International Journal of Recent Technology and Engineering (IJRTE)
ISSN: 2277-3878 (Online), Volume-8 Issue-6, March 2020

# Implementation of Active Noise Cancellation for Small Confined Spaces

**Prathamesh N. Narkhede, Narendra S. Pandit, Sahil A. Palaskar, Varsha Harpale, Deepti Khurge**

***Abstract:** Noise cancellation has slowly become a necessity instead of luxury. This is due to the increasing levels of noise pollution in today's society. Exposure to prolonged and excessive noise has resulted in serious health problems ranging from stress, poor concentration and fatigue from lack of sleep, to more serious issues such as cardiovascular disease, cognitive impairment, tinnitus and hearing loss. Such noise pollution levels are also harmful to fetuses as low frequencies are not attenuated in the same way as high frequencies are attenuated by the mother's womb. Recent techniques such as noise cancelling headphones, earbuds are being implemented that provide noise cancellation limited to a single person and not to a small room as a whole. Thus our paper proposes a duct system, which will be using the concept of active noise cancellation with the application of digital adoptive feedforward control, where an anti-noise signal is generated, having same amplitude as that of the noise signal but of reverse phase which when added together will cancel each other out. The adaptive filter is controlled by a microphone located in the duct to sense the noise reduction and adjust the entire system for optimum operation. The noise cancelling headphones that are being in use nowadays also operate on the same principle that is being implemented by our proposed system. But due to complex designs in the circuits such devices usually end up being too costly. So our system aims at providing a low cost solution. The system is mainly aimed at lower range of frequencies up to 1 KHz. Active noise control is being preferred as passive silencers are bulky and the attenuations achieved for low frequencies are relatively small.***

***Keywords:** Active Noise Cancellation (ANC), Anti-Noise, Duct System (DS), Least Mean Square (LMS), Filtered X-LMS (FxLMS), Noise Pollution.*

Manuscript received on March 15, 2020.
Revised Manuscript received on March 24, 2020.
Manuscript published on March 30, 2020.

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## I. INTRODUCTION

Noise pollution or simply noise has become an inseparable part of our lives. In this day and age, noise pollution has increased to an extent, matching levels with that of water and air pollution. Major contributors leading to such prominent levels of noise pollution are outdoor noise sources like machines, public address systems, automotive, community festivals. Poor urban planning which puts industrial, residential and commercial areas in close proximity also increase the noise pollution levels. Vehicle horns are the major contributors. They mostly generate noise which is in the range of 300 – 500 Hz [4]. Similarly, locomotives create noises of frequencies between 80 – 500 Hz [6]. These are generated due to continuous rolling contact, impact noise and the squeal generated by friction on tight curves [6].

Noise pollution can be avoided or cancelled by two major techniques such as Passive noise cancellation and Active noise cancellation. Passive noise cancellation deals with the higher range of frequencies normally more than 1 KHz. It is used to block out the noise level or simply provide isolation. This is achieved by using shielding, padding, muffler, sound absorbing tiles. This technique does not require use of any electrical or electronic circuits or a power source to reduce external noise [7]. Whereas active noise cancellation works better for frequencies below 1 KHz, but requires adaptive filters for cancellation thus, this technique requires a power source. It consists of three methods,

- Feedforward Cancellation
- Feedback Cancellation
- Hybrid Cancellation

All these methods require a microphone signal be digitized to allow DSP-based algorithms to be executed on the microphone's digitized signal [2]. This paper proposes a system which uses the broadband feedforward ANC system with Filtered-X Least Mean Square Algorithm, for a white noise generated in a PVC duct, using speakers and compare the input noise with the error signal to verify noise cancellation.

## II. ACTIVE NOISE CANCELLATION

Noise cancellation can be classified into two types i.e. passive and active noise cancellation. Passive cancellation technique consists of enclosures, barriers, and silencers to attenuate the noise. This is done by achieving impedance change and energy loss caused by sound propagation via sound-absorbing material.

However, such methods require large spaces, have high costs, and are ineffective at low frequencies, making the passive approach to noise reduction often impractical [7]. To overcome these problems active noise control comes into picture. Active noise control uses the principle of destructive interference of the sound waves, where in order to cancel an undesired noise a sound wave with inverse sound pressure is generated.

The system consists of an electroacoustic device that cancels the unwanted sound by generating a signal of equal amplitude and opposite phase. The original, unwanted sound and the anti-noise acoustically combine, resulting in the cancellation of both sounds. Fig.1 explains the concept of active noise cancellation. It is evident from the figure that the effectiveness of cancellation of the primary noise depends on the accuracy of the amplitude and phase of the generated anti-noise [2].

The major challenge for such a system to work is to identify the original signal and generate an inverse without delay in all directions where noise interact and superimpose.

Fig. 2 shows how the system can be implemented inside a duct to cancel the noise. This is achieved using an electronic controller which generates the anti-noise.

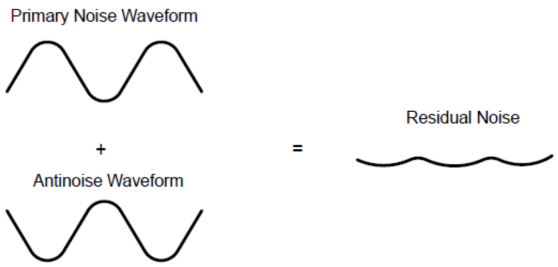


**Fig. 1. Physical Concept of Active Noise Cancellation [2].**

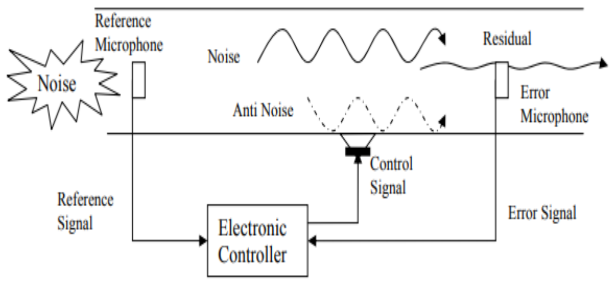


**Fig. 2.ANC System implementation in a duct [2].**

## III. ANC ALGORITHMS

### A. Least Mean Square Algorithm

Least Mean Square algorithm is a class of adaptive filters, which operates by finding the filter coefficients that relate to producing the least mean square of the error signal. LMS incorporates an iterative procedure that makes successive corrections to the weight vector which leads to the minimum mean square error. The algorithm starts by assuming small weights (zero in most cases) and, at each step, by finding the gradient of the mean square error, the weights are updated. Due to its simplicity and efficiency LMS is generally used for signal enhancement, it is the most efficient in terms of storage requirement and indeed computational complexity, the basic LMS algorithm updates the filter coefficients after every sample [2].

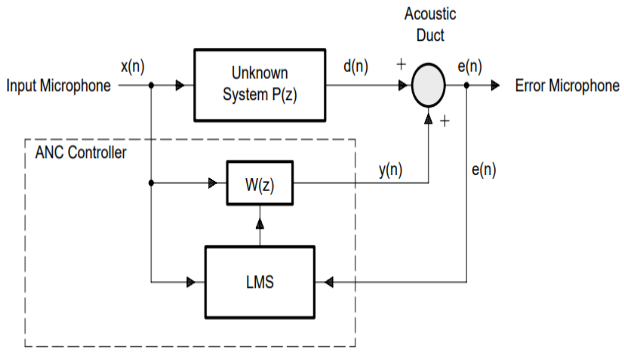


**Fig. 3.Feedforward structure for LMS Algorithm[2]**

Considering the frequency domain representations, error signal can be representated as,

$$E(z) = D(z) + Y(z) = X(z)[P(z) \; + \; W(z)] \tag{1}$$

Where, E(z) is the error signal, X(z) is the input signal and Y(z) is the adaptive filter output. After the adaptive filter W(z) has converged, E(z) = 0 thus,

$$W(z) = \; -P(z) \tag{2}$$

Which implies that,

$$y(n) = \; -d(n) \tag{3}$$

Therefore, the adaptive filter output y(n) has the same amplitude but is 180° out of phase with the primary noise d(n). When d(n) and y(n) are acoustically combined, the residual error becomes zero, resulting in cancellation of both sounds based on the principle of superposition [2].

### B. Filtered-X LMS Algorithm

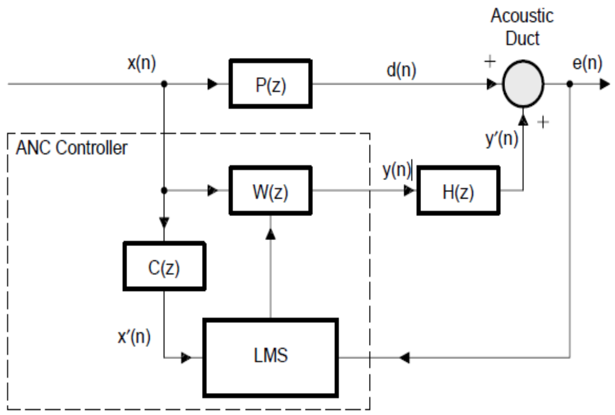


**Fig. 4. FxLMS Algorithm implementation [2].**

FxLMS is an algorithm very similar to the LMS algorithm discussed before, but the major difference between LMS and FxLMS is that the latter is used for identification of unknown system in the presence of a secondary path. The characteristics of the secondary path H(z) have significant effects on the performance of an ANC system. FxLMS Algorithm can be expressed as,

$$w(n+1) = w(n) - \mu e(n)x(n)h(n) \quad (4)$$

where μ is the step size of the algorithm that determines the stability and convergence of the algorithm and h(n) is the impulse response of H(z). Therefore, the input vector x(n) is filtered by H(z) before updating the weight vector. H(z) is unknown and must be estimated by the filter, C(z) [2].

## IV. TMS320 C6713 DSP STARTER KIT

The TMS320C6713 DSP Starter Kit (DSK) developed jointly with Spectrum Digital is a low-cost development platform designed to speed the development of high precision applications based on TI´s TMS320C6000 floating point DSP generations [1].

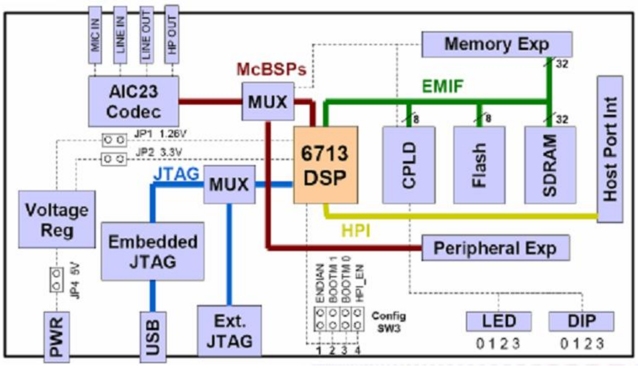


**Fig. 5. Block diagram of TMS320C6713 DSK[1]**

Key features of DSP Starter Kit:

- TMS320C6713 DSP core operating at 225 MHz.
- AIC23 Stereo Codec.
- 16 Mbytes Synchronous DRAM.
- 512 Kbytes non-volatile Flash memory.
- 4 user accessible LEDs and DIP switches.
- Software board configuration through registers implemented in CPLD.
- Configurable boot options.
- JTAG emulation through on-board JTAG emulator with USB host interface or external emulator.
- Single voltage power supply (+5V)

**Fig. 6. TMS320 DSP Starter Kit**

## V. METHODOLOGY

### A. Block Diagram

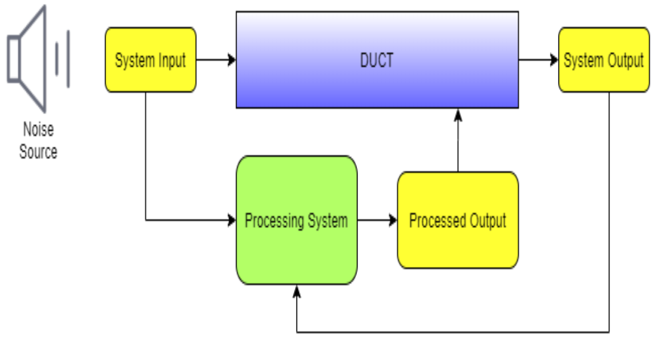


**Fig. 7. Block diagram of ANC System.**

Fig. 6 shows a generalized block diagram for the system. The block diagram consists of a noise source, system input, system output, a processing system, and a duct. Noise source can consist of any kind of noise such as traffic noise, construction noise, or noise generated through a signal generator, only constraint here is that the frequency of the noise should be below 2 KHz. As the active noise control is for a duct based system, the crucial element of the block diagram is the duct. For this system we have chosen a PVC duct basically due to its flexibility and light weight characteristics.

The system input consists of a microphone which captures the noise signals. It allows the signal to pass through the duct and also this signal is given to the processing system. The processing system performs calculation on the input signal based on the error signal captured at the system output. Similar to the system input, system output also consists of a microphone which is used to capture signals which are exiting the duct. Upon performing the calculations, the processor generates the anti-noise signal which is passed into the duct. Here the input signal is added with the processed signal and noise cancellation is achieved.

### B. Experimental Setup

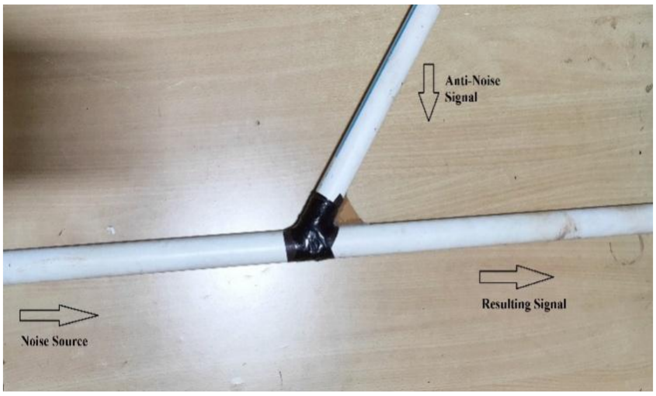


**Fig. 8. Experimental Setup**

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DOI: 10.35940/ijrte.F9870.038620
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A duct using polyvinyl chloride (PVC) is used as the experimental setup for the system. The duct consists of two speakers, i.e. one is placed at the end of the duct to be used to play/capture the noise source and the other speaker is placed on the side opening to apply the anti-noise signal. Two condenser microphones are also required in the setup, one near the source speaker to capture the input source and use it as a reference signal and second microphone is placed on the other end of the duct to capture the resulting signal produced on addition of the original source and anti-noise signal. The TMS320C6713 DSK is used here to perform operations on the noise signals.
The PVC duct system consists of a main path through which the noise source would be given as the input, this would be captured by a microphone, whereas a Y-section is joined to the main path to account for the secondary path effects. Through this Y-section the anti-noise signal would be added. In this way at the other end of the main path the resulting signal would be obtained. The main path is 41 inches, whereas the secondary path is of 12 inches. The secondary path is added to the main path at a distance of 21.5 inches from the input end and at an angle of 45 deg. The PVC duct is chosen due to its flexibility and lightweight characteristics. The anti-noise signal generated by the processor is added in the duct via the secondary path thus cancelling the noise and generating the resulting signal at the end of the duct.

### C. Flowchart

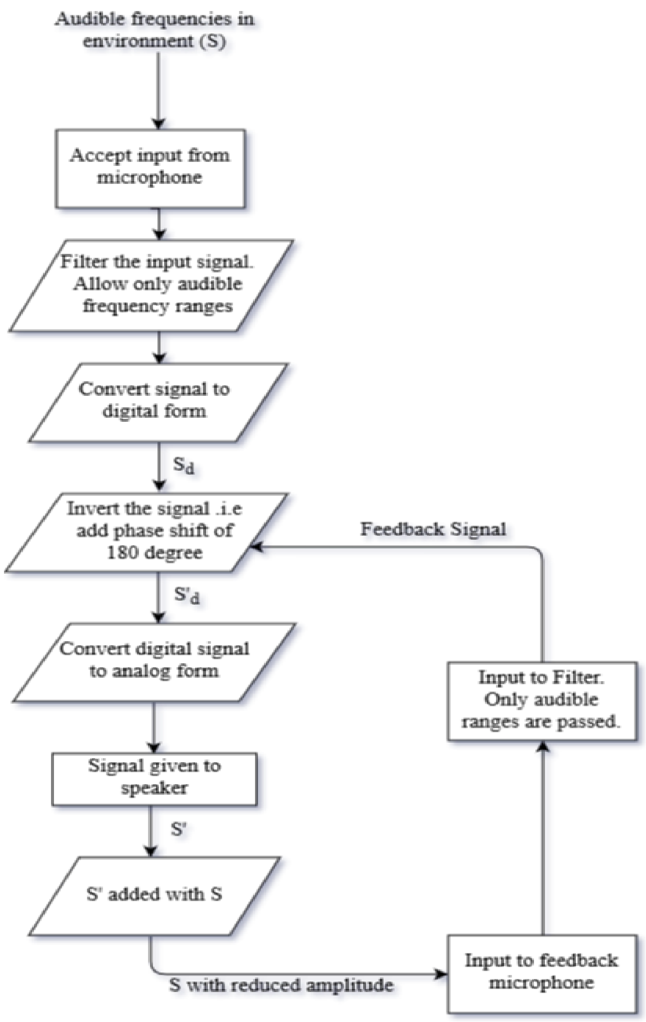


**Fig. 9. Flowchart**

## VI. RESULT

The following are the results generated using MATLAB as the simulation tool. The MATLAB version R2013a has been used for analysis. The function filter() is used here, which filters a data sequence using a digital filter which works for both real and complex values. The filter is a direct form II transposed implementation of standard difference equation. For the simulation a random white noise signal is generated with a duration of T=1000.
Fig. 10 shows the random white noise signal generated for the simulation, this signal is then given as a parameter to the filter() function. The output of the filter is shown in Fig. 11, FxLMS algorithm is applied on the output of the filter, with a step size equal to 0.1. Fig. 12 shows the comparison between the noise signal and the control signal generated using the FxLMS algorithm, through this we can observe that the control signal is almost equal to the noise signal. The noise residue keeps on reducing as the time duration increase. This is shown in Fig. 11.

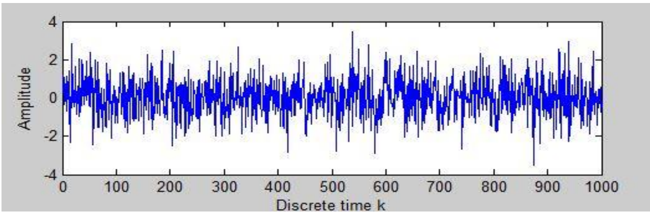


**Fig. 10. Random White Noise Signal**

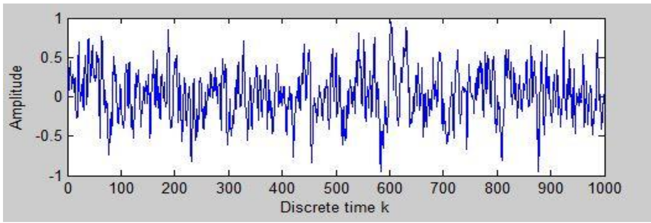


**Fig. 11. Filter Output**

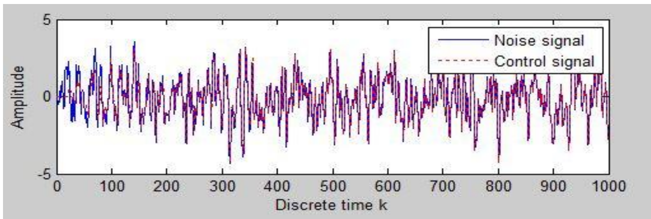


**Fig. 12. Comparison of noise and control signal**

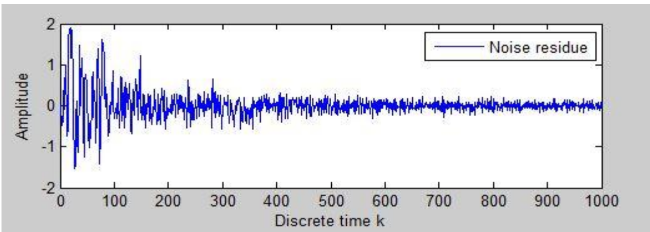


**Fig. 13. Residue Noise Signal**

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## VII. CONCLUSION

On observing the simulation results, it can be concluded that the proposed ANC system can be implemented. Simulation was performed in MATLAB 2013a for a random white noise signal. In the simulation it was observed that in the starting of the signal the filters output was not tracing the desired signal and the error signal was also high. But after some time duration the filter adapts to the signal and the output of the filter is almost tracing the desired signal and hence reducing the error signal to very low value. Using a TMS320C6713 DSP processor, better and faster signal processing can be performed, which will help in efficient noise cancelling. Proper placement of sensor devices is very crucial in such a system as it will determine how much delay will be added. Further improvements can be made by modelling the feedback path effects which is the path from summing junction to input. This helps in increasing the coherence of input signal. Thus it can be concluded that using FxLMS algorithm instead of LMS is also important if one were to consider the secondary path effects.

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