Characteristics and Properties of Audio Signal and Noise Cancellation Techniques: A Theoretical Review.

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***Abstract*— An increased concern for the environment and human health has generated great interest in noise control in recent times. The techniques used for blocking of unwanted sound called noise are known as noise cancellation techniques which may be classified as active and passive noise cancellation. The purpose of this study is to investigate the technique of active sound cancellation.**

Key Words: Audio Wave, Digital signal processing, active noise cancellation (ANC), Algorithms.

I. INTRODUCTION

Audio/Sound becomes noise when it is loud or unpleasant and causes a disturbance, else called a noise signal if it acts as a random signal that accompanies the desired signal and tends to obscure it. Sound cancellation is a means of sound reduction with a motive of reducing personal discomfort caused due to noise pollution. The two forms of implementation of sound cancellation are basically named as Active Sound Cancellation and Passive sound Cancellation. Sound controlled actively involves the use of a power source as compared to the use of insulation and sound absorbing materials or isolations when controlled passively. The basic principle behind active cancellation is to generate a sound field intentionally which has the property of canceling the undesired sound. Active sound cancellation system can be termed as adaptive as it has to adapt itself to the time-varying characteristics of the sound signal to be canceled [14]. The noise control system was firstly patented by Paul Lueg with U.S patent number 2,043,416 which described the process of canceling the sound waves in the form of sinusoidal tones produced in ducts by the technique of polarity inversion and advancing the phase of arbitrary sound waves in the region around the sound source. Other similar patents were filed in the year 1950 with U.S Patent number 2,866,848, 2,920,138 and patent number 2,966,549 which were filed by Lawerence J. Fogel in which noise cancellation techniques were applied to cancel noise in airplane and helicopter cockpits. Thereafter in 1957 Willard Meeker applied active noise cancellation to earmuffs achieving an attenuation bandwidth of nearly 50-500Hz with attenuation of maximum value around 20 dB, further in 1986 researchers Dick Rutan and Jeana used active noise cancellation enabled headphones manufactured by Bose in their flight. The research on the topic is still on with the latest application being Noise-cancelling headphones which use active noise control concepts in reducing the unwanted surrounding sounds as compared to soundproofing techniques employed by passive headphones. Various research publication concerned with the said topic is reviewed in the paper and various techniques adopted are comparatively analyzed.

II. AUDIO SIGNAL

An audio signal is a signal generated by vibration of particles in a medium received by our ears and converted into some information by neural actions of our brain. If the signal perceived is meaningful and follow some symmetrical pattern then our brain considers it as sound else if it is meaningless and doesn't have any symmetry it is considered as noise which causes uneasiness. Formally sound wave is defined as the propagation of a disturbance of particles through any medium without permanent displacement of particles themselves involving changes in the particle characteristics such as pressure, displacement, and velocity [15]. The thorough description and study of the properties of a sound wave are termed as Acoustics. The various important properties which characterize a sound wave are its wavelength which is defined as the distance the wave travels in one cycle, the frequency of sound wave which defines the number of compressions or refractions of air particle vibrations in a second, the velocity of sound defined as the product of the frequency and wavelength. At sea level and 70ºF, the velocity of sound is approximately 344 m/s [15]. A sound wave can be noise for one human being and at the same time pleasant and meaningful for another human being. This dual property of sound defines the need for active sound cancellation discussed in this review paper so that its effect as noise can be reduced without it causing any concern to the person who is enjoying the same.

III. BASIC CONCEPT BEHIND ACTIVE NOISE CANCELLATION

As discussed in the previous section sound become noise when it causes uneasiness, therefore, techniques adopted to reduce this uneasiness by canceling the effect of undesired sound is often termed as active noise cancellation technique. The basic concept behind active sound cancellation is the use of artificial signals to cancel undesired noise. That can be achieved by generating a second signal which can nullify the primary noise signal across the desired area. The cancellation sound signal is generated by one or more loudspeaker known as cancellation speaker which is controlled by a digital signal processor. The parameters of the controller are controlled by the adaptive algorithms. It is called adaptive because it has to adapt to the variations of the noise signals. The basic block diagram of active noise/sound cancellation is given in Fig 1.

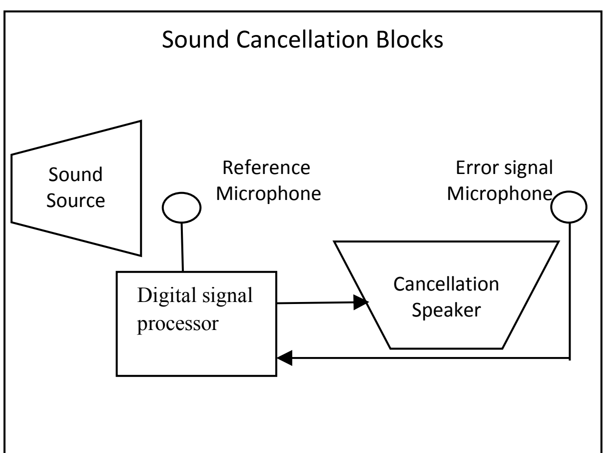

Fig. 1

As shown in the block diagram the active sound cancellation blocks comprises of a noise source which generates the unwanted or undesired sound signals, that are detected by the microphone which is connected to the digital signal processor which processes the signal using various adaptive algorithms and regenerates the sound signal for cancellation which is given as input to the sound cancellation speaker to cancel the incoming sound waves. There is a second microphone which may be referred to as an error signal generator as it generates the feedback signal for the digital signal processor. The DSP processor is the brain of the system which has the noise signal, error signal as input and feedback, processes them using adaptive algorithms and produce an output signal for the canceling speaker which converts it to sound waves in order to cancel the incoming noise. The overall action is depicted in the Fig. 2.

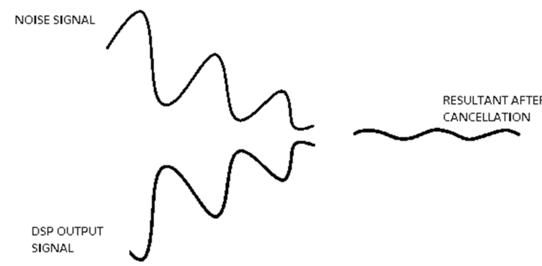

Fig. 2

The various algorithms used by the digital signal processor are described comparatively in the next section.

IV. ALGORITHMS

Adaptive algorithms act as an optimization tool for FIR and IIR digital filters since the characteristics such as frequency, amplitude, phase, and velocity of the undesired noise are time-varying. Many adaptive algorithms are available in the literature; the most frequently used being the least mean square (LMS), Recursive Least Mean Square (RLS) algorithms and Filtered X LMS (FXLMS) algorithm. In the LMS algorithm, a combiner is used to sum and weight a set of the input signal, where the weighting coefficients can be adjusted according to variations, the output signal is the product of the input vector with the weighting factor. The mean square error is the difference between the signal given by the error microphone which is the desired response as shown in Fig 1 and the actual response which is calculated as the product of the input and the weighting factors. The main purpose of this adaptive algorithm is to minimize the man square error by adjusting the weights of the adaptive linear combiner. To minimize the number of iterations performed in LMS algorithm an advanced LMS known as RLS algorithm can also be used[1].Various other modifications on the basic LMS algorithms are being implemented and analyzed by various researchers from time to time and has come under various publications some of which reviewed are the NLMS (Normalized Least Mean Square), NDNLMS (Normalized Data Non linearity

Least Mean Square algorithm),VSSLMS (Variable step-size least mean square algorithm),CSLMS (Constraint stability least mean square algorithm) and ENSSLMS (Error normalized step size last mean square algorithm) [2]. Besides this algorithm namely BLMS (Block least man square algorithm) having the advantage of simple design and more adaptability, higher complex ratio and fast convergence rate are highly suitable for use in real time processing applications [3]. The other approach being adopted is the use of the secondary path estimate in calculating the output signal to destructively interfere with the undesired noise where the reference signal being the noisy form of the undesired sound signal this approach is named as filtered X least mean square algorithm [4]. Reference [5] have implemented and analyzed the Multirate filtering algorithm in which multiple version of the observations at different sampling rates is used to estimate the desired signal. Reference [6] compares more than fifteen algorithms for reconstruction of noisy samples and categorized them into duplication and trigonometric approach which implements them through time series and polynomial models in which the dataset of sample signal of different songs divided in different genres have been used to analyze the algorithms and the findings of ARMA model found to be the best among all genres. Cosine interpolation has the lowest computational time and the AR model achieves the most effective interpolation in a limited time spam [7].Complex filtered X least mean square algorithm can work for sparse noise fields and the introduction of the constraint on the sound signal source can improve the algorithm efficiency [8].For the solution of multipoint and wave domain active noise cancellation more complex algorithms such as zero attracting multipoint and zero attracting wave domain complex filtered X least mean square algorithms can provide adequate solutions.Other additions to the existing theory of transfer domain sparse complex filtered X algorithms are the generalized form of zero attracting multi-point filtered X LMS algorithm and zero attracting wave domain filtered X LMS algorithm[8].Fuzzy rules employing the fuzzy adaptive filtered X algorithm can also be used for active noise cancellation application in ducts for cancellation of the undesired noise signal[9]. Use of fuzzy rules can reduce the complex acoustic models into simpler active noise cancellation systems by application of auto-tuning of the free parameters and change the conditional rules adaptively with the changes in the incoming information resulting in minimization of the residual noise [9]. Sigmoid function based adaptive algorithm may result in improvement of offline modeling and proved to be more efficient in removing undesired noise signal as compared to conventional filtered X adaptive algorithms [10]. From the review of the various research publications on active noise cancellation, it may be concluded that there are various algorithms which are been implemented for the said objective with each algorithm having its merits and demerits in the cancellation process. Active noise cancellation is been applied for various research and industrial applications some of the latest Publications in the concerned field are reviewed as under.

V. APPLICATIONS OF ACTIVE NOISE CANCELLATION: A REVIEW

Reference[10] depicts the application of active noise cancellation system is the development of sound signal processing algorithms being able to control sound signal in multi-dimensions especially in spherical quiet zones where the frequency spectrum of the noise to be canceled actively is analyzed and active noise control system for multi-dimensions is proposed. Reference [11] describes an approach of active noise cancellation based on adaptive control principles aiming at the reduction of noise around a noisy machine with the aim of creating a noise-free environment. The system that is described uses a single microphone as compared to the conventional two-microphone systems. The system thus designed reduces the output noise by generating the sequence of the noise to be canceled hypothetically. Reference [12] explain theoretically the decrease in convergence rate using low-level probe noise by a combination of specifically designed probe noise signals and probe noise enhancement techniques. The results show that despite using low-level probe noise signals, both the approaches improve the convergence behavior of the noise cancellation system significantly as compared to the conventional probe noise cancellation approach. Besides this one of the practical applications of the active noise cancellation is the active noise cancellation headsets for protecting human hearing ability and making communication possible in a noisy environment. It can be implemented using either feedback or feed forward technique. The feedforward ANC headset is effective in the open area with only one noise source in the vicinity and applicable to an enclosed area having very low-level reflections [13]. The above-given review is only a small insight of the vast research scope available on the topic active noise cancellation.

VI. CONCLUSION

The purpose of this paper was to present a small literature review on the topic active sound cancellation in order to gain an understanding of the basic concepts, to gain a knowhow of the algorithms being used and the digital signal processing techniques being adopted, the practical applications and the latest research going on. After the thorough review of the research publications, articles and other literature concerned with the topic of active sound cancellation it is concluded that it is a combination of multiple areas of study such as digital signal processing and sound wave analysis. Much research has been done but much more can be done as sound whenever canceled has been canceled in a small limited area like the noise cancellation headsets or limited area with real-time applications only limited to a few discrete points in space[11]. These restrictions have caused the creation of multidimensional and practically feasible quiet zones nearly impossible in practical aspects and day to day application, as it needs sound fields to be controlled in an infinite number of discrete points in space thus creating a noise-free or complete silence zone is still a fiction to be converted into reality which if conquered may lead to a noise-free environment in the benefit of mankind which presently is greatly affected by the problem of increasing noise pollution in various forms.

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