Aswathi Cherkkil. Int. Journal of Engineering Research and Application *www.ijera.com*
ISSN : 2248-9622, Vol. 7, Issue 12, (Part -6) December 2017, pp.74-77

RESEARCH ARTICLE OPEN ACCESS

Multifunction Intelligent Headset

Aswathi Cherkkil**, Sreelekshmi P.S*,Reshmi.S*
*** (B.tech Student, Department of Electronics and Communication Engineering, NSS College of Engineering, Palakkad.*
**(Assistant Professor, Department of Electronics and Communication Engineering, NSS College of Engineering, Palakkad.*
**(Assistant Professor, Department of Electronics and Communication Engineering, NSS College of Engineering, Palakkad*

ABSTRACT
Personal sound systems have become an overwhelming trend of today's lifestyle. Headsets have been developing throughout the years focusing on improving sound quality. The risks involved in using headsets especially on road is something that has been underrated. It has been found one in ten of road accidents are caused by headphone wearing pedestrians. In this paper we are proposing an idea for a family of intelligent headsets that help reduce this danger by using sound recognition to detect and warn the user in case of emergency vehicle sirens, alarms at crosswalks and sounds of incoming trains. Noise cancellation is implemented to achieve a better sound clarity as well.

***Keywords* –** noise cancellation, sound detection, sound recognition, voice detection, adaptive filtering.

Date of Submission: 13-12-2017 Date of acceptance: 28-12-2017

I. INTRODUCTION

Over the years the popularity and production of headsets have only witnessed a rampant growth. The manufacturers are dedicated to producing better versions of headsets with improved sound clarity and quality. The popularity of the product is such that every other you individual you see is using a headset. Although the qualities of the product vastly shadow the risks involved, recent studies have shown that the number of road accidents caused by headphone wearing pedestrians and drivers have been increasing. To find a solution to the problem we came up with the idea to develop a headset that detects particular sounds form the surrounding of the user and automatically controls the sound input to the earpiece so that any chance of accident which can be caused by the ignorance of the sound, can be avoided. The device is trained to detect sounds of fire engines, ambulances, incoming trains, other emergency vehicle sirens and also to identify the name of the person being called (all from a sufficient distance). And upon detection the sound input to the earpiece is either reduced or entirely cutoff automatically. Adaptive filtering is used to improve the sound quality.

Sound detection and recognition is a technology that has immense potential to be tapped for applications in various fields. Here it is used to detect sounds in the background that may go unnoticed by a headset user. Speech recognition modules are readily available for use. The module can also be trained to identify machine sounds and alarms. We have used EasyVR version 3.0 speech/voice recognition module. The analog voice controller of the headset is replaced by digital potentiometer M62429 for sound controlling. Adaptive noise cancellation using NLMS algorithm is employed for improving sound quality. The noise cancellation is implemented using SIMULINK-MATLAB.

II. SOUND DETECTION AND RECOGNITION

Speech recognition comes under Natural Language Processing, it enables the computer to derive meaning out of input language. It is a field of computer science, artificial intelligence and linguistics involved in the interaction between language and computer systems. Sound detection modules available are unable to detect the sounds on grounds of any other feature other than the intensity of the sound. The idea proposed required the detection of different sounds of same intensity or different intensities. Therefore, a high quality voice recognition module the EasyVR 3.0 is trained to detect and recognize a variable array of warning signals and sounds, along with voice recognition for volume control as well as detection of the users' name being called. The module has recognition which is both speaker/source dependent as well as independent which makes it a suitable for sound detections.

Speech recognition system is a continuous mixture density hidden Markov model (HMM) and the parameters are estimated by Viterbi training. The steps involved in speech recognition are as follows:

first the normalization is done, to find the cesptral coefficient, by taking Fourier transform of a short time window speech and then decorating the spectrum using inverse Fourier transform to recognize the pattern for different speakers and recording conditions[1]. Phenomes are recognized on the basis of pattern in the second step. The database will have different sets of patterns representing the states required for completion of a phenome. Then the multilevel checking of the audio input from the desired audio is done by converting the data into a string by finding the ASCII transformation for each data generated which is used to encode the voice string pattern for extraction of the required pattern. The patterns are matched from the ones in the database. After the threshold value, which is calculated by summing up the weight associcated with the states of automata, is crossed; the task associated with the command is done.

The EasyVR 3.0 is a multipurpose speech recognition module with 26 speaker independent and 32 speaker dependent commands that can be stored. It has ease of training and PC compatibility by means of the EasyVR commander. The machine sounds including the sound of fire engine alarm, ambulance siren, train sound, loud train and vehicle Horns are all stored under a single group of speaker independent commands. For training the sounds are played to the microphone and the command is thus stored for pattern checking. In order to facilitate voice control feature the "volume up", "volume down" commands are stored in the speaker dependent group, dedicated to the user alone. Also the name of the person is stored in the speaker independent group. So that if a person is being called from behind the name is detected and the volume is decreased so that he may be able to respond.

III. NOISE CANCELLATION

Filtering is the basic method used in noise cancellation. Filters can be configured accordingly for the extraction of desired signal. Fixed coefficient filters have disadvantages when working in a dynamic environment hence adaptive filters are preferred. Adaptive filters, which are based on negative feedback, estimate the noise and apply appropriate weights accordingly for eliminating the noise. Least mean square algorithm (LMS) was introduced by windrow and hoff and has a quasi periodic nature of speech. The coefficient update equation is given by: (1).

$$w(k+1)= w(k)+\mu * x(k) * e(k) \quad (1)$$

Here $w(k+1)$ is the filter coefficient of the next iteration, $w(k)$ is the coefficient for the present iteration,α is the step size which determines the convergence, $x(k)$ is the input value, $e(k)$ is the error value. Instability of input signal power, changes the value of step size, and results in a change of convergence rate, which is a limitation to LMS algorithm.[2] Normalised LMS algorithm was developed to solve the problem. The problem is solved by normalizing the input power. The variable step size is taken as: (2)

$$\mu= \beta/((c+ | |I (n) | |^2) \quad (2)$$

where the step size variation is now dependent on adaptation constant β and (0< β <2) and c is the normalization constant (<1). The variation in step size is what results in the optimized convergence rate [2]. The updated step size in the weight vector equation is used to achieve faster convergence rate and stability of algorithm.

The noise cancellation section is implemented using Simulink developed by Mathworks, which is a graphical programming environment for modeling, simulating and analyzing multi domain dynamic systems. It is integrated with MATLAB.

IV.PRINCIPLE OF OPERATION

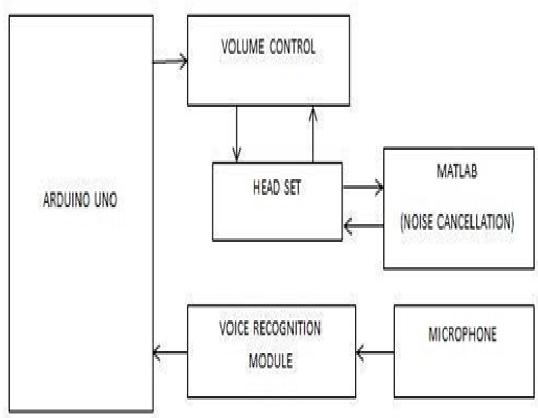

Fig 1. Block diagram

The system block diagram in fig 1consists of Arduino Uno micro controller, a voice recognition module and its microphone, MATLAB, headset and a volume control block. The Arduino Uno board is used for micro controller interface. In the proposed system, a stereo headset with an adjustable voice controller is used. The analog volume control is then replaced by a relay circuit. MATLAB is used for noise cancellation in the headset. A voice recognition module; the Easy VR is used. The VR module has a microphone which is used to collect the input voice.

V.EXPERIMENTAL SETUP AND SIMULATION

The EasyVR module that is used comes with a microphone and user friendly interface, EasyVR commander which is the software that is used to train the module in detecting sounds and human commands. The digital potentiometer IC M62429 is connected to the headset via the Arduino uno microcontroller as shown in Fig.2 The outputs from the digital potentiometer pins 2 and 7 are given to the left and right channels of the headset. Seven commands were stored in group 1 of the VR module. Two of which were Volume up and Volume down used to control the volume of the audio played through the headset. Upon detection of these commands the arduino will sent input to m62429 which will either increase of decrease the resistance for decreasing or increasing the volume, respectively. The data input is of the from D1 to D10 where D0 is used to select any one channel, and D1 is for using both channels at a time or a single channel. D2 to D8 is the data that controls the volume the volume code for different attenuations were selected from the datasheet and the arduino was programmed appropriately.

The same process is involved in completely cutting off the audio played in the headset, where when the VR module detects and recognizes the sounds trained which are alarms and emergency sirens or the name of the user, the potentiometer is set to the maximum attenuation so that the audio is nearly cutoff. Because emergency signals need immediate and complete attention. The user can simply increase the volume after the danger has passed.

The noise cancellation is simulated in Simulink. An acoustic environment model is first created. For the simulation the regular environment noise in a home ambience is initially stored as a "wav." file. A bandpass filter passes the real time sound input to the microphone and adds it to the noise which is then passed to the users mic output port. The acoustic environment produces two output signals , the signal output at the exterior mic port which is the noise signal. And the signal output at the users mic port composed of noise added to the signal from wav. file. An adaptive filter removes noise from users mic signal. Normalized LMS algorithm is used and the filter length is 40, The step size μ is chosen as 0.4, this can be varied for slow or fast adaptation. An audio device reader is used to record the sound coming in the microphone and save it as a wav. file. This noise is set as the background noise of the speaker, this along with the audio input device is connected to the adaptive LMS filter block. The output is heard through the headphone where the noise signal is heard to be present initially, but as the model keeps working, the filter adapts itself to the nosie and finally the noise is reduced to very low levels, unrecognizably.

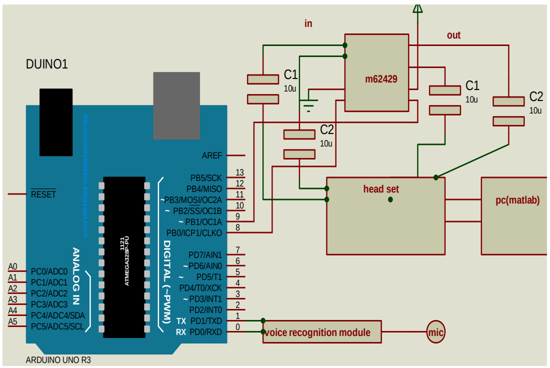

Fig.2 circuit diagram

IV. RESULTS AND CONCLUSION

The hardware testing was successfully completed with satisfying results. The VR module detects both voice commands as well as machine sounds. The audio output was clear and the volume control was distinct with 4 different volume levels. Sound detection has an accuracy of 85%. Sometimes the noises similar to the machine sounds trained interfere with detection. However the model delivered acceptable performance. Simulation of noise cancellation was successfully completed. The simulation was done real-time the noise collection using the microphone and the final audio play out was done real-time with satisfactory results.

The distance of voice recognition in the detection module used is limited to 1.5-3 meters maximum. Recognition systems with better accuracy at farther distances can vastly improve the performance of the headset. Further scope includes the usage of dry EEG sensors for sleep detection also, which can regulate the volume of the audio played with the intensity of sleep so that there is a gradual decrease in the volume until the person completely sleeps.

REFERENCES

[1]. Anshul guptha, Nileshkumar Patel, Shabana Khan "Automatic speech recognition technique for voice command", IEEE 32331.
[2]. Ms.Mugdha.M.Dewasthale "Comparative Performance Analysis and Hardware Implementation of Adaptive filter algorithms for Acoustic Noise Cancellation" 2015 International Conference on Information Processing (ICIP) Vishwakarma Institute of Technology. Dec 16-19, 2015
[3]. Anuja N,Kishori S, "Survey on Noise Cancellation Techniques of Speech Signal by Adaptive Filtering", 2015 International Conference oPervasiveComputing (ICPC)
[4]. Stefan Liebich, Carlotta Anem¨uller, PeterVary, "Active Noise Cancellation in Headsets by Digital Robust Feedback Control"

2016 24th European Signal Processing Conference (EUSIPCO)
[5]. Raj Kumari ,Rajesh Mehra, ”Effective Adaptive Noise Canceller Design Using Normalized LMS”, 2015 1st International Conference on Next Generation Computing Technologies (NGCT-2015)

International Journal of Engineering Research and Applications (IJERA) is **UGC approved** Journal with Sl. No. 4525, Journal no. 47088. Indexed in Cross Ref, Index Copernicus (ICV 80.82), NASA, Ads, Researcher Id Thomson Reuters, DOAJ.

Aswathi Cherkkil "Multifunction Intelligent Headset." International Journal of Engineering Research and Applications (IJERA) , vol. 7, no. 12, 2017, pp. 74-77.