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Digital Signal Processing Voice Recognition in Matlab Source Code

Farheen Bibi on February, 6th 2013 in Computing

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Introduction

Voice recognition process the word spoken is taken as input to program. In this process it is important to that virtual reality because it provides a very natural & intuitive way by controlling the simulation while is allows the user to remain free.In this article we will dive into the uses of voice recognition by using virtual reality and then we will examine how recognition of voice is performed.

Characteristics of an individual's voice were thought to be able to characterize the uniqueness of an individual much like a fingerprint. The early systems had many flaws and research ensued to derive a more reliable method of predicting the correlation between two sets of speech utterances. Speaker identification research continues today under the realm of the field of digital signal processing where many advances have taken place in recent years.
The efficiency of speech recognition system is dependent on word error rate in speech recognition which is in % and is measured for some kind of specific technologies for example if a task is given and you can say in a specified mode and specified word vocabulary is also considered.Sometimes robust speech recognition systems may be applied to gain the high accuracy.

Objectives

The objective of this project is to recognize voice uniquely and compare it against a database to check weather person is authorized on not. But a problem is that a human can easily recognize a voice and a program cannot recognize this easily so this task is very difficult for the computer to recognize the different features of voice.Many Complex

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problem arise while writing algorithm for different kind of voice recognition system.This problem may arise because to say a same word at different times becomes difficult.Some of the factors that are vary or change time to time in the human speech that how fast word is being spoken.

Approach

In this section, the first step is to define time frame for recording command words having duration=40000 mille seconds, frequency fs=8000 HZ. The next step is to record key word sample using "wavrecord" command, take value above 0.1 magnitude voice sample, calculate the difference and store the file using "wavwrite" command. To store other samples for key words procedure is same as previous. In the second step read the file and take above 0.1 magnitude value for the current voice sample. Calculate the difference and store in a variable which is then compared with pre define time frame if it match then give the output. The time frame to speak and store key words is as shown in table-I below.
Step First:
1) file_reverse=wavrecord (duration1,fs); This command is used to record command word with
parameters: time frame "duration1=40000" and frequency "fs=8000".
2) [x_reverse y_reverse]=find(file_reverse>.1);
This command is used to take above 0.1 magnitude speech sample and discard below.
3) diff_reverse=max(x_reverse)-min(x_reverse);
This command finds the difference between maximum and minimum value of speech sample and store in a variable.
4) wavwrite(file_reverse,'c:\voice\reverse.wav');
This command is used to store voice sample in memory location of the computer.
Then plot the graph between time and magnitude axis.
Step Second:
In this step we compare the recorded sample with the real time speech.

Implementation

Two approaches are used recognition the voice this can be done by dividing into two classes: first is **Template Matching** while other is **Feature Analysis**. Template matching is very straight, simple, easy and possess the very highest accuracy & efficiency & when it is used very properly but it it problem is that it has many limitations. In voice recognition system in 1st step is for the user who gave input by speaking a word just like speaking in a microphone .Now speech signal which is analog electrical signal is converted into digital signal by using A2D converter and then this result is stored into memory.Now this digitized data is matched with template to determine the features of sound.Since voice of every person is changed so program is not so efficient to store voices of all persons so it is made learning agent to learn the voice of person who mostly used these services .

Matlab Implementation

```
'Get ready to record your name'
for i = 1:1
file = sprintf('%s%d.wav','rec',i);
input('Press enter when ready to record your name');
y = wavrecord(88200,44100);
soundsc(y,44100);
wavwrite(y,44100,file);
end
name = input ('Enter the name that must be recognized:','s');
for j = 1:1
file = sprintf ('%s%d.wav','rec',j);
path='rec1.wav';
file=cat(2,path,file);
[t, fs] = wavread (file);
% Crop recording to a window that just contains the speech
s = abs (t);
```

```
end
‘now the comparison algo’
input (‘Press enter when ready’)
usertemp = wavrecord (88200,44100);
sound ( usertemp,44100 );
rec = input ( ‘ Press button 1 to record again or press enter to proceed:’);
while rec == 1
rec = 0;
input (‘Press enter when you ready’)
usertemp = wavrecord (88200,44100);
soundsc (usertemp,44100);
rec = input (‘Press 1 again to record again : ‘);
end
% Crop recording to a window that just contains the speech
s1 = abs (usertemp);
%s1=usertemp;
subplot(2,1,1);
plot(s);
title(‘Your Original voice’);
subplot(2,1,2);
title(‘Tested voice’);
plot(s1);
if s==s1
fprintf(‘You are %s’,name);
else
fprintf(‘You are not %s’,name);
end
```

Diagram

Dsp voice recognition

CONCLUSION

The implemented algorithm and control system control fan speed, temperature of heater and robot direction using the voice key word. It demonstrates its reliability and ease of future development. Based on obtained experimental results it demonstrates that the proposed algorithm is indeed functional and it can be used in voice key word control of home appliances and industrial robots. Percentage of correct recognition of commands is high enough. The main contribution of this study is that it presents the idea of key word recognition and control system. The experiments also show that the approach is good for key word recognition.

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Farheen Bibi

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kuldeep kumar saini

on March 30, 2013 06:39:36
Reply

This Matlab code is vry helpful..

ammadkhan

on March 30, 2013 07:58:58
Reply

Thanks For Appreciation.

ahmed alhanbly

on April 17, 2013 20:03:09
Reply

did this s=project work in r 2012a

M Amu Khan Baloch

on April 17, 2013 21:13:59
Reply

yes

Amr

on April 21, 2013 19:27:49

Reply

Hello bro. !!
how can i get the code ?!
if there is the source code it would help me in My Graduation Project

Amr

on April 21, 2013 19:55:39

Reply

can we merge it with a C# Application ?!

M Amu Khan Baloch

on April 24, 2013 12:39:01

Reply

Well everything is possible just follow this link http://www.mathworks.com/matlabcentral/fileexchange/12987-integrating-matlab-with-c

HJ

on July 13, 2013 12:54:33

Reply

Hi, thanks for providing this but I have few queries.

Firstly, based on your explanation the following command is used to take above 0.1 magnitude speech sample and discard below. '2) [x_reverse y_reverse]=find(file_reverse>.1);'

May I know why do we have to take above 0.1 magnitude speech sample and what does discard below means?

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