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Research Activities of the Speech Processing Department

November 1987 through December 1988

ATR Interpreting Telephony Research Laboratories

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ATR Interpreting Telephony Research Laboratories

Research Activities of

the Speech Processing Department

November 1987 through December 1988

Abstract:

The research areas of the Speech Processing Department now include:

(A) Phoneme Recognition and Segmentation.

(1) Feature-Based Approaches.

(2) Hidden Markov Models.

(3) Neural Network Approaches.

(B) Speaker Adaptation.

(1) Speaker Adaptation by Vector Quantization.

(C) Language Models.

(1) Word Trigram Models.

(2) Word Prediction by Neural Networks.

(D) Noise Reduction.

(1) Noise Reduction by Neural Networks.

(E) Speech Synthesis by Rule.

(1) High Quality Japanese Speech Synthesis by Rule.

(2) Modal Information and Prosodies.

(F) Voice Conversion.

(1) Voice Conversion by Parameter Mapping through Vector Quantization.

(2) Personal Characteristics Analysis by Analysis-by-Synthesis.

(3) Speech Analysis and Synthesis Algorithm.

(G) Phonetics

(1) Analysis of Allophones and Coarticulation.

(H) Speech Database

(1) Japanese Large-Scale Speech Database with Phonetic Transcriptions.

(2) Speech Database Retrieval System.

(3) Speech Workbench.

For additional information, contact:

ATR Interpreting Telephony Research Laboratories

Inuidani, Seika-cho, Souraku-gun, Kyoto 619-02, JAPAN

Telephone: +81−7749−5−1311, Telefax: +81−7749−5−1308

E-mail: atr-la.atr.junet ! shikano @ uunet.UU.NET (from USA arpanet).

Contents

1. General

The necessity for automatic telephone interpretation research is described in brief. We introduce the Speech Processing Department of the ATR Interpreting Telephony Research Laboratories, and provide an overview of its research activities from November 1987 to December 1988.

1.1. Automatic Telephone Interpretation Project [Kurematsu-88-01, 88-11]

The Automatic Telephone Interpretation system is a facility which enables a person speaking in one language to communicate readily by telephone with someone speaking another language. It does so by automatically and simultaneously transforming the dialogue from the speaker's to the listener's language. At least three constituent technologies are necessary for such a system: speech recognition, machine translation and speech synthesis.

Since this is a new concept, a number of feasibility studies and evaluations must still be made. A high degree of performance from each of the constituent technologies and system user friendliness are also essential. According to a feasibility study report published in 1985 by the Japanese Ministry of Posts and Telecommunications, realizing a system like this will require at least fifteen years. One more important requirement of this ambitious project is the ongoing cooperation of research institutions in Japan and abroad.

1.2. Speech Processing Department

The ATR Interpreting Telephony Research Laboratories, Dr. Akira Kurematsu, president, have three departments. They are the Natural Language Understanding Department, the Knowledge and Database Department and the Speech Processing Department. At least three basic technologies, which include speech recognition, machine translation and speech synthesis, are necessary for such an interpreting telephony system. Moreover, integrated research into these technologies is also very important. We propose an interpreting telephony model shown in Figure 1-1. In this model, language processing is split into a language source modeling stage and a language analysis stage. Main targets of our research laboratories are 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.These laboratories were founded in April of 1986 with the support of the Japan Key Technology Center, ATR International, NTT, KDD, NHK and other Japanese enterprises.

The Speech Processing Department, headed by Dr. Kiyohiro Shikano, is one of this endeavor. The research areas of the Speech Processing Department are speech recognition , speech synthesis and speech databases. These research areas are aimed at advanced technologies capable of recognizing continuous speech from any speaker and synthesizing high quality speech with the characteristics of various speakers. These technologies, which are indispensable to realize an

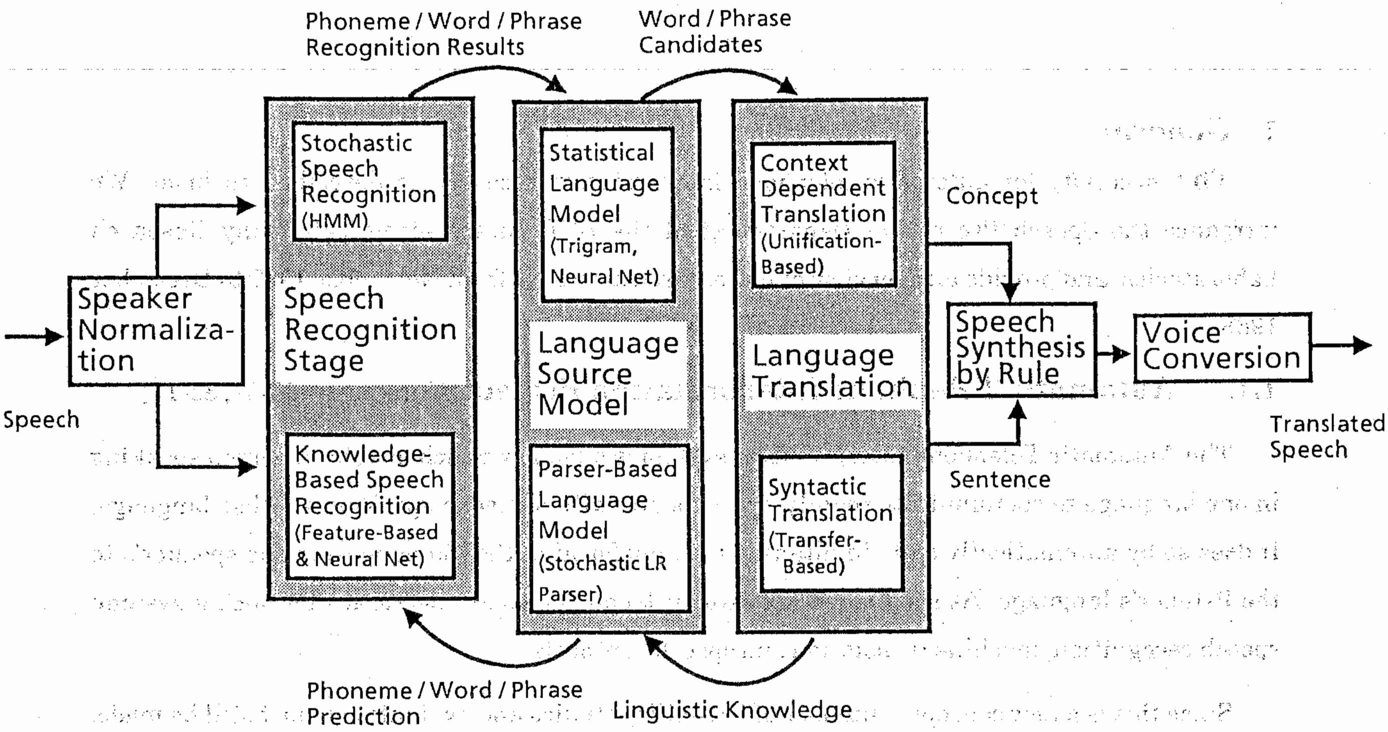

Figure 1-1. Proposed Interpreting Telephony Experimental System.

interpreting telephony system, must be considerably improved by investigating these fields and exploring inovative techniques and break through technologies.

The Speech Processing Department is comprised of two major groups: a speech recognition group and a speech synthesis and phonetics group. The speech recognition group is mainly directed by Drs. K. Shikano, A. Waibel, T. Kawabata and H.Sawai. The speech synthesis and phonetics group is mainly directed by Drs. H. Kuwabara and Y. Sagisaka.

As of December, 1988, efforts aimed at speaker-dependent phoneme recognition and speaker-independent phoneme segmentation have resulted in dramatically improved phoneme recognition performance and continuous speech recognition system implementation. We are still pursuing three approaches: (1) *Feature-Based* approach, especially for phoneme segmentation, (2) *Hidden Markov Model* approach, and (3) *Neural Network* approach. Continuous speech recognition and word spotting efforts have also continued. For speaker-independent speech recognition, a speaker adaptation approach has been undertaken using the concept of *Fuzzy Vector Quantization and Spectrum Mapping*. In the area of language models, the *N-Gram* approach using Neural Networks and a language source modeling approach using the *generalized LR* parser have been followed.

Analysis efforts for the exploration of *Allophones and Coarticulation* phenomena have been undertaken utilizing a large-scale speech database. In order to realize a high quality rule-based speech synthesis system, we are working in four areas, such as a speech synthesis by rule based on *Multiple Speech Units* from a phoneme-labelled speech database, high quality speech analysis and synthesis algorithms, voice conversion algorithms from one speaker to another based on

Vector Quantization and Mapping, and a relationship analysis of *Modal information and Prosodics*. Moreover, non-parametric noise reduction algorithms using *Neural Networks* have been investigated, and cross-linguistic voice conversion from MITALK to a Japanese speaker has just been initiated.

A large-scale speech database with phonetic transcriptions, mainly for research in speech recognition and speech synthesis, has been developed with considerable support from other speech research laboratories. A speech workbench and a database retrieval system have been implemented on a micro Vax station in order to provide researchers with a tool for easy access and manipulation of speech data.

2. Research Activities [Speech-88-12]

Research activities in the Speech Processing Department from November, 1987, through December, 1988, are summarized below, and related technical publications are quoted.

2.1. Phoneme Recognition and Segmentation

Reliable phoneme recognition and segmentation algorithms have been studied leading to considerable improvements over conventional approaches. We have been pursuing three approaches: a *Feature-Based* approach, a *Hidden Markov Model* approach, and a *Neural Network* approach. These improvements resulted in the successful implementation of a continuous speech recognition system such as *HMM-LR* by combining HMM with the generalized LR parsing algorithm.

2.1.1. Feature-Based Approach [Hatazaki, Komori, Kawabata, Tamura, Murayama, Tanaka]

A spectrogram reading seminar taught by Prof. Victor Zue was held in February of 1986. Since that time, the spectrogram reading session has been convened on a weekly basis. Many researchers in our department can now read phonemes of Japanese utterances with relatively high accuracy.

In order to clarify the differences of spectrogram reading features between English and Japanese, statistical analysis has been carried out based on the large vocabulary speech database with phonetic transcriptions.

In particular, phoneme boundaries can be identified correctly in Japanese utterances using expert's knowledge. Now we are developing an expert system for phoneme segmentation using the expert tool, *ART*. A Symbolics lisp machine for running *ART* and a micro Vax station for feature extraction of input speech are connected and used to implement a phoneme segmentation expert system. The phoneme segmentation knowledge for consonants, excluding contracted consonants, has been described by ART. The phoneme segmentation expert is integrated with a *TDNN* neural networks for consonant discrimination.

Technical Publications: [Hatazaki-88-08, 88-01, Kawabata-87-12], [Progress will be presented at ICASSP89]

2.1.2. Hidden Markov Models [Hanazawa, Kawabata, Maruyama, Kita, Shikano]

Hidden Markov models have been studied to determine how to best use models in the training stage on the forward-backward algorithm. The number of states, tied probabilities of transition and output, parameter smoothing techniques, initial probability settings and duration control techniques were studied based on the task of phoneme recognition in our large vocabulary speech database described in section **2.8.1**. After that the HMM phoneme models have been improved by introducing *Separate VQ* (multiple codebooks) and *Fuzzy VQ* techniques. These improvements resulted in a 7.5% phoneme recognition rate improvement, from 86.5% to 94.0%. The HMM phoneme model approach is also applied to English word recognition.

The Hidden Markov models based on phoneme units have now also been applied successfully to word spotting in continuous speech. Moreover, the HMM phoneme models are combined with a generalized LR parser to efficiently recognize a Japanese phrase input, where the LR parser can predict phoneme candidates to the HMM phoneme models. This system is called *HMM-LR*. The HMM-LR can recognize Japanese phrase input with a phrase recognition accuracy of 83%.

Technical Publications: [Hanazawa-88-11, 87-12, Kawabata-88-11, 88-06, Kita-88-10b], [The HMM-LR and the continuous speech recognition system based on word spotting will be presented at ICASSP89]

2.1.3. Neural Network Approaches [Waibel, Sawai, Miyatake, Haffner, Shikano]

Using Neural Networks, phoneme recognition in continuous utterances has been investigated. A time-delay neural network (*TDNN*) now achieves high phoneme recognition rates for the task of speaker-dependent discrimination not only among the voiced consonants, /b/,/d/, and /g/ but also for full consonants. All consonants were extracted from the phonetically labelled large vocabulary database , i.e., from 5240 common Japanese words spoken by three speakers. The TDNN attains a 98.6% phoneme recognition rate for the /b/,/d/, and /g/ task, and 96.7% for the full consonant task. Phoneme spotting by TDNN aimed at determining a continuous speech recognition approach by neural networks has been initiated.

Efforts to speed up the back-propagation algorithm have resulted in a 600 times speedup of the TDNN. This speedup and the use of an Alliant mini-supercomputer make it possible to challenge larger scale neural networks.

Technical Publications: [Sawai-88-12, Waibel-87-12, 88-04a, 88-04b, 88-05, 88-10, 88-11a, 88-11b], [Consonant recognition networks and the phoneme spotting based on TDNN will be presented at ICASSP89], [Related ATR technical reports are also available on request]

2.2. Speaker Adaptation

Aiming at a general preprocessor for speaker normalization, speaker adaptation research using vector quantization techniques has been studied. Discrete spectrum space representation by vector quantization makes it possible to realize sophisticated speaker adaptation / normalization.

2.2.1. Speaker Adaptation by Vector Quantization[S. Nakamura, Shikano]

As a general preprocessor to a phoneme recognizer, speaker normalization algorithms have been developed. The algorithms adopt vector quantization as a discrete representation of spectral space. Discrete representation makes it possible to carry out sophisticated spectral normalization or mapping from one speaker to another. In previous studies, algorithms using a single codebook were developed, and a complex spectral distortion measure which is composed of a spectrum term, a differenced spectrum term and a differenced power term was adopted. The complex measure improved the word recognition results, but this resulted in significant distortion degradation of the spectrogram distortion. In order to reduce this degradation, algorithms with multiple codebooks (separate VQ) have been investigated. These algorithms are called "speaker adaptation by separate vector quantization", and attain good performance in experiments evaluating spectrogram distortion and in phoneme recognition experiments. These algorithms are successfully applied to voice conversion, as described in section **2.6**. Moreover, *fuzzy vector quantization* techniques have been introduced to realize more accurate speaker adaptation.

These technologies lead us to an HMM speaker adaptation algorithm greatly improved over previously reported algorithms. These algorithms will be also applied to neural network speech recognition and feature-based speech recognition as a speaker normalization preprocessor.

Technical Publications: [S.Nakamura-88-02, 88-08a, 88-08b, 88-12], [Speaker adaptation using fuzzy VQ will be presented at ICASSP89]

2.3. Language Models

The ATR Interpreting Telephony Laboratories also include two other departments which are studying language models more deeply than the Speech Processing Department. Nevertheless, the Speech Processing Department has been studying language models based on bottom-up word prediction and statistical word prediction in collaboration with the other departments. We have been taking two approaches: a word trigram approach and a neural network approach (*Netgram* model), to predict the next word set. Such bottom-up word prediction approaches should be combined with top-down approaches in language processing.

As another top-down language model, the generalized LR parser has been adopted with the aim of better language source modeling.

2.3.1. Word Trigram Models [Saito, Shikano]

Trigram language models based on word categories were introduced in order to improve word recognition results for English sentences uttered word by word. Probabilities of the trigrams of word categories were estimated using the Brown Corpus Text Database. Moreover, high probabilities of common word sequences such as frozen word sequences were extracted from the Brown Corpus Text Database. These probabilities were integrated using a dynamic programming algorithm, and improved word recognition from 80.9% to 89.1%.

The generalized LR parser is introduced to predict next words/phonemes. The LR parser can be regarded not only as a parser for input word sequences but also as a language source model for word/phoneme prediction. The LR parser is successfully integrated with HMM phoneme models to recognize a Japanese phrase utterance, where the LR parser predicts phoneme candidates according to the LR table. The LR parser will be developed for better language source modeling by introducing the probabilities of phoneme sequences and word sequences automatically estimated from the large-scale text database.

Other low-level linguistic information is contained in phoneme sequences. The use of phoneme sequence information will be started aiming at source modeling of Japanese speech inputs.

Technical Publications: [Saito-88-06, 88-08, Kita-88-10b], [The HMM-LR will be presented at ICASSP89]

2.3.2. Word Prediction by Neural Networks [M. Nakamura, Shikano]

A neural network approach, *the Netgram*, has been developed to predict words using the Brown Corpus Text Database. The results have been compared to the results of trigram modeling. The analysis of internal representation of the Netgram reveals some correspondence to language categories. In order to realize this approach practically and quickly, further improvements to our learning network simulator have been made. Application to model Japanese phrase inputs will be started shortly.

Technical Publications: [M.Nakamura-88-06, 88-11], [The Netgram will be presented at ICASSP89], [Related ATR technical reports are available on request]

2.4. Noise Reduction

In order to bring speech recognition technologies into the field, reduction of various noises must be implemented. We are now looking for new noise reduction mechanisms using various kinds of neural networks.

2.4.1. Noise Reduction by Neural Networks [Tamura, Waibel]

Noise reduction algorithms using neural networks have been under investigation using the back-propagation algorithm on multi-layered feed-forward networks. During training, the networks develop their internal representation for a mapping from a set of noisy signals to the set of noise-free signals with noise-added speech signals as inputs and noise-free speech signals as the target outputs. The training is carried out using the digitized speech wave itself. Noise reduction algorithms already effectively eliminate computer room noise. Effectiveness for other noise and other speaker is also being studied. The internal representation of the noise reduction networks is being investigated. This investigation has somewhat clarified the noise reduction mechanism.

Technical Publications: [Tamura-88-01, 88-03, 88-04], [Related noise reduction networks are also presented at ICASSP89]

2.5. 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 is under development, and speech synthesis algorithms are evaluated to realize a better synthesis system. Also, rules between speaking styles and prosodic control have been studied.

2.5.1. High Quality Japanese Speech Synthesis by Rule [Sagisaka, K. Abe, Takeda, Poser]

In speech synthesis by rule, various kinds of synthesis units such as phonemes, dyads, CV-syllables and CVC units have been proposed. Our proposed synthesis scheme is different from ordinary systems in two ways. One is the flexible use of non-uniform synthesis units, and the other is the choice of optimal units of multi-templates using criteria such as feasibility of unit concatenation and fitness of units to target context. The strength of coarticulation for CV syllables is also investigated in depth. The place of articulation is much more dominant than the manner of articulation from the viewpoint of coarticulation phenomena.

Prosody control models to generate natural Japanese synthesized utterances are studied based on the speech database with pitch information.

Speech synthesis algorithms are evaluated over LPC, Cepstrum, and PSE. OLA(overlapping add) synthesis algorithms are also evaluated.

A prototype speech synthesis-by-rule system has been implemented, and efforts at improvement are continued.

Technical Publications: [Sagisaka-88-04, 88-03, K.Abe-88-11, Takeda-88-01a, 88-11], [Related ATR technical reports are available on request]

2.5.2. Modal Information and Prosodics [Miyatake, Sagisaka]

Effects of speaking styles on prosodic parameters were analyzed to clarify the interrelationship of pitch patterns, power patterns and durations. The analysis shows that differences among speaking styles are strongly reflected in pitch and power patterns. The results suggest the possibility of a new prosody control model for speech synthesis. Prosodic models have been studied to realize more natural synthesized speech by rules.

Technical Publications: [Miyatake-88-05a], [Related ATR technical reports are available on request]

2.6. Voice Conversion

Voice conversion from one speaker to another is an important aspect in realizing an automatic telephone interpreting system. Voice conversion algorithms based on vector quantization and spectrum mapping have been developed. We have also started to analyze personal voice characteristics using analysis-by-synthesis methods. Moreover, cross-linguistic voice conversion from MITALK to a Japanese female voice is tried with some success. A new speech analysis and synthesis algorithm based on short-time Fourier transform has been investigated in order to deal with more personal voice characteristics.

2.6.1. Voice Conversion by Parameter Mapping through Vector Quantization [M. Abe, S. Nakamura, Kuwabara, Shikano]

A new voice conversion technique through vector quantization has been studied. The algorithms used in this voice conversion are basically the same as the spectrum mapping described in section 2.2. The algorithms have three codebooks, for spectrum, power and pitch. Voice conversion from male voice to female voice was tried and a listening test shows that the converted voice was uniformly judged to be female. In order to improve the converted voice, vector quantization of residual waves and use of fuzzy vector quantization techniques are now being explored.

Moreover, a preliminary experiment in cross-linguistic voice conversion from MITALK to Japanese female voice is successfully carried out.

Technical Publications: [M.Abe-88-04, 88-02]

2.6.2. Personal Characteristics Analysis by Analysis-by-Synthesis [Segot, Kuwabara]

Aiming to improve the quality of speech, the modification of formant frequencies and bandwidths has been studied. These modification techniques will be applied to analyzing personal voice characteristics. Attention is now focused on the dynamic aspects of spectral parameters that convey personal voice characteristics.

2.6.3. Speech Analysis and Synthesis Algorithm [M. Abe, Tamura, Kuwabara, K.Abe]

A new speech analysis and synthesis algorithm based on short-time Fourier transform has been investigated aiming at better separation between spectral envelope information and vocal source information not only for high quality synthesis by rule system but also voice conversion.

Technical Publications: [M.Abe-88-09], [The algorithm will be presented at ICASSP89]

2.7. Phonetics

Analysis of speech phenomena has been started using our large vocabulary speech database in order to clarify allophonic and coarticulatory phenomena.

2.7.1. Analysis of Allophones and Coarticulation [Takeda, Kuwabara]

As a first step in analyzing allophonic and coarticulatory phenomena, devocalization of vowels was studied using our large vocabulary speech database. The contexts of the devocalization are predicted by rules. Next, phoneme durations have been studied using our large speech database.

Technical Publications: [Takeda-88-11, Kuwabara-88-05b]

2.8. Speech Database

A speech database with phoneme labels is essential and necessary not only for speech recognition research but also for speech synthesis research.

2.8.1. Japanese Large-Scale Speech Database with Phonetic Transcriptions [Takeda, Kuwabara, M. Abe]

A large-scale Japanese speech database has been constructed at ATR with the help of several other laboratories. For multiple transcriptions, three types of categories are considered: phonetic labels, acoustic events and allophonic variations. To date, about 8,500 words uttered by seven professional announcers have been collected and transcribed. The speech database has been effectively used to perform speech research at ATR.

A continuous utterance speech database with phonetic transcription has been constructed, in which phoneme balance is considered from the viewpoint of entropy of CV and VC occurances. Prosody labels have been tried on the continuous utterance speech database.

Technical Publications: [The ATR database will be introduced at ICASSP89], [Related ATR technical reports are available on request]

2.8.2. Speech Database Retrieval System [Takeda, Kuwabara]

To easily access the speech database, a database management system was developed. The system was implemented using the relational database *UNIFY*. The easy access language, *EAL*, was implemented using the host language interface on *UNIFY*. *EAL* is now widely used at ATR.

Technical Publications: [A technical report on EAL is available on request]

2.8.3. Speech Workbench [Maruyama, Tanaka, Murayama, Kawabata]

A speech workbench for speech researchers has been developed on a micro Vax station based on x-window. The main features are high quality spectrograms, easy extension of functions and easy access to the speech database.

Technical Publications: [Related ATR technical reports are available on request]

3. Research Staff

The research staff is mainly composed of members from the research institute and laboratories which support ATR, and visiting scientists. They are listed below:

Dr. Kiyohiro Shikano,	*Department Head*	*(NTT,*	*1986.6 −)*
Dr. Hisao Kuwabara,	*Supervisor*	*(NHK,*	*1986.10−)*
Dr. Yoshinori Sagisaka,	*Senior Researcher*	*(NTT,*	*1986.4 −)*
Dr. Takeshi Kawabata,	*Senior Researcher*	*(NTT,*	*1986.9 −)*
Dr. Hidefumi Sawai,	*Senior Researcher*	*(Ricoh,*	*1988.4 −)*
Dr. Alex Waibel,	*Invited Researcher*	*(Carnegie-Mellon,*	*1987.5 − 1988.8)*
Dr. William Poser,	*Invited Researcher*	*(Stanford Univ.,*	*1988.9 −)*
Shin'ichi Tamura,	*Researcher*	*(Sony,*	*1986.9 −)*
Kaichiro Hatazaki,	*Researcher*	*(NEC,*	*1986.12−)*
Masanori Miyatake,	*Researcher*	*(Sanyo,*	*1986.9 −)*
Masami Nakamura,	*Researcher*	*(Sumitomo Metal,*	*1987.9 −)*
Yasuhiro Komori,	*Researcher*	*(Canon,*	*1988.9 −)*
Katsuo Abe,	*Researcher*	*(Toyocom,*	*1987.3 −)*
Satoshi Nakamura,	*Researcher*	*(Sharp,*	*1986.9 −)*
Masanobu Abe,	*Researcher*	*(NTT,*	*1987.4 −)*
Kazuya Takeda,	*Researcher*	*(KDD,*	*1986.8 −)*

Katsuteru Maruyama,	*Researcher*	*(Nitsuko,*	*1987.3 –)*
Toshiyuki Hanazawa,	*Researcher*	*(Mitsubishi Elec.,*	*1987.3 –)*
Hiroaki Saito,	*Visiting Researcher*	*(Carnegie-Mellon,*	*1988.3 – 1988.7)*
Kooichi Murayama,	*Engineer*	*(DEC Japan,*	*1987.9 – 1988.8)*
Takaharu Tanaka,	*Engineer*	*(DEC Japan,*	*1988.9 –)*
Hubert Segot,	*Intern student*	*(ENST, France,*	*1987.4 – 1987.11)*
Patrick Haffner,	*Intern student*	*(ENST, France,*	*1988.4 – 1988.11)*

4. Research Facilities

The Speech Processing Department facilities include a number of computer systems which are connected through Ethernet. The computer systems include a Vax 8600, a Vax 8700, a Vax 8800 and an Alliant 9800 (eight-CPU array processor). Every researcher also has a micro Vax station or a C Vax with an AD/DA device. Four Masscomps (MC5600) and one micro Vax II which are equipped with 16 bit AD conversion devices are used to collect speech data. Three S3670s (Symbolics) with *ART* are used for implementing a phoneme segmentation expert system.

Technical Publication List
at the Speech Processing Department,
ATR Interpreting Telephony Research Laboratories.

November, 1987 through December, 1988

(Abbreviations in the technical publication list)

ASA : The Acoustical Society of America

ASJ : The Acoustical Society of Japan

IEICE : The Institute of Electronics, Information and Communication Engineers (Japan)

FASE : Federation of Acoustical Society of Europe

ICASSP : IEEE International Conference on Acoustics, Speech, and Signal Processing

COLING : International Conference on Computational Linguistics

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(Bold letter titles show papers written in English or with an English abstract)

Ref.ID	Title(題名)	Authors (著者)	Conference /Journal (発表先)	Page
Haffner -88-10	**Fast Back-Propagation Learning Methods for Neural Networks in Speech**	P. Haffner A. Waibel K.Shikano	ASJ Fall Meeting, Hakata, (1988-10)	1
Haffner -88-11a	**DyNet, a Fast Program for Learning in Neural Networks**	P.Haffner	ATR Technical Report, TR-I-0059, (1988-11)	—
Haffner -88-11b	**Fast Back-Propagation Learning Methods for Neural Networks in Speech**	P.Haffner A.Waibel H.Sawai K.Shikano	ATR Technical Report, TR-I-0058, (1988-11)	—
Hanazawa -87-12	Hidden Markovモデルを用いた日本語有声破裂音の識別 **(Recognition of Japanese Voiced Stops Using Hidden Markov Models)**	T.Hanazawa T.Kawabata K.Shikano	IEICE Technical Report SP87-98 (1987-12)	3
Hanazawa -88-01	Hidden Markov Model を用いた日本語有声破裂音の識別 (Phoneme Recognition of Japanese Voiced Bursts by HMM Phone Modeling)	T.Hanazawa T.Kawabata K.Shikano	ATR Technical Report TR-I-0018 (1988-01)	—
Hanazawa -88-03	HMMを用いた音韻認識における出力確率の平滑化手法の検討 (Output Probability for HMM Phoneme Recognition)	T.Hanazawa T.Kawabata K.Shikano	ASJ Spring Meeting, Tamagawa Univ., (1988-03)	9
Hanazawa -88-06	HMM音韻認識におけるモデル学習の諸検討 **(Studies for HMM Phoneme Recognition)**	T.Hanazawa T.Kawabata K.Shikano	IEICE Technical Report SP88-22, Sendai, (1988-06)	11
Hanazawa -88-10a	**Duration Control Methods for HMM Phoneme Recognition**	T.Hanazawa T.Kawabata K.Shikano	ATR Technical Report, TR-I-0050, (1988-10)	18
Hanazawa -88-10b	HMM音韻認識におけるセパレートベクトル量子化の検討 (Study of Separate Vector Quantization for Phoneme Recognition)	T.Hanazawa T.Kawabata K.Shikano	ASJ Fall Meeting, Hakata, (1988-10)	28
Hanazawa -88-11	**Duration Control Methods for HMM Phoneme Recognition**	T.Hanazawa T.Kawabata K.Shikano	ASA-ASJ Joint Meeting, Honolulu (1988-11) *(Handout)*	18
Hatazaki -88-01	スペクトログラム·リーディング知識を用いた音韻認識エキスパートシステム **(Phoneme Recognition Experts System Using Spectrogram Reading Knowledge)**	K.Hatazaki S.Tamura T.Kawabata K.Shikano	IEICE Technical Report SP87-117 (1988-01)	30
Hatazaki -88-03	スペクトログラム·リーディング知識を用いた音韻セグメンテーションの試み (Phoneme Segmentation Using Spectrogram Reading Knowledge)	K.Hatazaki S.Tamura T.Kawabata K.Shikano	ASJ Spring Meeting, Tamagawa Univ., (1988-03)	36

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(Bold letter titles show papers written in English or with an English abstract)

Ref.ID	Title(題名)	Authors (著者)	Conference /Journal (発表先)	Page
Hatazaki -88-08	**Phoneme Segmentation by an Expert System Based on Spectrogram Reading Knowledge**	K.Hatazaki S.Tamura T.Kawabata K.Shikano	Seventh FASE Symposium Edinburgh, (1988-08)	38
Hatazaki -88-10	スペクトログラム・リーディング知識に基づく無声摩擦音の検出 (Segmentation of Unvoiced Fricatives Using Spectrogram Reading Knowledge)	K.Hatazaki S.Tamura T.Kawabata K.Shikano	ASJ Fall Meeting, Hakata, (1988-10)	46
Iso -88-03	音声データベース用文セットの設計 (Design of a Japanese Sentence List for a Speech Database)	*K.Iso *T.Watanabe H.Kuwabara	ASJ Spring Meeting, Tamagawa Univ., (1988-03)	48
K.Abe -88-03	入力音韻系列に応じた音声素片の選択法 (A Synthesis Unit Selection Method Adapting to an Input Phoneme)	K.Abe Y.Sagisaka	ASJ Spring Meeting, Tamagawa Univ., (1988-03)	50
K.Abe -88-11	**On the Unit Selection Measure for Speech Synthesis by Rule Using Multiple Synthesis Units**	K.Abe Y.Sagisaka	ASA-ASJ Joint Meeting, Honolulu (1988-11) *(Handout)*	52
K.Abe -88-12	**On the Unit Selection Measure for Speech Synthesis by Rule Using Multiple Synthesis Units**	K.Abe Y.Sagisaka	ATR Technical Report, TR-I-0053, (1988-12)	52
Katagiri -88-01	スペクトログラムを用いた音声ラベリング **(Speech Labeling Using a Spectrogram)**	*S.Katagiri K.Takeda Y.Sagisaka	IEICE Technical Report SP87-115,(1988-01)	62
Kawabata -87-12	日本語スペクトルグラム特徴の英語との比較**(Sound Spectrogram Features in Japanese)**	T.Kawabata S.Tamura K.Hatazaki	IEICE Technical Report SP87-95, (1987-12)	70
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