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Speech Recognition System for Indian Languages

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Abstract:

There are many open source packages available for automatic speech recognition (ASR) worldwide. In this project, it is aimed to customize a selected open source package for our Indian (Hindi) and Indian users. The outcome of the project is a system developed on an embedded system platform with LINUX OS capable of recognizing a set of Hindi words in a selected domain. Currently we focused on Hindi digits and terminologies related to agricultural domain. The embedded board we consider is Raspberry pi and the open source ASR package is Sphinx supported by United Kingdom acoustic model.

Keywords: Automatic speech recognition system, feature extraction, Hidden Markova model, Hindi grammar file, Hindi dictionary

1. Introduction

1.1. Study Background

Many successful systems have been developed for the automatic speech recognition system. This is characterized by

- Large vocabulary speech recognizing
- High accuracy
- Speed performance

CMU sphinx tool is the general term to describe group of speech recognizer. It has many versions like Sphinx which is basically used for embedded applications. POCKET SPHINX is a library that depends on another library called Sphinx Base which provides common functionality across all CMU Sphinx projects. Automatic speech recognition system is widely used in many applications ranging from mobiles to space missiles. Yet the quality of this recognition is far below human ability. Added to that, many time critical applications are unable to use ASR because of the heavy latency in processing the speech with a large vocabulary size.

Automatic speech recognition system (ASR) is not to be simple or easy able than the isolated word recognition. Error rates increases from isolated – word to speech. But we important only to develop continuous speech recognition. Large vocabulary file consists of the above 1000 words. When the number of words increases then confuse between the words also increases.

We describe Pocket sphinx, a large vocabulary speaker-independent Automatic speech recognition system. Sphinx base is to be also provided service to the HMM (Hidden Markova models) which is used for speaker independent system. In the pocket sphinx tool is to be also used as the HMM model that is Hidden Markova Model. HMM's are the easy to suitable for the automatic speech recognition system as it is largely done possibly forward and backward estimation of the algorithm. Due to resource management task, HMM is represented as the phonetic HMM's model.

HMM model is provided to the natural framework for the constructing language model. It is heart of the automatic speech recognition system. When speech is given from microphone to your system then feature extraction is to convert the analog signal in to the acoustic vector. This process is to be called as the feature extraction. Such as $X1:t=X1, X2, \dots, Xt$, it also provides compact representation of the waveform. Then this technique is to be useful to avoid loss of information provided by the dictionary. Decoder is to use to finding the sequence of the words. Such as $S1:t=S1, S2, \dots, St$.

2. Pocket Sphinx Installation

The project completely depends on Pocket Sphinx (PS) and the PS libraries depend on many other libraries and the procedure is given below.

- Install and create binary for Python
- Install and create binary for Bison

- Install and create binary for pulse audio
- Install and create sphinx
- Install and create pocket sphinx

3. System Overview

Here Fig1. Shows the automatic speech recognition system developed by using pocket sphinx tool.

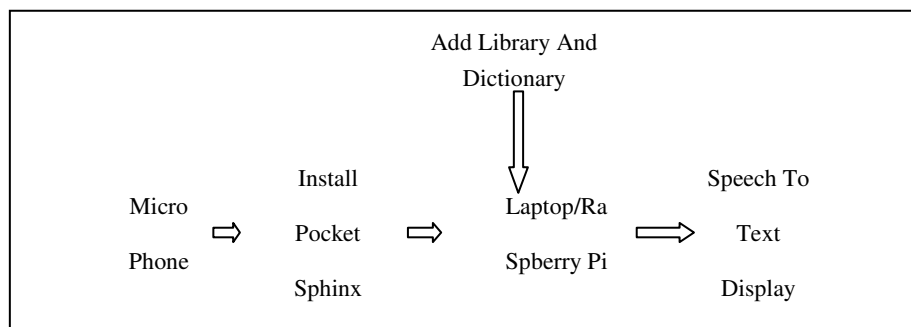


Figure 1: Block Diagram of ASR system

Development of speech recognition system in Indian Language for Indian users is to be developed on the basis of pocket sphinx tool software. Then first we can install the pocket sphinx into the system or the raspberry pi2. When we can give the speech from the microphone to the system. Given speech is in the analog signal. We are already install the pocket sphinx in the laptop or raspberry pi embedded board. When we can give the speech then it compares your pronunciation to the specific grammar file and dictionary file then display specific word on the screen. And hence it is very important to gives the specific path on the terminal window.

The output of the automatic speech recognition system in the Hindi language is display on the screen shown below:

```

INFO: fsg_search.c(1456): End node teen.20:29:61 (-1702)
INFO: fsg_search.c(1680): lattice start node <sil>.0 end node </s>.62
INFO: ps_lattice.c(1365): Normalizer P(O) = alpha(</s>:62:62) = -200396
INFO: ps_lattice.c(1403): Joint P(O,S) = -200396 P(S|O) = 0
000000026: teen
READY....
Listening...
Recording is stopped, start recording with ad_start_rec
Stopped listening, please wait...
INFO: cmn_prior.c(121): cmn_prior_update: from < 23.78 -0.82 1.82 -2.17 -2.38 -1
INFO: cmn_prior.c(139): cmn_prior_update: to < 24.36 -0.58 2.05 -2.08 -2.63 -1
INFO: fsg_search.c(1032): 76 frames, 1158 HMMs (15/fr), 3795 senones (49/fr), 199
INFO: fsg_search.c(1417): Start node <sil>.0:2:34
INFO: fsg_search.c(1456): End node <sil>.61:63:75 (-802)
INFO: fsg_search.c(1456): End node <sil>.61:63:75 (-802)
INFO: fsg_search.c(1456): End node teen.22:30:75 (-1432)
INFO: fsg_search.c(1680): lattice start node <sil>.0 end node </s>.76
INFO: ps_lattice.c(1365): Normalizer P(O) = alpha(</s>:76:76) = -191487
INFO: ps_lattice.c(1403): Joint P(O,S) = -191487 P(S|O) = 0
000000027: teen
READY....
Listening...
Recording is stopped, start recording with ad_start_rec
Stopped listening, please wait...
INFO: cmn_prior.c(121): cmn_prior_update: from < 24.36 -0.58 2.05 -2.08 -2.63 -1
INFO: cmn_prior.c(139): cmn_prior_update: to < 24.55 -0.18 1.92 -2.20 -2.43 -1
INFO: fsg_search.c(1032): 69 frames, 998 HMMs (14/fr), 3222 senones (46/fr), 182
INFO: fsg_search.c(1417): Start node <sil>.0:2:29
INFO: fsg_search.c(1456): End node <sil>.59:63:68 (-1096)
INFO: fsg_search.c(1456): End node <sil>.60:62:68 (-845)
INFO: fsg_search.c(1456): End node <sil>.60:62:68 (-845)
INFO: fsg_search.c(1456): End node noah.18:45:68 (-3723)
INFO: fsg_search.c(1456): End node don.18:32:68 (-2436)
INFO: fsg_search.c(1680): lattice start node <sil>.0 end node </s>.69
INFO: ps_lattice.c(1365): Normalizer P(O) = alpha(</s>:69:69) = -227825
INFO: ps_lattice.c(1403): Joint P(O,S) = -227839 P(S|O) = -14
000000028: don
READY....
  
```

Figure 2

3. Creation of Grammar Dictionary File

The original file of the grammar has the finite state grammar. Which has designing for the resource management task. The grammar file of the Hindi digits is to be shown in below:

#JSGF V1.0;

Grammar digit; public<digit>=yekldonlteenlcharlpachlchelsatlaatlnoa hl dus;

In this Hindi digits grammar file which is to be displayed by making dictionary file. Dictionary file is to be based on the pronunciation and make phonetics dictionary is to be shown in below:

- yek Y EH K
- yek Y EH K
- don D AO N
- teen TH IH N
- char CH AA R
- che CH EH
- sat S AE T
- aat AA T
- noah N OW AH
- dus DH AH S

The dictionary file is to be based on the pronunciation and make phonetics dictionary output with Raspberry has been shown below in fig 3.

```

192.168.137.2 - Remote Desktop Connection
INFO: continuous.c(275): Ready...
INFO: continuous.c(261): Listening...
INFO: cmn_prior.c(99): cmn_prior_update: from < 39.51 8.39 -7.46 -12.55 7.90 -2.36 -5.56 1.96 2.19 2.29 10.34 7.
INFO: cmn_prior.c(116): cmn_prior_update: to < 43.08 7.58 -8.37 -11.42 7.70 -3.02 -5.32 1.66 2.18 0.83 9.86 6
Input overrun, read calls are too rare (non-fatal)
INFO: cmn_prior.c(131): cmn_prior_update: from < 43.08 7.58 -8.37 -11.42 7.70 -3.02 -5.32 1.66 2.18 0.83 9.86 6
INFO: cmn_prior.c(149): cmn_prior_update: to < 41.99 6.59 -7.39 -11.55 6.96 -2.87 -5.07 0.91 2.52 1.19 9.50 6
INFO: fsg_search.c(843): 264 frames, 6123 HMMs (23/fr), 19153 senones (72/fr), 546 history entries (2/fr)

shirt
INFO: continuous.c(275): Ready...
INFO: continuous.c(261): Listening...
INFO: cmn_prior.c(131): cmn_prior_update: from < 41.99 6.59 -7.39 -11.55 6.96 -2.87 -5.07 0.91 2.52 1.19 9.50 6
INFO: cmn_prior.c(149): cmn_prior_update: to < 41.76 6.08 -6.84 -11.87 6.56 -2.85 -5.57 1.14 2.15 1.39 9.40 6
INFO: fsg_search.c(843): 77 frames, 1748 HMMs (22/fr), 5476 senones (71/fr), 245 history entries (3/fr)

shiva
INFO: continuous.c(275): Ready...
INFO: continuous.c(261): Listening...
INFO: cmn_prior.c(131): cmn_prior_update: from < 41.76 6.08 -6.84 -11.87 6.56 -2.85 -5.57 1.14 2.15 1.39 9.40 6
INFO: cmn_prior.c(149): cmn_prior_update: to < 41.32 6.33 -6.51 -12.10 6.10 -2.60 -5.80 0.03 2.00 1.83 9.34 6
INFO: fsg_search.c(843): 110 frames, 2683 HMMs (24/fr), 8307 senones (75/fr), 386 history entries (3/fr)

shirt
INFO: continuous.c(275): Ready...
INFO: continuous.c(261): Listening...
INFO: cmn_prior.c(99): cmn_prior_update: from < 41.32 6.33 -6.51 -12.10 6.10 -2.60 -5.80 0.03 2.00 1.83 9.34 6
INFO: cmn_prior.c(131): cmn_prior_update: to < 42.14 5.69 -6.91 -11.87 6.43 -2.87 -5.88 -0.06 1.99 1.76 9.24 6
Input overrun, read calls are too rare (non-fatal)
INFO: cmn_prior.c(131): cmn_prior_update: from < 42.14 5.69 -6.91 -11.87 6.43 -2.87 -5.88 -0.06 1.99 1.76 9.24 6
INFO: cmn_prior.c(149): cmn_prior_update: to < 43.78 3.12 -8.13 -11.14 7.65 -3.12 -7.04 -0.03 2.12 1.64 9.11 5
INFO: fsg_search.c(843): 161 frames, 3566 HMMs (19/fr), 11352 senones (62/fr), 492 history entries (2/fr)

shirt
INFO: continuous.c(275): Ready...
INFO: continuous.c(261): Listening...
INFO: cmn_prior.c(131): cmn_prior_update: from < 43.78 3.12 -8.13 -11.14 7.65 -3.12 -7.04 -0.03 2.12 1.64 9.11 5
INFO: cmn_prior.c(149): cmn_prior_update: to < 44.95 5.11 -7.51 -10.43 7.65 -3.35 -6.71 0.01 1.38 0.91 8.93 4
INFO: fsg_search.c(843): 170 frames, 3915 HMMs (23/fr), 12221 senones (71/fr), 408 history entries (2/fr)

shirt
INFO: continuous.c(275): Ready...
INFO: continuous.c(261): Listening...
INFO: cmn_prior.c(99): cmn_prior_update: from < 44.95 5.11 -7.51 -10.43 7.65 -3.35 -6.71 0.01 1.38 0.91 8.93 4
INFO: cmn_prior.c(116): cmn_prior_update: to < 44.88 5.17 -7.46 -10.41 7.66 -3.35 -6.75 0.05 1.32 0.96 8.91 5
INFO: continuous.c(261): Listening...
INFO: cmn_prior.c(131): cmn_prior_update: from < 44.88 5.17 -7.46 -10.41 7.66 -3.35 -6.75 0.05 1.32 0.96 8.91 5
INFO: cmn_prior.c(149): cmn_prior_update: to < 43.22 5.51 -6.82 -10.64 7.44 -3.01 -6.56 0.57 0.14 1.62 8.82 5
INFO: fsg_search.c(843): 92 frames, 2087 HMMs (22/fr), 6516 senones (70/fr), 257 history entries (2/fr)

shiva
INFO: continuous.c(275): Ready...
INFO: continuous.c(261): Listening...
INFO: cmn_prior.c(131): cmn_prior_update: from < 43.22 5.51 -6.82 -10.64 7.44 -3.01 -6.56 0.57 0.14 1.62 8.82 5
INFO: cmn_prior.c(149): cmn_prior_update: to < 45.24 7.03 -6.25 -9.89 7.54 -3.08 -6.74 0.20 0.01 0.71 8.44 4
INFO: fsg_search.c(843): 111 frames, 2717 HMMs (24/fr), 8397 senones (75/fr), 253 history entries (2/fr)

shiva
INFO: continuous.c(275): Ready...

```

Figure 3

4. Testing of the System

The In this paper a testing experiment is conducted for the grammar and dictionary created. The grammar consists of 10 words and tested with 5 persons with 20 trials word. Thus, there are 100 trials per word and following are the recognition accuracy we tested in a normal room environment

- 1) yek—98%
- 2) don-96%
- 3) theen -95%
- 4) char-92%
- 5) paach-90%
- 6) che-91%
- 7) sat-94%
- 8) aat-94%
- 9) noah-96%
- 10) dus 95%

It is found the result is speaker independent and also highly accurate. Our future work will be to develop grammar and dictionary for Hindi words related to agriculture.

5. Conclusion

In this paper, the speech recognition system in Hindi language is successfully done using Linux operating system and available open source pocket sphinx software easily. On the other hand, it observed that the performance of the Hidden Markova Model was very fatly and accruable. The recognition accuracy is tested to be 93.5% for Hindi digits. By using pocket sphinx software, the need of speaker independency is successfully done and it can be also used in the future application e.g. speech recognition system based on the home automation, some ASR based robotics system etc.

6. References

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