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Background Noise Suppressor for Speech Signal

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Abstract—: **Background noise is always the biggest matter of concern when it comes to the end user communication devices. These noises are unexpected and degrade the quality of communication. During the conversation over a mobile phone if the caller party is on a busy road, the called person hears the noises along with the voice of the caller party. In order to overcome this inconvenience we can use several noise cancellation mechanisms. Noise suppression algorithm is one of such mechanism that can be implemented in the mobile phones to eliminate or suppress the background noise and pass only speech signal of the caller party to the called party. This paper describes the technique used for the development of noise cancellation system to cancel the background noise from the speech signal. Our proposed system uses dsPIC controller that will suppress the background noise in end user communication. The noise suppression algorithm is burnt in the dsPIC using the programmer Pickit-3.**

Keywords: **Noise suppression algorithm, dsPIC, Pickit-3.**

I. Introduction

Quality of speech is one of the most important matters of concern when we talk about the end-to-end communication devices. An example of such a device is the mobile phones. In order to achieve a communication which is uninterrupted, the quality of speech signal should be excellent which means that the speech signal should be clear so that the person on the other side i.e. called party can hear properly and reply. This should happen on both sides. Real time noise suppression algorithm can be provided as an important feature in the mobile phones to provide a better quality of speech signal at the receiver end.

Let us consider an example where a person is trying to make a mobile call from a noisy environment such as a noisy street, crowded train stations or some musical restaurant. Under such conditions the background noise makes it impossible to hear the incoming call as shown in Fig.1. The Fig. 1 shows that background noise is added with the speech signal at the transmitter side due to which the receiver intercepts the noise along with speech. This makes the situation complicated or sometimes worse when the person is yelling into the phone with an attempt to hear. These noises cannot be totally eliminated but it can be suppressed using the noise cancellation techniques. These techniques use suitable filters along with noise suppression algorithm.

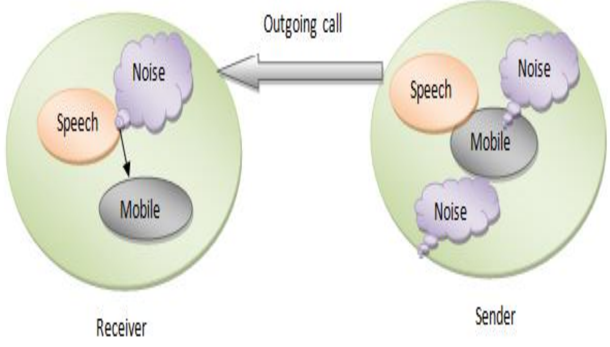

Fig. 1.1 Illustration of Phone Call in Noisy Environment

II. Literature Review

With reference to some papers we come to know that there are various techniques proposed for noise cancellation. Some research workers use the LMS algorithm; some use adaptive filters while many others use the microphone systems. Each of these techniques cancels the background noise using some principle.

A. Design and Development of Noise Cancellation System for Android Mobile [1]

The performance of the adaptive filter has been compared with different filter taps and step size for speech signal. A filter length 128 and step size 0.9 for the adaptive filter in the NCS provided the best performance. With these tuned parameters, the NCS application has been able to perform acceptable adaptive noise cancellation. The quality of signal can be further improved by enhancing the adaptive filter algorithm.

B. Speech Enhancement Algorithm to Reduce The Effect of Background Noise Signal in Mobile Phone[2]

A simple post processing scheme is proposed that can be applied to the output of STSA speech enhancement algorithm. The post processing algorithm is based on using spectral properties of noise in order to detect noisy time-frequency regions which are then attenuated using SNR based rule. A suitable suppression rule is developed that is applied to the suitable noisy regions so as to achieve significant reduction of noise with minimal speech distortion.

C. Noise Cancelling Headphones[3]

NCH reduce the noise from active noise cancellation. Headphone's ear cups are able to block out some high frequency noise because it is absorbed. Active noise cancellation system is a close loop system and can attenuate the noise below 1 KHz.

III. Proposed System

In most of the systems we have seen that various techniques including adaptive filter, spectral subtraction, etc. are used for suppressing the background noise signal. In this paper we have proposed a system that uses two microphones for the reception of the input signal. One microphone which is termed as primary microphone or main microphone will capture the speech signal as well as the background noise whereas the other microphone which is termed as the secondary microphone will capture the background noise. These inputs will undergo the processing which mainly includes the microphone biasing, filtering of signal and amplification of signal. The filtered signal is a high pass signal which is given as an input to the digital signal processor.

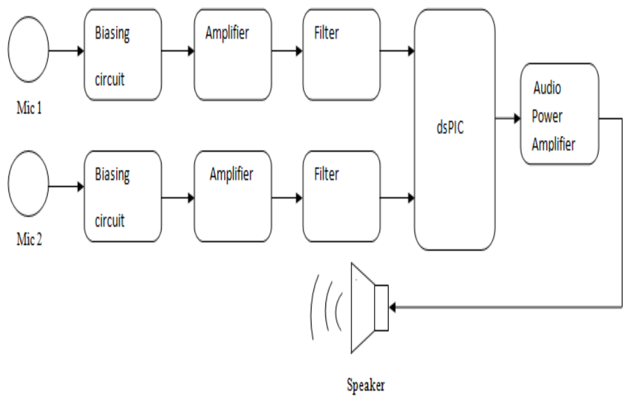

Fig. 3.1 Block diagram of noise suppressor

Biasing: The biasing is done so that microphone may function properly and in order to keep the transistor operating in the active region.

Amplifier: The output of microphone is very low amplitude signal near about 20mv and for processing we need to amplify it to interface with DSP. To increase amplitude level amplifier block used. The amplifier used is a transistor amplifier.

Filter: Audio signal have specific bandwidth which is 2 KHz to 20 KHz. But most probably the signal are of 5KHz frequency or greater than that. So we need to use a high pass filter which passes signal of 5KHz or greater and block other signals.

dsPIC: Digital signal processor will sample both input amplified signal and then based of different conditions of the input to both the microphone it will give the input to the DAC. This reduces all the remaining noise. The output of ADC is sent to DAC blocks which convert digital signal to analog signal.

Audio Power Amplifier: The strength of output signal from the DAC is very low so we cannot directly give this output to the speaker. So for this reason we use an audio power amplifier to amplify the strength of the audio signal..

IV. Methodology

The important feature of this system is using the dsPIC. Different cases are considered for the input to the digital signal processor. Based on this input combination of primary and the secondary microphone the output of the DAC is decided. In our implementation we have considered four different cases which are as follows:

Case I:

If input sample of primary microphone is less than a particular threshold level then output of the DAC will be the input of secondary microphone.
i.e,
If mainMic1stSample < 10
Then DAC output = subMic1stSample

Case II:

If input sample of secondary microphone is less than a particular threshold level then the output of the DAC will be the input of primary microphone.
i.e,
If subMic1stSample < 10
Then DAC output = mainMic1stSample

Case III:

If input sample to the primary and the secondary microphone is greater than the threshold level and are not equal then the output of the DAC will be substraction of the secondary microphone input from the primary microphone input.
i.e,
If mainMic1stSample > 10 and subMic1stSample > 10
Then
DAC output = mainMic1stSample – subMic1stSample

Case IV:

If input sample to the primary and the secondary microphone is greater than the threshold level and are equal then the output of the DAC will be the input of the primary microphone.
i.e,
If mainMic1stSample = subMic1stSample

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Then DAC output = mainMic1stSample.

V. Plan Of Implementation

A. Hardware

We have used digital signal processor for the implementation for of our project. The dsPIC is the main or the core entity of our project. Except dsPIC other hardware we have used are power supply, filters, amplifiers, audio power amplifiers, speakers, microphone, etc.

B. Software

For the software part of our project we are going to use software's are such as MPLAB exe, proteus 8 etc. We are using the software's for coding purpose as well as for the simulation purpose.

VI. Conclusion

Our proposed model avoids external man made and machine made noise and improves voice quality of end user. Filter block already suppress machine made noise whose frequency is above 20 KHz and power supply noise which is below 2 KHz. The rest of the noise will be removed by the dsPIC module. Thus, at the end of processing we get a noise free speech signal.

VII. Result

The results were checked for the below shown simulation done in proteus.

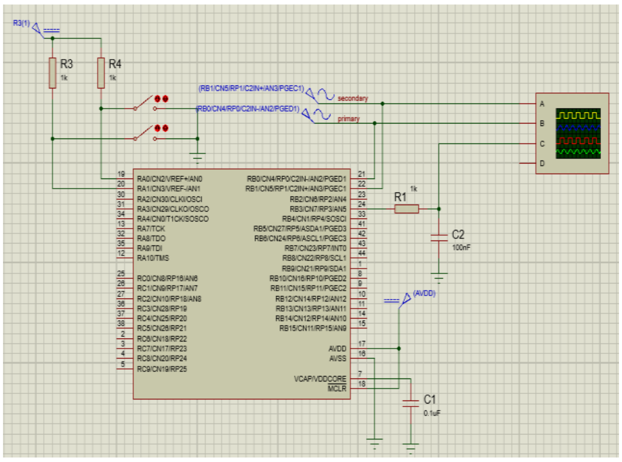

Fig. 4.1 Simulation in Proteus

The different cases mentioned in the methodology were tested and their results are as follows:

Channel A shown by yellow color represents secondary microphone input.

Channel B shown by blue color represents primary microphone input.

Channel C shown by pink color represents the DAC output.

Case I:

If mainMic1stSample < 10

Then DAC output = subMic1stSample

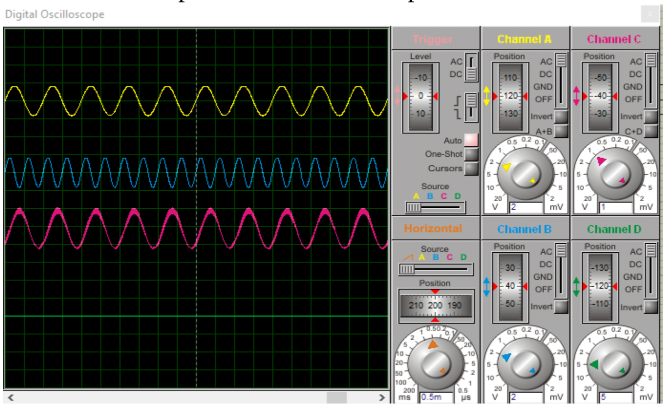

Fig. 4.2 Result of Case I

Case II:

If subMic1stSample < 10

Then DAC output = mainMic1stSample

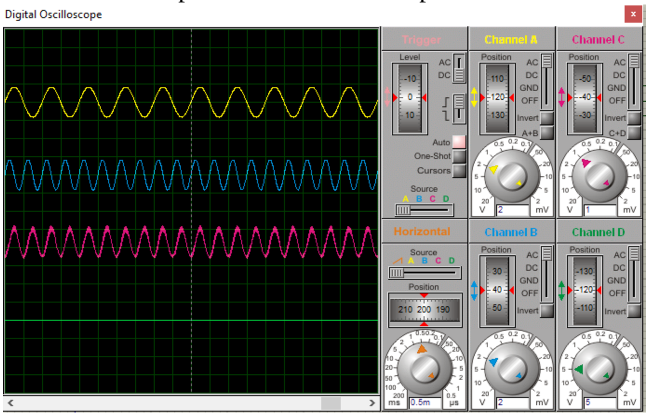

Fig. 4.3 Result of Case II

Case III:

If mainMic1stSample > 10 and subMic1stSample > 10

Then

DAC output = mainMic1stSample – subMic1stSample

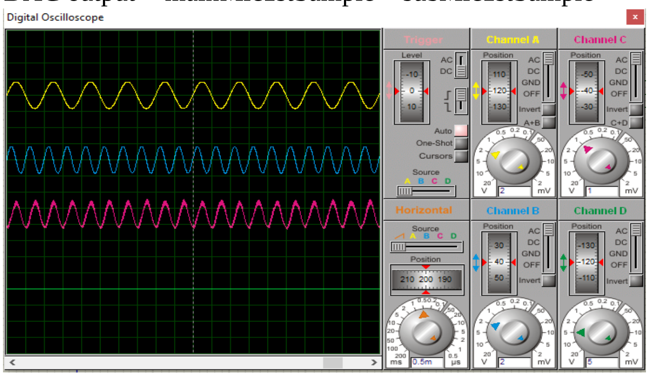

Fig.4.4 Result of Case III

Case IV:
If mainMic1stSample = subMic1stSample
Then DAC output = mainMic1stSample.

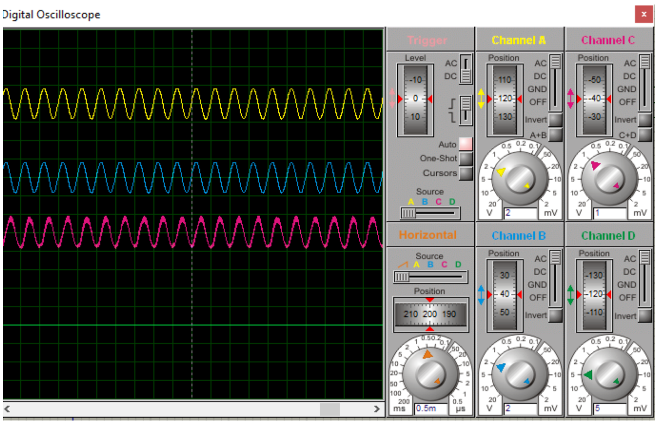

Fig.4.5 Result of Case III

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