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Design and Implementation of Speech Based Scientific Calculator

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Abstract: **In recent days speech recognition based systems are gaining more prominence and are used in a wide range of applications. Calculator has become an indispensable part of our lives, used at home, at local retail shops or at workplaces like banks, IT companies, medical laboratories etc. Speech recognized calculator is one of the speech recognition systems, in which the digits and operators spoken by the user with pauses in between are analyzed using various algorithms and finally one of the 10 digits (0 to 9) and one of the 5 operators(+, -, *, /, =) The objective of our paper is to design and implement the speech recognized calculator in MATLAB and Standalone Microcontroller. This designed system is able to take detect the digits and operators of the arithmetic expression whose result is to be calculated. In this paper we have used linear predictive method for recognizing the digit/operator in MATLAB and Analog Speech Recognition Chip.**

Index Terms—**Speech recognition, Analog Speech Recognition Chip, Microcontroller ,MATLAB,SRC**

I. INTRODUCTION

Speech is the vocalized form of human communication. It ranges from 90 Hz to 7,000 Hz. Each spoken word is created out of the phonetic combination of a limited set of vowel and consonant speech sound units. The voiced speech of a typical adult male will have a fundamental frequency from 85 to 180 Hz, and that of a typical adult female from 165 to 255 Hz. Thus, the fundamental frequency of most speech falls below the bottom of the "voice frequency" band as defined above.
People use calculators to perform calculations that have a numeric keypad to input the data and a display to get the result. To overcome this, speech signal can be used to input the data. Speech recognition is the process of automatic extracting and determining linguistic information conveyed by a speech wave using computers. This Speech recognized calculator can be implemented by using linear predictive coding (LPC) method.

The developed system would be able to take operands and the operation commands as voice inputs and perform the mathematical operation. This system includes two main stages.

First stage is the training phase which consists of feature extraction using linear predictive coding (LPC) and storage of extracted features as training data in the form of reference templates.

Second stage is the testing phase in which the features of real time voice inputs are extracted and compared with the reference template using Euclidean distance criterion to recognize the input digit. In this paper we are design and implementing speech recognition calculator by two one by the help of MATLAB and A standalone calculator microcontroller using.

II. BLOCK DIAGRAM OF THE SYSTEM

a. Block Diagram Of Matlab Based SRC

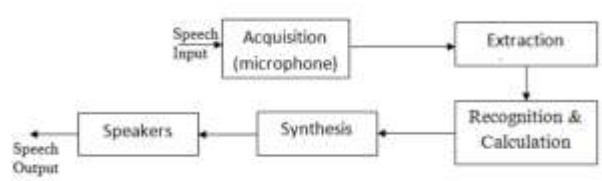


Figure 1: Above figure shows the MATLAB based Speech Recognised Calculator

The Figure 1. Shows the MATLAB design which is explained below. There are various steps involved in the implementation of Speech recognized calculator and the steps are:

1. Speech Acquisition
2. Speech Extraction
3. Speech Recognition
4. Pattern Training and Decision Logic

1. Speech Acquisition

Speech acquisition refers to the process of acquiring the required speech input from the user. A close talking microphone is used for recording the speech signals and it replaces the numeric keypad.

During speech acquisition, speech samples are obtained from the speaker in real time and stored in memory for preprocessing. As the articulation bandwidth of voice is 300 Hz to 3 kHz. According to the Nyquist Sampling Theorem (Fs ≥ 2*Highest frequency) ,the sampling frequency is 8kHz as the minimum sampling rate. Here in our project we have chosen the sampling rate to be 10 kHz so that more samples can be obtained which will be a greater advantage in further processing.

2. Speech Extraction

After speech acquisition stage the speech signal or the recorded speech interval is fed to the extraction stage where the uttered digit or the operator is extracted from the entire recording interval. The recording interval here consists of voiced segments and unvoiced segments, the unvoiced segments are usually referred as the pause periods which usually do not convey any sort of information. To extract the voiced segment from the recording interval, we will first locate the maximum speech amplitude in the recording interval. Later, we will extract 3000 samples on either side of the maximum speech amplitude sample, from this one can rely that for each uttered digit there will be 6001 samples and the maximum speech amplitude sample resides at the 3001 sample. These samples are enough for the recognition of digits and the operators.

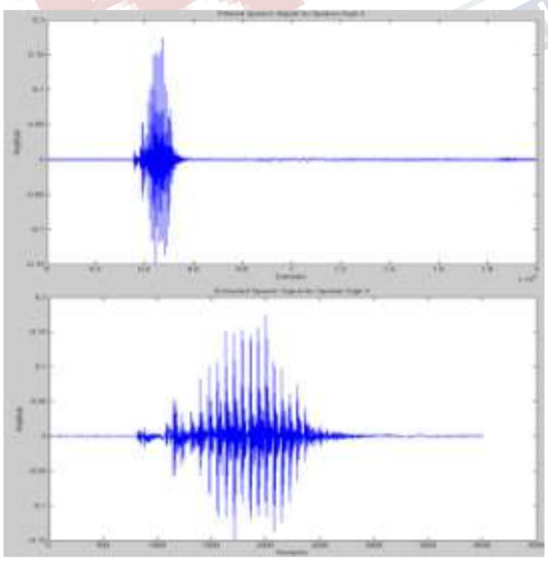


Figure 2: Above figure is Original Speech Signal and below figure is Extracted Speech Signal

3. Speech Recognition

Speech recognition is also defined as the process of extraction of linguistic information from speech signals. It is the heart of this paper . There are many approaches for speech recognition. Feature extraction is one of the approach for speech recognition which we are using in this paper .The extraction of features from speech signal is achieved by some spectral analysis technique such as Filter bank analyser, Linear predictive coding analysis, or a Discrete Fourier Transform analysis. We are employing the linear predictive coding technique to estimate the features of speech signal in this paper.

III. LINEAR PREDICTIVE CODING (LPC)

Linear prediction analysis of speech is historically one of the most important speech analysis techniques. The basis of LPC model is that, for a given speech sample at time n, s(n), can be approximated as a linear combination of the past p speech samples, such that

$$\tilde{s}(n) \approx a_1 s(n-1) + a_2 s(n-2) + a_p s(n-p) \quad (1)$$

where the coefficients a1, a2,.... ap are assumed to be constant over the speech analysis frame in the equation represents the order of the linear predictor.

The LPC can be achieved by some method such as Auto Correlation Method, Covariance Method and Lattice Method in this paper we are implementing LPC by Auto-Correlation Method.

4. Pattern Training & Decision Logic

In pattern training, one or more test patterns corresponding to speech sounds of the same class are used to create a pattern representative of the features of that class. The digits from 0 to 9, operators +, -, * and / and equals are referred as CLASS and there are 15 classes. These each class contains one or more patterns of spoken digit or operator.

To compare speech patterns (which consists of a sequence of spectral vectors), we require local distance measure (LDM). The local distance is defined as the spectral distance between two well defined spectral vectors.

And decision of spoken digit or operator is chosen by class with the minimum distance of respective digit or operator in database.

These above steps are implemented on the MATLAB for Speech Based Calculator

IV. SYSTEM IMPLEMENTATION

Block Diagram Of Microcontroller Based SRC

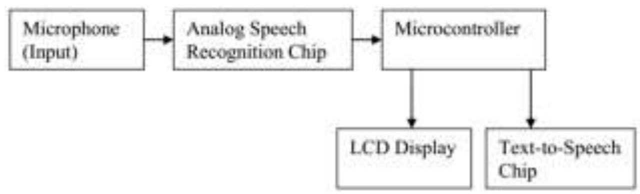


Figure 3: Above figure shows the Microcontroller based SRC.

Figure 3 shows a block diagram schematic of the entire Speech Recognition Calculator system. Voice signal input is provided via a microphone, converted by the analog SR (speech recognition) chip into digital signals, and then fed into the microcontroller for actual computation and display.

The analog speech recognition chip is trained for different function of calculator operators, numbers etc. The analog speech recognition chip assign different value corresponding to its operators, numbers.

These values are taken to microcontroller and programed to perform calculation and display on LCD after that text is spell out by text to speech chip.

V. APPLICATION, ADVANTAGE AND LIMITATIONS

A. Application

1. Interactive voice response system (IVRS) Nearly PBX/Voice Mail devices permit callers to speak commands instead of pressing buttons to send specific voice.

2. Voice-dialling in mobile phones and telephones Some new cellular phones comprise C&C speech recognition that permits spoken such as ”Call Home”. This could be a main factor in the future of ASR.

3. Hands-free dialling in wireless Bluetooth headsets (its almost the same as above two applications)

4. PIN and numeric password entry modules This finds a great application in security related issues. The access to a system can be provided based on user identification which can be done based on the recognition of speech.

5. Automated teller machines (ATMs) It finds application in communication systems by getting the voice input from the user at one side, translating the text into other language for the understandability of the other user and then conveying the desired output voice signal for the user at another side

B. Advantages

Following are the advantages of voice activated calculation:

1. Does not require on-line training
2. Relatively cheaper
3. High reliability and flexibility
4. Hands free calculation Following are the 4. Hands free calculation

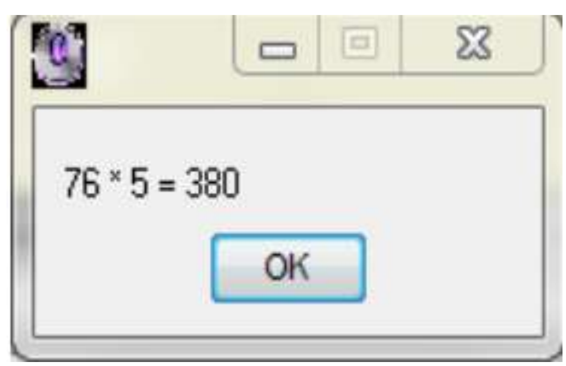
C. Limitation

Following are the limitations:

1. Alters based on different health or mental conditions.
2. Larger off-line training.
3. Not suitable for noisy environment

VI. RESULTS

Design and Implementation of Design and Implementation of Speech Based Scientific Calculator was implement both in software and standalone hardware i.e MATLAB and Microcontroller



VII. CONCLUSION

In this paper, basic principles and properties of human speech were investigated and digital signal processing techniques on speech signal were studied. Finally, this implementation for standalone is done using a microcontroller.

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