Development of Noise Cancellation Device

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Abstract - *This paper work proposes an Active Noise Cancellation (ANC) system framework for various noise generating machineries to prevent workers from different side effects which arises due to noise. The ideology behind every innovation is to improve the adaptability and comfort of human beings. Ergonomics and economics are the distinguishable factors on which the development of any technology is based upon. The objective of this paper is to develop the framework of an ANC system such that its efficiency will increase and can be compatible with different noisy industrial appliances. This study was carried out based upon literature cited and the data which was gathered from literature survey. ANC systems which were reviewed, includes different complex data processing systems. These systems provide some unwanted characteristics and consume space of the system as well as increasing the cost of the noise cancellation device. Complexity of such systems was reduced in this research work up to substantial results, indirectly reducing the cost of the device. Two frameworks were designed to simplify this ANC system. These experimental setups were differed on the basis of data processing system. We described and discussed their functionality, implementation and runtime environment along with their pros and cons. Finally, it is observed that theoretically implemented microcontroller framework confirmed its efficient applicability in ANC system.*

***Key words*: ANC, Audio power amplifier, Carbon microphone, DSP, NIHL.**

1. INTRODUCTION

Provision for suitable work environment for every individual's, whether, it be a Doctors, Engineers, Workers, Teachers, Students, etc. is necessary. Construction work in a running city like Mumbai increases noise sound pressure levels. To increase capability of an individual, it is for fact to remain healthy. Noisy working conditions have negative effects on the worker's morale and adversely affect their safety, health and performance. It is brought to the knowledge of all concerned that Noise is emerging as an important and challenging health hazards for working individuals. With increase in mechanization of construction operations and use of heavy machinery, the noise levels in city have increased over the years. Repeated or prolonged exposure to excessive noise levels leads to Noise Induced Hearing Loss. Potential sources of noise emissions include compressors, drilling machines, crushers, and other mechanical equipment used at a construction, renovations, and automobile servicing sites. Increasing the distance between the noise source and the listener is often a practical method of noise control. Where such noise control measures are not possible, personal using protection devices, such as approved ear plugs or ear muffs, should be worn by every person exposed to SPL more than 85 dBA [1]. Many studies suggest, after certain period of time, a human ear has loss of hearing due to various factors; whereas, a new born baby has a good hearing quality than the middle-aged man. Few of those factors, as we discussed earlier, are the same reasons to affect the human ear. To individuals working on those machines for prolonged exposure causes hearing impairment and deafness.

The results obtained after investigation indicated that the sound pressure levels of various machineries were higher than the acceptable limits i.e. >85dBA. Therefore, control measures should be adopted at the sites for machinery as well as hearing protection aids should be supplied to the workers in order to protect the workers from NIHL [2]. High SPL equipment's such as engines, fans, blowers, transformers, turbines, cutting machines, compressors, etc. create a sound at certain frequency which is irritating to human ears. Many passive techniques are utilized to reduce its ill effects such as earplugs, ear-protectors, sound insulation walls, mufflers, and sound-absorbing materials. And as stated earlier, the boons over curses increases with inclusion of the fact that all these passive methods require costly material which has a high sound absorbing coefficient and are ineffective at sound level pressure more than 100dBA [1].

This paper describes the design and fabrication of an active noise cancellation device to help reduce the sound initiated by machines, equipments or domestic appliances used in industries, educational institutes and houses; respectively. The first patent on Active Noise Control (ANC) was granted to Paul Lueg in 1936 [3]. Lueg appended a drawing to his patent, describing how two sinusoidal sound waves can cancel each other while using a principle of destructive interference. The technique has been implemented and devices are available which can cancel the noise entering an area of interest. For example, a device developed for home application to cancel the noise of mixers and grinders and avoid entering a room. Many developments are focused on the area wherein, equipment is placed in sound proof surrounding but these devices are either too costly or are not capable of cancelling low frequency noise. Active Noise Control

(ANC) is best described as the attenuation or reduction of a certain frequency by emitting the same frequency from a loudspeaker, but with the phase inverted. The inverted phase tone essentially cancels out the unwanted noises. The basic design is represented in Fig. 1.

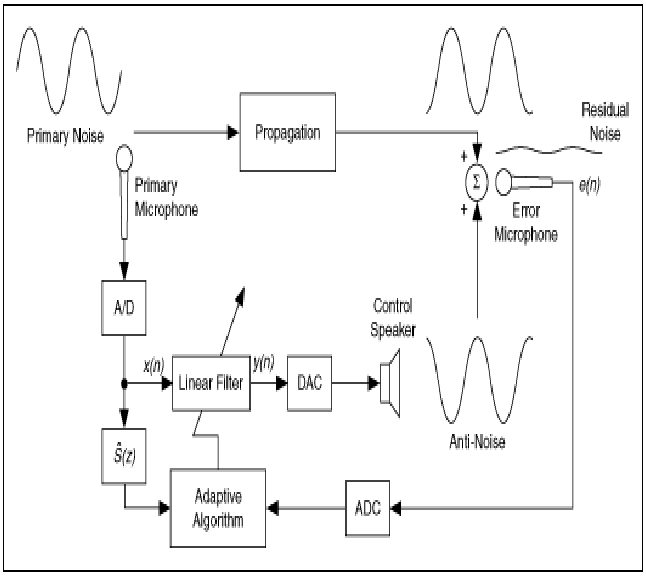

Fig. 1: Basic Design of an Active Noise Cancellation System.

An active noise control system needs two microphones and one speaker. One microphone is placed close to the noise source and the other microphone is placed at a suitable location near to the noise source where the noise needs to be minimized. The microphone close to the noise source is known as the primary microphone. The microphone in the sound field is known as the error microphone. The speaker, known as the control speaker, is used to transmit the anti-noise signal that the adaptive filter generates. Audio sound active control is a signal-processing methodology that reduces the effective sound amplitude to improve signal-to-noise ratio (SNR) so that unwanted noise is less perceptible. It uses amplifiers and microphones inside vehicles, along with digital signal processing (DSP), to cancel the noise. The sound can be described as a pressure wave consisting of an amplitude and phase.

DSP kit requires complicated algorithms to generate effective rate of SNR. This thesis mentions to two plans of approach for development of device and one of them uses active control system but after studying the conditions and effectiveness for understanding of digital signal processor, use of DSP was not preferred. And the other approach uses same techniques ideologies with the use of microcontroller, Op-amps and audio amplifier for transmission, necessary filtering and reduction of noise. The noise control embedded system will use one microphone and one speaker/ earphones/ headphones which ever functionality is necessary can be adapted by making device compatible to those audio output devices (i.e. speaker, earphones, headphones, etc.). The controlling unit for amplitude levels and adaptive algorithms is microcontroller which stores the sound data from microphone and uses initiated set for reduction values for comparing it with stored data, thus, reducing noise sound pressure level at significantly.

Noise levels before enclosed press line at center of pressing machine was around 103 dBA, in surroundings, sound pressure levels more than 90 dBA were measured and around 35 meter radius from the machine reported around 83-85 dBA with enclosed environment data reported was in the range of 0-80 dBA but enclosed machine environment also has many flaws in itself. An individual working in that close to machine would still induce NIHL [4]. In an occupational health survey conducted for industrial workers, more than 35% of workers showed the evidence of Noise Induce Hearing Loss (NIHL). Noise Induced Hearing Loss was observed among workers of age group of 19-55, but the prevalence was highest among workers from age group 36 -40 years [5]. To ensure safety of workers against high intensity SPL, Indian Noise Standards regulated limits to exposure of different levels to SPL as broached in Table-1.

Table-1: SPL and Their Corresponding Permissible Exposure Time Limit for the Workers [6]

Total Time of exposure (continuous or short term of exposure) per day, in hours	Sound Pressure Level in dB(A)
8	90
6	92
4	95
3	97
2	100
1.5	102
1	105
0.75	107
0.5	110
0.25	115

The occurrence and severity of NIHL was related to the degree of exposure to noise and years of service in the mine. More exposed a worker is to the noise; more are his chances of NIHL. Apart from this, various other consequences of lesser potential are caused, but are potentially strong to cause a fatal accident.

2. NOISE CANCELLATION TECHNIQUES

In this paper, a specially designed device is proposed, which is to be used by all the workers who are exposed to sounds of SPL greater than 90dB. This device will be of help in a special manner to allow only the human voice to reach the ears. So as to filter out all the sounds having frequencies outside the human audible frequency range and will cancel out the undesired noise within human audible range. Basic principles behind the device are Active Noise Cancellation to cancel out the undesired noise within the audible frequency range and a Band pass filter to filter out noise which are outside the human audible frequency range.

2.1 Digital Signal Processor

There are mainly two approaches for reducing the noise levels: active and passive methods. Active noise control (ANC) is also referred to as active noise cancellation. Due to the advancement of Digital Signal Processor (DSP), it was introduced with ANC systems to obtain more accurate results with minimum error.

DSP was used in active noise cancellation systems to cancel the noise which was at low frequency. In this case close loop transfer function was made and implemented digitally with DSP [7]. Different active noise cancellation algorithms like H∞ control algorithm, filtered-X radial basis function (FX-RBF) algorithm, etc. were used by researchers to achieve efficient DSP based system [7]-[9].

2.2 Amplifier

Amplifier plays an important role in anti-noise cancellation systems. Hence, it is very important to optimize the capacity of amplifier. Optimization is based on the factors like space occupied on PCB, operating voltage, time taken to respond to the signal, cost, etc. These amplifiers are able to provide functions like triangle wave generator, audio modulation comparator, switch controller and output stage, low pass filter, etc. Audio modulation and low pass filter capabilities of amplifier makes the ANC circuit simpler. The PIC24FV16KM202's wide range of analog and peripherals allows it to be used to create complete solutions for circuit including a class D amplifier. Because of its peripherals class D design minimizes PCB area and overall cost [10].

2.3 MATLAB Simulations

MATLAB is very useful simulation platform. Mathematical frameworks which were designed for active noise cancellation system can be simulate in the MATLAB by implementing real time conditions. Results obtained from MATLAB simulations are very useful to understand the behavior of the system and changes can be made accordingly. Many researchers used this method to model a active noise cancellation system. Features like Monte Carlo Simulations in MATLAB minimizes the complexity of the system. File created using MATLAB can be easily implemented to any web application. Comparison of different algorithms with different combinations can be achieved using this software [11]-[14].

2.4 Active Control System

In active noise control system basic principle of destructive interference is used. In this method of noise cancellation system, anti-noise waveforms will be generated. Arrangement was made in such a manner that noise coming from source will be cancelled by anti-noise waveform. For better results closed loop feedback dependent transfer function is preferred in this system. Predictable white noise or quiet zones is the advantage of this method. Maximum accuracy is required while adjusting the working setup of this kind of system [15].

3. DESCRIPTION OF EXPERIMENTS

In this paper two experimental setups were discussed. One is based on DSP and other is based on microcontroller.

3.1 Experimental Setup with DSP

Various different technologies have been used for development of the cancellation device. The use of 'anti-noise' wave for cancelling the undesired sound is one of the ANC techniques. Two technologies influx are to be used for software simulation of device. The first one is the band pass filter and the second one is ANC (Active Noise Cancellation). Active Noise Cancellation (ANC) is a method for reducing noise levels. ANC is achieved by introducing a cancelling 'anti-noise' wave through linear filters. These linear filters are interconnected through DSP using a specific signal processing algorithm for the particular cancellation scheme. The proposed method is to build a Noise-cancelling device by means of active noise control and band pass filter to be used by workers.

Band pass filter conditions the primary source wave signal. Band pass filter are stacked low pass filters and later transmitting a certain frequency range to high pass filter which gives an attenuated signal of primary source wave signal. And then the attenuated signal is cancelled out by using ANC technology. Essentially, this involves using a microphone, placed near the ear, and electronic circuitry which generates an "anti-noise" sound wave with the opposite polarity of the sound wave arriving at the speakers, but such condition would not be ideal for use of machine. This research demonstrates the approaches that we take on tackling the noise cancellation effects, along with results comparison. Noise Cancellation makes use of the notion of destructive interference. When two sinusoidal waves superimpose, the resulting waveform depends on the frequency amplitude and relative phase of the two waves (Fig. 2, illustrates use of destructive interference).

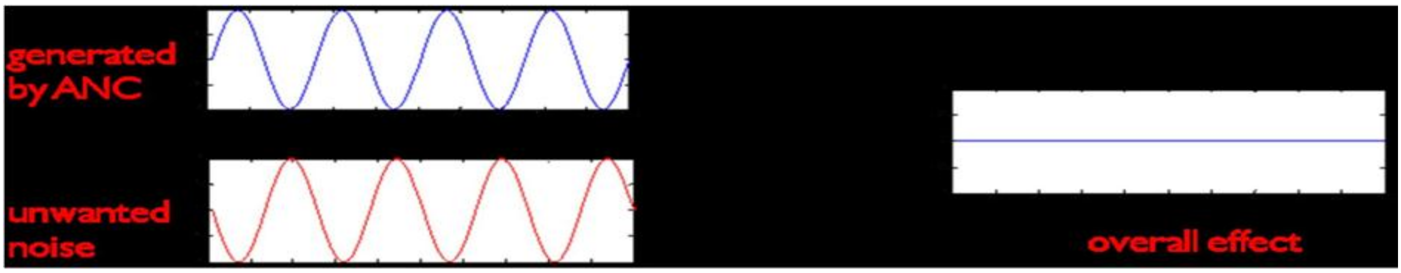

Fig. 2: Signal Cancellation of Two Waves 180° Out of Phase

If the original wave and the inverse of the original wave encounter at a junction at the same time, total cancellation occurs. The challenges were to identify the original signal and generate the inverse without delay in all directions where noise interacts and superimposes. ANC is developing rapidly because it permits improvements in noise control, often with potential benefits in size, weight, volume, and cost.

The ANC is achieved by using LMS algorithm which is also interconnected to a DSP. Use of DSP with adaptive filtering results the occurrence of original and inverse of original waveform to meet at summing point in same interval of time. Adaptive filter ensures to remove any lags or leads from an original and inverse of original waveforms, then a summing point signal is transmitted to speaker. The transmitted signal gives an output signal which has reduced noise levels. If unexplained parameters from primary source of waveform are identified and implemented into simulating model that would result in complete cancellation of noise levels.

A noise-cancelling system is incorporated within an audio device that emits a wave with equal amplitude, but a phase of 180° (inverted phase, also known as anti-phase) of the original wave. The recombination process of the two waves is based on physical principle of destructive interference. A simulation of the noise cancellation setup was used to determine circuitry requirement for generation of the inverted wave for noise cancellation. The following figure shows simulation results and circuit (from EveryCircuit Android app) in Fig. 3 and Fig. 4, respectively.

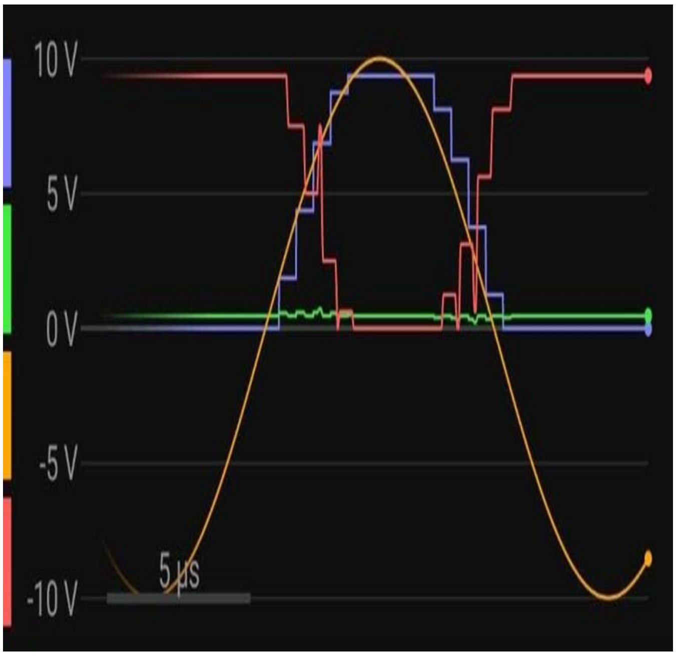

Fig. 3: Simulated Output Waveforms

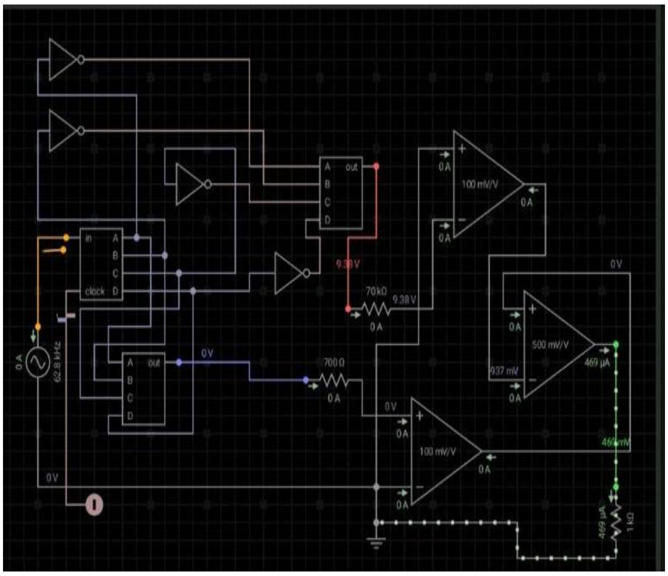

Fig. 4: Simulation Circuit

Every system has its own processor which processes the data provided by user and gives the output by interpreting the data. In active noise control system, Digital Signal Processor was used to interpret primary source signal. DSP is an electronic module which works on electrical signals. Microphone was used in the system which senses the incoming noise and converts that analog sound into digital electronic signal. Digital signal coming from microphone is given as an input to linear band pass filters which was interconnected with DSP which processes this data and generates an attenuated signal. Generation of an attenuated signal by the DSP also requires a programming code to be written and be burnt onto the processor for specific execution.

Inverse of original waveform is also generated by DSP by use of its embedded system to implement an interrupt signal which allows and adaptive algorithm to ensure equal time interval of both waveforms. Since the characteristics of the acoustic noise source and the working environment are time varying, the frequency content, amplitude, phase, and sound velocity of noise are non- stationary. ANC system must therefore be adaptive in order to cope with these variations. Adaptive filters adjust their coefficients to minimize an error signal and can be interpreted as (transversal) finite impulse response (FIR), (recursive) infinite impulse response (IIR), lattice, and transform-domain filters. The most common form of adaptive filter is the transversal filter using the least mean-square (LMS) algorithm.

After simulation the experimental setup was implemented to observe the resulting outputs and is as shown in Fig. 5. It consists of Microphone, Digital Signal Processor, Cathode Ray Oscilloscope and Speaker.

Fig. 5: Experimental Setup

Fig. 6, shows a framework of adaptive filter. It consists of an adjustable filter with input X and output Y. The goal was to minimize the difference between 'd' and 'Y', where 'd' is the desired signal. Once the difference is computed, adaptive algorithm will adjust the filter coefficients with the difference. There are many adaptive algorithms available in literature, the most popular ones being LMS (least mean-square) and RLS (Recursive least squares) algorithms. In the interest of computational time, LMS was used.

One of the main constraints in the choice of an adaptive algorithm is its computational complexity. For the application of ANC, it is desired to choose an algorithm which is computationally very fast. Taking this into consideration, LMS algorithm became an obvious choice over RLS.

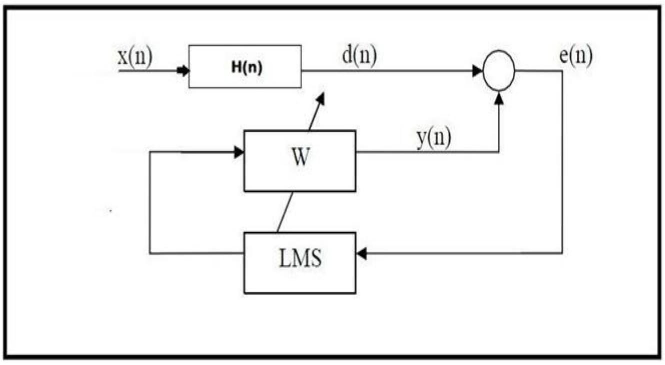

Fig. 6: Frame Work of Adaptive Filter

The output signal of the sound wave with a phase shift can be seen on the display of the digital CRO. By using connecting wires, output coming from DSP is transferred to the speaker. Speaker converts the digital inverted sound wave into analog inverted sound waves.

The proposed set-up mentioned above is not preferred due to use of DSP kit which increased complexity of device. Because, a single DSP kit incorporates adaptive filtering and many other functions. Since, other functions are not necessary for development of device. A different approach for processing and filtering of signal in the development of device was adopted. The new proposed technology was similar to the one we used, instead of using DSP kit; microcontroller and an audio amplifier were used.

3.2 Experimental Setup with Microcontroller

The second approach for a noise cancelling device also utilizes same concepts of band pass filtering and active noise cancellation. Differences between both the concepts are only on basis of incorporation of different components from one another. Use of microcontroller instead of DSP gives an advantage for small accumulation of space on a device. A primary source of a sound wave is non-linear parameter and modeling of non-linear parameter using non-linear optimization techniques may or may not determine any parametric value(s) to define a sound wave model. Since no parametric values of waveforms can be determined in a deterministic time, use of non-linear adaptive filter just for serving the purpose of converging original sound wave and inverse original sound wave is implemented. Least Mean Square (LMS) and Recursive Mean Square (RMS) are two from many types of non-linear adaptive filtering algorithms but due to computational complexity of recursive mean square algorithm is not preferred.

Basic significant components used for approaching microcontroller-based noise cancellation are shown in Fig. 7.

Microcontroller based noise cancelling approach assimilates a carbon microphone to sense the primary analog signal from any noise initiating equipments. Carbon microphones were generally used in telephones for communication purpose, these microphones produce a harmonic frequency dominated by peaks around 2 kHz to 3 kHz range. They should be placed close to sound generating equipments to ensure maximum reduction as much as possible.

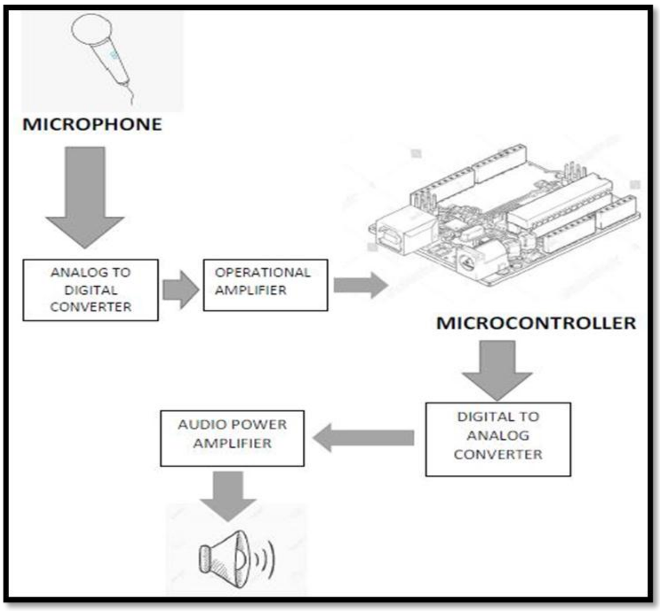

Fig. 7: Block Diagram of 2nd Experimental Setup

The primary analog signal sensed by carbon microphone will produce undulating current which is ultimately converted to a digital signal for further processing of data by an Analog to Digital Convertor. Further conditioning and filtering of signal is processed by band pass filters which are operational amplifiers. Conditioned signal would manage to remove any electrical noise produced by carbon microphone. The original filtered signal is transmitted to microcontroller which accumulates LMS adaptive filtering algorithm will process both original and inversed original signals by utilizing interrupt conditions and adaptive filters convergence time is programmed at a summing point. After convergence of signal at summing point, signal is transmitted to audio power amplifiers to make sure of amplification factor required for signal to be transmitted by speaker.

3.3 MATLAB Framework for Simulations

LMS algorithm uses the estimates of the gradient vector from the available data. LMS incorporates an iterative procedure that makes successive corrections to the weight vector in the direction of the negative of the gradient vector which eventually leads to the minimum mean square error. Compared to other algorithms, LMS algorithm is relatively simple; it does not require correlation function calculation nor does it require matrix inversions.

$$w(n+1) = w(n) + 1/2\, \mu[-\nabla(E\{e^2\,(n)\})]$$

Where, w(n) is the coefficient of adaptive filter, μ is the step-size parameter and controls the convergence characteristics of the LMS algorithm; $e^2(n)$ is the mean square error between the output and the reference signal.

Error is given by:

$$e^2\,(n) = [d*\,(n) - w^h\,x(n)]^2$$

The basic LMS algorithm fails to perform well in the ANC framework. This is due to the assumption made that the output of the filter Y (n) is the signal perceived at the error microphone, which is not the case in practice. The presence of the A/D, D/A converters and anti-aliasing filter in the path from the output of the filter to the signal received at the error microphone cause significant change in the signal Y(n). This demands the need to incorporate the effect of this secondary path function S (z) in the algorithm. One solution is to place an identical filter in the reference signal path to the weight update of the LMS algorithm, which realizes the so-called filtered-X LMS (FXLMS) algorithm. The FXLMS algorithm has been observed to be the most effective approach among all other solutions.

This can potentially lead to delayed convergence and possible non-convergence of the algorithm. Yet the use of S (z) path function in the algorithm meant the use of DSP which was the tried in the 1st set-up, the use of filtered-X LMS filtering method is also an alternative instead of DSP, but it was not implemented and simple LMS algorithm filtering method was used because embedded C coding for LMS algorithm methods reduced computational complexity of device.

In this research work, a proposed method is carried out using MATLAB simulations to counter noise using different Active Noise Cancellation techniques. Adaptive Filters have been used to implement ANC Techniques. A noisy environment may contain noise varying linearly or non-linearly. Depending upon various constraints likes linear-noise or non-linear noise, efficiency, budget, environment of the noise there can be different algorithms to update the filter coefficients. Various methods like LMS Algorithm, FXLMS algorithm, Particle Swarm Optimization (PSO) techniques can be used for the reduction of noise actively. The linearly varying noise is filtered using LMS Algorithm or FXLMS. A theoretical solution to workable ANC device for both artificial and real-world noise is proposed and due to lack of availability of hardware components (due to Covid-19 lockdown), a device was not developed.

4. CONCLUSION

The use of Digital Signal Processor (DSP) for filtering and processing of signal was an ideal case and preferable method to achieve reduced SPL, but cost of a DSP kit was not acceptable to our constraints. A DSP kit incorporates many more functions including adaptive sensing and filtering of various signal, as well as noise signal. Since, such a diverse and complex function is not necessary for noise cancellation device; a different approach with componential changes was adopted to implement noise cancellation technique. This research work implements two different componential approaches for use of an adaptive noise cancellation technique. Study, mentioned most of techniques implemented by researchers, which suggested an approach of active noise control system with use of Digital Signal Processor. Literature citied in this paper already discussed noise regulations adopted by India and also indicates NIHL impacts on individuals working in and around industrial area which is indicated based on researcher's survey data. With use of microcontroller approach complexity of device was much reduced. The 2nd componential changed approach with use of microcontroller was theoretically implemented and suggested a MATLAB framework for simulations. A theoretically implemented microcontroller framework was progressive enough to follow into noise cancellation techniques. Audio power amplifier circuit was first designed with use of different passive electrical components but without any specific parametric values.

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