Industrial Noise Cancellation System-Model

Gourab Moitra
Department of Electrical Engineering, University Institute of Technology, University of Burdwan, West Bengal , India
Email: moitragourab@rediffmail.com

***Abstract*—The active noise cancellation (ANC), also known as "anti-noise" and "active noise control" involves the electro-acoustic generation (usually with loudspeaker) of a sound field to cancel an unwanted existing sound field. A typical single-channel active noise system consist of: a microphone reference sensor to sampled the disturbance to be cancelled, an electronic control system to process the reference signal and generate the control signal, a loudspeaker driven by the control signal to generate the cancelling disturbance, an error microphone to provide the controller with information so that it can adjust itself to minimize the resulting sound field. In this paper I will present a model of the above technology which can be used in industries in order to prevent noise pollution and provide safety to the workers working under excessive noise.**

***Index Terms*—basic principle, control system design, feedback control, working and implementation of ANC**

I. Introduction

As industries are increasing, the workers are also increasing. But the safety and protection of workers are not taken into account. Today in most industries workers work under high noise which can be dangerous to their health. Noise control is the field of acoustical engineering that deals with reducing unwanted sound in the environment.

Three basic principles are: reducing the energy of noise near a listener, creating a sound barrier between the source of noise and a listener and reducing the power of a noise at its source. The energy density of sound waves decreases as they spread out so the trivial solution is to increase the distance from source, but that is not always an option.

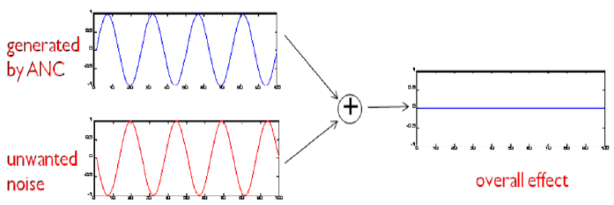

Figure 1. Cancelling "anti-noise" wave.

Conventional passive methods include sound insulation, silencers, vibration mounts, damping and absorptive treatments and mufflers. Active Noise Cancellation (ANC) is a method for reducing undesired noise. ANC is achieved by introducing a cancelling "anti-noise" wave (Fig. 1) through secondary sources. These secondary sources are interconnected through an electronic system using a specific signal processing algorithm for the particular cancellation scheme [1].

Manuscript received February 25, 2014; revised September 24, 2014.

A. Need of ANC System in Industries

In heavy industries where the production is high, the safety measures should be accurate. Noise pollution in an industry can be caused due to various reasons for example-crushing operations, manufacturing processes, machine operations, and production process etc.

Workers working under this high noise condition can develop hearing problem in course of their work. In some extreme cases they can develop permanent deafness. So in order to protect worker's health, the application of ANC system should me made mandatory in those industries which are working under heavy noise.

II. Description of Sound Waves

The Sound waves are any disturbance that is propagated in elastic medium. In fluids, that is liquids and gases, sound waves can only be longitudinal which is associated with compression and decompression of fluid in the direction in which wave travels.

In solids however we also get transverse waves, due to shear deformation of elastic medium perpendicular to the direction of travel. In this paper we only deal with sound waves in air which is a compressible fluid. A variation in pressure above and below atmospheric pressure is called sound pressure measured in Pascal [Pa]. A person with normal hearing can detect sound pressure as audible in the frequency range from 15Hz to 16kHz and can detect pressures as low as about 20 µPa at frequencies between 3000 and 6000Hz where the ear is most sensitive. The sound wave equation for pressure field $p(r,t)$ is written as

$$\frac{\partial^2 p}{\partial t^2} - c^2 \nabla^2 p = 0 \tag{1}$$

where c is the speed of sound which can be written with thermodynamic definition of compressibility X_S as

$$c = \sqrt{\left(\frac{\partial p}{\partial \rho}\right)_S} = \sqrt{\frac{\rho_0}{X_S}} \tag{2}$$

where p_0 is density of air. At 20^0C the speed of sound is 331.5m/s. A monochromatic plane sound wave can be represented by the equation for sound pressure as

doi: 10.12720/ijeee.3.5.411-415

$$p(r,t) = \mathrm{Re}\left[p_0 \exp\left(i(kr - \omega t)\right)\right] \quad (3)$$

where p_0 is the maximum amplitude, v is the frequency so that $\omega = 2\pi v$ and k is the wave vector defined as,

$$k = \left(\frac{\omega}{c}\right) n \quad (4)$$

where n is the unit vector in direction of propagation. The time from t = 0 to t = 1/v = T is known as the period.

Propagation of acoustic wave is a linear process and the principle of superposition applies. Therefore monochromatic waves are important because a sound wave may be viewed as a combination of harmonically related and unrelated single monochromatic waves with various frequencies and wave vectors.

$$p_{rms} = \sqrt{\frac{1}{T}\int_0^T p^2 dt} \quad (5)$$

For non-periodic waves such as noise T is not the period of the wave and we must calculate the limit of integral as $T \to \infty$. Because the range of pressure amplitudes and intensities can span over many orders of magnitude we use logarithmic measures called levels with units dB. The most used is sound pressure level defined as

$$SPL = 20\log\left(\frac{p_{rms}}{p_{ref}}\right) dB \quad (6)$$

where the reference pressure is $p_{ref} = 20\mu Pa$. If sound wave travels through boundary between two mediums with different propagation speeds it undergoes refraction and reflection [2].

In first medium we get a combination of reflected and incident wave, in the second medium we get only refracted wave. The relation between these waves depends on boundary condition at the surface of separation, where the pressure and normal velocities of fluid of these waves must be equal.

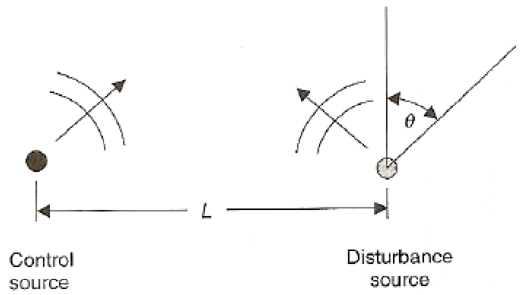

Figure 2. Disturbance source and control source

A. Basic Principle

The basic idea of active noise control is illustrated in Fig. 2 where we have two acoustic sources placed at a distance L from one another. One is called disturbance source and is the source of unwanted noise. The other, called control source is the loudspeaker with which we want to create sound that will be out of phase with the sound from disturbance source in order to cancel it as much as possible due to destructive interference between two waves.

We can observe the basic limitation of active noise control. Cancellation of the sound radiation in all directions is only possible if the distance between control and disturbance is small compared to acoustic wavelength, because the phases of the two sound fields surrounding the two sources are nearly the same at every location in space which is shown in Fig. 3 [3].

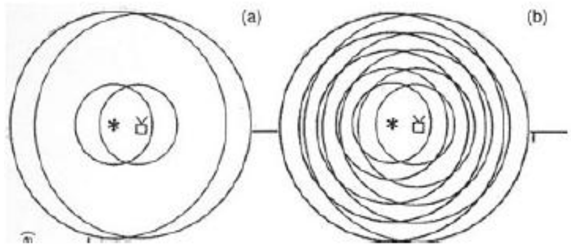

Figure 3. The wavefronts from primary and secondary source

On the left the source separation is small and on the right the source separation is large compared with acoustic wavelength. When a monitor microphone which measures primary source at some point and a secondary source loudspeaker are positioned close together they will be well coupled and only a modest drive voltage is required to achieve cancellation at these points so that sound pressure at other points further away will not be significantly affected.

We get a zone of quiet around the monitor microphone where we have reductions in the primary sound level greater than 10dB. Dimensions of the zone are approximately one tenth of a wavelength. This is useful at low frequencies. A frequency of 100Hz with $\lambda = 3.4$ m in air, translates to a zone of quiet with dimensions of 0.34m. However, the full three-dimensional shape of the zone of quiet is much more complicated and is out of scope of this paper.

III. CONTROL SYSTEM DESIGN

The basic elements of active noise control systems are sensors for example microphones, controllers like digital filters and control sources that are usually loudspeakers in acoustic applications. The active element of a control source might be used to move a speaker cone, modulate airflow or even deform its structure. A sufficient control authority, that is sufficient space and power must be available in order to have an adequate performance of such a system.

For simple example the diameter of a speaker cone must not be small compared to the wavelength of sound it produces. The propagation of an acoustic wave is a very nearly linear process unless its amplitude is corresponding to extremely loud noise. Most nonlinearity is usually due to problems with loudspeaker design.

For example, when producing a low-frequency the cone of the speaker might undergo considerable excursions which can generate higher frequency harmonics. These harmonics will not be canceled and may become audible. In signal processing theory we describe system response with transfer function H(s), that are defined as the ratio of Laplace transformation of input X(s) and output Y(s) signal [4].

$$H(s) = \frac{X(s)}{Y(s)} = \frac{L(x(t))}{L(y(t))} \qquad (7)$$

Active noise controllers are basically filters. Its inputs are signals from sensor microphones and its outputs are the drive signals to the control surfaces. When required magnitude and phase responses of the filters are relatively simple function of frequency we can use analog controllers.

However in most cases, especially when characteristic of these filters are required to change over time and are made adaptive, using digital controllers has many advantages over analog controllers in terms of flexibility, accuracy and cost.

A. Feedback Control

A very basic configuration of a feedback control approach is illustrated in Fig. 4 where we have just a sensor microphone connected to an amplifier that drives the control speaker. In Fig. 5 we have the block diagram of signal path in such setup.

There we can see that we basically have a simple feedback loop where we drive the signal measured with microphone denoted as residual e through control filer W and plant P back to the disturbance signal of the original noise d. Both signals are summed together at the point where we have our microphone. With the transfer function of the filter W we describe what happens with the signal as it travels through electrical path of the system. With transfer function of the plant P we describe the physical path of the signal that is response of the loudspeaker that generates the control sound signal and any effects on sound wave as it travels from the speaker to the microphone. We can write the transfer function of such system as ratio between disturbance and measured error.

$$\frac{E(s)}{D(s)} = \frac{1}{1+P(s)W(s)} \qquad (8)$$

For periodic signals we can use operator $s = i\omega$.

In the ideal case where the frequency response of the plant P and electronic filter W would be relatively flat and free from phase shift we could use a simple inverting amplifier $W(i\omega) = A$ with $A \to \infty$ and therefore causing the overall transfer function to become very small [5], [6].

In the real case this cannot be done, because the electro-acoustical response of a moving coil speaker induces considerable phase shift near its mechanical resonance frequency. Some delay is also inevitably introduced due to acoustic propagation time between loudspeaker and microphone. When the phase shift approaches π the negative feedback becomes positive and the system can become unstable, which means that the amplitude of the output signal grows beyond all means, thus amplifying the noise.

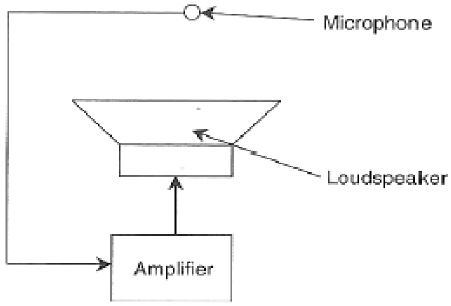

Figure 4. Basic active noise control system using feedback control.

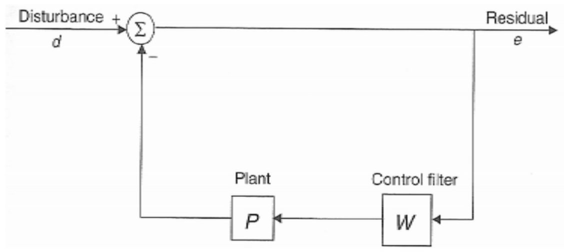

Figure 5. Block diagram of the signal path.

We can look at filter system as a regulator part $H(i\omega)$ with added compensating filters $G(i\omega) \approx P(i\omega)^{-1}$ that are designed to approximate the plant inverse and compensate for its amplitude and phase shifts. We can write $W(i\omega) = G(i\omega)H(i\omega)$. Entering this into equation above.

We can see that the transfer functions simplifies to $1/H(i\omega) \ll 1$ when $|H(i\omega)| \gg 1$. When $|H(i\omega)| \ll 1$ equation above is equal to 1 which means no gain or reduction of input signal. This means that we can design a regulation filter with gain that is large for frequency range where control is desired [7].

For frequencies that are not controlled, regulator gain remains small and we do not amplify any noise outside regulation band. Stability is described with Nyquist criterion, which in this particular case states that the phase of $H(i\omega)$ should not exceed π until its magnitude is less than 1.

The most successful application of active noise control using feedback system is active noise cancellation headsets mainly used by aircraft pilots. In very loud environments the performance of such headset is also limited because the small speaker can only generate limited sound level within the shell. A feedback controller must be designed to provide negative feedback (to reduce the amplitude of the undesired noise) rather than positive feedback, over the frequency range of interest.

The bandwidth over which control will be effective (negative feedback) is fundamentally limited by the delay in the path between the controller output to the loudspeaker and the controller input from the error sensor. The bandwidth of effective control is directly proportional to the reciprocal of the delay. The phase shift associated with this delay always changes the system from a negative feedback system to a positive feedback system at high frequencies. This is the main disadvantage of feedback controllers. When the phase shift (or delay) through the control system (including the path from the control source to the error sensor) exceeds 180°, and the overall gain exceeds unity, the system will become unstable, producing positive feedback instead of negative feedback, resulting in ever increasing noise levels that are only limited by the output capacity of the loudspeaker and its amplifier. This can cause serious acoustic noise problems in the presence of high frequency noise or if the physical system being controlled changes too much from the design condition (for non-adaptive feedback control) or too rapidly between states (for adaptive feedback control.

The instability problems of feedback controllers are usually minimized by keeping the controller gain within reasonable bounds, (which has the effect of limiting the controller noise reduction performance) and using low pass filters to attenuate incoming high frequency signals that cannot be controlled.

Unfortunately if the amplitude “roll-off” characteristic of the high pass filter is too steep, the phase at lower frequencies is affected adversely and this further limits the useful bandwidth of the controller. To minimize acoustic delays and thus maximize feedback control system performance and stability, the physical locations of the control source and error sensor should be as close together as possible.

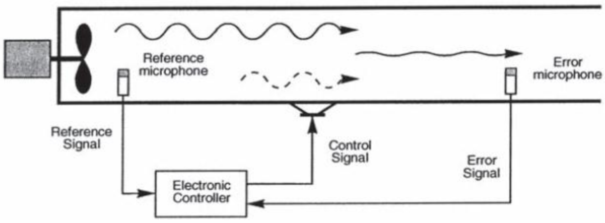

Figure 6. Model for use of ANC system in industries

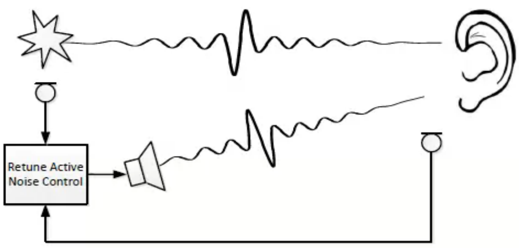

Figure 7. Feedback of error signal by error microphone

IV. Working and Implementation of ANC System Model in Industrial Sector

In Fig. 6, there are two microphones on the loudspeaker, one outside the loudspeaker and the other inside the loudspeaker. The outside microphone called the reference microphone is used to measure the noise near the noise generating source which is inputted to the ANC feedback control unit. We use feedback control as feedback systems aim to attenuate the residual effects of the disturbance after it has passed [8].

The feedback control unit adaptively tries to produce an inverse noise by processing the input. The inside microphone called the error microphone will perceive the error between the noise from the noise source and the inverse signal generated by the control unit as shown in Fig. 7. This error signal will be feedback to the control unit, which updates the filter coefficients based on this feedback [9].

V. Example of Practicle ANC System

Fig. 8 illustrates a MATLAB simulation of noise cancellation. The first row is the original noise, the second row is the noise observed at the headset, third is the inverse noise generated by an adaptive algorithm using the control unit, and the last row is the error signal between the noise observed at the headset and the inverse noise generated by the control unit. The last plot shows that the error signal is initially very high, but as the algorithm converges, this error tends to zero thereby effectively cancelling any noise generated by the noise source

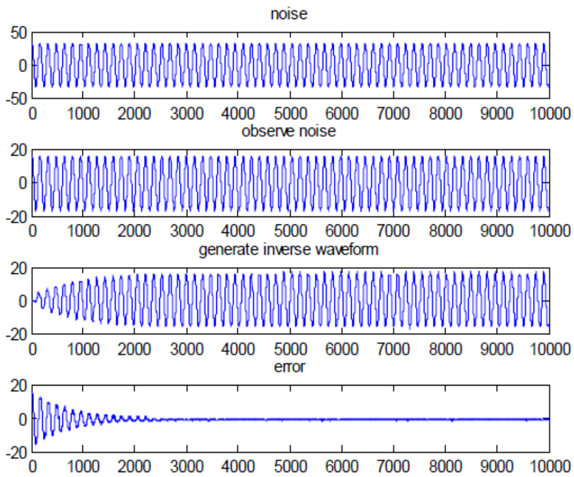

Figure 8. MATLAB result

VI. Conclusions

In conclusion, we have proposed a workable ANC system model for Industrial Sector. It works best for sound fields that are spatially simple for example low frequency sound waves traveling through a duct which is a one dimensional problem.

Active control systems are more locally oriented as often reducing noise in some local region causes increase in the noise elsewhere. It does not reduce noise globally unless sound fields are very simple and the primary mechanism is impedance coupling.

Another important factor in active noise control is whether or not the disturbance can be measured before it reaches the area where noise reduction is desired. This so called feed forward control is sometimes not possible and the control signal can only be calculated from error sensor measurement. This feedback control can be very unstable under some circumstances and is usually even less effective at higher frequencies as feed forward control.

Sometimes it is possible to reduce the noise of a system by reducing its vibration which may cause unwanted noises. For example when dealing with unwanted noise in rooms and buildings, the source of outside noise for an observer in a room are actually the walls of the room. If we reduce vibrations of these walls that are caused by outside noise sources we can reduce the transfer of noise from outside into a room. So by adapting this technology in industries we can provide safety to those workers working under heavy noise condition.

REFERENCES

[1] A. V. Oppenheim and A. S. Willsky, *Signals and Systems*, 2nd ed., Prentice Hall, 1997.
[2] D. T. Blackstock, *Fundamentals of Physical Acoustics*, New Jersey: John Wiley & Sons, 2000.
[3] S. J. Elliott and P. A. Nelson, "Active noise control," *IEEE Signal Processing Mag*., vol. 10, no. 4, pp. 12-35, 1993.
[4] J. G. Proakis and D. G. Manolakis, *Digital Signal Processing*, 3rd ed., New Jersey: Prentice-Hall, 1996.
[5] B. C. Kuo, *Automatic Control System*, Englewood Cliffs: Prentice-Hall, 1975.
[6] K. Ogata, *Modern Control Engineering*, 3rd ed., New Jersey: Prentice Hall, 2000.
[7] S. Haykin, *Adaptive Filter Theory*, 2nd ed., Englewood Cliffs: Prentice-Hall, 1991.
[8] M. Kuo and D. R. Morgan, "Active noise control: A tutorial review," *Proceedings of the IEEE*, vol. 87, no. 6, 1999.
[9] I. L. Ver and L. L. Beranek, *Noise and Vibration Control Engineering*, New Jersey: John Wiley & Sons, 2006.

Gourab Moitra is currently studying in the fourth year of the four year Degree course in Bachelor of Engineering in Electrical Engineering in University Institute of Technology under University of Burdwan. His research area includes power electronics, control system, digital electronics and digital signal processing.