AUTOMATIC SPEECH GENERATION AND UNDERSTANDING

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ABSTRACT

This paper reviews the work done on automatic speech recognition in the last three years and reports some new achievements on speech synthesis.

1. Introduction

This paper reviews the work done on automatic speech recognition in the last three years and reports some new achievements on speech synthesis.
A comphrensive review of the work done up to 1979 have been published by De Mori (1979). More recent reviews are contained in the books by Lea (1980), Suen and De Mori (1982) and in the papers by Rabiner and Levinson (1981), Haton (1982) and Shwartz (1982).
The review presented in this paper complements the above mentioned works and introduces a new approach based on methods from Artificial Intelligence. This approach is suitable for designing distributed systems for continuous speech recognition.
A set of Tables summarizes the most recent achievements in the solution of various problems.

2. Isolated Word Recognition

Systems are actually available on the market allowing to achieve fairly high recognition rates in speaker-independent way and with a lexicon of about 100 words. Costs of these systems depend on their performances and can vary from hundreds to tenth of thousands of Canadian Dollars.

There are many applications for such systems in the area of voice-activated tooling machines, systems for baggage or parcel distribution, quality control, aid for the handicapped.

Another area of applications is in voice-data entry with applications in carthography, methereological and geological data acquisition, stock management.

The research has been developed in the last years primarily towards two lines, usually:

1) speaker-dependent recognition of a large lexicon,
2) speaker-independent recognition of a small, well-specified set of words.

One of the main objectives of the first type of researches is that of developing a voice-activated typewriter capable of accepting a vocabulary of the order of 5000 words.

The second type of research has the purpose of developing speaker-independent recognizers of hundreds of words suitable to be used in office and banking applications.

2.1 Recognition of a large vocabulary of isolated words

The researches in this area follow a "passive model" for speech recognition. This model is "passive" because it contains two main components, namely feature extraction and pattern classification.

Usually there is no feed-back on the feature-extraction component, classification is based on pattern matching, time alignment between data and prototypes is based on dynamic programming or on the Viterbi algorithm and words are represented by networks of phonemes.

Features are extracted from a time-frequency-energy representation of speech or from parameters obtained from a model of the vocal tract.

Popular time alignment algorithms are based on Dynamic Programming (DP).

DP algorithms compute the distance D(A,B) between two sequences A and B of elements.

Let:

$$A = (a1, a2, \dots, ai, \dots aI)$$

be a prototype and (1)

$$B = (b1, b2, \dots, bj, \dots bJ)$$

be a representation of the data to be recognized.

The elements ai ($1 \leq i \leq I$) and bj ($1 < bj \leq bJ$) are vectors of features. They can be, for example, the energies at the output of a filter bank in a 10 msec interval.

The distance D(A,B) is computed by composing elementary distances d(ai,bj) between a feature vector of the prototype and a feature vector extracted from the data.

Pairs (ai,bj) ($1 \leq i \leq I$, $1 \leq j \leq J$) can be represented by points of coordinates (i,j) in a plane IJ. There are many paths in the plane IJ going from the point (1,1) to the point (I,J). A global distance D(p) can be associated to each path p of this type. Let P be the set of these paths. The distance D(A,B) is defined as follows:

$$D(A,B) = \min_{p \in P} D(p) \qquad (2).$$

Rabiner and Levinson (1981) have extensively reviewed this problem.

There are two problems related to such an approach. The first one concerns the algorithm for computing the elementary distance d(ai,bj) in such a way that relevant similarities or differences are properly weighted.

The second one is related to the composition of elementary distances for obtaining a global distance.

Sakai and Okochi (1982) have proposed a method for distance composition with interesting properties such as:

commutativity:	D(A,B) = D(B,A)
·reversibility:	D(A,B) is the same if computed from left-to-right or from right-to-left.

consistency with linear compression.

When the path is one that would be obtained by a linear compression of the data, then the distance computed with dynamic programming is the same as the one computed with linear compression.

Patterns of "best matching" paths have been investigated at Fujitsu (Nara et al., 1982).

A deformation model of these paths has been used for speeding up the calculation of distances between patterns. Real-time performances have been obtained on a speaker-dependent system having a lexicon of 1,000 japanese words. The reported recognition accuracy is 95.8%.

Similar results (94.6%) on 1,000 words were obtained on the same task by KOHONEN et al. (1980) who designed the OTANIEMI real-time speech recognition system based on the application of Associative-Memory concepts.

Other interesting results are summarized in Table I.

These systems are speaker-dependent, require the collection of a large number of templates, are based on algorithms having a complexity proportional to the square of the length of the input sentence and weight the same way peices of information having different importance.

2.2 Speaker-independent recognition of isolated words.

Two approaches have been taken in the design of speaker-independent systems for the automatic recognition of isolated words.

The first approach, mostly followed in U.S.A. and Japan, consists in collecting clusters of templates from different speakers.

The second approach, taken in Europe, is based on the extraction of speaker-independent features from time-frequency-energy representations of speech.

Practical systems are based on well chosen vocabularies the size of which do not exceed a few hundred words.

Recent achievements are summarized in Table II.

3. Recognition of Phonetic Features

Recognition of phonetic features is a fundamental step for designing speaker-independent systems capable of accepting continuous speech of considerably large protocols.

This task is still object of research.

An approach that becomes more and more popular is based on the concept of Expert Systems developed in Artificial Intelligence.

These systems have a knowledge made of rules involving acoustic cues and phonetic features.

Other approaches are based on segmentation of the speech signal into syllabic or phonenic segments and segment labelling with dynamic programming algorithms.

Table III summarizes the most recent achievements in this field.

4. Recognition of Continuous Speech

The concepts of dynamic programming have been applied to the recognition of limited squences of connected words.

Contributions along this line have been provided by Sakoe (1979) in Japan, Bridle (1982) in Europe, Rabiner and Levinson (1981) in U.S.A., Vintsjuk in the Soviet Union.

Another approach based on Markov chains has been proposed by the Speech Research Group at IBM (Bahl et al., 1981). This approach uses also the Viterbi algorithm for speech decoding as proposed by White (1978).

Among the most interesting applications it is worth mentioning the recognition of connected digits (Rabiner and Levinson, 1981), the design of an automated travel reservation system (Levinson, 1980 in U.S.A., Shikano and Kohda in Japan, 1978), computer interfaces for spoken BASIC (Niimi, 1979) and spoken FORTRAN (Shigenaga and Sekiguchi, 1979).
Techniques of Artifical Intelligence have been applied for speech decoding and lexical access (De Mori 1982, Shipman and Zue, 1982, Smith and Erman, 1981).
Table IV summarizes the recent achievements in this field.

5. Special Architectures for Automatic Speech Recognition

Special architectures have been proposed for Automatic Speech Recognition.
Lyon (1982) at Fairchild has developed a new VLSI chip that model that signal processing operations in the human cochlea.
Broderson (1982) has developed another VLSI chip for performing in real-time dynamic programming algorithm on a set of more than 1000 word templates.
Yoder and Siegel (1982) have investigated models based on Single Instruction Multiple Data (SIMD) machines for implementing DP algorithms in real-time.
Bisiani (1983) at Carnegie Mellon University has proposed new architectures for problem solving.

De Mori (1983) has developed a system to speech decoding based on the Expert System metaphaze. Each Expert is a reasoning program having a Long Term Memory (LTM) and a Short Term Memory (STM). Each LTM contains the knowledge of an Expert. This knowledge is described by a frame language. Knowledge instantiations are created into STM's for interpreting input data and for growing hypotheses.
Experts also create interpretation processes that communicate among them as in Data Flow Machines.

TABLE I

Institution	Country	Research
Univ. Kyoto (OKOCHI,SAKAI (1982))	Japan	trapezoidal DP matching
Fujitsu (NARA et AL., 1982)	Japan	Model of typical distortions for DP matching
Acad. Science Pol. (WEZLAK et AL.,1982)	Poland	Use of fuzzy algebra
KTH-Stockholm (ELENIUS et AL.,1982)	Sweden	Use of temporal constraints
Bell Lab (R.BROWN,1982)	U.S.A.	Beam search techniques
Helsinki U. of Techn. (KOHONEN et AL., 1980)	Finland	Associative memories
Hewlett-Packard GREER et AL., 1982)	U.S.A.	Beam search
Royal signal and Radar Esta-blishment (MOORE et AL., 1982)	England	Use of fuzzy algebra
Bell Northern (MERMELSTEIN, 1982)	Canada	Comparison of several acoustic parameters for DP matching
Purdue Univ. (KASYAP,1979)	U.S.A.	Stochastic model
IBM Yorktown (SILVERMAN et DIXON,1980)	U.S.A.	Use of acoustic constraints in DP matching
IBM Watson Center (BAKIS et DIXO) 1982)	U.S.A.	Recognition by Synthesis

TABLE I continued

Institution	Country	Research
Bell Labs (BROWN et RABINER,1982)	U.S.A.	Use of graph theory methods for matching
Univ. de Kiev (VINTSJUK,1980)	URSS	Use of symbols describing phones in DP-matching

TABLE II

Institution	Country	Research
Univ. de Novosibirsk (KELMANOV et AL.,1982)	URSS	Search of speaker in-dependent cues
CNET (MERCIER, 1981)	France	Search of speaker in-dependent cues
NTT Musashino (FURUI,1980)	Japan	Clustering of speaker dependent templates
Nippon Electric Co. (CHIBA et AL., 1978)	Japan	Clustering of speaker dependent templates
Bell Laboratories (RABINER,1978)	U.S.A.	Clustering of speaker dependent templates

TABLE III

Institution	Country	Research
Univ. De Mexique (MARTINEZ et AL., 1982)	Mexico	Source coding for phonenic recognition
Univ. Stanford (TURNER, 1982)	U.S.A.	Algorithm for the recognition of plosive sounds
BB (Cambridge) (ROUCOS et AL., 1982)	U.S.A.	Use of dyphons
Hitachi (KOMATSU et AL.,1982)	Japan	Phoneme recognition
TOSHIBA (WATANABE et AL.,1982)	Japan	Syllabic sounds
Universite de Munich (RUSKE,1982)	Germany	Syllabic sounds
MIT (LAMEL et ZUE, 1982)	U.S.A.	Phonetic constraints
Hertz Inst. Berlin (SCHULZE,1982)	Germany	Preclassification of phonetic features
Electrotechnical Lab. (NAKAJIMA et AL., 1982)	Japan	Phoneme recognition
Carnegie-Mellon Univ. (COLE et AL., 1982)	U.S.A.	Use of phonetic features
Universite de Turin and	Italy	" "
Concordia University (DE MORI,1982)	Canada	" "
Academie Sciences Moscow (KNIPPER,1980)	URSS	" "

TABLE III Continued

Institution	Country	Research
GURA et al. 1980 Univ. of Lvov	URSS	Phoneme recognition
Academy of Science Moscow (IVANOVA et AL., 1980)	URSS	plosive sounds recognition
Universite de Minsk (LOBANOV,1981)	URRS	" "
Universiute Tokyo (FUJISAKI,1982)	Japan	" "
Universite Canberra	Austrailia	" "
Turin (DEMICHELIS et AL.,1982)	Italy	" "
Concordia University (SUEN and SANTERRE,1981)	Canada	" "
Concordia University (DE MORI,1982)	Canada	Expert system for speech decoding
Centre National Etudes Tele-communications (MERCIER,1983)	France	Expert system for speech decoding

TABLE IV

Institution	Country	Research
Nippon Electric Co. (SAKOE,1979)	Japan	Automated travel inform-ation and reservation system
JSRU (BRIDLE el AL., 1982)	England	Two levels DP
Naval Res. Lab. Washington (SHORE et BURTON, 1982)	U.S.A.	Two levels DP

TABLE IV Continued

Institution	Country	Research
Carnegie Mellon Univ. (SMITH et ERMAN, 1981)	U.S.A.	Large lexicon in continuous speech
Kyoto Institute of Technology (NIIMI,1979)	Japan	Spoken basic
NTT (SHIKANO KOHDA, 1978)	Japan	Automated travel information and reservation system
Bell Labs (LEVINSON,1980)	U.S.A.	Automated travel information and reservation system
Auricle (WHITE,1978)	U.S.A.	Viterbi algorithm
IBM Watson Center (BAHL et AL., 1981)	U.S.A.	Markov chains
University of Kiev (VINTSJUK,1980)	URSS	Dynamic programming
University of Yamanashi (SHIGENAGA,1979)	Japan	Spoken FORTRAN
Bell Labs (RABINER et LEVINSON,1981)	U.S.A.	Use of syntactic constraints in DP matching
Concordia University (DE MORI,1982)	Canada	Expert system for speech decoding. Networks for lexical access.

TABLE V

Institution	Country	Research
Logica LTD (PECKMAN et AL., 1982)	England	Machines for computing DP matching in real-time
Purdue Univ. (YODER et SIEGEL)	U.S.A.	SIMD machine for DP matching
Fairchild (LYON,1982)	U.S.A.	Cochlear model
Brown University (SILVERMAN,1982)	U.S.A.	Machine for computing DP matching with acoustic constraints
Concordia University (DE MORI, 1982)	Canada	Network architectures for lexical access
University California Berkeley (BRODERSON,1982)	U.S.A.	VLSI DP matching
Carnegie Mellon University (BISIANI, 1983)	U.S.A.	Architectures for problem solving

6. Speech Synthesis

The most interesting progresses achieved in the last years are related to text-to-speech conversion systems and to the development of new integrated circuits for speech synthesis.

The most impressive achievements to text-to speech have been obtained by Klatt (1982) at MIT and by Carlson et al. (1982) at the Royal Institute of Technology in Stockholm (Sweden).

Carlson and his colleagues have designed a speech synthesis board connected with a keyboard. The user specifies with a code a language to be synthesized among five possible choices (English, French, German, Italian, Swedish). Then every sentence typed from the keyboard will be synthesized in the selected language.

The quality of the synthesized speech is good because rules have been inserted which were carefully selected by a linguist. The most popular integrated circuits are based on an all-pole model of the vocal tract driven by Linear Prediction Coefficients. Researches are in progress for selecting matrices of coefficients corresponding to phonemes, diphones and syllables and for finding rules for their concatenation.

Researches are in progress also in Canada, especially at Bell Northern-INRS (Montreal) and Concordia Univeristy.

A recent review on speech synthesis has been provided by Liénard (1981).

A description of novel integrated circuits has been provided by Brantingham (1982).

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