# Active Noise Control Of Wireless Networks

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## ABSTRACT

If sound is annoying or loud enough to damage hearing, it is called noise. Noise has many effects on people. In addition to the most obvious, hearing loss, it disturbs speech communication, interferes with sleep, and lowers performance in mental and physical tasks. Diminishing noise pollution and its effects on human beings is slowly but steadily attracting more interest. Conventionally, noise reduction has been implemented with vibration isolators, acoustical absorbing materials, enclosures, barriers, and vibration damping materials. Generally, these approaches to the noise control are a relatively inexpensive way to reduce noise. They are called passive noise control methods because they do not need power to operate. Fortunately, active noise control can complement passive noise control. The idea behind active noise cancellation is to generate opposite phase sound that destructively interferes with the existing sound field. Active noise control has a broader meaning than just cancellation of noise because the ANC can be used for boosting the desired noise. Active methods are more cost-effective and capable of adapting noise levels at low frequencies, but at higher frequencies of, say, over 500 Hz, complex sound fields and the larger computational load needed in a control system due to higher sampling rates limits system performance greatly. Comprehensive noise control solutions often combine both of these approaches. Active noise control is a combination of acoustics, signal processing, and mechanics [2].

## Keywords

ANC, FX- LMS, RLS, adaptive filter, FIR filter.

## 1 INTRODUCTION

Active Noise Cancellation (ANC) is a method for reducing undesired noise. ANC is achieved by introducing a canceling “antinoise” wave through secondary sources. These secondary sources are interconnected through an electronic system using a specific signal processing algorithm for the particular cancellation scheme. Essentially, this involves using a microphone, placed near the ear, and electronic circuitry which generates an antinoise sound wave with the opposite polarity of the sound wave arriving at the microphone. This results in destructive interference, which cancels out the noise within the enclosed volume of the headphone. 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. If the original wave and the inverse of the original wave encounter at a junction at the same time, total cancellation occur. The challenges are to identify the original signal and generate the inverse without delay in all directions where noises interact and superimpose.

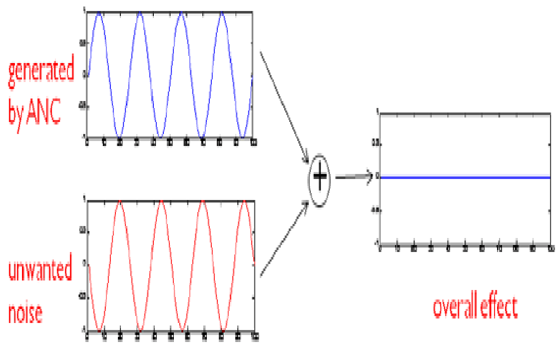



**Figure 1: Signal cancellation of two waves 180° out of phase[2].**



## 2 ERROR REDUCTION AS SPECIAL CASE OF ACTIVE NOISE CONTROL

Acoustic problems in the environment have gained attention due to the tremendous growth of technology that has led to noisy engines, heavy machinery, pumps, high speed wind buffeting and a myriad other noise sources. Exposure to high decibels of sound proves damaging to humans from both a physical and a psychological aspect. The problem of controlling the noise level in the environment has been the focus of a tremendous amount of research over the years. According to ANC theory, there are four major components of an ANC system:1)Reference microphone: the microphone that receives the noise to be cancelled (unwanted noise) and forwards it to the controller.2)Error microphone: the microphone that senses the noise at the point at which noise reduction is required and monitors how well the ANC system performs.3) Speaker: the device that physically does the work of producing anti-noise.4)Controller: a signal processor (usually digital) that tells the speaker what to do.

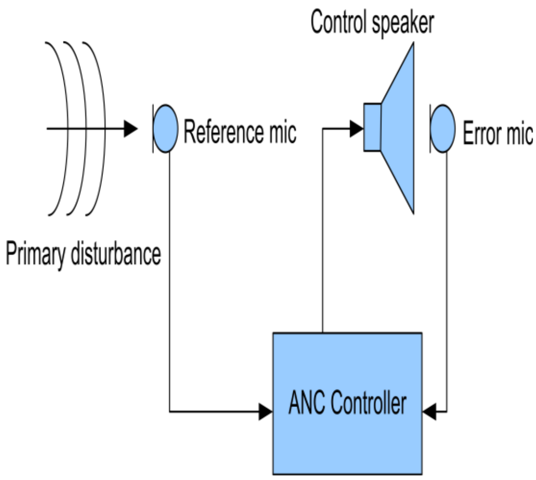



**Figure 2: Block diagram of ANC Controller [2].**

In the simplest case, the DSP controller multiplies the reference microphone signal by minus one and sends it to the speaker to produce the anti-noise. By detecting the unwanted noise with the reference microphone, the ANC system can automatically generate the correct signal to send to the speaker, which will produce the anti-noise to cancel out the unwanted noise. The size of the quiet zone created near the error microphone depends on the wavelength of the noise. The effectiveness of the noise cancellation depends on the size and shape of the room in which the system is applied, the size of the source of the unwanted noise, the position of the speaker, the accuracy of the controller and many other parameters.

## 2.1 ANC Categories

Traditionally, ANC is classified as either global ANC or local ANC. In global ANC the goal is to achieve satisfactory noise reduction in a large area where the listeners are able to move around freely, while local ANC targets only noise reduction at specific points in space, usually inside a head phone or at the ears of a listener with restricted head movement. Although global ANC seems more beneficial generally than local ANC, it is regarded as much more complicated to achieve due to wavelength and acoustic field considerations. Referring to the primary and secondary noise sources as to point sources, the maximum distance between the two sources is highly dependent on the maximum frequency of the noise to be reduced . As a rule-of-thumb, while the two sources are arbitrarily positioned in space, noise the error microphone with dimensions of approximately one tenth of a wavelength. From the opposite standpoint, to achieve homogenous noise reduction of more then 10 dB over all the space, the distance between the two point sources should not exceed one-tenth of the wavelength associated with the maximum frequency. Therefore, achieving a 10 dB noise reduction in a frequency band of up to 1 kHz, requires a distance that is less than 3.4 cm, which is certainly small in light of the typical speaker dimensions. However, in a real ANC application, the situation is little bit complicated since the noise sources are not point sources and other physical considerations must be made. The above terminology does not reflect the diverse characteristics of the noise source. Similarly, the two types of quiet zones are described with the terms global and local and the noise source can be regarded as a point source or a spatial source. This distinction warrants adoption of the following four terms to describe both the noise source and quiet zone characteristics:

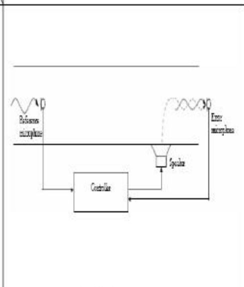
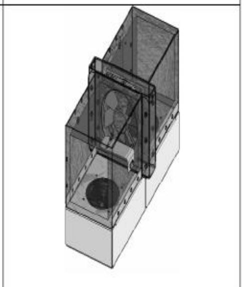
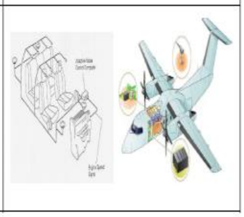
Point-to-Point ANC: In which the noise source is positioned in a small limited area in space and the quiet zone obtained is local as well. These systems are easy to install and are very effective for educational and demonstration purposes.

Zone-to-Point ANC: Where the noise source is distributed over a region and the required quiet zone is local. This category includes active headphones and active passenger seats in an airplane.

Point-to-Zone ANC: In this category, the noise source is localized and the required quiet zone is spatial. ANC systems in this category are usually embedded in the noise-generating application or at least are very close to it.

Zone-to-Zone ANC: This is referred as the holy grail of ANC. A robust system that is positioned in a room and reduces the disturbing noise no matter what the sources, without of course disturbing conversation in the room. Studies under these conditions demonstrate noise reduction in tonal or narrowband frequencies and require a significant number of speakers and sensors**.**

**Table 1: Four categories of ANC[5].**

| Quiet zone / Noise source | Point | Zone |
|---|---|---|
| Point |  |  |
| Zone |  |  |

## 2.2 Benefits of active noise control

The primary objective of most active noise control systems is the reduction of noise for personal comfort, reduction of stress and increased concentration. Another major benefit is low-frequency quieting for applications where passive insulation would be too expensive, inconvenient or heavy. For example, the lead-impregnated sheets used to reduce aircraft cabin propeller noise impose a severe weight penalty, but active control can perform just as well at a much smaller weight. Active noise control also reduces the vibrations induced into mechanical structures, thus prolonging lifetime and increasing efficiency. Compared to passive noise control, active noise control is more effective at low frequencies. Conventional methods of suppressing acoustic noise using passive sound absorbers generally do not work well at low frequencies. This is because at these low frequencies the acoustic wavelengths become large compared to the thickness of a typical acoustic absorber. It is also difficult to stop low frequency sound being transmitted from one space to another unless the intervening barrier is very heavy. Active noise control also less bulky compared to passive method as it is used electronic approach to attenuate the sound. These problems are sometimes difficult to solve using passive methods since the solutions are expensive in terms of weight and bulk. However, industrial acoustic noise often has its main power on lower frequencies, where the wavelength of sound is so long that passive techniques are no longer cost-effective because they require material that is too bulky and heavy, such as the silencer of a car. In aircraft cabins noise reduction, numerous speakers (or headsets) and microphones are used. In the noise reduction system built into new Boeing aircraft, it appears that the noise reducing apparatus has been integrated into the entertainment system. This has the benefit of reducing the cost of the noise reducing system, since it is being added to an existing acoustic system. The existing entertainment system uses headphones, signal processors, and cabling that can be leveraged by the active noise control (ANC) design [1].

Various fields in which active noise control techniques are used are: Control of aircraft interior noise by use of lightweight vibration sources on the fuselage and acoustic sources inside the fuselage, reduction of helicopter cabin noise by active vibration isolation of the rotor and gearbox from the cabin, reduction of noise radiated by ships and submarines by active vibration isolation of interior mounted machinery (using active elements in parallel with passive elements) and active reduction of vibratory power transmission along the hull, using vibration actuators on the hull, reduction of internal combustion engine exhaust noise by use of acoustic control sources at the exhaust outlet or by use of high intensity acoustic sources mounted on the exhaust pipe and radiating into the pipe at some distance from the exhaust outlet.

## 3 ADAPTIVE ACTIVE NOISE CONTROL TECHNIQUES

One of the major features of active noise cancellation schemes is the identification of external feedback paths and cancellation of this feedback signal. Continuous adaption to changing environment requires the use of adaptive systems for the identification purposes. Various techniques used for adaptive system identification have been implemented in active noise cancellation problems.ANC systems are based either on feed-forward control where a coherent reference noise input is sensed or feedback control where the controller does not have the benefit of a reference signal. Further, ANC systems are classified based on the type of noise they attempt to cancel as either broadband or narrowband. A brief overview of the various approaches to ANC follows next.

### 3.1 Adaptive filter concept

Adaptive filter is a filter that self-adjusts its transfer function according to an optimizing algorithm. Because of the complexity of the optimizing algorithms, most adaptive filters are digital filters that perform digital signal processing and adapt their performance based on the input signal. By way of contrast, a non-adaptive filter has static filter coefficients (which collectively form the transfer function).Standard fixed FIR and IIR filters are incapable of removing noise from a signal if the noise is subject to changes in frequency, phase, amplitude, or some combination of all three (Noor Fadzlynda Abdullah Ali, 2007)[4]. For some applications, adaptive coefficients are required since some parameters of the desired processing operation (for instance, the properties of some noise signal) are not known in advance. In these situations it is common to employ an adaptive filter, which uses feedback to refine the values of the filter coefficients and hence its frequency response. Generally speaking, the adapting process involves the use of a cost function, which is a criterion for optimum performance of the filter (for example, minimizing the noise component of the input), to feed an algorithm, which determines how to modify the filter coefficients to minimize the cost on the next iteration. As the power of digital signal processors has increased, adaptive filters have become much more common and are now routinely used in devices such as mobile phones and other communication devices, camcorders and digital cameras, and medical monitoring equipment.

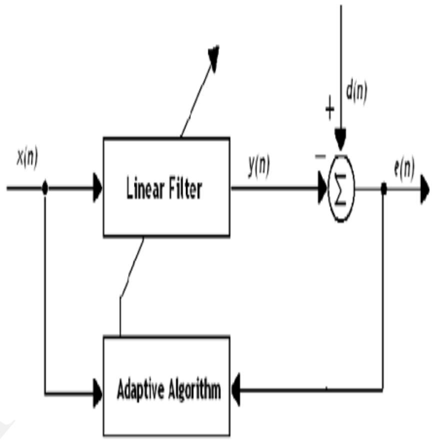



**Figure 3: Adaptive Filter Concept [4].**

Where :x(n) is the input signal to a linear filter at time n.Y(n) is the corresponding output signal.D(n) is an additional input signal to the adaptive filter.E(n) is the error signal that denotes the difference between d(n) and y(n).

#### 3.1.1Broadband feedback active noise control

These are systems that have a single secondary source, a single reference sensor and a single error sensor. The reference signal is processed by the ANC system to generate the control signal to drive a loudspeaker. The error microphone is used to monitor the performance of the ANC system. The objective of the controller is to minimize the measured acoustic noise

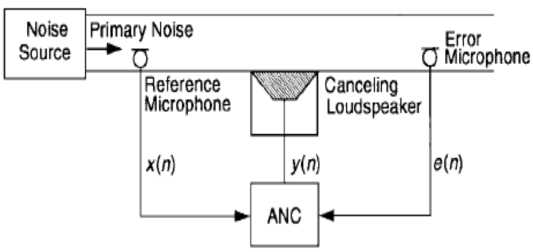



**Figure 4: Broadband feed-forward Active Noise Control[4].**

The basic broad-band ANC system shown in Figure 4 is described in an adaptive system identification framework illustrated in Fig. 3.3, in which an adaptive filter W(z) is used to estimate an unknown plant P(z) The primary path P(z) consists of the acoustic response from the reference sensor to the error sensor where the noise attenuation is to be realized. If the plant is dynamic, the adaptive algorithm then has the task of continuously tracking time variations of the plant dynamics. The most important difference between Figure 5 and the traditional system identification scheme is the use of an acoustic summing junction instead of the subtraction of electrical signals.

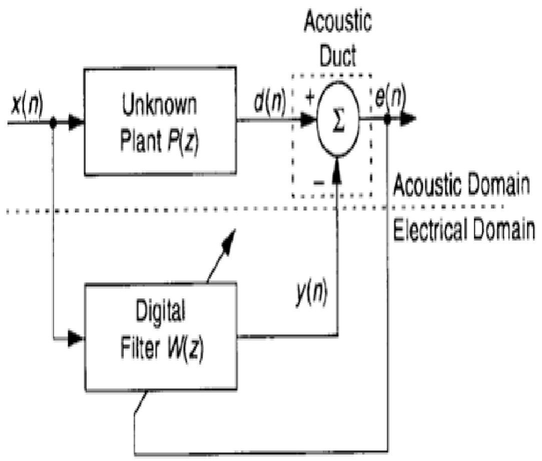



**Figure 5: System identification viewpoint of ANC[4].**

### 3.1.2. Filtered-X LMS Algorithm

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 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. Also this algorithm appears to be very tolerant to errors made in the estimation of S(z) thereby allowing offline estimation of S(z) as the most apt choice. Besides, the use of FIR filters to design W(z) makes this system very stable.But the downside is the use of high order filters that will make the algorithm run slow, and also the convergence rate of this algorithm depends on the accuracy of the estimation of S(z). The major disadvantage of this algorithm is the presence of acoustic feedback. The coupling of the acoustic wave from the canceling loudspeaker to the reference microphone will cause this acoustic feedback problem, resulting in a corrupted reference signal x(n). This can potentially lead to delayed convergence and possible non-convergence of the algorithm.

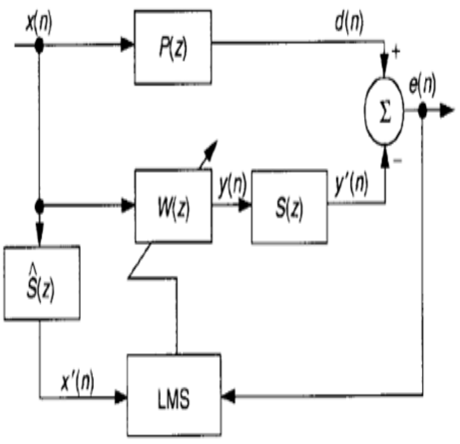



**Figure 6: ANC System using FXLMS Algorithm[2].**

### 3.1.3 Narrowband Feed-forward ANC

Noise sources are periodic in nature such as engines, compressors, motors, fans, etc. In such cases, direct observation of the mechanical motion using an appropriate sensor is used to provide an electrical reference signal which consists of the primary frequency and all the harmonics of the generated noise. The basic block diagram is as shown in figure below.

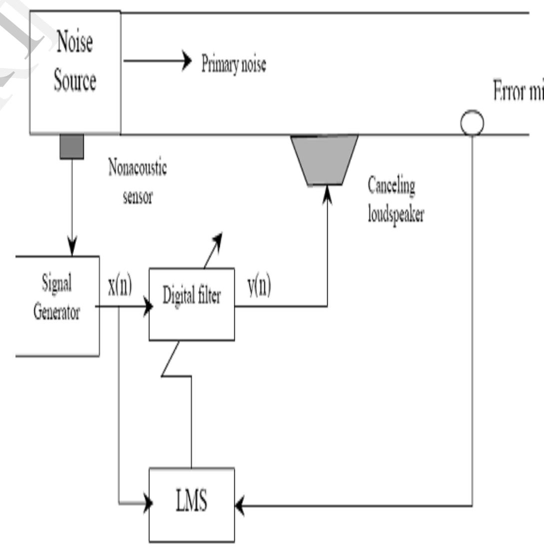



**Figure 6: Narrowband feed-forward ANC system [2].**

This technique avoids the undesired acoustic feedback to the reference sensor, as well as nonlinearities and aging problems with acoustic microphones. The periodicity of the noise removes the causality constraint, as each harmonic can be controlled independently and a much shorter FIR filter can be used to model the secondary path. There are two techniques for narrowband ANC i.e. the waveform synthesis method, which uses an impulse train with a period equal to the inverse of the fundamental frequency of the disturbance. The second technique uses an adaptive notch filter with a sinusoidal reference signal.

### 3.1.4 RLS Algorithm for ANC .

RLS algorithm can be used with an adaptive transversal filter to provide faster convergence and smaller steady-state error than the LMS algorithm. The fast transversal filter is an efficient version of the RLS algorithm, which reduces the required operations to Approximately 7L . Haykin has given a detailed treatment of RLS algorithms and the fast transversal filter

## 4. MODELING AND SIMULATION OF ACTIVE NOISE CONTROL .

Model used to simulate the active noise control system is A single channel feed-forward active noise control system based on the RLS and FxLMS algorithms.

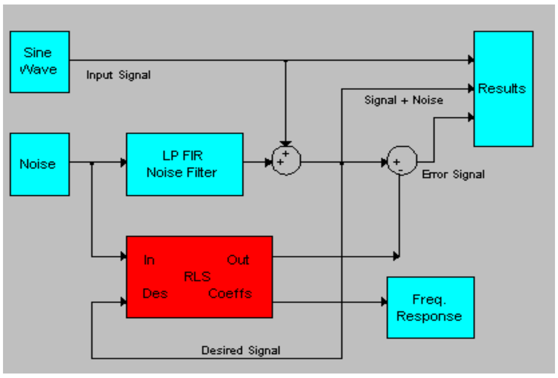



**Figure 7: Model of ANC using RLS algorithm.**

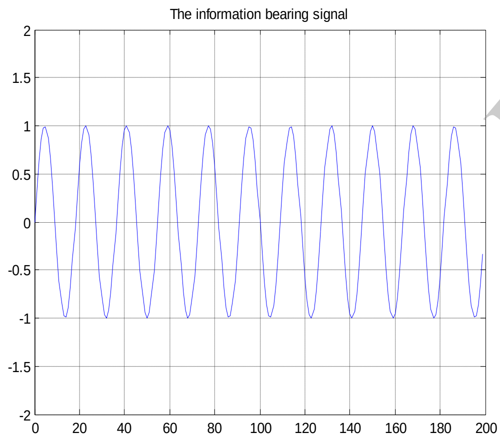



**Figure 8: Simulation of Information bearing signal.**

The information bearing signal is a sine wave of 0.055 cycles/sample. This is the sampling frequency. The noise picked up by the secondary microphone is the input for the RLS adaptive filter. The noise that corrupts the sine wave is a low pass filtered version of (correlated to) this noise. The sum of the filtered noise and the information bearing signal is the desired signal for the adaptive filter. In this code, randn(1000,1) gives a 1000 by 1 matrix.

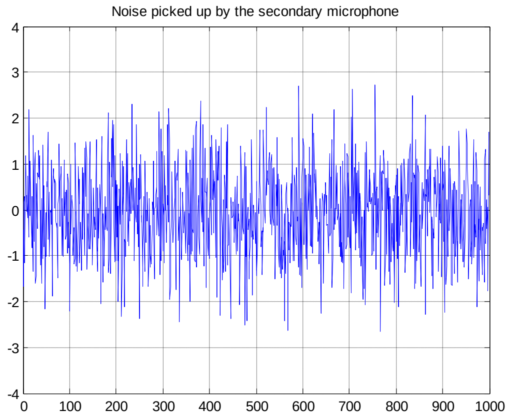



**Figure 9: Simulation of noise signal picked up by secondary microphone.**

Now, add correlated white noise to signal. To ensure that the noise is correlated, pass the noise through a lowpass FIR filter, and then add the filtered noise to the signal.The noise corrupting the information bearing signal is a filtered version of noise. Desired signal for the adaptive filter (sine wave + filtered noise): here fnoise is the correlated noise and d is now the desired input to the sign-data algorithm.

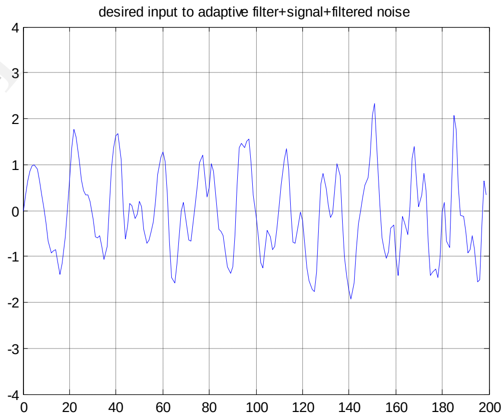



**Figure 10: Simulation of desired input to adaptive filter.**

To prepare the 'adaptfilt' object for processing, set the input conditions coeffs and mu for the object. The values set for coeffs and mu determine whether the adaptive filter can remove the noise from the signal path.Set and initialize RLS adaptive filter parameters and values. Running the RLS adaptive filter for 1000 iterations. The plot shows the convergence of the adaptive filter response to the response of the FIR filter. The closer one set the initial filter coefficients to the expected values, the more likely it is that the algorithm remains well behaved and converges to a filter solution that removes the noise effectively.

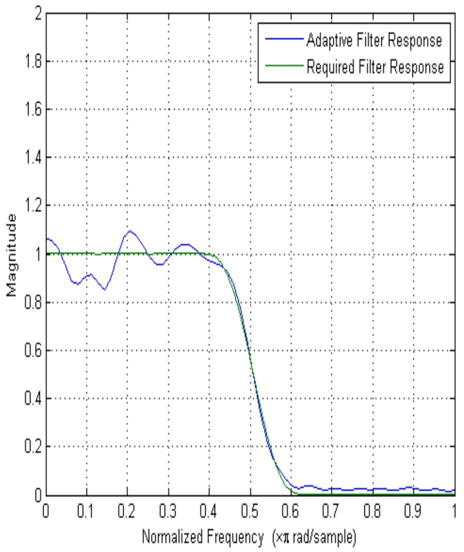



**Figure 11 : Simulation of adaptive filter response.**

As the adaptive filter converges, the filtered noise should be completely subtracted from the "signal + noise" signal and the error signal 'e' should contain only the original signal.

## CONCLUSION

ANC cancels the unwanted noise by generating antinoise of equal amplitude and opposite phase through the secondary sources. This paper has emphasized the practical aspects of ANC systems in terms of adaptive algorithms The primary objective of most active noise control systems is the reduction of noise for personal comfort, reduction of stress and increased concentration. Another major benefit is low-frequency quieting for applications where passive insulation would be too expensive, inconvenient or heavy. For example, the lead-impregnated sheets used to reduce aircraft cabin propeller noise impose a severe weight penalty, but active control can perform just as well at a much smaller weight.

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