-C: a database of a large number of speakers (mainly for speaker-independent/speaker-adaptive speaker recognition researches), containing 520 words (a subset of 5240 words in set-A), 216 phonetically-ballanced words, 15 numerics, and 150 sentences (a subset of set-B), with an ultimate goal of 200 speakers, half of which will be phonetically labeled.

These are recorded and sampled at 20 kHz, digitized in 16 bits, and phonetically labeled by well-trained workers by hand.

The former two, (1) and (2), have been already completed and are gradually being put to public availability.

The speech database has effectively been used to perform speech research at ATR and other laboratories. Data of several speakers are already available in CD-ROMs. This speech database is now being accepted as the standard in Japan.

# 9 Techical Publications from November 1989 through December 1990

Folowing pages are the list of technical publications from the Speech Processing Department, ATR Interpreting Telephony Research Laboratories, published during the period from November 1989 through December 1990, and reprints of these publications.

Abbreviations used in the list are as follows:

- ASJ: the Acoustical Society of Japan
- IEICE: the Institute of Electronics, Information and Communications Engineers (Japan)

# List of Technical Publications from Speech Processing Department

(November 1989 through December 1990)

**(Bold titles indicate papers written in English)**

| Ref.ID | Title(題名) | Authors(著者) | Journal(掲載誌) | Page |
|---|---|---|---|---|
| Abe,K 89-11 | 音韻環境に応じた音声合成素片の接続方法の検討<br><br>[On the Concatenation of Speech Synthesis Units according to Unit Extraction Context] | 安部　勝雄<br>武田　一哉<br>匂坂　芳典<br>[K. Abe<br>K. Takeda<br>Y. Sagisaka] | 電子情報通信学会技術研究報告 SP89-66, pp.17-22, 1989.<br><br>[IECIE Technical Report, SP89-66, pp.17-22, 1989] | 1 |
| Abe,K 90-2a | 波形重ね合わせ法による合成音の品質について<br><br>[Quality Evaluation for Synthesized Speech Using Wave Overlap Adding] | 安部　勝雄<br>匂坂　芳典<br>桑原　尚夫<br>[K. Abe<br>Y. Sagisaka<br>H. Kuwabara] | ATR Technical Report TR-I-0135 | - |
| Abe,K 90-2b | 音韻環境に応じた音声合成素片の接続方法の検討<br><br>[On the Concatenation of Speech Synthesis Units According to Unit Extraction Context] | 安部　勝雄<br>武田　一哉<br>匂坂　芳典<br>[K. Abe<br>K. Takeda<br>Y. Sagisaka] | ATR Technical Report TR-I-0136 | - |
| Abe,K 90-3 | 規則合成におけるパワー制御の検討<br><br>[A Study on Power Control of Speech Synthesis Units] | 安部　勝雄<br>武田　一哉<br>匂坂　芳典<br>[K. Abe<br>K. Takeda<br>Y. Sagisaka] | 日本音響学会 平成2年度 春季研究発表会講演論文集, 1-4-12, pp.201-202, 1990.<br>[Proc. of ASJ Spring Meeting, 1-4-12, pp.201-202, 1990] | 7 |
| Abe,M 89-5 | **Voice Conversion Through Vector Quantization** | M. Abe<br>H. Kuwabara<br>S. Nakamura<br>K. Shikano | The Journal of the Acoustical Society of Japan, pp.71-76, 1989. | 9 |
| Abe,M 89-11 | **Fundamental Frequency Database with Linguistic and Phonetic Information** | M. Abe<br>Y. Sagisaka<br>H. Kuwabara | The Journal of the Acoustical Society of America, Supplement 1, Vol.86, O8, pp.S36, Fall, 1989. | 15 |
| Abe,M 89-12 | **Cross-Language Voice Conversion** | M. Abe | ATR Technical Report TR-I-0126 | - |
| Abe,M 90-2 | 言語間にわたる声質変換<br><br>[Cross-Language Voice Conversion] | 阿部　匡伸<br><br>[M. Abe] | 電子情報通信学会技術研究報告 SP89-123, pp.31-38, 1990.<br>[IECIE Technical Report, SP89-123, pp.31-38, 1990] | 16 |
| Abe,M 90-3a | 声質変換の研究<br><br>[A Study on Voice Conversion] | 阿部　匡伸<br><br>[M. Abe] | ATRジャーナル, No.7, pp.8-11, 1990.<br>[ATR Journal, No.7, pp.8-11, 1990] | 24 |
| Abe,M 90-3b | **Cross-Language Voice Conversion** | M. Abe | 日本音響学会 平成2年度春季研究発表会講演論文集, 3-4-11, pp.289-290, 1990.<br>[Proc. of ASJ Spring Meeting, 3-4-11, pp.289-290, 1990] | 28 |

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| Ref.ID | Title(題名) | Authors(著者) | Journal(掲載誌) | Page |
|---|---|---|---|---|
| Abe,M 90-4 | **Cross-language Voice Conversion** | M. Abe<br>K. Shikano<br>H. Kuwabara | Proceedings of 1990 International Conference on Acoustics, Speech, and Signal Processing, S6a.14, pp.345-348, Albuquerque, 1990. | 30 |
| Abe,M 90-6 | **Voice Conversion for an Interpreting Telephone** | M. Abe<br>K. Shikano<br>H. Kuwabara | Proceedings of the Tutorial and Research Workshop on Speaker Characterization in Speech Technology, pp.40-45, Edinburgh, 1990. | 34 |
| Abe,M 90-9a | 音声セグメントを変換の単位とする声質変換<br>[A Segment Model Based Approach to Voice Conversion] | 阿部　匡伸<br>嵯峨山　茂樹<br>梅田　哲夫<br>[M. Abe<br>S. Sagayama<br>T. Umeda] | 日本音響学会 平成元年度秋季研究発表会講演論文集, 3-6-11, pp.287-288, 1990.<br>[Proc. of ASJ Fall meeting, 3-6-11, pp.287-288, 1990] | 40 |
| Abe,M 90-9b | 研究用日本語音声データベース利用解説書(連続音声データ編)<br>[Speech Database User's Manual] | 阿部　匡伸<br>匂坂　芳典<br>梅田　哲夫<br>桑原　尚夫<br>[M. Abe<br>Y. Sagisaka<br>T. Umeda<br>H. Kuwabara] | ATR Technical Report TR-1-166 | - |
| Abe,M 90-11 | **Statistical Study on Voice Individuality Conversion across Different Languages** | M. Abe<br>S. Sagayama | Proceedings of 1990 International Conference on Spoken Language Processing, 5.8, pp.157-160. Kobe,1990. | 42 |
| Campbell 90-11a | **Evidence for a Syllable-based Model of Speech Timing** | Nick Campbell | Proceedings of 1990 International Conference on Spoken Language Processing, 1.3, pp.9-12, Kobe, 1990. | 46 |
| Campbell 90-11b | **Duration, Pitch and Diphones in the CSTR TTS System** | Nick Campbell<br>S.D. Isard<br>A.I.C.Monaagham<br>J.Vechoeveen | Proceedings of 1990 International Conference on Spoken Language Processing, 19.16, pp.825-828, Kobe, 1990. | 50 |
| Cheveigné 90-2a | **The MapSignal Remote Speech Editor** | Alain de Cheveigné | ATR Technical Report TR-I-0137 | - |
| Cheveigné 90-2b | **Experiments in Pitch Extraction** | Alain de Cheveigné | ATR Technical Report TR-I-0138 | - |
| Cheveigné 90-2c | **Auditory Nerve Fiber Spike Generation Model** | Alain de Cheveigné | ATR Technical Report TR-I-0139 | - |

# List of Technical Publications from Speech Processing Department

(November 1989 through December 1990)

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| Ref.ID | Title(題名) | Authors(著者) | Journal(掲載誌) | Page |
|---|---|---|---|---|
| Dang 90-1a | シフト不変型決定論的ボルツマンマシンによる音声認識<br><br>[Shift-invariant Deterministic Boltzmann Machines for Phoneme Recognition] | Jean-Claude Dang<br>田村　震一<br>沢井　秀文<br>[S. Tamura<br>H. Sawai] | 電子情報通信学会技術研究報告 SP89-98, pp.1-6.1990.<br><br>[IECIE Technical Report, SP89-98, pp.1-6, 1990] | 54 |
| Dang 90-1b | シフト不変型決定論的ボルツマンマシンによる音声認識<br><br>[Shift-invariant diterministic Boltzmann Machines for Phoneme Recognition] | Jean-Claude Dang<br>田村　震一<br>沢井　秀文<br>[S. Tamura<br>H. Sawai] | ATR Technical Report TR-I-0130 | - |
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