



Extra Large Vocabulary Continuous Speech Recognition Algorithm based on Information Retrieval

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Abstract

This paper presents a new two-pass algorithm for Extra Large (more than 1M words) Vocabulary Continuous Speech recognition based on the Information Retrieval (ELVIRCOS). The principle of this approach is to decompose a recognition process into two passes where the first pass builds the word subset for the second pass recognition by using information retrieval procedure. Word graph composition for continuous speech is presented. With this approach a high performance for large vocabulary speech recognition can be obtained.

Index Terms: large vocabulary continuous speech recognition, phoneme recognition, information retrieval

1. Introduction

There exists a necessity for speech recognition with a huge numbers of alternatives without any syntax restriction for input words sequence.

For example, during the voice control of a computer it is impossible to predict the subsequent word on the basis of several previous ones because it is defined by control logic, instead of text properties. From the other hand there is a necessity for growth of the volume of the dictionary to capture all possible synonyms of the same command caused by difficulty for users to remember the single command name variant.

The next example concerns the text dictation. The application of such systems is limited by the texts, which are statistically similar to one where statistics were collected. Additional editing of the collected text demands the presence of all words in the actual dictionary.

Thus, there are applications where it is desirable to have a dictionary as large as possible, in future to cover all words for the given language.

The additional information to restrict the number of alternatives can be received from a speech signal immediately. For this purpose it is proposed to execute preliminary trial recognition by using the phonetic transcriber. Phonemes sequence analysis allows to build the queries flow. Applying the information retrieval approach considerably limits the number of alternatives for recognition.

The next sections describe a new two-pass algorithm. First, the baseline recognition system is presented to compare with the proposed algorithm. Then, two variants of algorithm for isolated and continuous speech are described. The fulfilled experiments have shown efficiency of the offered techniques.

2. The baseline recognition system

The approach proposed in this paper is applicable for any recognition system where phonemes and phoneme recognition (phonetic transcriber) are present but the number of phonemes no more than approximately 500 units.

As a reference system HMM-based HTK toolkit [1] is used.

2.1 Feature Extraction

In the feature extraction unit the speech signal is converted into a sequence of vector parameters with a fixed 25 ms frame, the frame rate is 10 ms. Then each parameter is pre-emphasized with filter $P(z) = 1 - 0.97z^{-1}$. Later, the Hamming window is applied. A fast Fourier transform is used to convert time domain frames into frequency domain spectra. These spectra are averaged into 26 triangular bins arranged at equal mel-frequency intervals. 12 dimensional mel-frequency cepstral coefficients (MFCCs) are obtained from cosine transformation and lifter. The log energy is also added as the 13th front-end parameter.

These 13 front-end parameters are expanded to 39 front-end parameters by appending first and second order differences of the static coefficients.

Finally, cepstral mean normalization is applied to deal with the constant channel assumption.

2.2 Acoustic Model

47 Russian context-free phonemes are modeled by 3 state left-to-right HMMs with skip transition. Each state is represented by a mixture of 64 Gaussian probability density functions with diagonal covariance matrices.

The pronunciation dictionary was created automatically from word orthography using a set of Russian context sensitive rewrite rules.

The Viterbi algorithm is used for the pattern matching.

2.3 Baseline Performance

All acoustic models were trained by using 12000 utterances from 2037 words dictionary. Training set consists of 2000 isolated words and 1000 continuous phrases. All utterances were collected by the single speaker. The recognition experiments are carried out on the PC P-IV 2.4GHz.

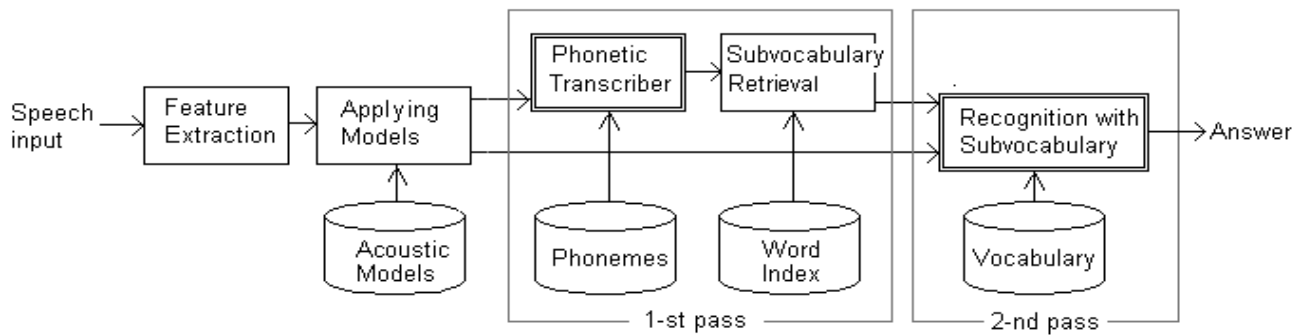


Figure 1: The architecture of ELVIRS recognition system

1000 utterances from the same speaker for isolated word recognition tests were collected. The word recognition accuracy and average time for different vocabulary sizes are given in Table 1. The recognition time per one second of speech is linearly dependent on the vocabulary size. The recognition time with the vocabulary of 1987K words can be evaluated as 2300 seconds.

Table 1: Baseline recognition performance for isolated word

Vocabulary size	1K	15K	95K
Accuracy, %	99.9	97.9	94.7
Time, sec	1	16	115

For continuous speech recognition an additional set of 1000 utterances with numbers from 0 up to 999 were collected. The word recognition accuracy and average time for different vocabulary sizes are given in Table 2.

Table 2: Continuous speech performance

Vocabulary size	1K	15K	95K
Accuracy, %	98.0	96.5	92.6
Time, sec	2.1	29	205

3 ELVIRS Algorithm for Isolated Words

3.1 Architecture

The architecture of the system is shown in Figure 1. The *features extraction* and *acoustic models* blocks are reused from the baseline system. Common *pattern matching* unit with subset of vocabulary is used on the second pass. Changes are concentrated in the new first recognition pass when *phonetic transcriber* is applied to make the sequence of phonemes. Then *information retrieval procedure* builds the sub-vocabulary for the second pass.

3.2 Phoneme Recognizer

The phonetic transcribing algorithm [2, 3, 4] builds a phonetic sequence for speech signal regardless to the dictionary. For this purpose a phoneme generative automaton was constructed which can synthesize all possible continuous speech model signals for any phoneme sequence. Then the phoneme-by-phoneme recognition of unknown speech signal is applied.

The same context-free phonemes as in baseline recognition system are used.

The experimental accuracy of finding phoneme at the right place equals to approximately 85%.

3.3 Sub-Vocabulary Retrieval Procedure

Preliminary transcription dictionary is prepared to build phoneme triples. The index entry key is a phoneme triple, thus, the index consists of M^3 entries where M is the number of phonemes in the system. Each index entry contains the list of transcriptions that include key phoneme triple. Additional memory usage is approximately 50 MB for vocabulary with 1 M words.

Sub-vocabulary retrieval process is illustrated in Figure 2. Phoneme recognizer output is split into overlapping phoneme triples. Resulting phoneme triple becomes the query. Now, in this system the simple query is used where phoneme triple and query are the same. In the future it should be modified to take into account the insertion, deletion and substitution of phoneme sequence by using *Levenshtein* dissimilarity. Thus phonetic sequence produces the query flow for database.

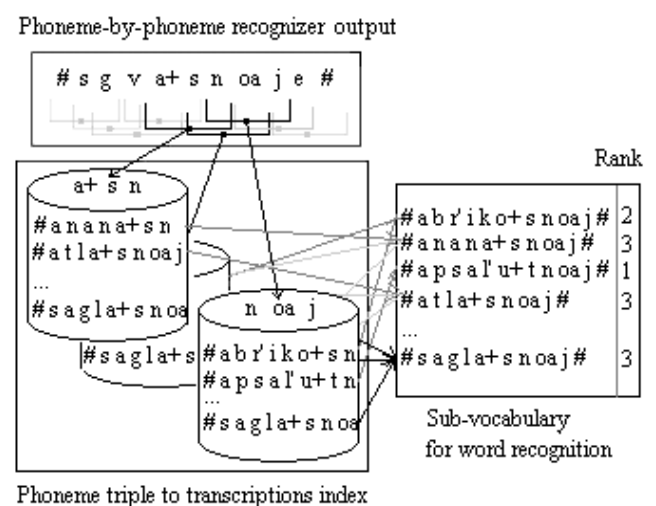


Figure 2 Sub-vocabulary retrieval process

The query answer consists of the list of transcriptions in which the given triple is included. Next queries produce new transcription portions to be copied into the sub-vocabulary for the second pass. The counter for word repetition is supported to make the rank of word.

All transcriptions in resulting sub-vocabulary are arranged according to the word rank (repetition counter). First N transcriptions are copied into a final sub-vocabulary for the second pass recognition. Thus the recognition sub-vocabulary consists of transcriptions of highest ranks but the vocabulary size does not exceed a fixed limit N.

3.4 ELVIRS Algorithm Overview

The ELVIRS algorithm [5] works as described in the following.

Preparation stage:

1. Prepare the recognition vocabulary.
2. Chose the phoneme set and build transcriptions for words from vocabulary by rules.
3. Create database index from phoneme triple to transcriptions.
4. Train the acoustic models from collected speech signals.

Recognition stage:

1. Apply phoneme recognizer for input speech signal to produce a phoneme sequence.
2. Split the phoneme sequence into overlapping phoneme triples.
3. Make queries from phoneme triples.
4. Retrieve transcription lists by queries from database index.
5. Arrange transcriptions by the rank.
6. Chose N-best transcriptions for recognition sub-vocabulary.
7. Recognize the input speech signal with sub-vocabulary.

4 The Information Consideration

The phoneme recognizer output can be considered as a correct phoneme sequence passed through a noisy channel and converted into an output sequence. Denote a right phoneme in output sequence as 1 and wrong one as 0. Let 1 occurs with probability u . The probability P to find k and more successive 1 in a binary set with length of n can be computed with the help of the following recurrent expression:

$$P_n = \begin{cases} 0, n < k \\ u^k, n = k \\ P_{n-1} + u^k(1-u)(1-P_{n-k-1}), n > k \end{cases}$$

Probabilities P to find three and more successive 1 in a binary sequence for different lengths n and probabilities u are shown in Table 3. Average transcriptions length is equal to approximately 8 and the accuracy of finding phoneme at the right place for known utterance is equal to approximately 85%. For these values the probability to find right word in chosen sub-vocabulary is equal 0.953.

Table 3: Probability to find three and more successive 1 in a binary sequence with length of n

$n \backslash u$	0.75	0.8	0.85	0.9
6	0.738	0.819	0.890	0.948
7	0.799	0.869	0.926	0.967
8	0.849	0.908	0.953	0.982
9	0.887	0.937	0.971	0.991
10	0.915	0.956	0.981	0.995

5 ELVIRCOS Algorithm for Continuous Speech

5.1 Architecture

After transcriptions list retrieval procedure an additional procedure – *word graph composition* is applied. It produces a word network for second pass recognition.

5.2 Word graph composition

The word graph composition procedure is illustrated in Figure 3. Word network starts from vertex S and ends in vertex F. Each triple from phoneme output burns intermediate vertexes with numbers synchronous the occurrence time. On the other hand, each triple became query to data base index, which returns the transcription list as result. Transcriptions are interlaced with intermediate numbered vertexes as base vertexes so that burning phoneme triples are placed in coordination.

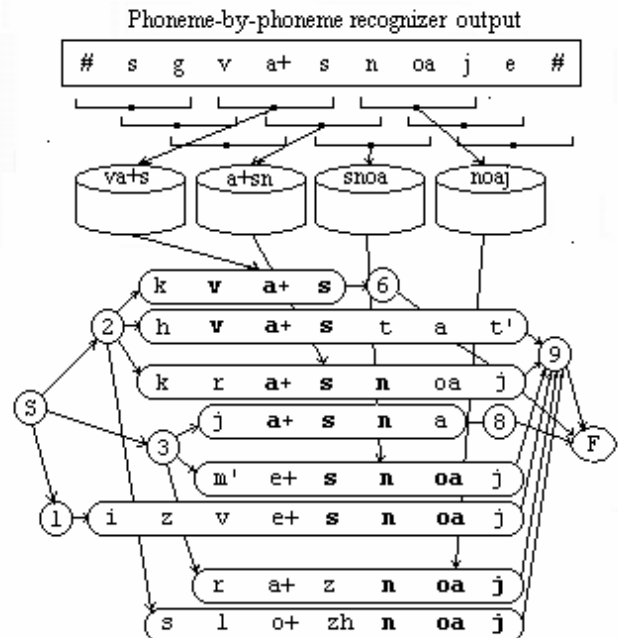


Figure 3: Word graph composition for continuous speech

The rank of transcription is increased in case when intersection between same transcriptions burned from different phoneme triple occurs. For each moment of time (synchronous with phoneme sequence) the number of involved transcriptions may be calculated.

In order to reduce the word graph complexity, the fixed limit N is applied. For each moment of time transcriptions with small ranks are removed from word graph so that only N transcriptions remain.

The word graph is composed from left to right, that is why it is possible to construct one in real time with the delay is equal of largest transcription length.

5.3 ELVIRCOS Algorithm Overview

The ELVIRCOS algorithm works as follows.

Preparation stage is the same as ELVIRS algorithm.

Recognition stage:

1. Apply phoneme recognizer to the input speech signal to produce a phoneme sequence.
2. Split the phoneme sequence into overlapping phoneme triples.
3. Make queries from phoneme triples.
4. Retrieve transcription lists by queries from database index.
5. Compose word graph network.
6. Recognize the input speech signal with composed word net.

6 Experimental Results

For experiments, some modifications of HTK toolkit were necessary to take into account the first pass of algorithm.

In case of isolated words the influence of the sub-vocabulary limit N on the average recognition time and word recognition accuracy for different vocabulary sizes was investigated (see Table 4). With the increase of vocabulary volume the size of N has to be increased in order to achieve high recognition performance.

The considerable reduction of the recognition time with relatively small accuracy degradation (approximately 5%) in comparison with baseline system has been achieved. The accuracy degradation has a good agreement with the information consideration.

Table 4: ELVIRS recognition performance

Vocabulary size Subvocab. limit N	15K		95K		1987K	
	Acc, %	Time, sec.	Acc, %	Time, sec	Acc, %	Time, sec
50	92.2	1.4	81.0	1.4	69.2	1.6
200	94.6	1.6	87.6	2.1	76.0	1.9
500	95.5	1.9	90.1	2.5	80.0	3.3
1000	96.0	2.1	90.7	3.1	82.7	4.4
2000	96.0	4.4	92.0	4.5	84.8	6.8
5000	96.0	4.6	92.9	8.3	86.4	12.0

Several preliminary experiments for continuous speech were executed in case when limit N is equal to the vocabulary size. Results for different vocabulary sizes are shown in Table 5.

Table 5: ELVIRCOS recognition performance

Vocabulary size	15K	95K	1987K
Accuracy, %	85.3	84.1	80.4
Time, sec	4.5	22	442

Average time reduction is not so considerable as in isolated word recognition because some phoneme sequences produce word graphs with very large sizes. It seems that introducing the limit N decreases essentially the recognition time.

7. Conclusions

This paper has presented the new approach for large vocabulary speech recognition and a preliminary experimental evaluation. The importance of information retrieval for speech recognition should be underlined. It was shown that additional information source from analysis of phoneme sequence allows to restrict the search space. These new restrictions lead to speech recognition with vocabularies cover practically all words for given language.

The approach proposed in this paper is useful for zero language model where all words from vocabulary are reachable for recognition. More complex language models can be applied in future works to achieve new features.

8. References

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