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TR-I-0281

Research Activities
of the
Natural Language Understanding Department
and the
Data Processing Department
for **Apr. 1991~Mar. 1992**

ATR Interpreting Telephony Research Laboratories

September 30 , 1992

ATR Interpreting Telephony Research Laboratories

Abstract

This report summarizes the research activities of the language-related departments in the ATR Interpreting Telephony Research Laboratories: the *Natural Language Understanding Department* and the *Data Processing Department*. Also contained are reprints of the related technical publications during Apr. 1991.~ Mar. 1992.

The research areas of the Data Processing Department are

(1) Integration of Speech and Language Processing
- HMM-LR Speech Recognition
- Speech Recognition Using Stochastic Language Models
- Sentential Speech Recognition Using Two-Level LR Parsing
- Unknown Word Processing in Speech Recognition
- Construction of Sentential Grammar for Speech Recognition
- Evaluation of Stochastic Language Model for Speech Recognition
- Linguistic Knowledge for Spoken Dialogue Processing

(2) Language Processing for Spoken Dialogue Translation
- Analysis of Japanese
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- General Overview
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- Construction of Linguistic Database and its Management
- Statistics on Collected Linguisticbase

(5) Knowledge Extraction and Lexical Investigation based on Corpora
- Extraction of transfer and thesaurus knowledge from LDB
- Investigation of Some Linguistic Phenomena Specific to Japanese Dialogue

The current research areas of the Natural Language Understanding Department include:

(6) Dialogue Modeling and Context-Based Inferences
- Plan Recognition Model for Dialogue Understanding
- Interpreting and Generating Elliptical Sentences in Machine Translation
- Identifying and Understanding Noun Phrases in Dialogue

- Translating Feature Structures to a Kind of Logical Form

(7) Distributed Natural Language Processing and Machine Translation

- Transfer-Driven Machine Translation(TDMT)
- Generation of English Dialogue
- Example-Based Machine Translation (EBMT)
- Local Cohesive Knowledge for Dialogue Translation
- Parallel Parsing Algorithms

(8) Memory-Based Massively-Parallel Spoken Language Processing

- MT Architecture Based upon Massively-Parallel Graph-Based Constraint Propagation

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1. Research Organizations:

ATR Interpreting Telephony Research Laboratories

An 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. At least three constituent technologies are necessary for such a system: *speech recognition*, *machine translation*, and *speech synthesis*. Integrated research into these technologies is also very important. A feasibility study, published by the Japanese Ministry of Posts and Telecommunications, says that realizing such a system will require at least fifteen years.

Basic research on each of the above technologies has already started at the ATR Interpreting Telephony Research Laboratories of which Dr. Akira Kurematsu is the president. These laboratories were founded in April, 1986 with the support of the Japan Key Technology Center, ATR International, NTT, KDD, NHK and other Japanese enterprises.

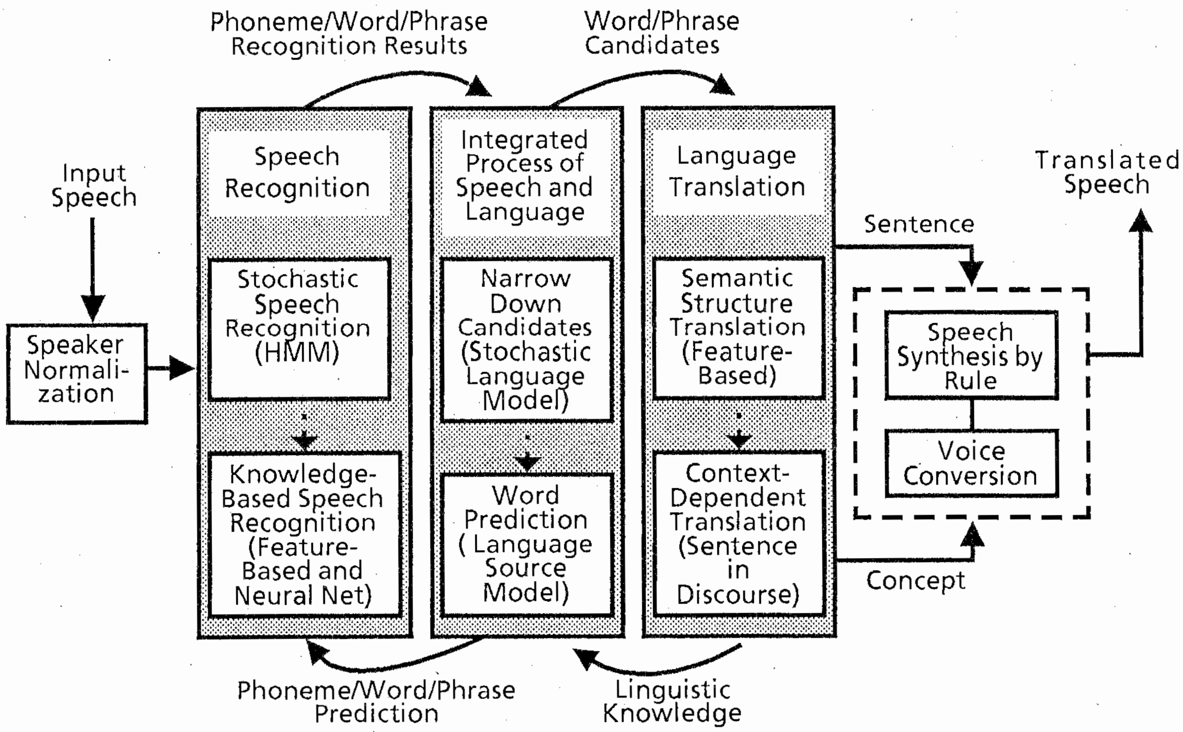

Figure 1. Proposed Automatic Telephone Interpretation System.

The ATR Interpreting Telephony Research Laboratories have three departments: the Speech Processing Department, the Natural Language Understanding Department and the Data Processing Department. These three departments cover the respective research areas to demonstrate the feasibility of an automatic telephone interpretation system shown in **Figure 1.** In this figure, the speech processing department is concerned with speech recognition, speech synthesis, speaker normalization, and voice conversion. The main research area of the natural language understanding department is language translation, and that of the Data processing department is integrated process of speech and language.

Technical Publications: [Kurematsu 91-05][Kurematsu 91-06-1][Kurematsu 91-06-2][Kurematsu 92-01][Kurematsu 92-03]

2. Research Activities

Research activities and the related technical publications for 1989 are summarized in Sections **2.1** through **2.5** for the Data Processing Department, and in Sections **2.6** through **2.8** for the Natural Language Understanding Department.

2.1. Integration of Speech and Language Processing

2.1.1 HMM-LR Speech Recognition

One of the major problems in speech recognition is coping with large search spaces. As search space size increases, recognition performance decreases. Syntactic constraints are effective in reducing the search space and hence increase processing speed and recognition accuracy. HMM-LR is an efficient speech recognition algorithm using Hidden Markov Models (HMMs) and predictive LR parsing. Predictive LR parsing is an extension of generalized LR parsing, and makes it possible to predict next phonemes in speech according to a context-free grammar. Predicted phonemes are then verified by using corresponding HMMs. The beam-search technique is also used to reduce the search space.

We implemented a Japanese phrase recognition system using the HMM-LR algorithm. For accurate phoneme recognition, multiple codebooks (separate vector quantization), fuzzy vector quantization, and HMM state duration control were used. A speaker adaptation algorithm based on codebook was also incorporated. The system attained an average phrase recognition rate of 98.5% and 99.2% for the top five choices in the speaker-dependent condition. In the speaker-adapted condition, rates were 81.6% and 98.0%, respectively.

The following topics are also studied:

- Phoneme context-dependent HMM-LR.
- HMM-LR using A*-search algorithm.
- Hardware implementation of HMM-LR.

The LR parser is also successfully combined with the Sphinx speech recognition system developed at Carnegie Mellon University (CMU) under a research collaboration between ATR and CMU.

Technical Publications : [Kita 91-05] [Nagai-91-06] [Nagai-91-09] [Nagai-91-10] [Yamaguchi-92-03]

2.1.2 Speech Recognition Using Stochastic Language Models

To take into account the stochastic characteristics of a language, we incorporated stochastic language models into the HMM-LR speech recognition system. Three stochastic language models were investigated.

1. Trigram model of Japanese syllables.
   Word bigram/trigram models are extensively used to correct speech recognition errors and improve recognition accuracy. The general idea of a trigram model was applied to Japanese syllables.
2. Stochastic LR parsing.
   The HMM-LR system uses the LR parser to deal with syntactic constraints supplied by a context-free grammar. In a traditional LR parser, each shift/reduce action is treated equally. However, some actions occur frequently, others rarely. Stochastic LR parsing was introduced to take their frequency of occurrence into account when calculating the recognition likelihood.
3. Trigram model of rewriting rules.
   Using the co-occurrence of rewriting rules, it is possible to avoid CFG rewriting rules generating incorrect word/phoneme sequences. This mechanism was implemented as a trigram model of CFG rewriting rules.

By utilizing these stochastic language models, the phrase recognition rate of the HMM-LR was improved from 88.2% to 93.2%.

Technical Publications : [Kita 90-05-2]

2.1.3 Sentential Speech Recognition Using Two-Level LR Parsing

The grammatical structure of Japanese has two levels: intra-phrase level and inter-phrase level. Thus, using two kinds of grammars, namely an intra-phrase grammar and an inter-phrase grammar, is sufficient for recognizing Japanese sentences. Two-level LR parsing provides a mechanism to use these two grammars stepwise during speech recognition. First, the inter-phrase LR parser predicts the next phrase categories, and then the HMM-LR system recognizes phrase candidates belonging to the predicted categories. The system attained a word accuracy of 95.9% and a sentence accuracy of 84.7%.

We also introduced a new parsing algorithm, called LR parsing with a category reachability test (the LR-CRT algorithm), that serves for an efficient implementation of two-level LR parsing. By utilizing the LR-CRT algorithm, a word accuracy and a sentence accuracy were improved to 97.5% and 91.2%, respectively.

Technical Publications : [Kita 91-07] [Kita-91-10]

2.1.4 Unknown Word Processing in Speech Recognition

Current speech recognition systems essentially ignore unknown words. Systems are designed to recognize words in the lexicon. However, for using speech recognition systems in a real application, it is very important to process unknown words. Preliminary research for unknown word processing has been initiated.

In our approach, two kinds of grammars are used. The first grammar is a normal grammar which describes our task. The lexicon for the task is embedded in this grammar as phoneme sequences. The second grammar describes the Japanese phonetic structure, in which constraints between phonemes are written. These two grammars are merged and used in the HMM-LR system. The HMM-LR system outputs words in the lexicon if no unknown word is included in the speech. If an unknown word is included, the system outputs a phonetic transcription that corresponds to the unknown word. Experiment results showed that this approach is very promising.

Technical Publications : [Kita 91-07-2]

2.1.5 Construction of Sentential Grammar for Speech Recognition

In the area of speech recognition applying syntactic rules is drawing a great deal of attention because of its potential for raising recognition accuracy. At the same time, this application reveals the lack of detailed study of syntactic phenomena, especially regards as spoken Japanese. We are therefore investigating building syntactic constraints which are effective for speech recognition and can handle spoken Japanese properly.

In contrast to traditional intuitive syntactic classification, we examine the behavior of syntactic categories quantitatively, using the dialogue data collected at ATR. Based on the retrieval results together with the speech recognition results, we are trying to find the best trade-off. In this way we have particularly studied sentence final conjunctive postpositions. The implementation of the study has very high filtering effect. By applying the constraints the acceptance rate goes down from 70% to 40%. Further we are studying sentence internal conjunctions and postposition deletion in nominal phrases, which are also features of spoken Japanese.

Technical Publications : [Hosaka 91-05] [Hosaka 91-09]

2.1.6 Evaluation of Stochastic Language Model for Speech Recognition

It is generally understood that introducing stochastic characteristics of language is useful in speech recognition. However, quantitative evaluation of ability of each stochastic model itself has not been performed so far. As for this problem, first an ability of the grammar which is used in our speech recognition system was investigated by applying it to a large amount of texts extracted from ATR dialogue database. By this investigation, it turned out that only less than 1% of all the grammar rules were actually used. This indicates that introducing a more tight grammar or an equivalent stochastic grammar can improve the speech recognition accuracy. Next issue to be made clear is to evaluate quantitatively the relationship between target texts and learning texts. It might be obvious that, for instance, a language model which is made from newspaper articles is not adequate to apply to spoken sentences, but how would the speech recognition accuracy be when a language model made from some domain is applied to a slightly different domain?. To evaluate this issue, we introduced a method which maps every texts into an n-dimensional Euclidean space, in which every texts are located at the points as they reflect the distances between them. The distance between two texts can be calculated from either the occurrent frequency of same words, or the usage frequency of same grammar rules. From this analysis, it was observed that the speech recognition accuracy has strong correlation with the degree of how many neighboring texts are used as learning texts.

2.1.7 Linguistic Knowledge for Spoken Dialogue Processing

Our previous speech recognition system was using only intra-phrase syntactical constrains, and output several hypotheses for each phrase. Then most of the sentences simply constructed by combining each phrase hypotheses are ill-formed. For improving of this kind of erroneous recognition, analysis on the points what kind of linguistic information is necessary to suppress/ detect such incorrect hypotheses. and how well they can work are investigated. From the analysis, the following points are made clear:

(a)Most of the incorrect hypotheses could be eliminated by introducing inter-phrase syntax.

(b)To suppress/detect the rest of them, semantics, pragmatics or contextual knowledge should be used. In particular, contextual knowledge, such as environment, cooperativeness or structure of dialogue, play an important role, because Japanese short post-positional particles or adverbial particles are likely to be mis-recognized, even though they control the total meaning of the sentence.

Currently, we are carrying researches along with this line.

Technical Publications : [Takezawa 91-05]

2.2 Language Processing for Spoken Dialogue Translation

2.2.1 Analysis of Japanese

The principal objective in this period was to extend the analysis grammar: from the existing 12 files of model dialogues (conference registration task) to the final target sentences for examination. The final target sentences were created artificially, but with supposing that most of those could be used in the conference registration. The others are composed purely for checking the grammatical functions implemented in the grammar. These sentences includes the target vocabulary to be contained in the final system. The basic words (approx. 1100, ranked as A,B,C) were originally selected according to the statistical data from the ATR dialogue database of conference registration. Besides, a number of proper nouns were collected and classified: eg. family names, first names, cities, prefectures, organizations, etc. In the final stage of the project, the above basic words plus a certain subset of pronouns will be prepared for demonstrating the feasibility of 1,500-word system.

In the course of extending the treatable linguistic expressions, we faced two major problems. One of them is the deterioration of performance when a larger grammar is used with all the target words dictionary. The second is the question of generalization of semantic representation. The heavy cost of unification of disjunctive feature structures was anticipated in the last research periods and the algorithm has been improved within the Active Chart Parser. The refinement of grammatical description was also attempted and the so-called medium grained grammar was examined as effective. However it turned out unavoidable to provide a newly organized grammar for the final target. The basic design has been launched and the new grammar system will be implemented in the next period.

Technical Publications : [Nagata 91-10-2] [Nagata 91-10-3]

2.2.2 Transfer from Japanese into English

The transfer module is also required to provide the necessary transfer rules for covering the target sentences as mentioned in the previous subsection. The feature structure rewriting system (RWS), the core transfer engine of the prototype system, was improved mainly for supporting an efficient environment for developing transfer rules. The major points are as follows:

1. Refinement of registering mechanism of transfer rules

2. Relaxations of the constraints in rule descriptions

A basic study based on a new paradigm of translation, which was performed in the last period, was summarized in a few papers. In this framework, a semantic network is supposed to reflect the context where an utterance is spoken. This network is divided and managed through the use of partitions such as theme vs. rheme or new vs. old information. For producing a proper dialogue sentence in the target language, a repartitioning of the network could be performed. This basic idea shows the necessity of introducing context processing into the speech translation system. Accordingly, a preliminary investigation on the specifications was performed for the discourse processing within the existing feature based system.

Technical Publications : [Suzuki 91-04] [Suzuki 91-10]

2.2.3 Generation of English

A new processing module was implemented based on a newly designed architecture with unification algorithm and it was regarded as feasible and scalable. The module uses two kinds of generation rules:

1) Phrasal Descriptions with constraints (PDs)

2) Morphological Synthesis rules

PDs are used for determining the syntactic structure. Some kinds of expressions are frequently used in goal-oriented dialogues such as conference registration task are idiomatized as fixed phrases. These cannot be generated from ordinary lexicon and general grammar rules. Instead, using a flexible production of surface expressions suited for performative expressions, etc. The processing is divided into three phases : generation of the basic syntactic structure, refinement of the structure and morphological generation. During the top-down traverse of a input feature structure in the first phase, the constraints are checked with unification. In some phrasal descriptions context sensitive constraints are considered for dynamical production of an expression affected by the previous utterance. In ordered to realize such a processing (also mentioned in the transfer phase), a Discourse Translation Manager (DTM) was designed and to be implemented. The detail will be reported in the next period.

In parallel with developing a new generation module, an interesting survey was performed for generating various kinds of surface utterance in English. It is on determining surface form for indirect speech acts. The study was performed for describing the syntactic and pragmatic factors involved, and a formalization is proposed from the view point of providing generation rules.

Technical Publications : [Kikui 91-10] [Fais]

2.3 Development of a Prototype System for Spoken Dialogue Translation

2.3.1 General Overview

The prototype system, SL-TRANS, was changed in several aspects, especially in the speech processing part. The former system processed only the pre-arranged voice date of a standard speaker(announcer). The SL-TRANS II was implemented so that it can accept experimenters' speech through a microphone. Besides, a speaker adaptation technique was combined and it enables the higher rate of speech recognition. A special hardware for HMM-LR is contributing to the high speed. Accompanied with these changes, a new demo-interface was deve-loped for displaying the content of the processing. It will be refined in the next period.

2.3.2 The the current status and the future plan

The final objectives expected in the last period are as follows.

1. From the viewpoint of allowable processing time, the whole run time should be within a minute in maximum.

2. The recognition rate of the input speech should be more than 85 % in sentence level with speaker adaptation.

3. The target words are expected to be 1,500, which was estimated with the statistics of frequent words in the collected corpus.

4. The analysis grammar should cover the basic polite expressions used for goal-oriented conversations. As the processing difficulties depend the expressions, their priority was decided through the survey of Japanese basic grammar.

5. During the course of translation, the selection of suitable expressions should be performed as possible, when there are alternations. Context sensitive decisions will also be needed.

The scale of the system has been enlarged. As for the translation, the coverage is expanded for all the twelve model dialogues in the demo system. We still have to overcome the problem of processing time. And a discourse management module is scheduled to be implemented. For the speech recognition module, a larger grammar should be prepared. In the final stage of the project, the system will be evaluated in a proper method. Thus, a study of evaluation procedure for spoken language translation system is important and a preliminary investigation was executed.

Technical Publications : [Morimoto 91-07] [Takezawa 91-09] [Takezawa 91-10]

2.4 Linguistic Database

2.4.1 Construction of Linguistic Database and its Management

The ATR Dialogue Database (ADD) has continuously grown to a large amount of dialogue corpora on several domains. Concerning the conference registration task (current target for the prototype system) and the inquiry to a travel agency, 200,000 words of dialogue texts were collected for each media, telephone and keyboard conversation. As a supplementary content of the ADD, Englishmorphological information is being added. Furthermore, the management system of the ADD was improved for various requirements of retrieval.

2.4.2 Knowledge Extraction and Lexical Investigation based on Corpora

Using the above mentioned linguistic database, various kinds of attempts were performed for extracting knowledge and investigating specific linguistic phenomena. In the last period, some experiments were performed on trying to extract transfer rules from the co-occurrences between a verbs and its subordinate nouns. As the result, it turned out that simple co-occurrence relationships cannot predict the suitable word selection in the target language. The problem is not so simple because the lexical choice is affected by various factors including contextual situation. For the present, however, it is worth investigating methodologies of extracting usable information semi/automatically from corpora, though the first attempts are limited on the cooccuurence within a sentence. For this purpose, hierarchical classification of nouns can be meaningful in a certain domain, taking ordinary lexicons into consideration. From this viewpoint, an automatic noun clustering technique was proposed as a procedure to create a knowledge base for MT system, and it will be applicable for actual translation rules.

Technical Publications : [Inoue 91-06] [Inoue 91-07] [Inoue 91-09]

2.4.3 Investigation of Some Linguistic Phenomena Specific to Japanese Dialogue

Investigations on some specific linguistic phenomena were continued for identifying the factors which affect the difference of meaning or usage. These analysis are also available for designing semantic representation of those linguistic phenomena, i.e. comparison, etc.

Technical Publications : [Tomokiyo 91-07] [Tomokiyo 91-10-1] [Tomokiyo 91-10-2]

2.5 Dialogue Modeling and Context-Based Inferences

It is our belief that the telephone interpretation system should be able to comprehend meaning in context. Major linguistic phenomena peculiar to Japanese spoken dialogues have been investigated from a linguistic viewpoint to construct a discourse-dialogue model that can be implemented on a computer. Among others, research topics concerning elliptical sentences, pragmatics, and intention are being studied, and the results obtained have been integrated step by step as a dialogue interpretation system. Considerable research has been focused on a plan recognition model for understanding a dialogue. Also a computational model for context processing using pragmatic constraints and circumstantial infrmation is being studied. A way to identify differently expressed noun phrases on the basis of domain knowledge is the first step toward dialogue meaning inference.

Technical Publications : [Iida 91-05] [Iida 91-12]

2.5.1 Plan Recognition Model for Dialogue Understanding

A multi-layered plan recognition model for dialogue structure construction and a method to predict the next utterance using the model was proposed.

The model employs three-typed universal pragmatics in addition to a set of the domain plan describing actions about the target world. The three-typed pragmatics are: "Interaction Plan", which describes a sequence of communicative acts for dialogue turn-taking, "Communication Plan", which determines how to execute or achieve an utterance goal or dialogue goals, and "Dialogue Plan" for establishing a dialogue construction, and a hierarchical order to apply these plans to recognize a goal of an input utterance can be determined in the line of Interaction Plan, Communication Plan, Domain Plan and then Dialogue Plan. The system implemented based on the model can efficiently construct appreciate dialogue structures step by step.

The method of next utterance prediction is based on the model. Using the constructed structure and the plans, the method can predict abstract information, about communicative act type and discourse entities of the next utterance, in the context-sensitive way. An experimental system to reduce the number of candidates of a speech recognition output was implemented and can improve the number of candidates using the contextual information and linguistic and pragmatic knowledge.

Technical Publications: [Iida 91-10][Yamaoka 91-09][Yamaoka 91-10] [Yamaoka 91-12]

2.5.2 Interpreting and Generating Elliptical Sentences in Machine Translation

(1) Identifying the Referents of Zero-Pronouns in Japanese based on Pragmatic Constraint Interpretation

A computational model has been developed to identify the referents of zero-pronouns by interpreting pragmatic constraints on the use of linguistic expressions under the context. The model exploits the constraints about honorific relationships, speaker's point of view and territory of information. These constraints are extracted from the usage of surface linguistic expressions. Therefore this method has the advantage of being less dependent on extra-linguistic knowledge in comparison with previous models of ellipsis resolution.

(2) A descriptive framework to Interpret and Generate Japanese Elliptical Sentences using CircumstantialInformation

A descriptive framework has been proposed to interpret and generate an elliptical / fragmentary sentence in the process of translating a sentence in a dialogue between Japanese and English.

In dialogues, especially Japanese dialogues, elliptical / fragmentary sentences are frequently used. Those sentences can be regarded as efficient in that they can convey a variety of informational contents according to the surrounding circumstances. Existing context processing models concentrate on the mechanism of interpreting elliptical sentences. However, previous methods do not work well in applying to the generation and translation process, because they fail in classifying the type of circumstances under which elliptical sentences can be used appropriately.

This framework aims at using efficient representations to describe the informational contents of elliptical sentences by guaranteeing that those efficient representations are embedded in the appropriate circumstances In addition, a variety of representations specified to different degrees of efficiency can be used to describe the same informational content.

To show that two representations convey the same informational content under an appropriate circumstance, a constraints is given as an equivalence relation between two representations with a condition on the type of circumstance. The process of interpreting a sentence can be modeled as replacing a representation with less efficient one through constraints by referring to circumstantial information as well as the generation process can be modeled as the converse process. Thus this framework has the advantage of applying to both interpretation and generation process. Moreover, the framework serves as an underlying model for the translation process at a variety of representations specified to different degrees of efficiency.

Some constraints has been exemplified to handle elliptical / fragmentary sentences in this framework. Those constraints exploit circumstantial conditions which concern social relationships, speaker's point of view and uniqueness of possible referents in a circumstance.

Technical Publications: [Dohsaka 92-03]

2.5.3 Model for Conversational Sentence Analysis

Developing the analysis method for conversation, as well as text, is very important for an application of the natural language processing. As for Japanese conversational sentences, the current constraint-grammar-based approach, such as HPSG, is not adequate due to its inflexibility in the treatment of the absence of complements, postpositions and verbs. The constraint grammar will reject such incomplete sentences because it is designed to restrict the structure of a sentence to only grammatical one; however, those sentences are completely adequate under the situation in which they are used.

In this research, we propose a new model, called information-based model, for the natural language analysis, which is flexible enough to analyze conversational sentences. The role of the grammar in this model is to extract the information from a sentence rather than to restrict the structure of a sentence. All possible structures are passed through to the interpretation module, and then the semantic interpretation of the information is done respecting the contextual and situational knowledge as well as linguistic knowledge. We formulate some interpretation rules and exemplify them with a couple of conversations.

Teechnical Publications: [Den 91-12][Den 92-01][Den 92-03-2]

2.6 Distributed Natural Language Processing and Machine Translation

In many conventional natural language processing, the analysis module is centered and the analysis results heavily affect following processes such as machine translation modules, foe example transfer and generation. An analysis-centered natural language processing does not always work well because analyzing relationships between words and phrases, or features on linguistic expressions are grasped from a certain monotonic view, in particular one grammar or one semantic category system. Various kinds of linguistic information must be handled simultaneously and satisfy any requirements made by each processing module which works as a sub-problem solver.

2.6.1 Transfer-Driven Machine Translation(TDMT)

The Transfer-Driven Machine Translation(TDMT) was proposed.

In many conventional machine translation systems, the analysis results heavily affect following processes such as transfer and generation. In contrast TDMT controls an entire translation by transfer, and the transfer engine and transfer knowledge, which is bilingual information, play central roles in this translation method, which aims at translation with high-speed, high quality, and high expandability. Although distributed cooperative processing is important to combine transfer knowledge systematically, as a first step sequential mechanism of TDMT was proposed. TDMT assumes that translation is performed by knowledge of various levels and their processes Transfer knowledge data is retrieved to determine how an input sentence should be translated. If necessary, when the retrieval fails the syntactico-semantic analyzer is activated. TDMT adopts the similarity calculation of Example-Based Machine Translation(EBMT)

Teechnical Publications: [Furuse 92-01][Furuse 92-03]

2.6.2 Example-Based Machine Translation (EBMT)

Study on Example-Based Machine Translation (EBMT) has been pursued in ATR since 1989 in order to overcome problems inherent in conventional Machine Translation. In EBMT, (1) a database which consists of examples (pairs of a source phrase or sentence and its translation) is prepared for translation knowledge; (2) an example whose source part is similar to the input phrase or sentence is retrieved from the example database; (3) by replacements of corresponding words in the target expression of the retrieved example, the translation is obtained.

We have adopted a hybrid architecture which incorporates EBMT as a subroutine into a conventional machine translation system. Because linguistic phenomena can be divided two parts: a regular part which is described well by rules; an irregular part which is not described well by rules. The former relates to model-oriented approaches and the latter relates to data-oriented approaches. It is more realistic to incorporate a data-oriented approach with a model-oriented approach than to use a single approach exclusively.

We aim to clarify what kind of phenomena EBMT is suited for and illustrate relationships between success rates and the example database parameters. We tested EBMT against selection problem of appropriate target expression for several linguistic phenomena which are frequent and polysemous: (Japanese to English direction), verbs, adnominal case particles and adverbial case particles; (English to Japanese direction), adnominal prepositions and adverbial prepositions. The numbers of examples are several thousands. The success rates are from about 80% to about 90%. EBMT feasibility has been proven through this successful experiments, so far. In addition, an interesting relationship between the weight computed with all examples in the example database and success rates, that is, "In general, the higher the weight, the better the quality." was observed in the experiments.

Technical Publications: [Sumita 91-06]

2.6.3 Multi-agent Mechanism for Natural-Language Processing

Working with multiple-agent systems is a basic research topic. Many approaches were examined. A package for supporting multi-processing on the Sequent parallel computer was developed and a manual written. The BEHOLDER series of algorithms for scheduling multiple tasks under limited resources were designed.

The problems of deciding which of multiple information sources to use and when to stop processing under uncertain conditions were investigated, using value-of-information theory and decision theory. This led to a new mathematics for explicitly representing uncertainty; the B-SURE system tore present situations and uncertain actions in multiple worlds; a system for planning with uncertain actions; and a new, optimal estimator for probabilities in discrete-outcome situations.The results are being applied to the TDMT system.

Finally, a top-down agent-based understanding system designed to solve specific problems in the interpretation demonstration was implemented. The system, known as ABDUCK, uses an extremely simple low-quality agent model to predict the illocutionary force, deep semantics, and surface semantics of the following utterance. This information is used to disambiguate candidates from speech recognition;determine the illocutionary force of utterances, including understanding syntactically-identical utterances such as "hai"; resolve zero-pronoun references;and understand "unagi-da" sentences. The quality of this understanding depends on the quality of the predictions. A high-quality intentional agent model is being prepared for use in this system.

2.6.4 Parallel Parsing Algorithms

The approach of parallel processing is effective for improving the performance of unification-based parsers. The characteristics of multiprocessore sytems vary according to their organization type such as tightly coupled or loosely coupled. So, it is necessary to consider whether the parallelism in application programs fits the characteristics of multiprocessor system. First, the parallelism in the processing of a unification-based parser is being studied.

A typical unification-based natural language processing system spends most of its processig times for graph unification. We are proposing a parallel quasi-destractive graph unification algorithm that avoids the problems on synchronizations for each recursive call into shared-arcs and on efficient management of lock /unlock scheduling of simultaneous accesses to global shared data structures.

Technical Publications: [Neuhaus 91-09]

2.7 Memory-Based Massively-Parallel Spoken Language Processing

It is very important to integrate speech proccesing and natural language processing for the sake of spoken language understanding and translation. A massively parallel constraint propagation network that propagates directed graphs to recognize spoken language inputs is being studied.

2.7.1 MT Architecture Based upon Massively-Parallel Graph-Based Constraint Propagation

We have developed an experiental architecture named MONA-LISA (Multimodal Ontological Neural Architecture for Linguistic Interactions and Scalable Adaptations) in which we introduced a massively parallel constraint propagation network that propagates directed graphs to recognize natural language input. The low-level signal input is desinged to be provided by the neural network acoustic recognition (TDNN and LPNN). The network is connected to a recurrent neural network which provides subsymbolic contextural recognition and predictions to the constraint propagation network. The model can be viewed as an architecture for symbolic and subsymbolic interactions during machine processing of massively-parallel cognitive activities as well as an experimental model for a new generation natural language processing and machine translation.

The integration of symbolic massive parallelism and subsymbolic neural net PDP processing provides a smooth a posteriori learning to the symbolic system and focused guided learning as well as strong constraints during recognition to the neural network. Through the use of graph-based representation and subsumption-ordered graph-unification, the graph-based constraint propagation network scheme solves the weaknesses of marker-passing-based massively-parallel natural language processing for handling structurally complex recognitions and dynamic configurational assignments. In essence, any modern linguistic theory that is representable using directed graphs can be captured in the network (such as LFG and HPSG).

As the result, the architecture provides the ability to handle strict and structured symbolic constraints during recognition while attaining a smooth contextual prediction ability applied with the least rigidity and learning of input regularities from actual samples. The domain researched so far in this project is natural language understanding for demonstration purposes;however, the architecture is expected to show equal strength in other modal channels such as visual inputs. We hope to extend the coverage of modal channels to other sensory inputs capturable by neural net recognition in the future. In fact, the input to our future systems is assumed to be through the real-world multi-dimensional modal channels of all human senses and the output of the system will provide signals to all human sensorty inputs, i.e., a creation of a virtual reality to the users of the systems.

Technical Publications: [Tomabechi 91-08][Tomabechi 91-10]

3. Research Staff

The research staff is mainly composed of members from the research institutes and laboratories which support ATR. Also, visiting foreign scientists are included. The following members have participated in language-related research until the end of the March in 1992.

Natural Language Understanding Department (Apr.,1986~Mar.,1992)

Name	*Position*	*Period*
Hitoshi Iida	Department Head	Apr., 1986 ~
Osamu Furuse	Senior Researcher	Feb., 1990 ~
Takashi Okada	Researcher	Apr., 1991 ~
Koji Dohsaka	Researcher	Mar.,1988 ~ Feb.,1992
Masaaki Nagata	Researcher	Mar., 1989 ~
Takayuki Yamaoka	Researcher	May.1989 ~ Mar.1992
Hiroshi Ohashi	Researcher	Jun., 1991 ~
Yasuharu Den	Visiting Researcher	Apr., 1991 ~
Yves Lepage	Visiting Researcher	Nov., 1991 ~
John K. Myers	Visiting Researcher	Sep., 1988 ~
Peter Neuhaus	Visiting Researcher	Apr., 1991 ~ Sep.,1991
Todd R.Kaufmann	Visiting Researcher	Jun.,1991 ~ Sep., 1991
Hideto Tomabechi	Visiting Researcher	Jun.,1991 ~ Nov.,1991
Susann LuperFoy	Visiting Researcher	Mar., 1992~May,1992

Data Processing Department (Apr.,1986~Mar.,1992)

Name	*Position*	*Period*
Tsuyoshi Morimoto	Department Head	Mar., 1987 ~
Terumasa Ehara	Supervisor	Jul., 1989 ~ Jun., 1991
Fumihiro Yato	Supervisor	Jun., 1991 ~
Noriyoshi Uratani	Supervisor	Jun., 1991 ~
Masami Suzuki	Senior Researcher	Aug., 1989 ~
Kenji Kita	Senior Researcher	Sep., 1987 ~
Toshiyuki Takezawa	Researcher	Oct., 1989 ~
Toshihisa Tashiro	Researcher	Sep., 1991 ~
Gen-ichiro Kikui	Researcher	Feb., 1990 ~
Junko Hosaka	Visiting Researcher	Oct., 1988 ~
Mutsuko Tomokiyo	Visiting Researcher	Apr., 1990 ~
Mark Seligman	Visiting Researcher	Jan., 1992 ~
Herbert S. Tropf	Exchange Researcher	Feb., 1992 ~

4. Research Facilities in the language-Related Departments

The two language-related departments have common computer systems which consists of VAX 8600/8800 with ULTRIX systems, various types of workstatons such as Symbolics 3675 / 3650 / 3620 / XL-1200 / Mac-Ivory, Xerox 1121, Explorer II, SUN 3/4, SPARC 2 , an iPSC parallel computer by INTEL Corp, and a Sequent / symmetry by Sequent Computer Systems, Inc. They are connected through the Ethernet network. Common Lisp and C are the major programming languages used in our departments.

List of Technical Publications of the Natural Language Understanding Department and the Data Processing Department

(April 1991 ~ March 1992)

Ref.ID	Title	Authors	Conference or Journal	page
DEN 91-12	情報伝達の観点から見た日常会話文の解析手法	伝　康晴 飯田　仁 [Y.DEN H.IIDA]	日本人工知能学会91年度合同シンポジウム「情報の意味と理解」 [JSAI Joint Symposium Dec. 1991]	1
DEN 92-01	情報伝達の観点から見た日常会話文の解析手法	伝　康晴 飯田　仁 [Y.DEN H.IIDA]	電子情報通信学会「自然言語処理の新しい応用」シンポジウム [IEICE Symposium , Jan., 1992]	11
DEN 92-03-1	ボトムアップチャート法に基づく並列文生成 [A Parallel Generation Mechanism based on the Bottom-up Chart Algorithm]	春野雅彦(京大) 伝　康晴 松本裕治(京大) 長尾　真(京大) [M.HARUNO Y.DEN Y.MATSUMOTO M.NAGAO]	情報処理学会 自然言語処理研究会 [WGNL Technical Meeting of IPSJ, 92-NL-88, pp.95-102, Mar. 1992]	21
DEN 92-03-2	情報伝達の観点から見た日常会話文の解析モデル [An Information based Analysis of Conversational Sentences]	伝　康晴 飯田　仁 [Y.DEN H.IIDA]	情報処理学会 第44回平成4年度前期全国大会講演論文集 [Proc. of IPSJ Spring, Meeting, Vol.3, pp.165-166, Mar. 1992]	29
DOHSAKA 92-03	ゼロ代名詞を取り巻く環境 [Circumstances surrounding the Usage of Japanese Zero Pronouns]	堂坂　浩二 [K.DOHSAKA]	情報処理学会 第44回平成4年度前期全国大会講演論文集 [Proc. of IPSJ Spring, Meeting, Vol.3, pp.195-196, Mar. 1992]	31

Ref.ID	Title	Authors	Conference or Journal	page
FURUSE 92-01	変換と解析の協調的処理による翻訳手法 -変換主導型翻訳手法- [Translation by Cooperation Between Transfer and Analysis -- Transfer-Driven Machine Translation --	古瀬　蔵 飯田　仁 [O.FURUSE H.IIDA]	情報処理学会 自然言語処理研究会 [WGNL Technical Meeting of IPSJ,NL-87-4, pp.27-34, Jan. 1992]	33
FURUSE 92-03	変換部主導型の対話翻訳機構 [Spoken-Dialogue Translation by Transfer-Driven Method]	古瀬　蔵 飯田　仁 [O.FURUSE H.IIDA]	情報処理学会 第44回平成4年度前期全国大会講演論文集 [Proc. of IPSJ Spring, Meeting, Vol.3, pp.127-128, Mar. 1992]	41
HOSAKA 91-05	対話データベースを利用した音声認識のための構文規則 [Constructing Syntactic Constraints for Speech Recognition Using Empirical Data]	保坂　順子 竹澤　寿幸 江原　暉将 [J.HOSAKA T.TAKEZAWA T.EHARA]	情報処理学会 自然言語処理研究会 [WGNL Technical Meeting of IPSJ, NL-91-83, pp.97-104, 1991]	43
HOSAKA 91-09	Utilizing Empirical Data for Postposition Classification Toward Spoken Japanese Speech Recognition	J.HOSAKA T.TAKEZAWA T.EHARA	Proc. of Eurospeech '91, pp.573-576, 1991	51
HOSAKA 91-10	話しことばの接続詞 [Classifying Conjunctions in Spoken Japanese]	保坂　順子 竹沢　寿幸 [J.HOSAKA T.TAKEZAWA]	情報処理学会第43回 平成3年度前期全国大会講演論文集 [Proc. of IPSJ Spring Meeting, pp.93-94, 1991]	53
IIDA 91-05	対話翻訳と高度自然言語処理	飯田　仁 [H.IIDA]	人工知能学会誌 [The Journal of JSAI Vol.6, No.3, pp.328-337, 1991]	55

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IIDA 91-10	文脈を考慮した対話の理解 [Context-Based Dialogue Understanding]	飯田　仁 [H.IIDA]	日本学術振興会 文字言語･音声言語の知能的処理第152委員会 [The Japan Society for the Promotion of Science, The No.152 Committee on Intelligent Processing of Literal Language and Spoken Language, 1991]	73
IIDA 91-12	音声対話の理解過程にみる情報の相互作用 [Interactions between Informations on Spoken Language Understanding Process]	飯田　仁 [H.IIDA]	日本人工知能学会91年度合同シンポジウム「情報の意味と理解」 [JSAI Joint Symposium Dec. 1991]	80
IIDA 92-01	ATR自動翻訳電話研究所紹介	飯田　仁 [H.IIDA]	日本機械翻訳協会 ニューズレター 「JAMTジャーナル」 [The Journal of JAMT, No.4, Jan. 1992]	90
INOUE 91-06	Automatic Noun Classification by Using Japanese English Word Pairs	N.INOUE	Proc. of ACL'91, pp.201-208, 1991	91
INOUE 91-07	階層的クラスタリング手法の訳語選択への応用 [Application of the Hierarchical Clustering Method to the Translated Word Selection]	井ノ上直己 (KDD) 竹澤　寿幸 江原　暉将 (NHK) [N.INOUE T.TAKEZAWA T.EHARA]	自然言語処理研究会言語理解とコミュニケーション研究会 [WGNLC Technical Meeting of IEICE and IPSJ, Jul., 1991]	101
INOUE 91-09	相手言語を考慮した概念抽出効果 [Effect of Considering Target Language in Concept Acquisition]	井ノ上直己 (KDD) 森元　逞 [N.INOUE T.MORIMOTO]	1991年電子情報通信学会秋季大会講演論文集 [Proc. of IEICE Fall Meeting, pp.6 49, 1991]	109
KIKUI 91-10	素性付き木構造を知識源とする文生成処理 [Feature Structure Configurator for Generation]	菊井玄一郎 [G.KIKUI]	情報処理学会第43回 平成3年度前期全国大会講演論文集 [Proc. of IPSJ Spring Meeting, pp.161-162, 1991]	110

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KITA 91-05-1	Incorporating LR Parsing into SPHINX	K.KITA Wayne H. Ward	Proc. of the ICASSAP'91 S5.5, pp.269-272, 1991	112
KITA 91-05-2	HMM Speech Recognition Using Stochastic Language Models	K.KITA T.KAWABATA T.HANAZAWA	Trans of ASJ Vol.12, No.3, pp.99-105, 1991	116
KITA 91-06	HMM-LR連続音声認識における音素環境依存型パーザの実現アルゴリズム [Algorithm for phoneme-environment-dependent parser in continuous speech recognition using HMM-LR method]	永井 明人 嵯峨山茂樹 北 研二 [A.NAGAI S.SAGAYAMA K.KITA]	電子情報通信学会 音声研究会 [WGSP Technical Meeting of IEICE, SP-91-23, pp.41-48, 1991]	141
KITA 91-07-1	Continuous Speech Recognition Using Two-Level LR Parsing	K.KITA T.TAKEZAWA T.MORIMOTO	Trans. of IEICE, Vol.E74, No.7,pp.1806-1810, 1991	149
KITA 91-07-2	Processing Unknown Words in Continuous Speech Recognition	K.KITA T.EHARA T.MORIMOTO	Trans. of IEICE Vol.E74 No.7,pp.1811-1816, 1991	154
KITA 91-08	GLP Parsing in Hidden Markov Model	K.KITA	"Generalized LR Parsing" Ed.M.Tomita	160
KITA 91-10	HMM-LR連続音声認識装置の開発と性能評価 [Hardware Development for HMM-LR Continuous Speech Recognition System]	永井 明人 北 研二 花沢 利行 鈴木 雅実 鹿野 清博 [A.NAGAI K.KITA T.HANAZAWA M.SUZUKI K.SHIKANO]	日本音響学会 平成3年度 秋季研究発表会 講演論文集 [Proc. of ASJ Fall Meeting, 1-5-23, pp.45-46,1991]	172
KITA 91-10	上位文法カテゴリへの到着可能性照合機構を備えたLR解析法とその音声認識への応用 [LR Parsing with a Category Reachability Test Applied to Speech Recognition]	北 研二 森元 逞 嵯峨山茂樹 [K.KITA T.MORIMOTO S.SAGAYAMA]	日本音響学会 平成3年度 秋季研究発表会 講演論文集 [Proc. of ASJ Fall Meeting, 2-P-17, pp.171-172,1991]	174
KUREMATSU 91-05	自動翻訳ー自動翻訳電話に向けて	榑松 明 [A.KUREMATSU]	通信工業 [Tsushin Kogyo, Vol.31, No.5, pp.20-25]	176

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KUREMATSU 91-06-1	自動翻訳電話	榑松　明 [A.KUREMATSU]	テレビジョン学会誌 [The Journals of the Institute of Television Engineers of Japan Vol.45, No.6, 1991]	187
KUREMATSU 91-06-2	自動翻訳のゆくえ	榑松　明 [A.KUREMATSU]	電気通信 [Denki Tsushin Vol.6, Jun., 1991]	208
KUREMATSU 92-01	Future Perspective of Automatic Telephone Interpretation	A.KUREMATSU	Trans. of IEICE, Vol.E75, No.1, pp14-19, Jan., 1992	221
KUREMATSU 92-03	Perspective View of Multi-Media Cross Language Communication	A.KUREMATSU	Proc. of International Workshop on Advanced Communications and Applications, pp.15-23, Munchen(Germany), Mar. 1992]	236
MATSUO 91-10	日本語対話文解析部における計算コストの削減方法(形態素解析部と構文解析部の分離とインターフェース) [A Method for Reducing the Computational Cost in Japanese Analysis Module]	松尾　秀彦[TIS] 谷田　泰郎(TIS) 永田　昌明 竹沢　寿幸 [H.MATSUO Y.TANIDA M.NAGATA T.TAKEZAWA]	情報処理学会第43回平成3年度前期全国大会講演論文集 [Proc. of IPSJ Spring Meeting, pp.149-150, 1991]	245
MORIMOTO 91-05	音声言語処理 [Spoken Language Processing]	森元　　逞 [T.MORIMOTO]	人工知能学会誌 [The Journal of JSAI Vol.6, No.3, pp.321-327, 1991]	247
MORIMOTO 91-06	自動翻訳電話の技術課題	森元　　逞 [T.MORIMOTO]	NTTアドバンストテクノロジ「技術移転」 [The Journal of "Technical Transfer", Vol.14,No.6,pp.26-28, NTT advanced Technology, 1991]	254

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MORIMOTO 91-07	Integration of Speech Recognition and Language Processing in a Japanese to English Spoken Language Translation System	T.MORIMOTO K.SHIKANO K.KOGURE H.IIDA A.KUREMATSU	Trans. of IEICE, Vol.E74, No.7, pp 1889-1896, 1991	258
NAGATA 91-10-1	疎結かつ階層的な音声言語インターフェイス:音声認識用文法と言語処理文法の段階的統合を目指して [A Loosely-Coupled Hierarchical Interface between Speech Recognition and Natural Language Processing]	永田　昌明 竹沢　寿幸 森元　逞 [M.NAGATA T.TAKEZAWA T.MORIMOTO]	情報処理学会第43回 平成3年度前期全国大会講演論文集 [Proc. of IPSJ Spring Meeting, pp.549-550, 1991]	285
NAGATA 91-10-2	日本語単一化文法における語順変化と主題化の効率的な解析手法 [Efficient Analysis Method of Word-Order Variation and Topicalization-Based Japanese Grammar]	衛藤　純司(日本IR) 永田　昌明 [J.ETOH M.NAGATA]	情報処理学会第43回 平成3年度前期全国大会講演論文集 [Proc. of IPSJ Spring Meeting, pp.131-132, 1991]	287
NEUHAUS 91-09	Unification-Based Parsing on Increasing Levels of Parallelism	P.NEUHAUS O.FURUSE H.IIDA	WGNL Technical Meeting of IPSJ, NL-85, pp.41-48, 1991	289
SUMITA 91-06	Experiments and Prospects of Example-Based Machine Translation	E.SUMITA H.IIDA	Proc. of ACL'91, pp.185-192, 1991	297
SUZUKI 91-04	Lexical Choice in Dialogue Translation	鈴木　雅実 [M.SUZUKI]	第2回日英計算言語学 共同研究ワークショップ [Proc. of The 2nd Japanese-English Joint Workshop on Computational Linguistic,1991]	305
SUZUKI 91-07	第5回ACL European Chapterに参加して [Report on Fifth Conference of the European Chapter of the ACL]	鈴木　雅実 中村　順一 [M.SUZUKI J.NAKAMURA]	電子情報通信学会 言語理解とコミュニケーション研究会 (共催情報処理学会自然言語処理研究会) [WGNLC Technical Meeting of IEICE and IPSJ, pp.1-6, Jul., 1991]	313

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SUZUKI 91-10	Repartitioning A Semantic Network for Translating Dialogue Utterances	M.SUZUKI	Proc. of Second Japan-Australia Joint Symposium on Natural Language Processing, pp.244-251, 1991	319
TAKEZAWA 91-05	Linguistic Constraints for Continuous Speech Recognition in Goal-Directed Dialogue	T.TAKEZAWA K.KITA J.HOSAKA T.MORIMOTO	Proc. of the ICASSP'91 S12.5, pp.801-804, 1991	329
TAKEZAWA 91-07	音素バランス文と案内タスクの連続音声データベース	板橋　秀一(筑波大) 速水　悟(電総研) 竹澤　寿幸 小林　哲則(早大) [S.ITABASHI S.HAYAMIZU T.TAKEZAWA T.KOBAYASHI]	講演討論会「音声認識の新潮流をめざして」(電子情報通信学会第2種研究会) [Open Forum on a New Wave of Speech Recognition IEICE Working Group Jul., 1991]	333
TAKEZAWA 91-09	日英音声言語翻訳実験システムSL-Trans2における音声対訳処理 [Interaction Processing in SL-TRANS2: an Experimental System for Translating Japanese Speech to English]	竹澤　寿幸 森元　逞 榑松　明 [T.TAKEZAWA T.MORIMOTO A.KUREMATSU]	1991年電子情報通信学会秋季大会講演論文集 [Proc. of IEICE Fall Meeting, pp.6 53-54, 1991]	337
TAKEZAWA 91-10	日英音声言語翻訳実験システムSL-TRANS2 [SL-TRANS2: an Experimental System for Translating Japanese Speech to English]	竹沢　寿幸 大倉　計美 森元　逞 嵯峨山茂樹 榑松　明 [T.TAKEZAWA K.OHKURA T.MORIMOTO S.SAGAYAMA A.KUREMATSU]	日本音響学会 平成3年度 秋季研究発表会 講演論文集 [Proc. of ASJ Fall Meeting, 1-5-24, pp.47-48,1991]	339

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TAKEZAWA 91-11	自然言語処理における統合の諸相 [Aspects of Integration in Natural Language Processing]	橋田　浩一 (ICOT) 竹沢　寿幸 [K.HASHIDA T.TAKEZAWA]	コンピュータ·ソフトウエア 1991.11月号 [The Magazine of "Computer Software", Vol.11, 1991]	342
TOMABECHI 91-08	Graph-based Constraint Propagation in Massively Parallel Memory: Toward Massively-pallel Natural Language	H.TOMABECHI H.IIDA T.MORIMOTO A.KUREMATSU	Proc. of Parallel Processing for Artificial Intelligence(PPAI) Workshop / IJCAI'91 Sydney,Australia, Aug., 1991	360
TOMABECHI 91-10	超並列制約伝播による自然言語処理方式 [Graph-based Constraint Propagation for Massively-Parallel Natural Language Processing]	苫米地英人 [H.TOMABECHI]	情報処理学会第43回 平成3年度前期全国大会講演論文集 [Proc. of IPSJ Spring Meeting, pp.143-144, 1991]	368
TOMOKIYO 91-07	日本語会話文における比較表現の分析 [Investigation of comparative form in Japanese Spoken Dialogue]	友清　睦子 鈴木　雅実 [M.TOMOKIYO M.SUZUKI]	電子情報通信学会 言語理解とコミュニケーション研究会 [WGNLC Technical Meeting of IEICE, pp.151-158, Jul., 1991]	370
TOMOKIYO 91-10-1	日本語会話文における複合文の調査と分析 [Investigation of Compound Sentence in Japanese Spoken Dialogue]	友清　睦子 鈴木　雅実 [M.TOMOKIYO M.SUZUKI]	電子情報通信学会 技術研究報告会 [WGNLC Technical Meeting of IEICE, PRU-91-62, pp.1-7, 1991]	378
TOMOKIYO 91-10-2	Classification of Adverbial Clause Markers in Spoken Japanese	M.TOMOKIYO T. MORIMOTO	Proc. of Natural Language Processing Pacific Rim Symposium, pp.113-120, Singapore, 1991	385
UEDA 92-01	タイプ付素性構造主導型生成 [Typed-Feature-Structure-Directed Generation]	上田　良寛 [Y.UEDA]	情報処理学会論文誌 [Trans. of IPSJ, Vol.33, No.1, 1992]	393

Ref.ID	Title	Authors	Conference or Journal	page
YAMAOKA 91-09	Dialogue Interpretation Model and Its Application to Next Utterance Prediction for spoken Language Processing	Y.YAMAOKA H.IIDA	Proc. of Eurospeech '91, pp.849-852, 1991	423
YAMAOKA 91-10	プラン認識アルゴリズムの制御 [A Controlable Plan Recognition Algorythm]	山岡　孝行 飯田　仁 [T.YAMAOKA H.IIDA]	情報処理学会第43回 平成3年度前期全国大会講演論文集 [Proc. of IPSJ Spring Meeting, pp.231-232, 1991]	427
YAMAOKA 91-12	問合せ対話における名詞句表現の構成 [Noun-Phrase Surface-Form Composition Model in Inquiry Dialogue]	山岡　孝行 飯田　　仁 有田　英一(三菱電機) [T.YAMAOKA H.IIDA E.ARITA]	電子情報通信学会 技術研究報告会 [WGNLC Technical Meeting of IEICE, NLC-91-42, pp.41-48, 1991]	429

IPSJ	Information Processing Society of Japan
IEICE	The Institute of Electronics, Information and Communication Engineers
ASJ	The Acoustical Society of Japan
ACH	Association for Computers and theHumanities
ALLC	Association for Literary and Linguistic Computing
ACL	Association for Computational Linguistics
JSAI	Japanese Society for Artificial Intelligence
WGNL	Working Group of Natural Language
WGNLC	Working Group of Natural Language Understanding and Communication
WGSP	Working Group of Speech Processing
ICASSP	International Conference on Acoustics,Speech and Signal Processing
ECAI	European Conference on Artificial Intelligence
PRICAI	Pacific Rim International Conference on Artificial Intelligence
ICSLP	Internaitonal Conference on Spoken Language Processing
COLING	International Conference on Computional Linguistics
IJCAI	International Joint Conference on Artificial Intelligence

List of ATR Technical Reports of the Natural Language Understanding Department and the Data Processing Department

(April 1991 ~ March 1992)

Number	Title	Authors	Date
TR-I-0215	単語の意味カテゴリーを用いた係り受け整合度の平滑化	Terumasa Ehara	1991-05
TR-I-0217	日本語形態素解析の細則	Terumasa Ehara Eiko Tatikawa Hisako Yamada	1991-05
TR-I-0218	言語データベースから抽出した知識データの分布	Terumasa Ehara	1991-05
TR-I-0220	協調的な目標指向型対話における文型式と発話者の意図との対応--文型式を重視した情報伝達行為の分類--	Tosiyuki Sadanobu Takayuki Yamaoka Hitoshi Iida	1991-07
TR-I-0224	音声認識結果の信頼性評価	Naohiro Kobayashi Toshiyuki Takezawa	1991-09
TR-I-0225	Report on Working discussions and ongoing research	Christian Boitet	1991-09
TR-I-0228	Tools for Monitoring Parallel Lisp Programs	Todd Kaufmann	1991-09
TR-I-0229	Unification-Based Passing on Increasing Levels of Parallelism	Peter Neuhaus Osamu Furuse Hitoshi Iida	1991-10
TR-I-0231	Research Activities of the Natural Language Understanding Department and the Knowledge and Database Department for November 1989 through March 1991	Tsuyoshi Morimoto Hitoshi Iida	1991-10
TR-I-0232	HMM-LR連続音声認識における音素コンテキスト依存型文脈自由文法の生成アルゴリズム	Hideaki Kikuchi Akito Nagai Shigeki Kageyama Kenji Kita	1991-12

Number	Title	Authors	Date
TR-I-0234	データ構造共有型複製方式準破壊型単一化手法 Quasi-Destructive Graph Unification with Structure-Sharing	Hideto Tomabechi	1991-12
TR-I-0235	Determining Surface Form for Indirect Speech Act in English	Laurel Fais Gen-ichiro Kikui	1991-12
TR-I-0237	Evaluation of Metrics for Text Comparison with Implication for Recognition	Nathalie Delfosse Tsuyoshi Morimoto	1992-01
TR-I-0240	音声認識のための構文規則ガイドブック	Masaaki Nagata Junji Eto Junko Hosaka	1992-02
TR-I-0241	SL-TRANSにおける音声認識のための構文規則の拡張	Junko Hosaka Toshiyuki Takezawa	1992-02
TR-I-0244	The Role of Identifiability and Accessibility in Accounting for Japanese Zero Pronouns	Kohji Dohsaka	1992-02
TR-I-0245	音声認識部、形態素解析部、構文解析部の品詞·語彙区切りの相違点の調査	Masaaki Nagata Junko Hosaka Masami Suzuki	1992-02
TR-I-0240	音声認識のための構文規則ガイドブック	Masaaki Nagata Junji Eto Junko Hosaka	1992-02
TR-I-0246	HMM-LRユーザーズ·マニュアル	Kenji Kita	1992-02
TR-I-0247	The Beholder TOOLBOX Manual Part1	John. K.Myers	1992-03
TR-I-0248	HMM-LR連続音声認識における適応的探索方法	Kunifumi Okamoto Kouichi Yamaguchi Shigeki Sagayama Kenji Kita	1992-03
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