

UNITED STATES PATENT AND TRADEMARK OFFICE

BEFORE THE PATENT TRIAL AND APPEAL BOARD

GOOGLE LLC,
Petitioner,

v.

BUFFALO PATENTS, LLC,
Patent Owner.

Case No. IPR2023-01386
U.S. Patent No. 6,904,405

Declaration of Shauna L. Wiest Regarding *Suhm*

I, Shauna L. Wiest, state and declare as follows:

I. Introduction

1. I have prepared this Declaration in connection with Google LLC's ("Petitioner") Petition for *Inter Partes* Review of U.S. Patent No. 6,904,405, which I understand will be filed concurrently with this Declaration.

2. I am a senior research analyst with the Research & Information Services team at Finnegan, Henderson, Farabow, Garrett & Dunner, LLP located at 3300 Hillview Avenue, Palo Alto, CA 94304 ("Finnegan").

3. I am over eighteen years of age, and I am competent to make this Declaration. I make this Declaration based upon my own personal knowledge, and my professional knowledge of library science practices.

4. I earned a Master of Science in Library Science degree from the University of North Carolina at Chapel Hill in 1999, and a Bachelor of Arts in Political Science degree from the University of California at San Diego in 1989. I have worked as a law librarian for over eighteen years. I have been employed in the Research & Information Services Department at Finnegan since 2021. Before that, from 2000-2015, I was employed as a Law Librarian at Stoel Rives LLP, and from 2015-2016, I was employed as a Competitive Intelligence Specialist for Nossaman LLP.

II. Standard Library Practice for Receiving, Cataloging, Shelving, and Making Materials, including Conference Publications, Publicly Available

5. I have knowledge of and experience with standard library practices regarding receiving, cataloging, shelving, and making materials, including conference publications, available to the public. I am fully familiar with and have knowledge of and experience with the Machine-Readable Cataloging (MARC) system, an industry-wide standard that libraries use to catalog materials.

6. The MARC system was developed during the 1960s to standardize bibliographic catalog records so they could be read by computers and shared among libraries. By the mid-1970s, MARC had become the international standard for cataloging bibliographic materials and is still used today. Many libraries provide public access to their MARC records via the Internet and/or their electronic cataloging systems at the library. In a MARC record, each field provides specific information about the cataloged item, including how materials are held and made available to the public.

III. MARC Records

7. The MARC record system uses a specific three-digit numeric code (“field tags”) (from 001-999) to identify each field in a catalog record. For example, MARC field tag 008 provides the six-digit date the item was received

and catalogued (*Date entered on file*). The first six characters of field tag 008 are always in the “**YYMMDD**” format. Descriptions and definitions of all of the character positions of field tag 008 are outlined here:

<https://www.loc.gov/marc/bibliographic/bd008a.html>

8. As is relevant to this Declaration, MARC field tag 245 identifies the title and statement of responsibility for the work. MARC field tag 260 identifies the place of publication, name of publisher, and date of the publication. MARC field tag 362 sets for the dates of publication and/or sequential date(s) of publication designation for a work. And MARC field tag 550 provides information about the current and former issuing bodies of a continuing resource including notes containing editing, compiling, or translating information.

9. Based on standard library practice, when a library receives an item, it generally stamps (and/or labels) the item with the library name, barcode, often with a date that is within a few days or weeks of receipt. Next, the library will catalog the item within a matter of a few days or weeks of receiving it. As a general practice, cataloguing is centralized and performed by a cataloguing department within a library or university setting. In certain circumstances the catalogued item may be subsequently sent to a library location within the library or university setting where it may be stamped and/or labeled after it has been catalogued.

10. Generally, after an item is cataloged, the public may access the item by searching a library catalog, browsing the library shelves, and either requesting or electronically accessing the item from the library. Standard library practice is to make the item available to the public within a few days or weeks of cataloging it.

IV. Cataloguing of Conference Proceedings (Monograph v. Serial Treatment)

11. Conference proceedings often consist of collections of papers presented at a meeting, program, symposium or conference held during a specific time-period in a designated geographic place. Bibliographic control of conference proceedings to promote public discovery and access has long been a priority of catalogers. Although some ongoing conference publications are catalogued as serials (if a conference publication exhibits continuity, regularity, and evidence of seriality), cataloguing a conference publication as a monograph is preferred for public retrieval purposes. Cataloguing a conference publication as a monograph is an optimal way for the public to easily access the conference resource, and its papers, by searching for a unique title and other descriptive data elements added by the cataloger. Monographs are efficient for public searching and offer the public an easier way to discover what an institution holds, while changes to the record (unlike serials) are not an issue. However, sometimes conference proceeding cataloging is performed

serially, and the MARC record will be updated after each serial publication is received.

V. Public Availability of *Suhm*

12. This Declaration relates to the dates of receipt and public availability of the following reference: B. Suhm, Petra Geutner, Thomas Kemp, Alon Lavie, Laura Tomokiyo-Mayfield, A.E McNair, Ivica Rogina, Tanja Schultz, T Sloboda, W Ward, Monika Woszczyna, and Alex Waibel, “JANUS: Towards Multilingual Spoken Language Translation” (“*Suhm*”) in Proceedings of the ARPA Spoken Language Systems Technology Workshop, Austin, TX, January 22-25, 1995 at pages 221-226. (“ARPA Publication”). I understand that *Suhm* has been submitted as Exhibit 1008 in this proceeding. That same reference is appended to my Declaration as **Appendix A**.

13. As detailed below, I have reviewed the print reference, public holdings information, and Iowa State University Library MARC record for the ARPA Publication containing *Suhm* to determine the date of public availability for this reference.

14. **Appendix A** to this Declaration is a true and accurate copy of the cover page, front matter, and date stamp for the copy of the Proceedings of the ARPA Publication containing *Suhm* held by the Iowa State University Library. The date stamp on **Appendix A** on the first page indicates that the ARPA Publication

containing *Suhm* was stamped by the Iowa State University Library of Science and Technology (also known as Iowa State University Library) on April 22, 1997.

Based on my experience as a librarian, and because the operative date of receipt set forth in the MARC record (**Appendix C**) is before April 22, 1997 (as explained in paragraphs 17-19), the April 22, 1997 date stamp was likely applied after the Iowa State University Library initially received and catalogued the ARPA Publication.

15. The ARPA Publication (**Appendix A**) states: “This document includes ‘copies of reports prepared for the ARPA SLT Workshop. Included are reports from ARPA sponsored programs and other materials prepared for use at the workshop.’” **Appendix A** also includes the specific pages in the publication comprising *Suhm*. The ARPA Publication (**Appendix A**) states that it was “Sponsored by Advanced Research Projects Agency Software & Intelligent Systems Technology Office” and distributed by “Morgan Kaufmann Publishers, Inc., 340 Pine Street, 6th Floor, San Francisco, CA 94104.”

16. **Appendix B** to this Declaration is a true and accurate copy of the Iowa State University Library public catalog record for its copy of the ARPA Publication containing *Suhm*, which was downloaded from https://quicksearch.lib.iastate.edu/permalink/01IASU_INST/174tg9m/alma990009235080102756 on September 7, 2023. The Iowa State University Library public catalog record sets forth the public holdings and onsite location information for

members of the public seeking a physical copy of the ARPA Publication containing *Suhm*. The public catalog record indicates that the resource’s “genre” is conference papers and proceedings, the “summary holdings” are dated 1995, and it is associated with the January 22-25, 1995 ARPA conference. It lists the author as “Advanced Research Projects Agency, Software & Intelligent Systems Technology Office,” and notes that the ARPA Publication was “Distributed by Morgan Kaufmann Publishers.” Additionally, the ARPA Publication containing *Suhm* is available in the Parks Library General Collection at Call Number TK7895 S65 .A77x. Based on my experience as a librarian, the public catalog record (**Appendix B**) references the ARPA Publication containing *Suhm* (**Appendix A**).

17. **Appendix C** to this Declaration is a true and accurate copy of the Iowa State University Library MARC record for its holdings of the ARPA Publication containing *Suhm*, which was downloaded from https://quicksearch.lib.iastate.edu/discovery/sourceRecord?vid=01IASU_INST:01IASU&docId=alma990009235080102756&recordOwner=01IASU_INST on September 7, 2023. The Iowa State University Library MARC record field tag 245 identifies the full title statement for the work as: Proceedings of the ARPA Spoken Language Systems Technology Workshop. MARC field tag 362 sets forth the dates of publication and/or sequential date designation as Jan. 22-25, 1995. MARC field tag 260 notes “Distribut[ion] by Morgan Kaufmann Publishers” and field tag

550 notes “Sponsor[ship] by Advanced Research Projects Agency, Software & Intelligent Systems Technology Office.”

18. **Appendix C** confirms the fixed data elements of MARC field tag 008 as 951024c19959999xxuar11a0engd. As discussed above, the first six characters “951024” are in typical “**YYMMDD**” format and indicate that the Proceedings of the ARPA Publication containing *Suhm* was first catalogued by the Iowa State University Library on October 24, 1995. Based on my experience as a librarian, the MARC record (**Appendix C**) references the ARPA Publication containing *Suhm* (**Appendix A**).

19. Based on the information in **Appendices A, B, and C** the Proceedings of the ARPA Publication containing *Suhm* was initially received, processed, and catalogued by the Iowa State University Library within a matter of a few days or weeks of October 24, 1995. This compilation of evidence shows that public visitors to the Iowa State University Library could have accessed the Proceedings of the ARPA Publication containing *Suhm* within weeks of October 24, 1995. Therefore, members of the general public could have discovered and learned the location of the Proceedings of the ARPA Publication containing *Suhm* by searching the Iowa State University Library catalog or by asking a library staff member and being directed to the Parks Library General

Collection at Call Number TK7895 S65 .A77x within a few weeks of October 24, 1995.

VI. Conclusion

20. In signing this Declaration, I understand it will be filed as evidence in a contested case before the Patent Trial and Appeal Board of the United States Patent and Trademark Office. I understand I may be subject to cross-examination in this case and that cross-examination will take place within the United States. If cross-examination is required of me, I will appear for cross-examination within the United States during the time allotted for cross-examination.

21. I declare that all statements made herein of my knowledge are true, that all statements made on information and belief are believed to be true, and that these statements were made with the knowledge that willful false statements and the like so made are punishable by fine or imprisonment, or both, under Section 1001 of Title 18 of the United States Code.

Executed on September 11, 2023.

A handwritten signature in black ink, appearing to read 'Shauna L. Wiest', written over a horizontal blue line.

Shauna L. Wiest

APPENDIX A

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Proceedings of the

ARPA

Spoken Language Systems

Technology Workshop



Barton Creek Resort Conference Center
Austin, TX
January 22-25, 1995

Sponsored by
Advanced Research Projects Agency
Software & Intelligent Systems Technology Office

This document contains copies of reports prepared for the ARPA SLT Workshop.
Included are reports from ARPA sponsored programs and other materials prepared for use at the workshop.

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JANUS: TOWARDS MULTILINGUAL SPOKEN LANGUAGE TRANSLATION

B. Suhm¹, P. Geutner², T. Kemp², A. Lavie¹, L. Mayfield¹, A. E. McNair¹, I. Rogina², T. Schultz²,
T. Sloboda², W. Ward¹, M. Woszczyzna¹, A. Waibel^{1,2}

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ABSTRACT

In our effort to build spoken language translation systems we have extended our JANUS system to process spontaneous human-human dialogs in a new domain, two people trying to schedule a meeting. Trained on an initial database JANUS-2 is able to translate English and German spoken input in either English, German, Spanish, Japanese or Korean output. To tackle the difficulty of spontaneous human-human dialogs we improved the JANUS-2 recognizer along its three knowledge sources acoustic models, dictionary and language models. We developed a robust translation system which performs semantic rather than syntactic analysis and thus is particularly suited to processing spontaneous speech. We describe repair methods to recover from recognition errors.

1. Introduction

JANUS [1, 2] has been among the first systems attempting to provide spoken language translation. While the previous JANUS-1 system processed syntactically wellformed read speech over a 500 word vocabulary, JANUS-2 operates on spontaneous human-human dialogs in a scheduling domain with vocabularies exceeding 2000 words. Currently, English and German spoken input can be translated in either English, German, Spanish, Japanese or Korean output. Work is in progress to add Spanish and Korean as input languages.

This paper reports on the current status of the system and ongoing efforts to extend and improve the recognition component. Then, we describe our new approach to robust translation of spoken language. We briefly describe and compare the alternative approach to parsing and translation we pursue, based on a generalized robust LR parser and an ILT. Finally we report on efforts to detect erroneous system output and provide interactive methods to recover from such errors.

2. Current Status of JANUS

2.1. Data Collection

Data collection to establish a large database of spontaneous human-human negotiation dialogs in English and German has started about 18 months ago. In the meantime, several sites in Europe, the US and Asia have adopted the Scheduling task

under several research projects and funding sources. Since the same calendars and data collection protocols are used the data elicited shares the same domain and procedural constraints.

English Scheduling		
recorded	dialogs 1984	words 505 K
transcribed	1826	460 K
German Scheduling		
recorded	dialogs 734	words 158 K
transcribed	534	115 K
Spanish Scheduling		
recorded	dialogs 340	words 79 K
transcribed	256	70 K
ATIS3		
transcribed	n./a.	250 K

Table 1: Comparison of Databases (as of December 1994)

Table 1 summarizes the current status of data collection. Since Scheduling utterances typically consist of more than one sentence, there is already more data available for English Scheduling than ATIS¹. More data collection will establish databases in size at least comparable to ATIS for all languages.

In Spanish, we have explored two different data collection scenarios: To allow only one person to speak at a time the *push-to-talk* scenario requires the speaker to push a button while talking to the system. The *cross-talk* scenario allows speakers to speak simultaneously without push button. The speech of each dialog partner is recorded on separate channels.

2.2. System Overview

The main system modules are speech recognition, parsing, discourse processing, and generation. Each module is lan-

¹The about 18000 utterances in English Scheduling correspond to some 30000 sentences.

guage-independent in the sense that it consists of a general processor that applies independently specified knowledge about different languages.

The recognition module decodes the speech in the source language into a list of sentence candidates, represented either as a word lattice or Nbest list. At the core of the machine translation components is a language independent representation of the meaning, which is extracted from the recognizer output by the parsing module. As last step, the final language independent representation is sent to the generator to be translated in any of the target languages. Figure 1 shows the system architecture.

After parsing, a discourse processor can be used to put the current utterance in the context of previous utterances, opening possibilities to integrate the speech and natural language processing components of the system to resolve parsing ambiguities and dynamically adapt the vocabulary and language model of the recognizer based on the current discourse state.

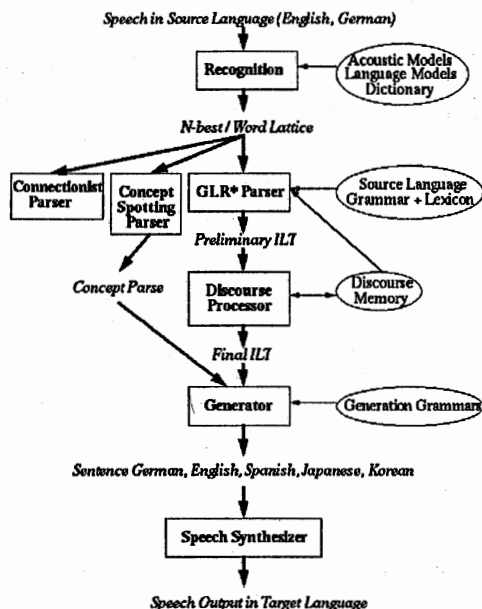


Figure 1: System Diagram

We explore several approaches for the main processes. For example, we are experimenting with TDNN, MS-TDNN [3], MLP, LVQ [4], and HMM's [5, 12] for acoustic modeling; n-grams, word clustering, and automatic phrase detection for language modeling [6]; statistically trained skipping parsing [7, 8], neural net parsing [9] and concept spotting parsing [10] for extracting the meaning; and statistical models

as well as plan inferencing for identification of the discourse state [11]. This multi-strategy approach should lead to improved performance with appropriate weighting of the output from each strategy.

2.3. Recognition Performance Analysis

The baseline JANUS-2 recognizer can be described as follows:

- *Preprocessing*: LDA on melscale fourier spectrum and additional acoustic features (power, silence)
- *Acoustic modeling*: LVQ-2 or phonetically tied SCHMM, no cross-word triphones, explicit noise models
- *Decoder*: Viterbi search as first pass, followed by a word-dependent Nbest search, standard word bigram language model, word lattice output

Current recognition results on the English, German and Spanish Spontaneous Scheduling Task (ESST, GSST, SSST) can be seen in table 2.

	ESST	GSST	SSST
Word Accuracy	66%	72%	61%

Table 2: JANUS-2 baseline recognition performance

The low absolute recognition accuracies are due to the challenging nature of human-human spontaneous speech. In the official evaluation of the German VERBMOBIL project on the GSST task, the JANUS-2 decoder outperformed all other participating systems. In addition, recent evaluations on the Switchboard task confirm that human-human dialogs are much more difficult to recognize than human-machine spontaneous speech (like ATIS). Participating systems achieved word accuracies between 30% and 50%.

Analysis shows that human-human dialogs (like Scheduling or Switchboard) are more difficult to recognize than human-machine dialogs (e.g. ATIS). Perplexities lie between 35 and 90 for ESST, SSST and GSST, and somewhat over 100 for Switchboard. Additionally, human-human dialogs are significantly more disfluent [8]. Large variations in speaking rates and strong coarticulation between words contribute significantly to the difficulty of recognizing human-human spontaneous speech.

3. Improving the Recognition Component

We describe efforts to improve the recognition component along its major knowledge sources acoustic models [12], dictionary [13] and language models [14].

3.1. Data-Driven Codebook Adaptation

We developed methods aimed at automatic optimization of the number of parameters for the semi-continuous phonetically tied HMM used in JANUS-2. Usually, a fixed number of codebook vectors is assigned to each of the phonemes. However, as the available training data differs between phonemes and the size of the feature space phonemes cover varies greatly, constant codebook size leads to suboptimal allocation of resources.

We have therefore suggested [12] to adapt the codebook size of each phoneme according to the amount and the distribution of the training data, similar to [15]. During training, the size of the codebook is incrementally increased. Some quality criterion determines when to stop the process of increasing the codebook. We compared a *variance* criterion based on the average distance between data points and their nearest codebook vector with a *prediction* criterion which tries to capture how well the modeling of the recognizer can predict unseen data.

Model	Codebook Size	Word Accuracy
baseline	4600	66.9%
variance	4201	69.9%
prediction	1677	67.8%

Table 3: Results for Codebook Adaptation (GSST)

Table 3 compares recognition accuracies and codebook sizes of the baseline models, with models automatically adapted using the variance and prediction criterion. As can be seen, codebook adaptation leads to significant error reduction if the same number of parameters is used. The number of parameters can be reduced by 40% with still better performance than the baseline system.

3.2. Dictionary Learning

Due to the enormous variability in spontaneous human-human dialogs creating adequate dictionaries with alternative pronunciations is crucial [16]. However, hand tuning and modifying dictionaries is time consuming and labor intensive. Pronunciations of a word should be chosen according to their frequency. Modifications of the dictionary should not lead to higher phonetic confusability after retraining. Therefore we have proposed [13] a data-driven approach to improve existing dictionaries and automatically add new words and pronunciation variants whenever needed.

The learning algorithm requires transcripts for the whole training set and a phoneme confusability matrix of the speech recognizer used. First, phonetic transcriptions for all appearances of each word are generated by help of a phoneme recognizer.

Then, variants which are infrequent or which would lead to erroneous training of confusable phonemes are eliminated. Finally, the acoustic models are retrained allowing for the newly acquired pronunciations variants.

As can be seen in table 4, our algorithm for adapting and adding phonetic transcriptions to a dictionary improves the recognition accuracy of the decoder significantly and leads to performance that is comparable to the context dependent results (cf. table 2). The baseline decoder for these experiments uses 69 context independent phoneme models. Evaluation using context dependent models is in progress.

Dictionary	Word Accuracy
baseline	61.7%
adapted	65.6%

Table 4: Results Dictionary Learning (GSST)

3.3. Morpheme Based Language Models

Based on our scheduling databases we noticed that in morphologically rich languages such as German and Spanish, dictionaries grow much faster with increasing database size, compared to English (cf. figure 2). This is due to the large number of inflections and compound words. One way to limit this growth with increasing dictionary sizes is to use other base units than words.

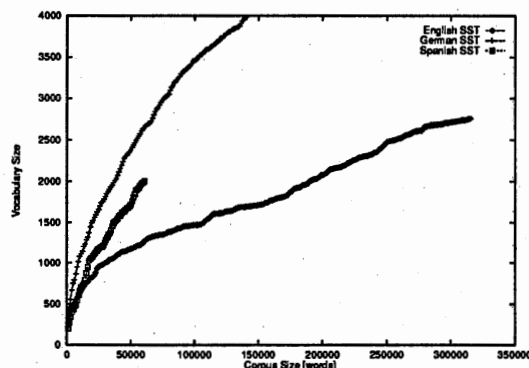


Figure 2: Vocabulary Growth

We compared three different decomposition methods:

- strictly *morpheme* based decomposition, e.g. weggehen (to go away) → weg-geh-en, Spracherkennung (speech recognition) → Sprach-er-kenn-ung
- decomposition in *root forms*, e.g. weggehen (to go away) → weggeh@, Spracherkennung (speech recognition) → Spracherkenn@

- combination of strictly morpheme based decomposition and root forms

Table 5 shows dictionary size, bigram perplexity and recognition accuracy using the respective decomposition method, based on 250 GSST dialogs. As can be seen, all decomposition methods significantly reduce vocabulary size and perplexity. The impact on recognition accuracy is still small. This may be due to the fact that the acoustic modeling suffers from smaller units and thus deteriorate the gain in the language model. In a real interface, however, this reduction in vocabulary growth leads to a reduction of new words. Further research will focus on finding more efficient and acoustically less confusable decompositions automatically, and also test the impact on translation.

	Dictionary	Perplexity	Accuracy
Baseline	3821	88	64.7%
Morphemes	2391	46	65.4%
Root Forms	3205	79	63.5%
Combined	2998	59	65.1%

Table 5: Comparison of Decomposition Methods (GSST)

4. Concept Based Speech Translation

We have developed a robust translation system based on the information structures inherent in the appointment scheduling task being performed, described in detail elsewhere [10]. The basic premise is that the structure of the information conveyed is largely independent of the language used to encode it. Our system tries to model the information structures in a task and the way these structures are realized in words in various languages. This system is an extension of the Phoenix Spoken Language System [18]. It uses the Phoenix parser to parse input into slots in semantic frames, and then uses these frames to generate output in the target language.

4.1. The Parser

Unlike individual words, semantic units used in a task domain are not language specific. Based on transcripts of scheduling dialogs, we have developed a set of fundamental semantic units in our parse which represent the different concepts a speaker would use. For instance, a typical *temporal* token could have *date* as subtoken, which could in turn consist of *month* and *day* subtokens. The *temporal* could be part of a statement of unavailability.

In contrast to previous speech translation systems, we presently don't perform syntactic analysis. Speaker utterances, as decoded by the recognizer, are parsed into semantic chunks which are concatenated without grammatical rules.

Original utterance:

THAT SATURDAY I'M NOT SURE ABOUT BUT YOU SAID
YOU MAY BE BACK IF YOU THINK YOU'LL BE BACK
THE THIS SUNDAY THE TWENTY EIGHTH I COULD SEE
YOU AFTER ELEVEN AM ON THAT IF YOU'RE BACK

Translated:

Saturday that's not so good for me Sunday the twenty eighth works for me after eleven a.m. (ENGLISH)

El sábado no me va demasiado bien pero el domingo veintiocho me va bien después de las once de la mañana. (SPANISH)

Samstag könnte ich nur zur Not aber Sonntag der Achtundzwanzigste geht bei mir ganz gut nach elf Uhr morgens. (GERMAN)

Figure 3: Translation Example

This approach is particularly well suited to parsing spontaneous speech, which is often ungrammatical and subject to recognition errors. This approach is more robust than requiring well-formed input and reliance on syntactic cues provided by short function words such as articles and prepositions.

4.2. The Generator

The generation component of the system is a simple left-to-right processing of the parsed text. The translation grammar consists of a set of target-language phrasings for each token, including lookup tables for variables like numbers and days of the week. When a lowest-level token is reached in tracing through the parse, a target-language representation is created by replacing tokens with templates for the parent token, according to the translation grammar. The result is a meaningful, although terse translation, which emphasizes communicating the main point of an utterance. An examples is illustrated in figure 3.

4.3. Results

We have implemented this system for bi-directional translation between English, German and Spanish in our scheduling task. Table 4 shows the performance of parser and subsequent generator on transcribed data. Evaluation of the system based on speech decoded by the JANUS-2 recognizer is still underway.

	Parsed from		Translated into
	token	utterance	utterance
English	95.6%	90.0%	90.2%
German	92.4	89.6	87.3
Spanish	88.8	58.3	82.2

Figure 4: End-to-End evaluation on transcribed data

One disadvantage of this approach is the telegraphic and repet-

itive nature of the translations. This could be overcome by providing multiple translation options for individual tokens in the target-language module, different levels of politeness, etc. However at present we feel that it is sufficient for intelligible communication.

5. GLR* Parser

In addition to the concept based Phoenix parser we pursue GLR* as robust extension of the Generalized LR Parser. It attempts to find maximal subsets of the input that are parsable, skipping over unrecognizable parts of the input sentence [7]. By means of a semantic grammar GLR* parses input sentences into an interlingua text (ILT) as language independent representation of the meaning of the input sentence, described in more detail elsewhere (e.g. [8]).

Compared to Phoenix parses the ILT generated by GLR* offers greater level of detail and more specificity, e.g. different speaker attitudes and levels of politeness. Thus, translation based on ILTs is more natural, overcoming the telegraphic and terse nature of concept based translation.

A drawback of GLR* was that it expected input segmented into sentences for efficiency reasons. However, typical Scheduling utterances consist of 2-3 sentences. To integrate the parser with the speech decoder, we developed methods which extend the parsing capabilities from single sentences to multi-sentence utterances. We extended the grammar with a high-level rule that allows the input utterance to be analyzed as a concatenation of several sentences and developed two methods to constrain the number of sentence breaks that are considered by the parser. The first is a heuristic which prunes out all parses that are not minimal in the number of sentences. The second is a statistical method to disregard potential sentence breaking points that are statistically unlikely.

For the English analysis grammar, time efficiency thus improved by about 30%. As an additional benefit, the parse quality improved because strange sentence breaks are rejected in favor of a more reasonable location.

6. Handling Unreliability

Although research has boosted performance of speech recognition and spoken language translation technology, recognition and translation errors will persist. To build a system for use in real applications we need repair methods to recover from errors in a graceful and unobtrusive way. We have developed a speech interface for repairing *recognition* errors by simply respeaking or spelling a misrecognized section of an utterance. While much speech "repair" work has focused on repairs within a single spoken utterance [19], we are concerned with the interactive repair of errorful recognizer hypotheses [20].

6.1. Identifying Errors

To be able to repair an error its location has to be determined first. We pursue two strategies to identify misrecognitions as subpieces of the initial recognizer hypothesis.

The *automatic subpiece location* technique requires the user to respeak only the errorful subsection of the (primary) utterance. This (secondary) utterance is decoded using a vocabulary and language model limited to substrings of the initial erroneous hypothesis. Thus, the decoding identifies the respoken section in the hypothesis. Preliminary testing showed that the method works poorly if the subpiece to be located is only one or two words long. However, this drawback is not severe since humans tend to respeak a few words around the error.

A second technique uses *confidence measures* to determine for each word in the recognizer hypothesis whether it was misrecognized. First, we applied a technique similar to Ward [21], which turns the score for each word obtained during decoding into a confidence measure by normalizing the score and using a Bayesian updating technique based on histograms of the normalized score for correct and misrecognized words. Since we found this not to work well on our English scheduling task, we are currently developing different methods to compute confidence measures based on decoder, language model and parser scores.

6.2. Robust Speech Repair

After locating and highlighting erroneous sections in the recognizer hypothesis misrecognitions are corrected.

The *spoken hypothesis correction* method uses Nbest lists for both the initial utterance and the respoken section. The Nbest for the highlighted section of the initial utterance is rescored using scores from decoding the secondary utterance. Depending on the quality of the Nbest lists, most misrecognitions can be corrected.

The *spelling hypothesis correction* method requires the user to spell the highlighted erroneous section. A spelling recognizer decodes the spelled sequence of letters. By means of a language model we restrict the sequence of letters to alternatives found among the Nbest from the located section.

To date, we have evaluated our methods over sentences from the Resource Management task. Table 6 shows the improvements in sentence accuracy, based on recordings from one speaker of the February and October 1989 test data. We selected a subset of erroneous utterances; therefore the accuracy of the baseline system is significantly lower than the 94% performance our system achieves on the whole test set. The results indicate that repeating or spelling a misrecognized subsection of an utterance can be an effective way to repair recognition utterances.

No Repair (baseline)	63.1%
Respeak	83.8%
Spell	88.5%
Respeak + Spell	89.9%

Table 6: Improvement of Sentence Accuracy by Repair

7. Conclusions

We have made significant advances towards building a multi lingual translation system for spontaneous human-human dialogs. Beyond speech recognition of spontaneous speech JANUS provides a framework to investigate important areas like robust parsing, machine translation of spoken language and developing methods to recover from recognition and parsing errors. To achieve acceptance in real applications, we have to embed the spoken language technology in a sensible and useful user interface that is carefully designed around human factors and common needs. To be flexible and robust, such interfaces should not only recognize speech but also recognize other communication modalities, provide freedom from headset and push-buttons, allow for graceful recovery from errors and miscommunications, know what they don't know, and model what the user does or doesn't know [23].

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APPENDIX B

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