DEPARTMENT OF ELECTRICAL AND COMPUTER ENGINEERING

VOICE INTERFACE TO AN ON-LINE DICTIONARY

by

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ABSTRACT

In the era of natural language recognition machines, the access of electronic equipment through a speech interface will go a long way in making state-of-the-art technology available to a larger class of users. A typical application useful to a significant group of people (e.g. students) is an on-line dictionary that can be accessed using voice commands. Currently, no such dictionaries exist for UNIX-based computer systems. Some personal computers offer this feature to a limited extent, but these are constrained by the amount of memory required for a large vocabulary recognition system. In this project, we design an interface that uses a public-domain speech-recognition software to recognize the specified words and accesses a dictionary that is available on-line. The resulting system will be publicly available from the ISIP home page.

WHY VOICE INTERFACE TO A DICTIONARY?

MORE NATURAL TO SPEAK THAN TO PROGRAM

- Database query requires complicated programming languages
- Interface by speaking is natural
- Definition is found easier
- Writing process speeds up
- Test bed for other data base queries
  - Library Resources
  - Telephone Directory
  - Television Listings

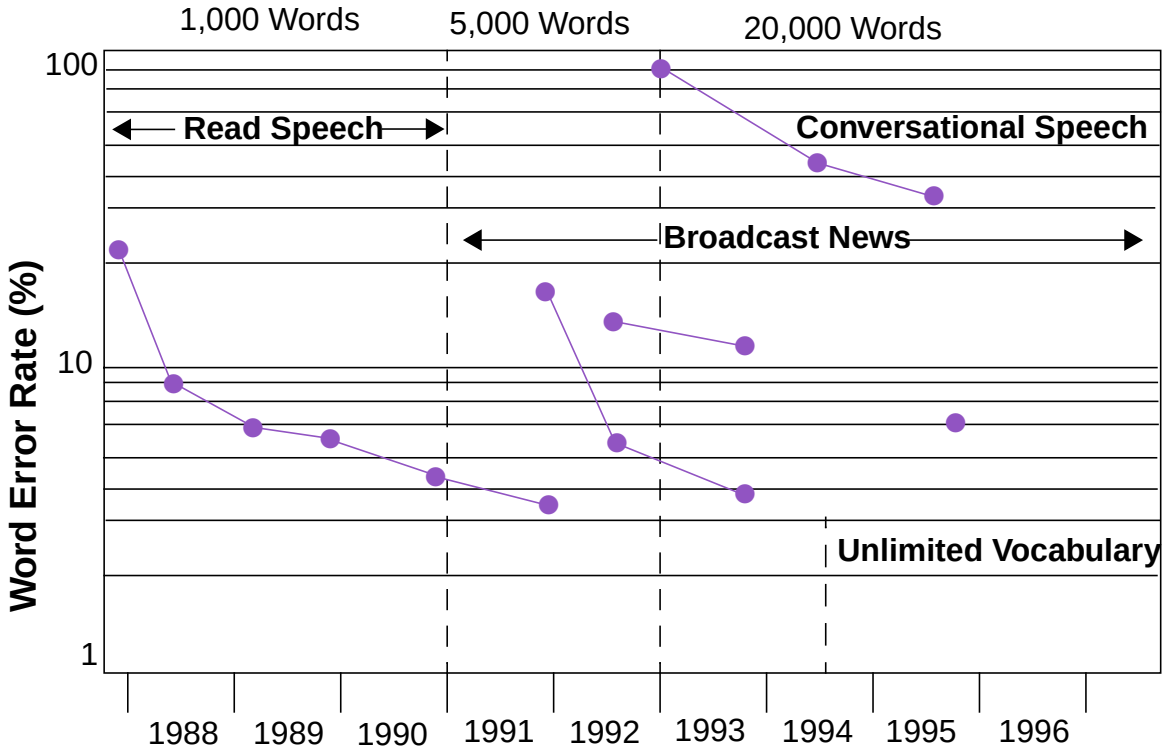
STATE OF THE ART
SPEECH RECOGNITION / UNDERSTANDING

When a system comprehends what is spoken

Challenges:

- Word spacing
- Coarticulation / Context
- Dialect
- Speaking rate / style

Performance:



- **Results - Speaker Independent**

WHY ABBOT?

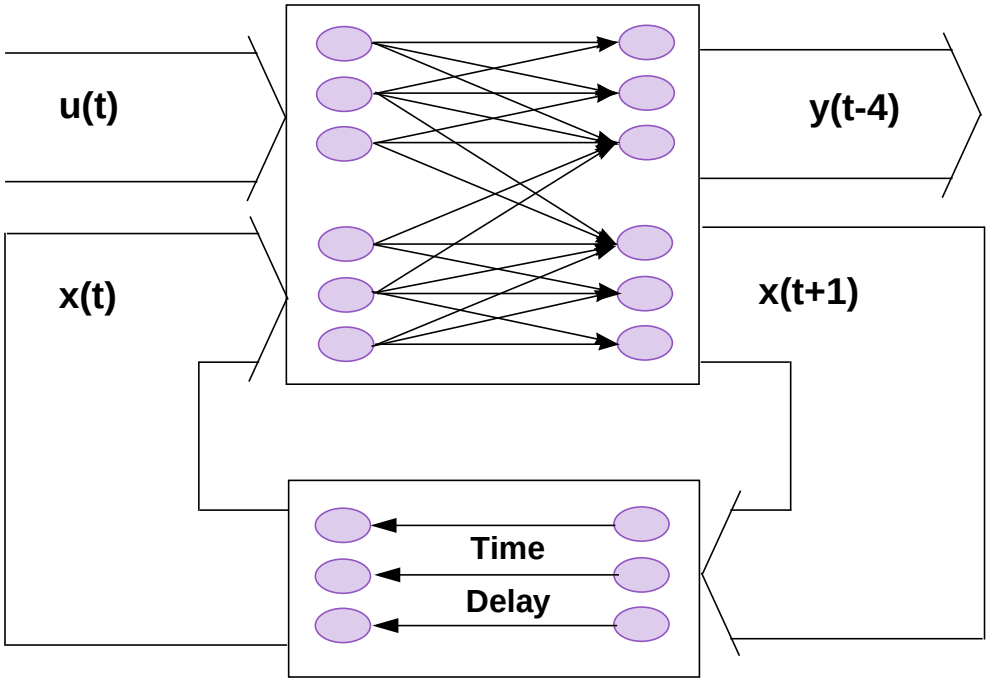
The competition - Cambridge HTK System

a generic HMM recognizer

ABBOT	**Cambridge HTK**
Hybrid connectionist hidden HMM	Gaussian HMM
Context - independent	Context - dependent
Recurrent Network System	Tied - State System
Cost - **Free**	Cost - **$100,000**

VS.

ACOUSTIC MODELING IN ABBOT



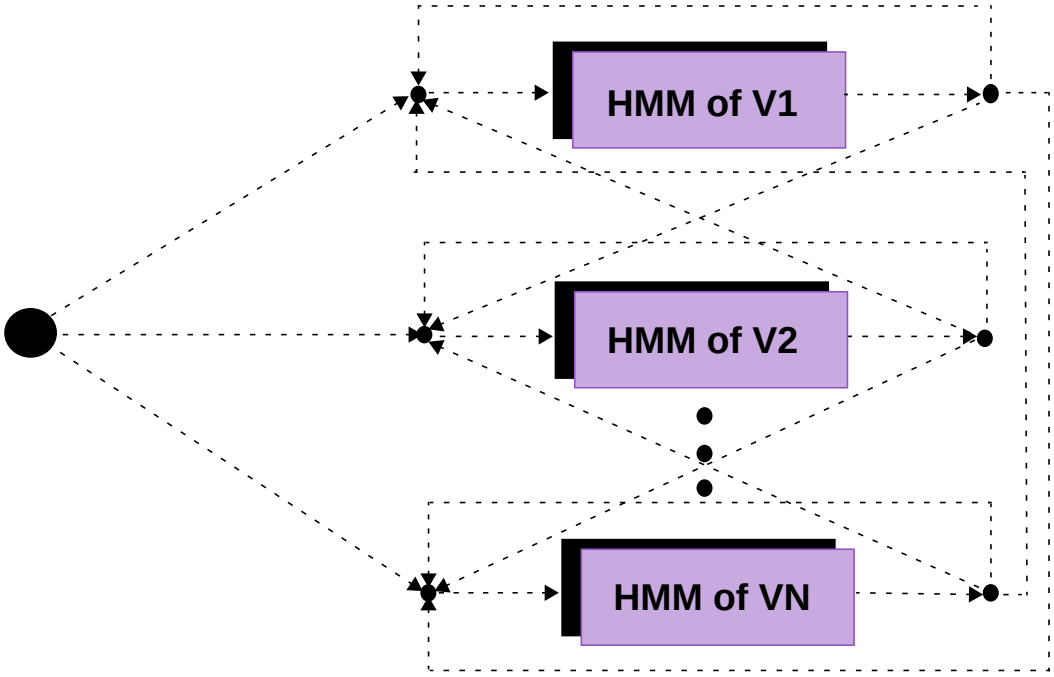
Phonetic Context-Independent Recurrent Neural Network

Input: acoustic vector u(t)
current state x(t)

$$y_i(t) \cong P_r\left(q_i(t) \middle| u_1^{t+4}\right)$$

Output: output vector y(t-4)
next state vector x(t+1)

LANGUAGE MODELING IN ABBOT



- Phone Set - 79 phone symbols, vowels have 3 levels of stresses
- Connectionist component - trained phone classifier
- Models - context & gender independent

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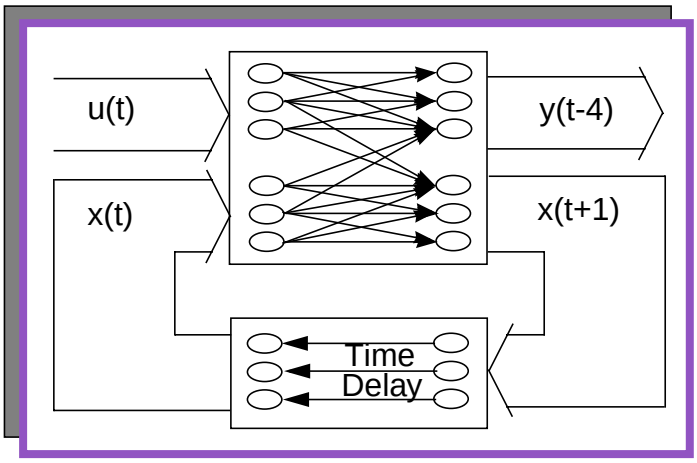
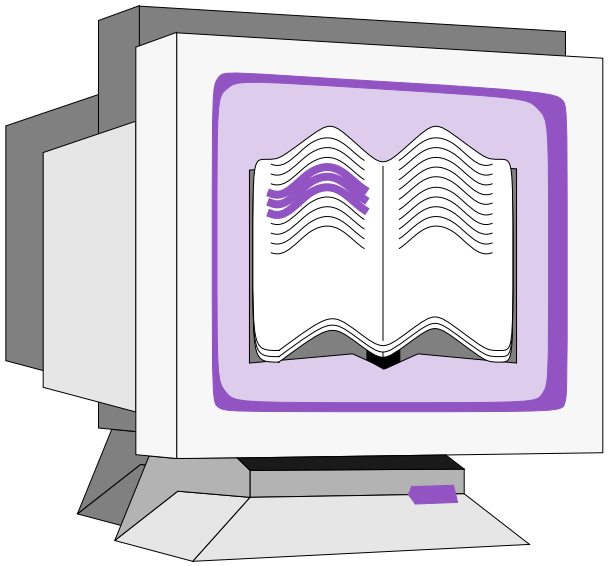
THE ABBOT DEMO

- Record at 16 kHz
- Determine the endpoints
- Convert to ASCII
- Normalize audio-gain



- Prints best guess to word & recognition continues
- The recognized word comes at the end



- Strip the recognized word from the end of the process
- Look up the word in the on-line dictionary

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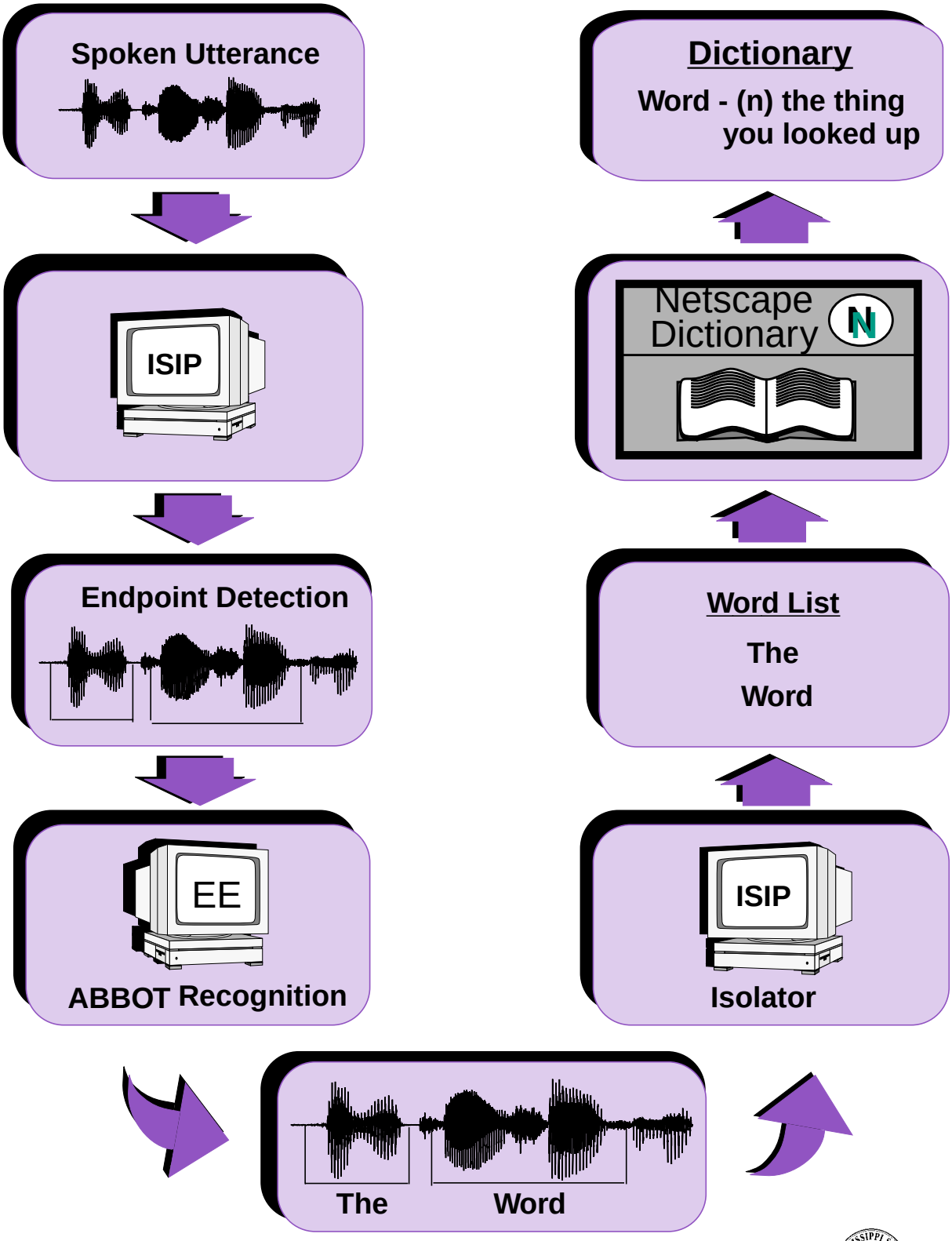
ARCHITECTURE OF CURRENT SYSTEM



WEBSTER DICTIONARY - WEB BASED SOURCE

Dictionary

- Systems of makeup
  - lexicon
  - grammar
  - semantic
  - phonology
- 160,000 entries
- Pronunciations

The Web

- Limited release of access
- Service for a fee
- Current version is first attempt
- CD Rom limited interface control

INTERFACING TO THE WEBSTER DICTIONARY

The Ultimate Interface

Natural Language Interface to the Dictionary

The Netscape Version
Point - and - click interface

- Type the word
- Hit return
- Retrieve definition

LANGUAGE MODELING ISSUES IN THE INTERFACE (GUI DESIGN)

Obstacles Encountered

Every word in dictionary recognized

ABBOT is a CSR

Dictionary takes only word roots

Data transported between machines

Word transported to Netscape Dictionary

Practical Solutions

Triphone based recognizer

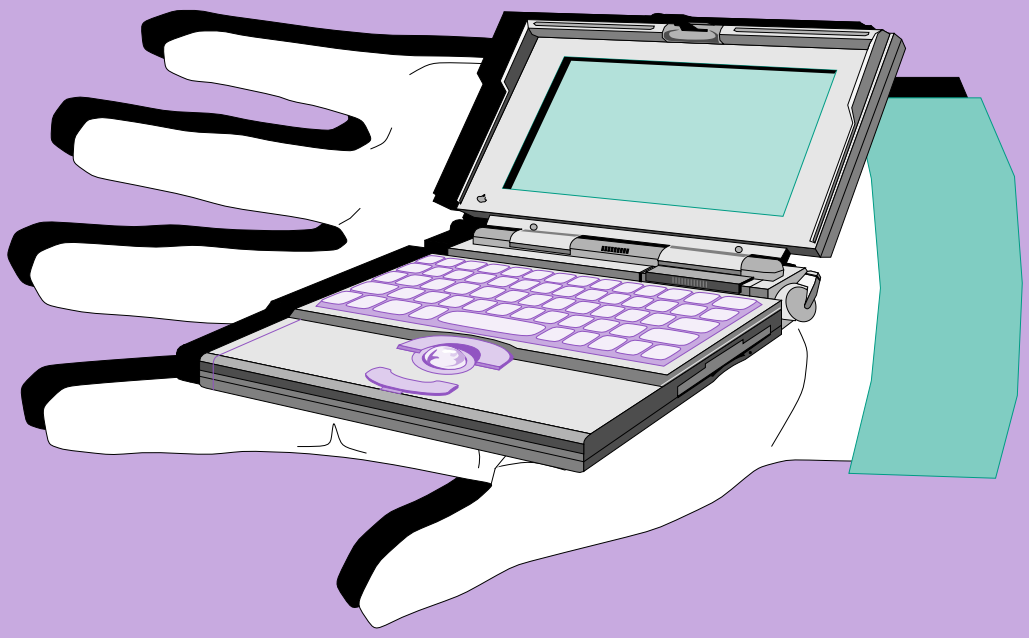
Language model changed to ISR

Portion recognizes prefixes / suffixes

Recognizer available locally

Dictionary available locally

BUILDING THIS DESIGN IN HARDWARE



Hand-Held Computer with DSP Chip and A/D Converter

- **Smaller than a credit card**

- **Plenty of memory for large recognition vocabulary**

SUMMARY

Designing a Voice Interface Dictionary

- An endpointed, spoken word
- A compatible speech recognizer
- An accessible, on-line dictionary
- A way to make all three work together

Future Enhancements:

- More adaptable recognizer (ISIP recognizer)
- Local dictionary access
- Cut down on real-time errors

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