Speech Recognition with Hidden Markov Model: A Review

Shivam Sharma

Abstract: The concept of Recognition one phase of Speech Recognition Process using Hidden Markov Model has been discussed in this paper. Preprocessing, Feature Extraction and Recognition three steps and Hidden Markov Model (used in recognition phase) are used to complete Automatic Speech Recognition System. Today’s life human is able to interact with computer hardware and related machines in their own language. Research followers are trying to develop a perfect ASR system because we have all these advancements in ASR and research in digital signal processing but computer machines are unable to match the performance of their human utterances in terms of accuracy of matching and speed of response. In case of speech recognition the research followers are mainly using three different approaches namely Acoustic phonetic approach, Knowledge based approach and Pattern recognition approach. This paper’s study is based on pattern recognition approach and the third phase of speech recognition process ‘Recognition’ and Hidden Markov Model is studied in detail.

Keywords: Automatic Speech Recognition (ASR), HMM model, human machine interface.

1. RECOGNITION

Recognizers is the third phase of speech recognition process deal with speech variability and account for learning the relationship between specific utterances and the corresponding word or words [1]. There has been steady progress in the field of speech recognition over the recent years with two trends [2]. First is academic approach that is achieved by improving technology mainly in the stochastic modeling, search and neural networks. Second is the pragmatic, include the technology, which provides the simple low-level interaction with machine, replacing with buttons and switches. A second approach is useful now, while the former mainly make promises for the future. In the pragmatic system emphasis has been on accuracy, robustness and on the computational efficiency permitting real time performance with affordable hardware. Broadly speaking, there are three approaches to speech recognition [3] [4].

(a) Acoustic-phonetic approach: Acoustic-phonetic approach assumes that the phonetic units are broadly characterized by a set of features such as format frequency, voiced/unvoiced and pitch. These features are extracted from the speech signal and are used to segment and level the speech.

(b) Knowledge based approach: Knowledge based approach attempts to mechanize the recognition procedure according to the way a person applies its intelligence in visualizing, analyzing and finally making a decision on the measured acoustic features. Expert system is used widely in this approach.

(c) Pattern recognition approach: Pattern recognition approach requires no explicit knowledge of speech. This approach has two steps – namely, training of speech patterns based on some generic spectral parameter set and recognition of patterns via pattern comparison. The popular pattern recognition techniques include template matching, Hidden Markov Model [5].

2. HIDDEN MARKOV MODELS (HMM)

HMM is doubly stochastic process with an underlying stochastic process that is not observable, but can only be observed through another set of stochastic processes that produce sequence of observed symbols. The basic theory behind the Hidden Markov Models (HMM) dates back to the late 1900s when Russian statistician Andrej Markov first presented Markov chains. Baum and his colleagues introduced the Hidden Markov Model as an extension to the first-order stochastic Markov process and developed an efficient method for optimizing the HMM parameter estimation in the late 1960s and early 1970s. Baker at Carnegie Mellon University and Jelinek at IBM provided the first HMM implementations to speech processing applications in the 1970s [6]. Proper credit should also be given to Jank ferguson at the Institute for defense Analysis for explaining the theoretical aspects of three central problems associated with HMMs, which will be further
discussed in the following sections [7]. The technique of
HMM has been broadly accepted in today’s modern state-or-
the art ASR systems mainly for two reasons: its capability to
model the non-linear dependencies of each speech unit on
the adjacent units and a powerful set of analytical
approaches provided for estimating model parameters [8]
[9].

3. DEFINITION
The Hidden Markov Model (HMM) is a variant of a finite state
machine having a set of hidden states Q, an output alphabet
(observations) O, transition probabilities A, output (emission)
probabilities B, and initial state probabilities II. The current
state is not observable. Instead, each state produces an
output with a certain probability (B). Usually the states Q,
and outputs O, are understood, so an HMM is said to be a
triple (A, B, II).

4. DESCRIPTION OF HMM
For the description figure 1 shows an example of Hidden
Markov Model, The model consists of a number of states,
shown as the circles in figure. At time t the model is
in one
of these states and outputs an observation (A, B, C or D) [10]
[11] At time t+1 the model moves to another state or stays in the
same state and emits another observation. The transition
between states is probabilistic and is based on the transition
probabilities between states which are given in state j at time t+1.
Notice that in this case A is upper triangular. While in a general
HMM transitions may occur from may state to any other state, for
speech recognition applications transitions only occur from left to
right i.e. the process cannot go backwards in time, effectively
modeling the temporal ordering of speech sounds. Since at each
time step there must always be a transition from a state to a state
each row of A must sum to a probability of 1. The output symbol
at each time step is selected from a finite dictionary. This process
is again probabilistic and is governed by the output probability
matrix B where Bjk is the probability of being in state j and
outputting symbol k. Again since there must always be an output
symbol at time t, the rows of B sum to 1 [12]. Finally, the entry
probability vector πi, is used to described the probability of
starting in described by the parameter set λ = [π, A, B]

Figure 1: A Five State Left-Right, Discrete HMM for Four Output Symbols.
A HMM is characterized by the following:

- N, the number of states in the model. The individual states are denoted as $S = \{S_1, S_2, \ldots, S_n\}$ and the system state at time $t$ as $q_t$.
- $M$, the number of distinct observation symbols per state, i.e. the discrete alphabet size. The individual symbols are denoted as $V = \{v_1, v_2, \ldots, v_m\}$.

6. **The transition probability distribution** $A = \{a_{ij}\}$ where, each $a_{ij}$ is the transition probability from state $S_i$ to state $S_j$. Clearly, $a_{ij} \geq 0$ and $\sum_{j=1}^{N} a_{ij} = 1$.

- The observation symbol probability distribution $B = b_{jk}$ where, each $b_{jk}$ is the observation symbol probability for symbol $v_k$, when the system is in the state $S_j$. Clearly, $b_{jk} \geq 0$ and $\sum_{k=1}^{M} b_{jk} = 1$.

- The initial state distribution $\pi = \{\pi_i\}$ where, $\pi_i = P(q_1 = S_i)$, $1 \leq i \leq N$. HMM model can be specified as $\lambda = (A, B, \pi, M, N, V)$. In this thesis, HMM is represented as $\lambda = (A, B, \pi)$ and assume $M, N$ and $V$ to be implicit.

5. **USE OF HMM IN SPEECH RECOGNITION**

HMM can be used to model a unit of speech whether it is a phoneme, or a word, or a sentence. LPC analysis followed by the vector quantization of the unit of speech, gives a sequence of symbols (VQ indices). HMM is one of the ways to capture the structure in this sequence of symbols. In order to use HMMs in speech recognition, one should have some means to achieve the following:

**Evaluation:** Given the observation sequence $O = \text{user}$

$(o_1, o_2, \ldots, o_T)$ and a HMM $\lambda = (A, B, \pi)$ to choose a corresponding state sequence $Q = q_1, q_2, \ldots, q_T$ which optimal in some meaningful sense, given the HMM.

**Training:** To adjust the HMM parameters $\lambda = (A, B, \pi)$ to maximize $P(O \mid \lambda)$.

The following are some of the assumptions in the Hidden Markov Modeling for speech.

- Independent and therefore the probability of sequence of $T$ $\ldots, o_T) = \prod_{i=1}^{T} P(o_i)$.

Markov assumption: The probability of being in a state at time $t$, depends only on the state at time $t-1$.

The problems associated with HMM are explained as follows:

**a) Evaluation:** Evaluation is to find probability of generation of a given observation sequence by a given model. The recognition result will be the speech unit corresponding to the model that best matches among the different competing models. Now to find $P(O \mid \lambda)$, the probability of observation sequence $O = (o_1, o_2, \ldots, o_T)$ given the model $\lambda$ i.e. $P(O \mid \lambda)$.

**b) Decoding:** Decoding is to find the single best state sequence $Q = (q_1, q_2, \ldots, q_T)$ for the given observation sequence $O = (o_1, o_2, \ldots, o_T)$. Consider $\delta_t(i)$ defined as

$$
\delta_t(i) = \max_{q_t} P(q_1, q_2, \ldots, q_T \mid i, o_1, o_2, \ldots, o_t)
$$

(c) **Training (Learning):** Learning is to adjust the model parameters $(A, B, \pi)$ to maximize the probability of the observation sequence given the model. It is the most difficult task of the Hidden Markov Modeling, as there is no known analytical method to solve for the parameters in a maximum likelihood model. Instead, an iterative procedure should be used. Baum-Welch algorithm is the extensively used iterative procedure for choosing the model parameters. In this method, start with some initial estimates of the model parameters and modify the model parameters to maximize the training observation sequence in an iterative manner till the model parameters reach a critical value.

6. **CONCLUSION**

The conclusion of this study of recognition and hidden markov model has been carried out to develop a voice based machine interface system. In various applications we
7. REFERENCE:


8. ACKNOWLEDGEMENT:

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Shivam Sharma has completed his B.Tech, Computer Science and Engineering from JSS Academy of Technology, Noida PH-08010206467.
E-mail: sharmashivam2806@gmail.com