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**Research Activities of
the Speech Processing Department**

January through December, 1992

ATR Interpreting Telephony Research Laboratories

March, 1993

ATR Interpreting Telephony Research Laboratories

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This volume is the sixth in a 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

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 Construction

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Research Activities of the Speech Processing Department, January through December, 1992

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1 Introduction

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

The utterance of one party is to be translated into another language, which is heard by the person on the other end. The basic constituent technologies are speech recognition, language translation, and speech synthesis, which are not only essential for the 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.

2 Automatic Telephone Interpretation Project

2.1 History and General Concept

In 1983, the first experimental demonstration of on-line speech translation was given at TELECOM-83 by NEC Research Laboratories. In 1987, speech translation between English and French was conducted at British Telecom Research Laboratories. That system was based on a set of more than 400 common business phrases. At Carnegie-Mellon University (CMU), a speech translation system with spoken input was developed in 1988 in a simple doctor-patient conversation domain.

In Japan, ATR Interpreting Telephony Research Laboratories, a subsidiary of Advanced Telecommunications Research International, was established in 1986 to initiate basic research for telephone interpretation. Stimulated by these earlier activities toward speech translation, a considerable number of research organizations became interested in this research area. Some of them have started active research in this particular field. In the United States, Carnegie-Mellon University is working on English-Japanese speech

translation. AT&T Bell Laboratories is making an English-Spanish speech translation experiment with a relatively small vocabulary. In Germany, a new research project named "Verbmobil" has been started aiming at language translation with speech input. Karlsruhe University has begun a speech translation project between German and English. Korea Telecom also have a 10-year research project on automatic speech interpretation; its short-term target is Korean-Japanese speech translation. Mutual cooperation and collaborations among these research groups has been already started. On January 28 of 1993, ATR Interpreting Telephony Research laboratories, Carnegie-Mellon University, and Siemens AG + Karlsruhe University successfully performed jointly the first international experiment of interpreting telephony.

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

A simple combination of these elements of current technology, however, does not provide a satisfactory speech translation system. What is required of these components is somewhat different from how they are used in conventional applications. The major differences are as follows:

- 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. Although, in spoken language, sentences are generally shorter and their structures are not particularly complicated, spoken dialogues include elliptic and anaphoric expressions. They may also include many syntactically ill-formed expressions.

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

Although the ultimate goal of the telephone interpretation system is universal dialogue in an unlimited domain, the immediate goal should be more feasible, 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 expected vocabulary limited to approximately 1,500 words.

2.2 ATR's Experimental Speech Translation System

Figure 1 outlines the current experimental 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 a final choice is determined by linguistic analysis. This is 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. This stage is, however, not yet integrated into the system. The other 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 approach can be applied to the English-Japanese speech translation.

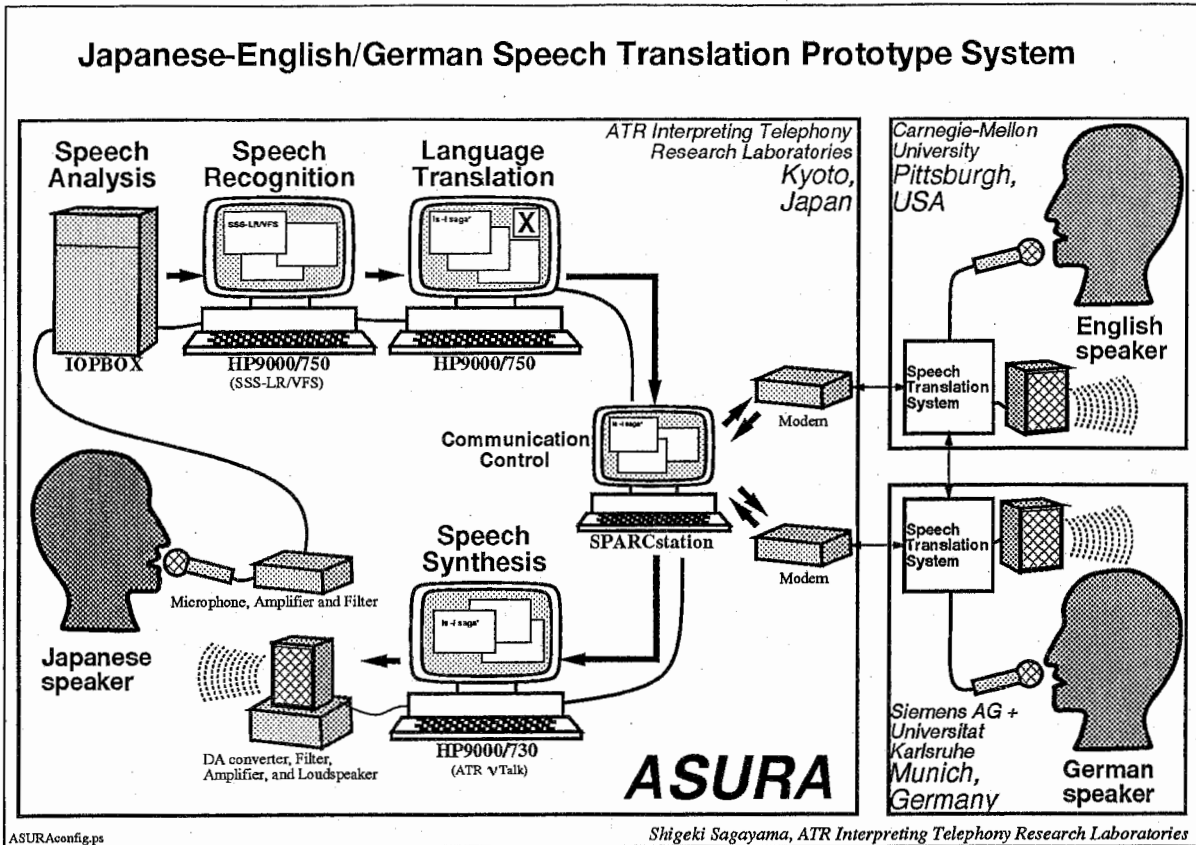


Figure 1: ASURA – ATR’s experimental interpreting telephone prototype system

3 Speech Research at ATR Interpreting Telephony Research Laboratories

3.1 Research Project

ATR Interpreting Telephony Research Laboratories (Akira Kurematsu, president) consist of three departments: Natural Language Understanding Department, Knowledge and Database Department, and Speech Processing Department. The laboratory was founded in April 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 the integration of speech and language processing technologies to demonstrate the feasibility of an automatic telephone interpretation system. As of 1991, the laboratory had a research budget of 2.6 billion yen.

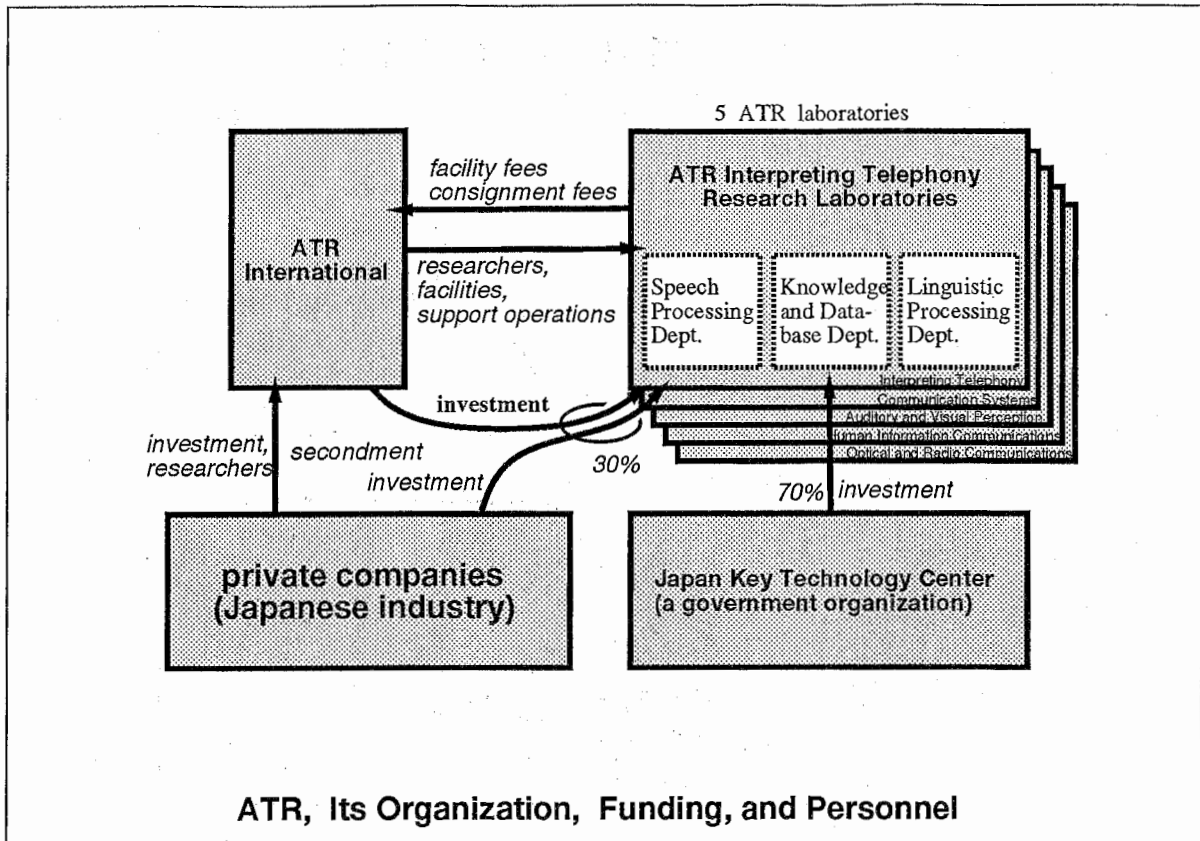


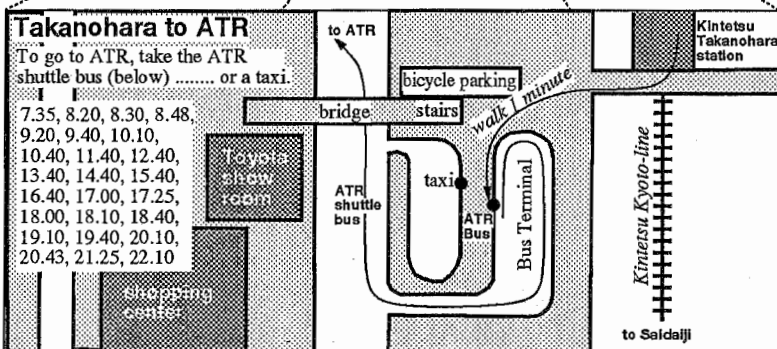
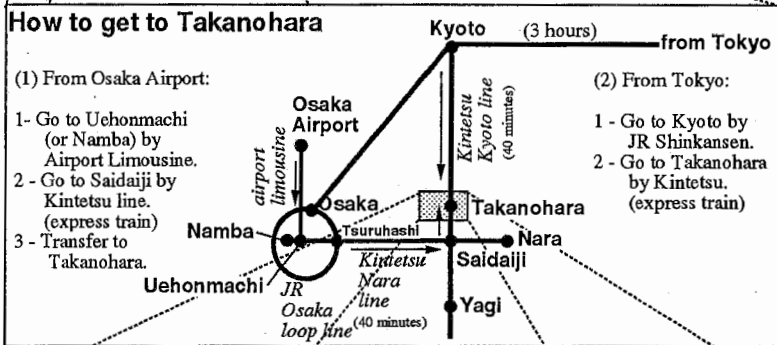
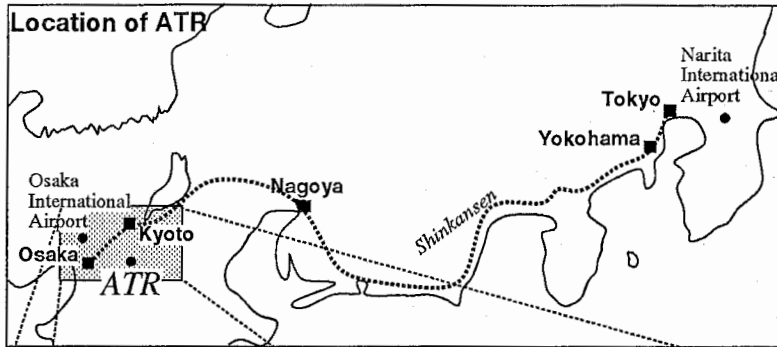
Figure 2: Funding and personnel of ATR Laboratories

3.2 Location

ATR Interpreting Telephony Research Laboratories is located in Kyoto Prefecture, although its geographical location is much closer to Nara than to Kyoto city. Nara was the capital of Japan more than 1,200 years ago. Promoted by Japanese government and industry, 12 separate areas in this district have been designated as Kansai Science and Technology Cities. They are expected to rapidly grow into Japan's new center of major scientific and technological research and education activities. ATR (including ATR International and its five subsidiary laboratories) was one of the earliest organizations to settle in this area.

ATR's transportation situation is not excellent at the moment. Figure 3 shows how to get to ATR Interpreting Telephony Research Laboratories. The approximate time from Kyoto to Takano-hara station of the Kintetsu Railway is 40 minutes by express train, which leaves Kyoto every 15 minutes. ATR shuttle buses or taxis are available at Takano-hara station and take 10 minutes to reach ATR.

Transportation to ATR



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 E-mail: saga@atr-la.atr.co.jp

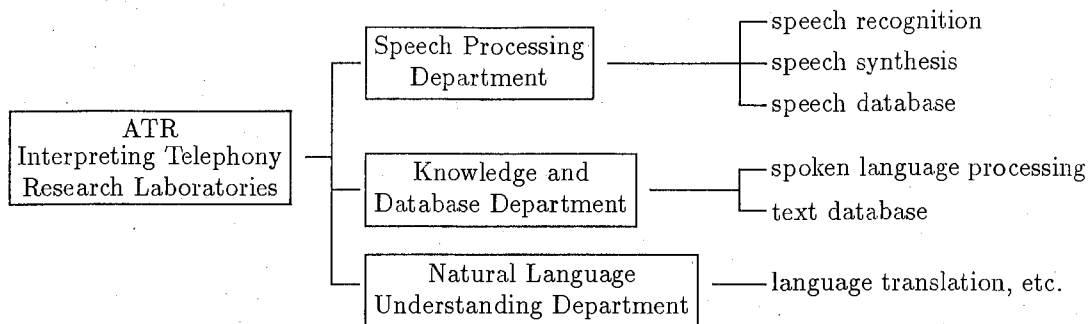
HowToGetToATR.ps

Figure 3: How to get to ATR Laboratories

3.3 Organization

ATR Interpreting Telephony Research Laboratories consist of three departments: Speech Processing Department (Shigeki Sagayama, Head), Knowledge and Database Department (Tsuyoshi Morimoto, Head), and Natural Language Understanding Department (Hitoshi Iida, Head).

An outline of the organization is shown below.



3.4 Research Staff

The research staff is mainly composed of members temporarily seconded from research institutes, laboratories, and industrial companies that support ATR, and visiting/invited researchers.

Currently (as of March, 1993), the Speech Processing Department has 23 members including a department head, two supervisors, 15 researchers, four visiting/invited researchers, and an engineer. Tables ?? and ?? shows former and current members of the research staff at the Speech Processing Department, including visiting/invited researchers. Table ?? shows the list of graduate students who worked or are currently working as their internships at the department.

Current members can be reached through E-mail account names followed by '@atr-la.atr.co.jp'. (For example, Sagayama's E-mail address is 'saga@atr-la.atr.co.jp'.)

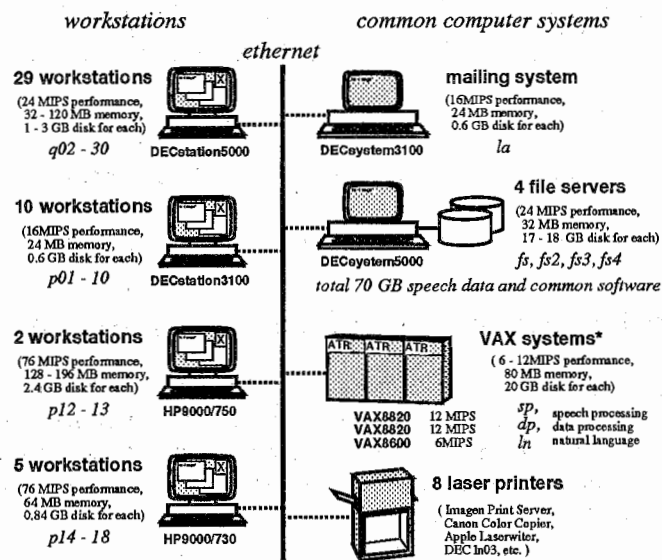
<u>name & title</u>	<u>Email</u>	<u>position</u>	<u>period</u>
Kiyohiro Shikano, Dr	-	Department Head	6/1986 - 1/1990
Shigeki Sagayama	saga	Department Head	2/1990 - 3/1993
Yoshinori Sagisaka, Dr	sagisaka	Supervisor	4/1986 - 3/1993
Hisao Kuwabara, Dr	-	Supervisor	10/1986 - 7/1989
Masahide Sugiyama, Dr	sugi	Supervisor	2/1990 - 3/1993
Tetsuo Umeda	-	Supervisor	7/1989 - 6/1990
Takeshi Kawabata, Dr	-	Senior Researcher	9/1986 - 2/1990
Shin-ichi Tamura	-	Senior Researcher	9/1986 - 2/1990
Hidefumi Sawai, Dr	-	Senior Researcher	4/1988 - 3/1991
Kazuya Takeda	-	Researcher	8/1986 - 2/1990
Satoshi Nakamura	-	Researcher	9/1986 - 8/1989
Masanori Miyatake	-	Researcher	9/1986 - 3/1989
Kaichiro Hatazaki	-	Researcher	12/1986 - 3/1989
Toshiyuki Hanazawa	-	Researcher	3/1987 - 2/1990
Katsuteru Maruyama	-	Researcher	3/1987 - 2/1990
Masanobu Abe	-	Researcher	4/1987 - 2/1991
Katsuo Abe	-	Researcher	3/1987 - 2/1990
Masami Nakamura	-	Researcher	9/1987 - 8/1990
Yasuhiro Komori	-	Researcher	9/1988 - 2/1992
Hiroaki Hattori	hiroaki	Researcher	5/1989 - 4/1992
Kazumi Ohkura	ohkura	Researcher	9/1989 - 3/1993
Nobuyoshi Kaiki	kaki	Researcher	11/1989 - 10/1992
Jun-ichi Takami	jun	Researcher	11/1989 - 3/1993
Kazuki Katagishi, Dr	katagisi	Researcher	2/1990 - 3/1993
Akito Nagai	nagai	Researcher	3/1990 - 3/1993
Keiji Fukuzawa	fukuzawa	Researcher	4/1990 - 3/1993
Katsuhiko Mimura	mimura	Researcher	5/1990 - 3/1993
Shingo Fujiwara	fujiwara	Researcher	5/1990 - 3/1993
Naoto Iwahashi	iwahashi	Researcher	10/1990 - 3/1993
Jin-ichi Murakami	murakami	Researcher	3/1991 - 3/1993
Yoshinaga Kato	kato	Researcher	4/1991 - 3/1993
Kouichi Yamaguchi	yamaguti	Researcher	5/1991 - 3/1993

Yasunaga Miyazawa	miyazawa	Researcher	7/1991 - 3/1993
Tetsuo Kosaka	kosaka	Researcher	9/1991 - 3/1993
Ryosuke Isotani	isotani	Researcher	4/1992 - 3/1993
Alex Waibel, Dr	-	Invited Researcher	5/1987 - 8/1988, 6 - 9/1989
Hiroaki Saito, Dr	-	Invited Researcher	2/1988 - 7/1989
William Poser, Dr	-	Invited Researcher	9/1988 - 2/1989
Alain de Cheveigné, Dr	-	Invited Researcher	2/1989 - 2/1990
Dieter Huber, Dr	-	Invited Researcher	3 - 6/1990
David Rainton, Dr	rainton	Invited Researcher	4/1990 - 3/1993
Wilhelm N. Campbell, Dr	nick	Invited Researcher	8/1990 - 3/1993
Harald Singer	singer	Invited Researcher	3/1991 -
Helmut Lucke, Dr	lucke	Invited Researcher	4/1992 - 3/1993
Wang Wern-jun	wang	Invited Researcher	2 - 7/1991
Jacqueline Vaissiere, Dr	-	Invited Researcher	7 - 8/1992
Paul Christopher Bagshaw	-	Invited Researcher	5 - 11/1992
Kouichi Murayama	-	Engineer	9/1987 - 8/1988
Takaharu Tanaka	-	Engineer	9/1988 - 8/1989
Kouji Kitagaito	-	Engineer	9/1989 - 8/1990
Shinobu Araki	-	Engineer	9/1990 - 8/1991
Yoshinori Ono	ono	Engineer	9/1991 - 3/1993
Hubert Segot	-	Student (ENST)	4 - 9/1987
Eiichiro Kitagawa	-	Student (Waseda Univ)	8/1987
Kenji Hashimoto	-	Student (Waseda Univ)	8/1987
Takahito Yamazaki	-	Student (Shizuoka Univ)	8/1988
Furati Mulad	-	Student (Shizuoka Univ)	8/1988
Patrick G. Haffner	-	Student (ENST)	3 - 11/1988
Yasuhide Hashimoto	-	Student (Toyohashi Univ of Tech)	7 - 8/1987
Takashi Endo	-	Student (Waseda Univ)	8/1987
Rei Furukawa	-	Student (Waseda Univ)	3 - 4/1989
Jean-Claude Dang	-	Student (ENST)	3 - 11/1989
Yoshimitsu Hirata	-	Student (Toyohashi Univ of Tech)	1 - 2/1989
Mitsuru Noda	-	Student (Toyohashi Univ of Tech)	1 - 2/1989
Yasuhiro Minami	-	Student (Keio Univ.)	3-6/1989, 8/1989-2/1990, 7-10/1990
Katsunobu Ito	-	Student (Tokyo Inst of Tech)	10 - 12/1989
Yoshio Ueda	-	Student (Toyohashi Univ of Tech)	1 - 2/1990
Takeshi Hirado	-	Student (Toyohashi Univ of Tech)	1 - 2/1990
Satoru Nakamura	-	Student (Keio Univ)	7 - 9/1990, 3 - 4/1991
Toshiyuki Nomura	-	Student (Nagoya Univ)	7 - 8/1990
Mamoru Watado	-	Student (Waseda Univ)	7 - 8/1990
Hidehiro Inagaki	-	Student (Waseda Univ)	7 - 8/1990
Yasushi Maruyama	-	Student (Shinshu Univ)	3-4/1990
Atsuhiko Kai	-	Student (Toyohashi Univ of Tech)	?/1990 - ?/1991
Alain Biem	-	Student (INT)	8/1990 - 1/1991
Hidefumi Kikuchi	-	Student (Waseda Univ)	8/1991
Masafumi Tamoto	-	Student (Tokyo Inst of Tech)	8/1991
Keisuke Doi	-	Student (Ryukoku Univ)	8/1991 - 9/1992
Tetsuya Yoshida	-	Student (Keio Univ)	8 - 9/1991
Kentaro Kurinami	-	Student (Keio Univ)	8 - 9/1991
Romain Brunias	-	Student (INT)	8/1991 - 1/1992

Richard Lengagne	-	Student (INT)	8/1991 - 1/1992
Tadashi Okamoto	-	Student (Toyohashi Univ)	1 - 2/1992
Satoshi Aki	-	Student (Toyohashi Univ)	1 - 2/1992
Hideyuki Watanabe	-	Student (Hokkaido Univ)	2 - 4/1992
Hiroki Yamamoto	xyama	Student (Waseda Univ)	3 - 9/1992
Ioana Donescu	-	Student (INT)	8/1992 - 1/1993
Edward David Peter Willems	-	Student (ENST)	8 - 12/1992
Masashi Mizuno	-	Student (Waseda Univ)	7 - 9/1992
Franck Martin	-	Student (Univ de Centre Paris)	8 - 9/1992
Yasuo Iwai	-	Part-time (Univ of Kyoto)	8/1989 - 3/1990
Toshiyuki Sadanobu	-	Part-time (Univ of Kyoto)	8/1989 - 3/1990
Natsuya Yoshida	-	Part-time (Univ of Kyoto)	8/1989 - 3/1990
Miwako Kurihara	-	Part-time (Doshisha Univ)	4/1990 -
Mechtild Tronnier	-	Part-time (Kansai Univ)	9/1990 - 4/1991
Ming-Yong Zhou	zhou	Part-time (Osaka City Univ)	10/1990 -
Kazue Kinugasa	xkazue	Part-time (Kansai Univ)	2/1992 -

3.5 Research Facilities

Now, each member of the research staff is provided a DECstation5000 (speed 24MIPS) or DECstation5500 (speed 50MIPS), with 24 - 120 Mbyte memory and a 600M - 3G byte disk, which is connected to an Ethernet. We have 4 other DECsystem5000s with over 30G byte disk memory each which are used as file servers to contain the large scale speech database and are accessed from workstations through the Ethernet. Some workstations are equipped with speech input/output devices. Older computer systems, including VAX8700, and VAX8820, are also connected to the Ethernet and accessed from workstations. The department has two Hewlett-Packard HP9000 model 750s and five model 730s which run at 76 MIPS for high-speed computations and experiments of interpreting telephony. In the international experiment of interpreting telephony among Japan, USA, and Germany, held on January 28, 1993, three HP9000s were used for speech recognition, language translation, and speech synthesis.



Research Facilities at Speech Processing Department

Figure 4: Computer Facilities of the Speech Processing Department, ATR Interpreting Telephony Research Laboratories

4 Interpreting Telephony

Speech recognition and speech synthesis research results were well combined with language translation and utilized in an experimental interpreting telephone system named "ASURA" (Advanced Speech Understanding and Rendering system of ATR). This system was demonstrated on January 28, 1993, in a three-party international experiment of interpreting telephone. The success of the first international experiment was widely reported in a number of TV news programs and newspapers. The system configuration is shown in 1.

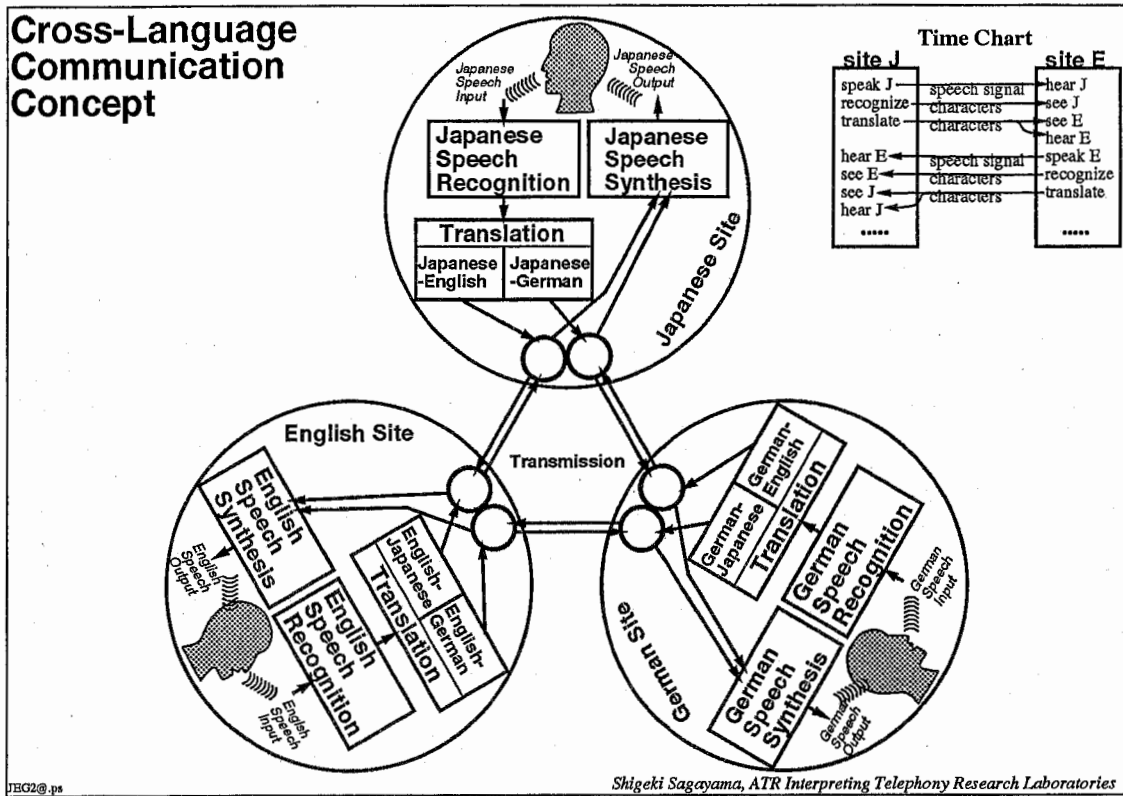


Figure 5: Conceptual diagram of interpreting telephony (Japanese, English, and German)

The task domain of the experiment was "international conference registration".

5 Technical Publications from January through December 1992

The following pages are a list of technical publications, published or coauthored by the researchers of the Speech Processing Department, ATR Interpreting Telephony Research Laboratories, published during the period from January through December 1991, including reprints of these publications. Some of these works were done in their previous affiliations and published after their joining ATR.

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
- ICSLP: International Conference on Spoken Language Processing

List of Publications, January through December, 1992

Araki92IPS03	1
<p>荒木 哲郎, 池原 悟, 村上 仁一: “音節ラティスに適用するビタービアルゴリズムの評価について,” 情報処理学会第44回全国大会講演論文集, 7N-2, pp. 2.163-164 (1992.03). T. Araki, S. Ikehara, J. Murakami: “Evaluation of Viterbi Algorithm Applied for Syllable Lattices,” Proc. of the 44th National Convention IPS Japan, 7N-2, pp. 2.163-164 (1992.03).</p>	
◆	
Biem92ACRCS10	3
<p>Alain Biem, Masahide Sugiyama: “A Hybrid Stochastic Connectionist Approach to Automatic Speech Recognition,” Proc. of 1st African Conference on Research in Computer Science, Vol. 2, pp. 605-617 (1992.10).</p>	
◆	
Campbell92ASJ03	16
<p>Wilhelm N. Campbell: “Gamma Modelling of English Segmental Durations,” 日本音響学会平成4年度春季研究発表会講演論文集, 1-2-18, pp. 241-242 (1992.03). ウィルヘルム N. キャンベル: “ガンマ分布を用いた英語音韻時間長のモデリング,” ASJ Spring Meeting, 1-2-18, pp. 241-242 (1992.03).</p>	
Campbell92ASJ03	18
<p>Wilhelm N. Campbell: “Prosodic Phrasing from Normalised Acoustic Measures,” 日本音響学会平成4年度春季研究発表会講演論文集, 1-2-19, pp. 243-244 (1992.03). ウィルヘルム N. キャンベル: “正規化した音響的尺度を用いた韻律フレーズング,” ASJ Spring Meeting, 1-2-19, pp. 243-244 (1992.03).</p>	
Campbell92SP03	20
<p>W. N. Campbell: “Synthesis Units for Natural English Speech” IEICE Technical Report, SP91-129, pp. 55-62 (1992.03). ウィルヘルム N. キャンベル: “自然な英語音声合成のための音声単位” 電子情報通信学会技術研究報告, SP91-129, pp. 55-62 (1992.03).</p>	
Campbell92ICSLP10a	28
<p>Wilhelm N. Campbell: “Prosodic Encoding of English Speech” Proc. of International Conference on Spoken Language Processing, Th.fPM.3.3, pp. 663-666, Canada (1992.10).</p>	
Campbell92ICSLP10b	32
<p>W. N. Campbell & C. W. Wightman: “Prosodic Coding of Syntactic Structure for Speech Synthesis,” Proc. of International Conference on Spoken Language Processing, Th.fAM.2.4, pp. 1167-1170, Canada (1992.10).</p>	
Campbell92ASJ10	36
<p>W. N. Campbell: “Labelling an English Speech Database for Prosody Control,” ASJ Fall Meeting, 1-P-8, pp. 315-316 (1992.10).</p>	
Campbell92SST12	38
<p>W. N. Campbell, Y. Sagisaka: “Automatic Annotation of Speech Corpora,” Proc. Fourth Australian International Conference on Speech Science and Technology, pp. 686-691 (1992.12).</p>	
Campbell92narakokusai	44
<p>W. N. Campbell: “Speech Timing in English and Japanese,” Mombusho International Symposium on Japanese Prosody, pp. 207-216, Nara Japan (1992).</p>	
Campbell92Book	54
<p>W. N. Campbell: “Syllable-based Segmental Duration,” Talking Machines: Theories, Models & Applications, Elsevier-North-Holland Pub. pp. 211-224 (1992).</p>	

Fujiwara92ISSPA08	68
Shingo Fujiwara, Yasuhiro Komori, Masahide Sugiyama: "An Integrated System for Automatic Labelling Based on HMM and Spectrogram Reading Knowledge," Proc. of International Symposium on Signal Processing and Its Applications, pp. 275-278, Australia (1992.08).	
Fujiwara92ICSLP10	72
Shingo Fujiwara, Yasuhiro Komori, Masahide Sugiyama: "A Phoneme Labelling Workbench using HMM and Spectrogram Reading Knowledge," Proc. of International Conference on Spoken Language Processing, Vol. 1, pp. 791-794 (1992.10).	
Fujiwara92ASJ10	76
藤原 紳吾, 小森 康弘, 杉山 雅英: "音素ラベリングワークベンチ," 日本音響学会平成4年度秋季研究発表会講演論文集, 2-Q-28, pp. 227-228 (1992.10).	
S.Fujiwara, Y.Komori, M.Sugiyama: "A Phoneme Labelling Workbench," ASJ Fall Meeting, 2-Q-28, pp. 277-278 (1992.10).	
◆	
Fukuzawa92SP01	78
福沢 圭二, 小森 康弘, 沢井 秀文, 杉山 雅英, "セグメントベース話者適応ニューラルネットワークとTDNN-LRを用いた文節音声認識," 電子情報通信学会技術研究報告, SP91-105, pp. 23-29 (1992.01).	
Keiji Fukuzawa, Yasuhiro Komori, Hidehumi Sawai, Masahide Sugiyama: "A Segment-based Speaker Adaptation Neural Network applied to TDNN-LR Continuous Speech Recognizer," IEICE Technical Report, SP91-105, pp. 23-29 (1992.01).	
Fukuzawa92ASJ03	85
福沢 圭二, 小森 康弘, 杉山 雅英: "TDNN-LR 連続音声認識における不特定話者 TDNN と話者適応ニューラルネットワークの性能比較," 日本音響学会平成4年度春季研究発表会講演論文集, 2-Q-21, pp. 199-200 (1992.03).	
Keiji Fukuzawa, Yasuhiro Komori, Masahide Sugiyama: "A Comparison between Multi-Speaker Trained TDNN and Speaker Adaptation Neural Network in a TDNN-LR Continuous Speech Recognition System," ASJ Spring Meeting, 2-Q-21, pp. 199-200 (1992.03).	
Fukuzawa92ICASSP03	87
Keiji Fukuzawa, Yasuhiro Komori, Hidehumi Sawai, Masahide Sugiyama: "A Segment-based Speaker Adaptation Neural Network Applied to Continuous Speech Recognition," Proc. of 1992 International Conference on Acoustics, Speech, and Signal Processing, 55.1, Vol. 1, pp. 433-436 (1992.03).	
Fukuzawa92IEICE09	91
福沢 圭二, 杉山 雅英: "階層的クラスタリングと Neural Network を用いた教師なし話者適応法," 電子情報通信学会秋季大会講演論文集, Vol. 1, pp. 271-272 (1992.09).	
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