UNITED STATES PATENT AND TRADEMARK OFFICE

BEFORE THE PATENT TRIAL AND APPEAL BOARD

GOOGLE LLC,
Petitioner,

v.

BUFFALO PATENTS, LLC,
Patent Owner.

Case No. IPR2023-01387
U.S. Patent No. 8,204,737

Declaration of Shauna L. Wiest Regarding *Vo*

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. 8,204,737, which I understand will be filed concurrently with this Declaration.

2. I am currently 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 310 sets forth the current stated publication frequency of either an item or an update to an item MARC field tag 362 sets for the dates of publication and/or sequential date(s) of publication designation for a work. MARC field tag 515 provides any numbering or publishing pattern peculiarities for the resource. And MARC field tag 991 provides local holdings information to assist the public in identifying the location of a desired resource.

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

V. Public Availability of *Vo*

12. This Declaration relates to the dates of receipt and public availability of the following reference: Minh Tue Vo and C. Wood, "Building an application framework for speech and pen input integration in multimodal learning interfaces," *("Vo*") found within the 1996 IEEE International Conference on Acoustics, Speech, and Signal Processing Conference Proceedings, Atlanta, GA, USA, May 7-10, 1996, pp. 3545-3548 Volume 6 ("1996 IEEE Publication"). I understand that *Vo* has been submitted as Exhibit 1014 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 Library of Congress MARC record for *Vo* 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 (including binding), title page, front matter, table of contents for the copy of the 1996 IEEE Publication containing *Vo* held by the Library of Congress. **Appendix A** also includes pages 3545-3548, which comprise the publication,

"Building an application framework for speech and pen input integration in multimodal learning interfaces," *i.e., Vo*. **Appendix A** notes that the "Copyrights Manager" is the "IEEE Service Center," with an address of "445 Hoes Lane, P.O. Box 1331, Piscataway, NJ 008855-1331" and has a Copyright date of 1996. It also explains that the conference was "Sponsored by the Signal Processing Society of the Institute of Electrical and Electronics Engineers, Inc."

15. **Appendix D** to this Declaration is a true and accurate copy of search results received from a search of the Library of Congress public catalog for records on International Conference on Acoustics, Speech, and Signal Processing (ICASSP) conferences, which was downloaded from https://catalog.loc.gov/vwebv/search?searchArg=IEEE+International+Conference+on+Acoustics%2C+Speech%2C+and+Signal+Processing&searchCode=GKEY%5E*&searchType=0&recCount=100&chkdTitles=0&sortBy=RELEVANCE&recPointer=0. The search results indicate that the Library of Congress public catalog holds 8 separate records for ICASSP conferences held since 1976. Some records are only associated with an ICASSP conference for one year, whereas other records are associated with ICASSP conferences from multiple years.

16. **Appendix B** to this Declaration is a true and accurate copy of the Library of Congress public catalog record for its copy of the 1996 IEEE Publication containing *Vo*, which was downloaded from

https://lccn.loc.gov/98655050 on September 5, 2023. I confirmed with librarians at the Library of Congress that the public catalog record is the correct and operative cataloguing record for the Library of Congress holdings for the 1996 IEEE Publication containing *Vo*.

17. The Library of Congress public catalog record (**Appendix B**) sets forth the public holdings and onsite location information for members of the public seeking a physical copy of the 1996 IEEE Publication containing *Vo*. The public catalog record indicates it is a "Conference proceedings" record and, specifically, identifies the meeting name as "ICASSP (Conference)," and the main title as "International Conference on Acoustics, Speech, and Signal Processing." The public catalog record notes that the publications associated with the record are (i) published "annual[ly]," (ii) published in 1995 and 1996, (iii) issued in five (5) or more parts, and (iv) should be requested in the Jefferson or Adams Building Reading Rooms at Call Number TK7882.S65 I37a. This information indicates serial cataloguing, as discussed above in paragraph 11. The public catalog record also notes a publication location in Piscataway, NJ, with copyright dates of 1995 and 1996, and IEEE as the publisher. It also explains that the conference was "sponsored by the Signal Processing Society of the Institute of Electrical and Electronics Engineers." Based on my experience as a librarian, the information provided in the public catalog record (**Appendix B**) indicates that the record

references multiple publications, including publications dated 1995 and 1996. Further, based on my experience as a librarian, the public catalog record (**Appendix B**) references the 1996 IEEE Publication containing *Vo* (**Appendix A**).

18. **Appendix C** to this Declaration is a true and accurate copy of the Library of Congress MARC record for its holdings of the 1996 IEEE Publication containing *Vo*, which was downloaded from https://catalog.loc.gov/vwebv/staffView?searchId=24554&recPointer=0&recCount=25&searchType=7&bibId=11501706 on September 5, 2023. I confirmed with librarians at the Library of Congress that the MARC record (**Appendix C**) is the correct and operative cataloguing record for the Library of Congress holdings for the 1996 IEEE Publication containing *Vo*. The Library of Congress MARC record sets forth information for members of the public seeking a physical copy of the 1996 IEEE Publication containing *Vo*. The Library of Congress MARC record field tag 245 sets forth the full title statement for the resource as: **a** Conference proceedings / |**c** the … International Conference on Acoustics, Speech, and Signal Processing; sponsored by the Signal Processing Society of the Institute of Electrical and Electronics Engineers. MARC field tag 260 sets forth the place of publication, publisher, and publication dates as: |**a** Piscataway, NJ : |**b** Institute of Electrical and Electronics Engineers, |**c** c1995-c1996. MARC field tag 310 indicates that the IEEE International Conference on Acoustics, Speech, and Signal

Processing Conference Proceedings is an annual publication. MARC field tag 362 indicates the sequential dates of publication for the resource as 1995-1996, indicating publication dates of both 1995 and 1996. MARC field tag 515 denotes numbering peculiarities for the resource as being issued in 5 or more parts. Finally, MARC field tag 991 sets forth the Library of Congress local holdings and call number information for the 1996 IEEE Publication containing *Vo* as General Collection, Call Number TK7882.S65 I37a (Serials). Based on my experience as a librarian, the MARC record (**Appendix C**) references the 1996 IEEE Publication containing *Vo* (**Appendix A**).

19. **Appendix C** confirms the fixed data elements of MARC field tag 008 as 950630d19951996njuar 1 0eng c. Data elements "d1995-1996" in field tag 008 indicates (i) new issues of the publication associated with this MARC record have published after 1996, or that a change in author or title has caused a successive MARC entry record to be created, and (ii) the first volume of the publication associated with the MARC record is dated 1995 and the last publication is dated 1996. As discussed above in paragraph 7, the first six characters "950630" are in typical "**YYMMDD**" format and indicate that the first volume of publication for the resource associated with the MARC record was received by The Library of Congress on June 30, 1995.

20. **Appendix E** to this Declaration is a true and accurate copy of the IEEE Xplore webpage associated with the 1995 IEEE ICASSP, which was downloaded from https://ieeexplore.ieee.org/xpl/conhome/3469/proceeding. This webpage indicates that the 1995 IEEE ICASSP was held on May 9-12, 1995. As explained above, the 1995 publication associated with the MARC record in **Appendix C** was received by The Library of Congress on June 30, 1995, a mere six weeks after the conclusion of the 1995 IEEE ICASSP.

21. **Appendix F** to this Declaration is a true and accurate copy of the cover, title and verso pages, and table of contents of the 1997 IEEE International Conference on Acoustics, Speech, and Signal Processing Proceedings, held by the University of Wisconsin at Madison, Kurt F. Wendt Library. **Appendix G** to this Declaration is a true and accurate copy of the University of Wisconsin at Madison MARC record for its holdings of the 1997 IEEE International Conference on Acoustics, Speech, and Signal Processing Proceedings, held by the University of Wisconsin at Madison, Kurt F. Wendt Library, which was downloaded from https://search.library.wisc.edu/catalog/999858382602121 (*See:* Library Staff Details -staff view) on August 22, 2023. **Appendix F** indicates that the conference related to the 1997 IEEE International Conference on Acoustics, Speech, and Signal Processing was held on April 21-24, 1997. The MARC record in **Appendix G** confirms the fixed data elements of MARC field tag 008 as

970707d19971997cauar 1 0eng d, indicating that the 1997 IEEE International Conference on Acoustics, Speech, and Signal Processing Proceedings was received on July 7, 1997, a mere ten weeks from the end of the 1997 conference.

22. Based on standard library practices and to assure the public can access desired resources, if the 1996 IEEE Publication containing *Vo* was received sometime in 1997, such information would be documented in the MARC record and/or the public catalog record in both the public holdings information and MARC field tags 008, 260, 362, or 991. Because no such information was included in the MARC record, it is my opinion that the Library of Congress received the 1996 IEEE Publication containing *Vo* before January 1, 1997.

23. Based on the information set forth in the appendices attached to this Declaration, standard library practices, and my experience as a librarian, the 1996 IEEE Publication containing *Vo* was received by the Library of Congress shortly after May 10, 1996 (the date of the 1996 conference), between June 1996 and July 1996, and at the latest by December 31, 1996.

24. Based on standard library practices and my experience, *Vo* would have been processed and catalogued by the Library of Congress within a matter of a few days or weeks after it was received in June or July of 1996 (or, at the latest by December 31, 1996). And, based on my experience as a librarian, **Appendix A** is the document the Library of Congress received, as recorded in the MARC record

(**Appendix C**), and was cataloged by the Library of Congress within a matter of a few days or weeks after receiving it.

25. Accordingly, *Vo* would have been made available to the public within a few days or weeks of being checked-in and catalogued. The public could have accessed *Vo* no later than a few days or weeks after December 31, 1996 by (i) searching the Library of Congress online catalog by, for example, searching by title, author, and/or subject keywords, or (ii) by asking a library staff member and being directed to General Collections at Call Number TK7882.S65 I37a.

VI. Conclusion

26. 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.

27. 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.

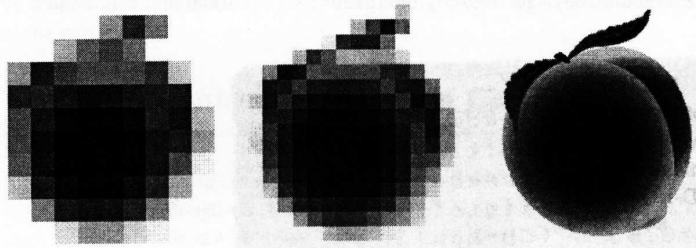


Shauna L. Wiest

APPENDIX A

1996 IEEE ICASSP VOL. 6





The 1996 IEEE International Conference on Acoustics, Speech, and Signal Processing Conference Proceedings

Sponsored by the Signal Processing Society of the Institute of Electrical and Electronics Engineers

May 7-10, 1996
Marriott Marquis Hotel
Atlanta, Georgia, USA

The 1996 IEEE International Conference on Acoustics, Speech, and Signal Processing Conference Proceedings

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BUILDING AN APPLICATION FRAMEWORK FOR SPEECH AND PEN INPUT INTEGRATION IN MULTIMODAL LEARNING INTERFACES

Minh Tue Vo and Cindy Wood

Interactive Systems Laboratories
Carnegie Mellon University
Pittsburgh, PA 15213-3890, U.S.A.
Email: {tue, caw}@cs.cmu.edu

ABSTRACT

While significant advances have been made in recent years to improve speech recognition performance, and more recently, gesture and handwriting recognition as well, speech- and pen-based systems have still not found broad acceptance in everyday life. One reason for this is the inflexibility of each input modality when used alone. Human communication is very natural and flexible because we can take advantage of a multiplicity of communication signals working in concert to supply complementary information or increase robustness with redundancy. In this paper we present a multimodal interface capable of jointly interpreting speech, pen-based gestures, and handwriting in the context of an appointment scheduling application. The interpretation engine based on semantic frame merging correctly interprets 80% of a multimodal data set assuming perfect speech and gesture/handwriting recognition; in the presence of recognition errors the interpretation performance is in the range of 35-62%. A dialog processing scheme uses task domain knowledge to guide the user in supplying information and permits human-computer interactions to span several related multimodal input events.

1. INTRODUCTION

Speech recognition is coming of age and is being incorporated into modern computer systems. Although pen input processing is still in its infancy compared to speech, pen-based systems have begun to appear in the form of personal digital assistants (PDAs) and pen laptops. However, current speech and pen systems are still not very popular and did not live up to the promise of bringing the power of computing to the population at large. Their failure is partly due to users' frustrations generated by inadequate recognition performance, especially for handwriting recognition. However, this failure is also an indication that current systems are too inflexible for the important task of facilitating the use of computers for everyone, especially people who may not be computer-literate. PDA users frustrated by gesture and handwriting recognition errors would naturally wish they could talk to their machines to correct the errors or simply to avoid having to write everything. Likewise users of speech-enabled systems quickly find out that there are tasks that cannot be conveniently expressed by spoken commands but would be enormously simplified by the ability to point to or circle objects on the screen in addition to speaking commands. These intuitive assertions were confirmed by a user study [1] conducted at Carnegie Mellon University, in which people interacting with a computer much preferred a combination of both speech and gestures over speech or gestures alone. Our own experiments with a calendar interface also show that when given a choice, people tend to use the communication method (or methods) most suitable for the task at hand and freely switch among available methods.

Research efforts at our Interactive Systems Laboratories (Carnegie Mellon University and University of Karlsruhe) are focused on producing a sensible and useful user interface by integrating multiple input modalities, rather than building better speech and gesture recognizers alone. An interface supporting the kind of highly flexible interaction we envision must be capable of integrating information from both speech and non-verbal input sources to arrive at a correct understanding of complete multimodal events. Some of our initial works along this line have been reported in previous publications [2][3][4][5]. This paper presents an approach to integrating speech, pen-based gestures, and handwriting, in the context of an appointment scheduling application. A multimodal interpretation engine jointly interprets information from all input sources by merging semantic frames. A domain-independent dialog processor maintains context information across input events. We evaluated the system with data collected in a user study conducted using the Wizard-of-Oz paradigm [6].

2. JEANIE: A MULTIMODAL CALENDAR

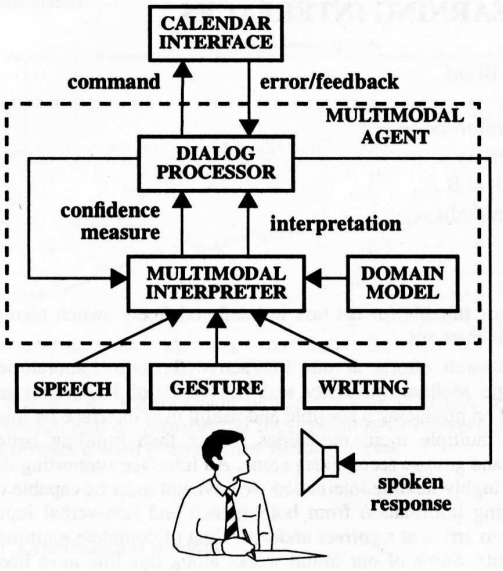
We have developed a prototype of a multimodal interface for an appointment scheduling program. A person using our Jeanie multimodal calendar can employ any combination of spoken input, gesturing with a pen on a touch-sensitive screen, or handwritten words to interact with the system. In typical scenarios, the user might say "Schedule a meeting on Monday" while at the same time drawing a box on the calendar to indicate where the new meeting should be inserted; write words on the newly scheduled meeting to annotate it; draw a cross on another meeting to cancel it; or point to a meeting and say "Reschedule this on Tuesday" or simply draw an arrow from that meeting to the new time slot on Tuesday.

An earlier version of this system was described in [4]. The interpretation engine in that system was based on an information-theoretic connectionist network [7] capable of incremental learning during use. Although that approach worked well for the preliminary system designed to explore multimodal interpretation, we found that the connectionist network was difficult to scale up when the task domain representation was significantly expanded in the new version. A semantic frame merging scheme was found to work much better on the larger and more complex action space, although we had to sacrifice the incremental learning capability.

Figure 1 shows a block diagram of the Jeanie system. The individual modality components (speech, gesture, and handwriting recognizers) are separate modules that can be replaced with ease. The calendar interface embodies domain knowledge and serves to isolate task-specific components. The heart of the system is the multimodal agent which interprets user input and performs

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FIGURE 1: The Jeanie multimodal calendar



requested operations through the calendar interface. This section describes the calendar interface and the modality processors. The next section presents the multimodal agent and the multimodal interpretation algorithm.

2.1. Calendar Interface

The object-oriented calendar interface is based on ICAL, a public domain calendar program developed by Sanjay Ghemawat at MIT. We extended ICAL to support a weekly view and client/server operations. The multimodal agent acting as a client of the calendar interface can query it for information such as gesture contexts (see Section 2.2) and ask the interface to carry out scheduling operations which include adding, removing, and changing appointments and notes.

2.2. Modality Processors

Inputs captured by a speech recorder and a touch-sensitive screen are processed by separate recognizers for each modality. Semantic frames extracted from recognizer outputs are merged by the multimodal interpreter to arrive at a unified interpretation as described in Section 3.

Speech. Our speech recognition subsystem is based on the recognition front-end of the JANUS speech translation system [8][9] which is capable of processing speaker-independent, spontaneous speech and was trained on human-human dialogs in the appointment scheduling domain. On a small data set of 128 utterances collected in user study experiments, the word recognition rate was 82%; if we take into account insertion, substitution, and deletion errors, the word accuracy is 76%.

Gesture. In earlier implementations we employed a gesture recognition module [2] based on a TDNN classifier [10]. The present implementation increases flexibility by decomposing gestures into sequences of strokes recognized as basic shapes such as lines, arcs, arrows, circles, crosses... Each gesture component is augmented by *gesture contexts* indicating spatial relationships between the gesture and nearby objects in the calendar interface. The current implementation of the gesture recognizer uses simple template matching and does not perform well on noisy input. A version based on the handwriting recognition algorithm described below is being developed and should greatly improve recognition performance.

Handwriting. Our handwriting recognizer developed by Stefan Manke at University of Karlsruhe based on the MS-TDNN [11] is capable of processing writer-independent, continuous (cursive) handwriting [12] at a recognition rate of over 90% on a 20,000-word vocabulary. Handwriting recognition is performed only when the gesture recognizer cannot identify the input strokes as basic shapes. This simple heuristic works fine for true cursive handwriting but can mistake writing strokes for gestures when it encounters printed block letters. This was unfortunately the case for some of the data collected in our user study; in addition most of the collected handwriting included uppercase letters for which the recognizer was not trained. These facts combined with gestures that were not adequately covered by templates led to a combined gesture/handwriting recognition rate of only 32%.

3. MULTIMODAL INTERPRETATION BY SEMANTIC FRAME MERGING

The multimodal interpreter is responsible for producing an interpretation of user intent (i.e., a command to send to the calendar interface) from the output of the modality processors. We represent this interpretation as a frame consisting of slots specifying pieces of information such as the action to carry out or the date and time of a meeting. Recognition output from the modality processors are parsed into partially filled frames that are merged together to produce the combined interpretation. The underlying algorithm is domain-independent although the encoding of information in frames must necessarily depend on the task domain.

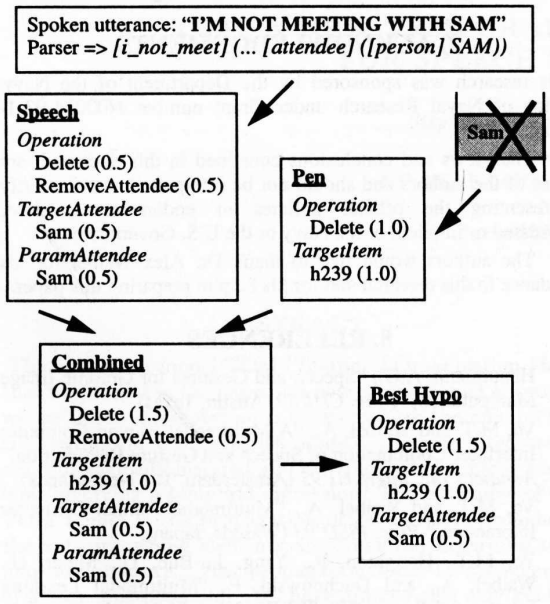
This frame merging technique represents a much extended and improved implementation of the approach sketched in [2]. It leads to uniform handling of high-level information from all input sources, which is very important for modularity and extensibility. To add another input modality we need only provide a module to convert low-level recognizer output to a partially filled frame to be merged with others. In addition, context information can be retained across input events by merging with previous interpretation frames as implemented in our dialog processor (Section 3.3).

3.1. Parsing Inputs From Individual Modalities

The text string output from the speech recognizer is processed by the Phoenix semantic parser developed by Ward [13]. The parser compiles a grammar specifying semantically meaningful fragments of text into an efficient recurrent transition network that identifies these fragments in the recognized utterance. It is capable of skipping unknown words and unmatched fragments and can therefore deal gracefully with ungrammatical sentences. The concepts identified by the parser serve to determine which slots of the speech frame should be filled and what scores should be assigned to the slot values.

The gesture/handwriting recognizers produce a sequence of gesture/handwriting components. The same gesture shape may mean different things depending on the calendar context surrounding it; for instance, a line across an appointment may indicate deletion but the same line falling on an empty time slot may visually describe a new meeting to be scheduled. In the present implementation of Jeanie, gestures and handwritten words are encoded into frames according to simple hard-coded heuristics. The development of a grammar to drive this process may become necessary for more complex gestures and operations.

FIGURE 2: Example of interpretation using frames



3.2. Merging Frames and Extracting Hypotheses

Frames are merged by taking the union of the sets of values filling each slots and adding corresponding scores. Embedded frames are merged recursively. This produces an aggregate frame encoding all alternative interpretation hypotheses. The multimodal interpreter extracts these hypotheses in descending order of aggregate scores and sends the best ones to the dialog processor. Figure 2 shows an example taken from our actual test data.

3.3. Dialog Processing

A dialog between a user and the system can span several related multimodal events, where each event may contain one or more spoken utterances and/or pen gestures. The dialog processor maintains a single frame per dialog to hold all information obtained since the start of the dialog; new information from each additional input event is merged into this frame and the scores are weighted appropriately so that frames produced by different input events in the same dialog contribute equally to the end result.

The domain-independent dialog processor also consults the domain interface (i.e., the calendar in this case) to filter out hypotheses that would cause domain errors such as missing or conflicting parameters. If it fails to find an acceptable hypothesis with a high enough score, it uses the error information returned by the domain interface to construct an appropriate feedback message to guide the user (only this process depends on the task domain). As soon as a good hypothesis is found, the dialog processor sends it to the domain interface as the operation to carry out and starts a new dialog. The user has the option of undoing the operation just performed if it is incorrect.

4. USER STUDY AND DATA COLLECTION

In order to create a useful and robust multimodal system, we need to find out how people would use a system with such capabilities. We have begun a series of experiments following the Wizard-of-Oz paradigm [6], in which the test subjects were presented with the Jeanie calendar interface and a hidden operator takes place of the real system in interpreting and carrying out multimodal commands. In parallel with the design of Jeanie, we conducted pilot experiments that contributed to the design of the frame representation for our task domain and yielded preliminary test data to evaluate the system.

Before the pilot experiment we conducted some small preliminary experiments to find out the types of gestures and speech people would use to make changes to a schedule, and to see if one modality was preferred over the other for certain tasks. That information was used to design pilot experiments that would illicit a mixture of speech and gestures from the test subjects. To get a variety of gestures that people might use the test subjects were first asked to make changes to a schedule using gestures only in any way they like. The same experiment was then repeated asking subjects to use speech only.

For the pilot experiments subjects were asked to make the changes by either giving voice commands or drawing on the touch screen or any combination of the two. Based on observations from the preliminary experiments we settled on instructing subjects by giving them a printed calendar with handwritten changes using as few words as possible and varying the wording to avoid speech contamination.

8 test subjects were used and a total of 13 trials were done for the preliminary and first set of pilot tests. The second pilot experiment consisted of 8 trials for 4 subjects using two completely different calendars to include a greater variety of tasks. In the pilot tests it was observed that some subjects preferred to use gestures more often, some preferred speech, others mixed the two. It was also observed that some tasks proved to be much easier to do by one method or the other; in those cases subjects chose the easier method regardless of general preference.

The pilot experiments showed, among other things, that test subjects often referred to a calendar event by using the name of a person or event, not just by a date and time; in addition, some subjects gave implied commands such as "On Tuesday I'm meeting with Bill, Tony, and Melanie" instead of "Add Tony and Melanie to the meeting with Bill on Tuesday". Test subjects with little computer experience tended to use implied commands most often, although we still do not have enough data to confirm such trends. The above observations significantly influenced the design of the frame representation.

5. PRELIMINARY PERFORMANCE EVALUATION

From out pilot Wizard-of-Oz experiments we obtained a small data set consisting of 185 multimodal events (77 speech-alone events, 57 pen-alone events, and 51 combination events). We transcribed each spoken utterance and pen gesture/handwriting as well as passing them through the modality recognizers. Results for all-transcribed inputs give us an idea of the performance of the multimodal interpreter, while results for recognized inputs offer some indication of performance degradation due to recognition errors.

The top-scored hypothesis produced by the interpreter for each input event is classified as *perfect*, *ok* (producing the intended result despite not being strictly correct), *ambiguous* (score too close to the next best hypothesis), *partially bad* (mostly correct except for one parameter), or *bad*. Table 1 shows the performance results for the pilot data. The pen data from the first pilot experiment could be transcribed but was unusable for recognition because of a flaw in the recording module which was cor-

TABLE 1: Multimodal interpretation performance

(TS/TP: transcribed speech/pen, RS/RP: recognized speech/pen)

	number of inputs	perfect (%)	ok (%)	ambi. (%)	partially bad (%)	bad (%)
TS/TP	185	78	2	7	3	10
RS/TP	185	57	5	16	6	16
TS/RP	100	49	2	26	9	14
RS/RP	100	28	7	29	12	24

rected for the second pilot experiment; because of this the bottom half of the table shows results for only the data set collected in this second experiment.

In the absence of recognition errors the multimodal interpreter will do the right thing 80% of the time (total of the *perfect* and *ok* columns), which is adequate for a usable system. At the other end of the spectrum the interpretation rate for all-recognized inputs is only 35%, a direct result of the poor 32% gesture/handwriting recognition rate.

The second row of the table shows that with a 76% word accuracy in speech recognition the interpretation rate drops to 62%. A closer analysis reveals that of the 18% that went from good to bad, about 15% occurred on speech-only events and only 3% occurred on combined speech/pen events. To put it another way, 27 of 77 (=35%) speech-only events caused interpretation errors due to speech recognition errors, compared to only 6 of 51 (=12%) combined events. On the pen-input side (only for the second data set of 100 input events), 11 of 18 (=61%) pen-only events caused interpretation errors due to gesture/handwriting recognition errors, compared to 19 of 39 (=49%) combined events. The sample is too small to assert any statistical significance, but this may be an indication of cross-modal redundancy at work.

We also observe that one significant effect of recognition errors is to increase the number of ambiguous outputs, more than the number of incorrect outputs. Recognition errors obviously reduce the amount of information available for interpretation, thereby causing more confusion. In interactive mode, the dialog processor can prompt the user with judicious guidance messages to supply clarifying information that could resolve ambiguities.

6. CONCLUSION

We have presented an architecture capable of joint interpretation of multimodal inputs and an example of a speech- and pen-enabled application using that architecture. The interpretation engine combines multiple input sources in a uniform and flexible way. The dialog processor maintains context information across input events and produces context-sensitive feedback to the user. The interpretation error rate is not unduly high but recognition errors may degrade performance significantly; however, we have seen indications that cross-modal redundancy can partly compensate for this degradation.

The pilot experiments have shown that the type of tasks we used in our Wizard setup will provide data for gesture, speech and a mixture of the two and will therefore be useful for the development and training of the Jeanie system. We are working on collecting more data to evaluate the system more fully, improve recognition rates, and refine parsing grammars. Experiments with the actual system in the loop rather than a Wizard operator are also being planned. Future studies should include experiments to investigate further the effect of cross-modal synergy in the presence of recognition errors to prove conclusively our intuitive assertions about the benefits of multimodal integration.

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APPENDIX B

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1. Conference proceedings

LCCN	98655050
Type of material	Periodical or Newspaper
Meeting name	ICASSP (Conference)
Main title	Conference proceedings / the ... International Conference on Acoustics, Speech, and Signal Processing ; sponsored by the Signal Processing Society of the Institute of Electrical and Electronics Engineers.
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Continued by	ICASSP (Conference). IEEE International Conference on Acoustics, Speech, and Signal Processing (DLC) 98655049 (OCoLC)37249153
Variant title	International Conference on Acoustics, Speech, and Signal Processing ... proceedings
Other title	ICASSP ... proceedings
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MULTIMODAL INTERFACES FOR MULTIMEDIA INFORMATION AGENTS

Alex Waibel, Bernhard Suhm, Minh Tue Vo and Jie Yang

Interactive Systems Laboratories
Carnegie Mellon University
5000 Forbes Avenue, Pittsburgh PA 15217 (USA)
{waibel,bsuhm,tue,yang+}@cs.cmu.edu

ABSTRACT

When humans communicate they take advantage of a rich spectrum of cues. Some are verbal and acoustic. Some are non-verbal and non-acoustic. Signal processing technology has devoted much attention to the recognition of speech, as a single human communication signal. Most other complementary communication cues, however, remain unexplored and unused in human-computer interaction. In this paper we show that the addition of non-acoustic or non-verbal cues can significantly enhance robustness, flexibility, naturalness and performance of human-computer interaction. We demonstrate computer agents that use speech, gesture, handwriting, pointing, spelling *jointly* for more robust, natural and flexible human-computer interaction in the various tasks of an information worker: information creation, access, manipulation or dissemination.

1. INTRODUCTION

Human-computer interfaces today are limited and inflexible and do not take advantage of the many communication channels humans use to communicate verbal and non-verbal ways. Humans speak, point, gesture, write, fixate, use facial expressions, head-motion, eye-contact, etc. to express ideas, intentions and feelings. To build computer systems that operate more flexibly and robustly, and that are more intuitive to use by anyone, computer interfaces must be able to understand and process this multiplicity of human communication cues. Recently many research projects have been conducted to address this problem (e.g. [1], [2], [3], [4]). The need for more intuitive interfaces becomes all the more pressing as multimedia presentation, that is output, is becoming commonly available.

In this paper we describe information agents that use multimodal interfaces to access, manipulate, create and disseminate information. In particular, we seek effective methods by which human users can easily interact with a computer by speaking, pointing, drawing, spelling, etc. We show how cross-modal cues can be used effectively to recover from errors or miscommunications, since confusions in one modality usually are unambiguous in another. We describe how multiple modalities can cooperate and complement each other in various phases of the preparation of multimedia documents. Moreover, the added flexibility and robustness significantly increases reliability and naturalness, and with it the usability and acceptance of information systems. We present QuickDoc as a prototypical application that combines our multimodal subsystems in a simple, but powerful way.

2. INTERPRETATION OF MULTIMODAL INPUT

2.1. Multimodal Components

Our labs have implemented recognizers and processing modules for various input modalities, most notably speech, pen input, and face tracking. These modality processors constitute the basic components of all our multimodal systems.

Speech

Our speech recognition subsystem is based on the recognition front-end of the JANUS speech translation system [5] which is capable of processing speaker-independent, spontaneous speech. The recognizer can be adapted to any task domain by retraining the language models and possibly tuning the acoustic models if the domain involves special vocabulary. We also have a high-performance, real-time continuous spelling recognizer for large lists of 100,000 or more names.

Gesture

Our approach to pen-based gesture recognition is to decompose pen strokes into sequences of basic shapes such as line, arc, arrow, circle, cross...[6] The same gesture shape may mean different things depending on the surrounding context, hence each gesture component is augmented by *gesture contexts* indicating spatial relationships between the gesture and nearby objects in the user interface.

Handwriting

Our MS-TDNN-based handwriting recognizer [7] is capable of processing writer-independent, continuous (cursive) handwriting at a recognition rate of 94% on a 20,000-word vocabulary. We employ simple heuristics to decide when to invoke handwriting recognition on pen input, e.g., when the gesture recognizer cannot identify the input strokes as basic shapes.

Face Tracking

We base our face-tracking subsystem on face color clustering and motion detection [8]. The face tracker can control a pan-tilt-zoom camera to follow a freely moving person, producing a constant-sized image of the face area in real time. Another software layer built on top of this face tracker can identify the eyes and other face features, build a model of the face, and estimate the gaze direction to track the head pose [9].

2.2. Joint Interpretation

In order to make sense of input from all available sources, we need a multimodal interpreter capable of producing an interpreta-

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tion of user intent (e.g., a command to execute in the application interface) from the output of the modality processors.

In our joint interpretation scheme, the user intent is represented by a frame consisting of slots specifying pieces of information such as the action to carry out or the parameters for that action. Recognition output from the modality processors are parsed into partially filled frames that are merged together to produce the combined interpretation as described in [6]. This technique leads to uniform handling of high-level information from all input sources, which is very important for modularity and extensibility. To add another input modality we need only provide a module to convert low-level recognizer output to a partially filled frame to be merged with others. In addition, context information can be retained across input events by merging with previous interpretation frames.

3. MULTIMODAL ERROR CORRECTION

Intensive research over recent years has boosted the performance of speech recognition technology significantly. Nevertheless it is widely believed that recognition performance will remain limited, at least for the foreseeable future. The potential for error with any speech interface will be balanced by the redundant and alternate ability to enter information or queries by spelled, handwritten or typed input. Rather than fighting a particular recognition error the user can therefore quickly circumvent the problem by choosing a different modality. Such alternates not only eliminate user frustration with potentially recurring errors, but also provide an avenue for background learning and adapting.

3.1. Multimodal Interactive Error Repair

Our approach is to have the user collaborate with the system in recovering from interpretation errors [10]. First, errors have to be identified, either initiated by the system, for instance based on some confidence measure, or by the user. If a graphical user interface (GUI) is available, the user can simply highlight erroneous words (the reparandum) in the recognition hypothesis displayed. For error correction, the user provides additional input, choosing among different correction methods: repair by repeating the reparandum by respeaking, spelling out loud, or handwriting; repair by paraphrasing the reparandum; repair by pen gestures (e.g. to delete or insert), in addition to the standard repair by typing or selecting among N-best alternatives.

The rationale for this multimodal approach to error recovery is twofold. First, it exploits the fact that different input modalities are orthogonal: words which are confusable in one modality, can be disambiguated in a different modality (e.g. "road" and "rote" spoken versus spelled). Secondly, recent studies [11] show that switching modality after system misinterpretation alleviates user frustration.

3.2. Error Repair for Multimedia Information Agents

The design of speech user interfaces is constrained by the context of the application, and so is the design of error repair. The application context varies along several dimensions: available modalities (speech only versus GUI), what is repaired (isolated words, phrases, sentences), goal of repair is (get verbatim every word versus get the intended action), dialogue metaphor (command controls versus conversational). For multimedia information agents, we see the following two application contexts as particularly relevant:

- (Mixed initiative) Spoken dialogue, for example in user queries and command control of the application. In this context, a GUI may be available, repair will typically be performed at the level of phrases, the goal of repair is to initiate the intended action (semantic repair), and a conversational dialogue is desirable.
- Dictation, for example to entry in fields, or to dictate reports. In this context, a GUI is very appropriate, repair will occur from isolated word to sentence level, the goal of repair is to get each word correct (verbatim repair).

We have built prototypical speech user interfaces with error repair capabilities for dictation and form filling tasks. In this application context, requiring the user to identify errors by highlighting them is appropriate, since the user naturally focuses on the actual recognition hypothesis. For error correction, we offer to repeat the input (potentially switching to another modality), and pen gestures to indicate where to insert words, or which word to delete. In form filling tasks, knowledge of the current field can provide powerful constraints for recognition by restricting vocabulary and language model accordingly.

3.3. Evaluating Interactive Error Repair

Given a set of error recovery methods feasible with current technology in a given application context, a crucial issue is to predict which methods a user will prefer. Based on the assumption of a rational user [13], we use the time to complete some input, including the time needed to correct errors, as an objective and easily quantifiable measure for the effectiveness of different error recovery methods, and main predictor of user preference. By assuming stochastic independence of the various repair attempts, the expected accuracy after a certain number of repair attempts can be estimated as a geometric series. Given a high level of accuracy sufficient for the application (e.g. 99%), the number of repair attempts can be estimated, and thus the total time required to input including repair [12].

A pilot evaluation on a form filling task [12] suggests that given current technology, repair by spelling and handwriting can be very effective, and significantly better than standard choice from N-best alternatives. Using the context of a repair, e.g. in rescoring N-best lists obtained by decoding repair input, can substantially improve the accuracy of repair. Although results are still preliminary, they show that our multimodal approach to interactive error recovery is very promising.

4. MULTIMODAL INFORMATION AGENTS

The objective for multmodal information agents is to quickly access, create, manipulate, and disseminate multimedia information. A multimodal agent can label multimedia information with appropriate classifications by voice or/and gesture annotations. All annotations are incorporated with original information. The agent will automatically attach hotlinks on the world-wide-web to broaden information sources. The agent then will generate an HTML format report. The agent can search all the information in the report including voice and gesture annotations and voice mails. The agent can repair errors and add more information. Finally, the agent can disseminate the report in electronic form or printed form.

4.1. Information Access

Multimodal information agents can speed up navigation of Information space and queries of a database. A city map, for example, can be queried by a combination of gesture, handwriting and speech. A query such as: "Show me all hospitals in this <circling-gesture> area," or "How far is from here to there <tracing-gesture>?" will access information that would otherwise have to be assembled and combined painstakingly in a series of steps. Multimodal agents will recognize gesture and speech concepts and map them onto a semantic representation. The semantic representation can be filled interchangeably by both gestures and verbal phrases using pattern recognition and chart parsing techniques. The parsing result is then sent to query the database. If interpretation errors occur, the user corrects them using multimodal correction techniques. The whole process is transparent to the user.

Searching multimedia data can be viewed as natural extension of multimodal database queries. For instance, voice annotations or video clips added during user interaction can be interpreted and indexed using multimodal interpretation components, and then accessed using database queries techniques.

4.2. Information Creation

A user often needs to add information to a database or modify existed information. Multimodal agents are also useful in such a process. For example, a doctor needs to mark his/her diagnoses on X-ray, CT, MR images and make comments on the case. He/she can use speech, gesture, and handwriting to do this. Similarly iconic gestures may be used on a touch sensitive display or writing pad, to quickly enter resources or objects in a database, such as a map, in non-verbal ways. Iconic gestures can be defined to represent objects that would be entered in the database. In the same way, handwriting symbols can be used to mark or tag objects on a screen. Multimodal input also need not always be interpreted, but can often be stored for later review. Thus a voice annotation, or a circling gesture can be simply stored to go along with an image for later review and retrieval. The efficiency of multimodal agents for information creation can illustrated by the following example. The scenario is an image analyst sitting in the front of a computer and classifying various targets. Images can be popped up quickly and in rapid succession and the analyst provides a quick classification. The image and its classification are entered into a database along with the analyst's spoken and gestured annotation. With the human in the loop to make decisions on intelligence relevance, but with the availability of a better (multimodal) interface, to do so quickly, the job of data creation and manipulation can be carried out with greater reliability and efficiency compared to a fully automated system.

4.3. Information Manipulation

Information will not only be requested to viewed, but will also need to be updated. A multimodal multimedia information system can offer new ways to maintain the underlying databases: the user can use speech, gesture, writing, spelling to update database information efficiently. manipulating visual representations on the display rather than having to learn and use a separate database manipulation system. Gestures play an important role in specifying object parameters (e.g. different iconic shapes to represent different object types) and spatial constraints (e.g., location and extent of objects), while speech is useful in specifying parameters not easily expressed visually. For example he/she might circle a bridge on a map and say "this bridge has been destroyed", or might cross out an object without words, etc.

In some situations, the answer for a query cannot be directly obtained from the database and the information from the database has to be future processed. For example, the query "Show me the nearest restaurant" requires not only database access but also other computations.

4.4. Information Dissemination

In addition to information access, creation and manipulation, the product has to be packaged and readied for later viewing. The world wide web can serve as a natural medium to disseminate multimedia information. The multimodal input signals are interpreted and compiled, and an HTML format report is automatically generated on the fly. It is then automatically available for viewing together with the original aligned speech waveform attached to it as hotlinks. Multimodal interaction can also be used to manipulate objects in a report, such as positioning objects on a page appropriately, or attaching appropriate hotlinks.

To quickly generate a multimedia report, speech dictation can be combined with point and click actions to arrange reports on the fly. Such a multimodal report generation would be faster than typing and can generate a hierarchy of facts and notions. The generation thus would consist of some dictation or typing, some multimodal object manipulation ("move this <point> here <point> and link it as an explanation for that <click>") and some handwriting or drawings that are to be attached. The result of these communicative acts will compile into HTML code, that is ready to be transferred. Time savings are possible in several ways: the report generation can be done more efficiently by speech, gesture, or handwriting, and by using references to already existing multimedia stickies, factoids, video footage or explanations, and the recipient of the report can probe and browse a hierarchical multimedia report in a non-linear fashion more efficiently, calling up aspects of the assessment as needed.

4.5. Controlling the Interface

Although the goal of a multimodal interface is to provide as transparent an access to functionality as possible, interaction to control various aspects of the system can't be eliminated.

For example for navigation within visually presented image data, the user typically modifies the view by operations such as zooming and panning. The integration of speech and gesture increases flexibility by allowing different attributes of an operation to be specified in whatever modality is more appropriate for each attribute; for instance, the effect of a spoken "zoom in to this <circle> area" accompanied by a circle drawn around the desired area is much more difficult to achieve using a single input modality

Additionally, multiple modalities can be used to enhance the reliability. For example, a gaze tracker can be used to detect the user's focus of attention. When the user is looking at a window on a screen, the window will be highlighted. No action is taken unless the user uses a voice command to confirm the selection. The voice commands could be ``select this window" or ``close window", etc. This can reduce unintended actions caused by the user randomly looking around.

5. THE QUICKDOC APPLICATION

QuickDoc is an example of an application that combines our multimodal subsystems in a simple but powerful way, letting the user perform a repetitive task with speed and convenience. The task is for a doctor to go through a series of images such as X-rays or computer-aided tomography scans, quickly identify an anomalous

area, label the area with the name of a disease or condition, and attach relevant comments. The end product is an HTML report that summarizes the doctor's findings in a compact table listing the annotated images, the corresponding preliminary diagnoses, and automatically generated hotlinks to relevant sites based on the diagnoses.

The QuickDoc user labels each presented image by circling an area on the image (drawing directly on a touch-sensitive screen) and speaking a disease name as well as a percentage representing the confidence level of the diagnosis. The gesture recognizer identifies the circle and generates an area marker on the image, attaching other parts of the gesture as written annotations (a future version may run the handwriting recognizer to turn those into text). The user can issue a spoken command to initiate a voice annotation attachment; the recorded audio file is also run through the speech recognizer to produce an automatic transcription. When all the images have been processed, another spoken command causes the HTML report to be generated. Optionally the collected data can also be edited in a multimodal dictation and repair facility before being processed into HTML.

The generated report contains thumbnails of the annotated images accompanied by disease label, confidence level, location in the image (automatically extracted from the circling gesture), and voice annotation playback icon if applicable. The thumbnail is linked to a more detailed page containing the full-size image and the automatic voice transcription. The name of the disease is looked up in a database and turned into a hotlink to a page listing Web sites relevant to that disease. A keyword search mode allows to search spoken keywords in the text of the report and the automatic voice transcriptions; entries containing hits are then highlighted in the report.

QuickDoc thus embodies multimodal creation, manipulation, dissemination, and access of multimedia information in a simple yet surprisingly useful application.

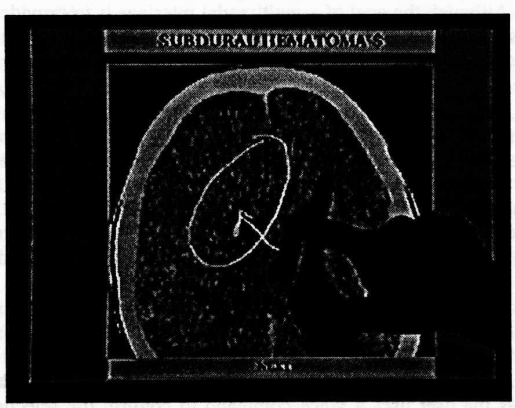


FIGURE 1: Multimodal information creation in QuickDoc: labeling some area by a circle gesture and speech ("subdural hematoma")

6. CONCLUSIONS

In this paper we have described systems that combine speech, gesture, handwriting, pointing, spelling, and other communication modalities into interfaces that are robust, flexible, and intuitive to use. We described how multimodal error recovery can ease the problem of unreliable automatic interpretation of communication modalities, in particular speech, thus removing a major obstacle in making multimodal interfaces truly usable. We demonstrated how these concepts can be combined for a more user friendly multimedia document production technology.

7. ACKNOWLEDGEMENTS

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APPENDIX G

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