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PERFORMANCE ANALYSIS OF LMS & NLMS AGORITHM FORACTIVE NOISE CANCELLATION

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Abstract- As we know environmental noise has been increasing since the industrial revolution. Noise affects our health and as well as interferes with the communication, so as in communication, it reduces our ability to detect the transmitted signal. In recent past to reduces the noise we use the passive components, which could control the high frequency noise and vibration but it was not so efficient at low frequencies.(below 500Hz).

Now a days we are using of Active Noise Cancellation technique to reduce the noise. In this technique noise is reduced by generating a cancelling anti-noise signal which is equal to(in magnitude), but 180 degrees out of phase with the noise. This anti-noise is the introduced into the environment such that it matches the noise in the region of interest. The two signals then cancel each other out, effectively removing a significant portion of the noise energy from the environment**.**

Keywords- Active & passive noise controller,
Acoustic system,Filter,Feedback & Feed forward control. Adavitive Noise filter LMS &NLMS

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Introduction

Conventional methods of suppressing acoustic noise using passive sound absorbers generally
don't work well at low frequencies. This is because at low frequencies, the acoustic wavelengths become large compared to the thickness of a typical acoustic absorber. A sound wave of frequency 100 Hz, for example, has a wavelength
of about 3.4m in air under normal conditions. It's also difficult to stop low frequency sound being transmitted from one space to another unless the intervening barrier is very heavy. For these reasons, a number of practically important acoustic noise problems are dominated by low frequency contributions. These problems are sometimes difficult to solve using passive methods since the solutions are expensive in terms of weight and bulk.

Active noise control exploits the long wavelengths associated with low frequency sound. It works on the principle of destructive interference between the sound fields generated by the original "primary" sound source and that due to other "secondary" sources, whose acoustic outputs can be controlled. The most common type of secondary source is the moving coil loudspeaker. Here, the acoustic output of the source is controlled by an electrical signal. It is the generation and control of the electrical signal to best reduce the acoustic field that is the signal processing task associated with active noise control.

In a far-sighted patent published in the United States in 1936, Paul Lueg first described the basic ideas of active noise control. The principle of measuring the sound field with a microphone, electrically manipulating the resulting signal and then feeding it to an electroacoustic secondary source are clearly described, as shown in Fig. 1, taken from this patent.

In diagram 1, the sound is initially considered to be traveling as plane waves in a duct, from left to right, originating from a primary source, A. The microphone, M, detects the incident sound wave and supplies the excitation to V, the electronic controller, which then drives the secondary loudspeaker, L. The object is to use the loudspeaker to produce an acoustic wave (dotted curve) that is exactly out of phase with the acoustic wave produced by the primary source (solid curve). The superposition of the two waves, from the primary and secondary sources, results in destructive interference.

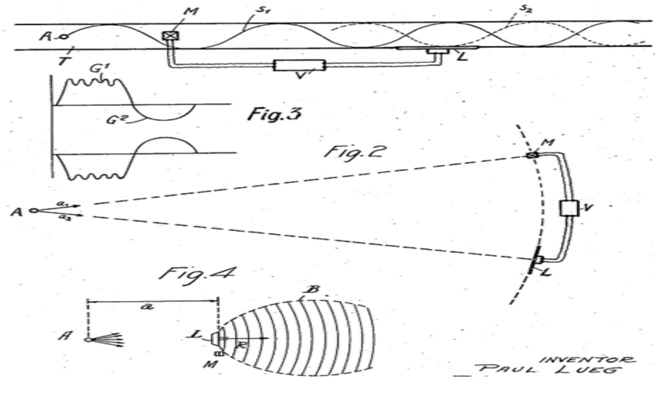


Fig. 1- Active Noise Control patent by Paul Lueg

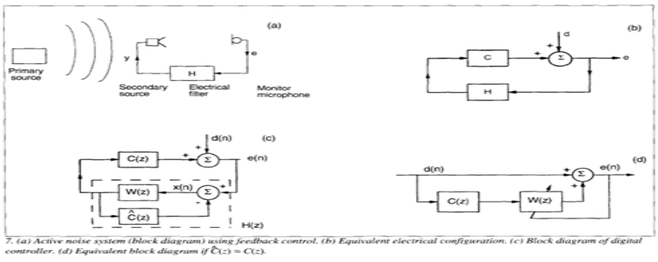
Thus, there is silence, in principle, on the downstream side (to the right) of the secondary source, L. The generation of a mirror image waveform for a non-sinusoidal acoustic disturbance is shown in diagram 3 in Fig. 1. Diagrams 2 and 4 illustrate Lueg's thoughts on extending the idea to an acoustic source propagating in three dimensions.

Acoustical Principles

All the strategies for active control listed earlier rely on the principle of superposition, which applies in any linear system. The propagation of an acoustic wave, with amplitude up to that corresponding to an extremely loud noise, is very nearly a linear process. The most significant cause of nonlinearity present in an active noise control system is usually due to the loudspeaker acting as the secondary source, although with good design this nonlinearity, too, can be made small.

The interference effects in acoustics are as follows: if the amplitude and phase of a pure tone signal driving one loudspeaker are adjusted relative to that driving another loudspeaker, then the acoustic pressure at a monitoring microphone, placed at any single point in the resulting sound field, can be driven to zero.

Unfortunately, it is also probable that at other points in the sound field, the two components of the pressure will be in phase and constructive interference will occur, increasing the sound level at these points. The philosophy suggested by Olson and May's arrangement of monitor microphone and secondary source is to position these



7. (a) Active noise system (block diagram) using feedback control. (b) Equivalent electrical configuration. (c) Block diagram of digital controller. (d) Equivalent block diagram if $\hat{C}(z) = C(z)$.

Fig. 2- Principle of feedback control

components close together. As a result, the secondary source will be very well coupled to the monitor microphone and only a modest loudspeaker drive voltage is required to achieve cancellation at this point. The pressure at other points, further away from the secondary source, will then not be significantly affected by this source.

Control Mechanisms

Feedback Control

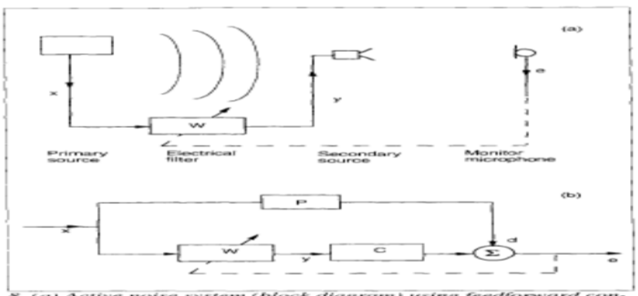
The feedback control approach is shown in Fig. 3. In figure, e represents the signal derived from the microphone due to the combined effect of the primary disturbance d and the feedback loop. The electrical transfer function of the feedback loop, H, was a simple gain and phase inversion described by Olson and May. The electrical transfer function from secondary loudspeaker input to microphone output, C, is called the secondary or error path. This system corresponds to the "plant" in conventional feedback control. Here, it contains the electroacoustic response of the loudspeaker, the acoustic characteristics of the path between loudspeaker and microphone, and the microphone's electroacoustic response.

The transfer function between the disturbance and measured error is thus

$$\frac{E(s)}{D(s)} = \frac{1}{1-C(s)H(s)}$$

Feedforward Control

A generic block diagram for such systems is shown in Figure 8a. The difference between this and the feedback approach is that a separate reference signal, x, is now used to drive the secondary source via the electrical controller W. This reference signal must be well correlated with the signal from the primary source. In systems for the control of broadband random noise, the reference signal provides advance information about the primary noise before it reaches the monitor microphone, which enables a causal controller to effect cancellation. In systems for the control of noise with a deterministic waveform, such as harmonic tones, this "advanced" information has little meaning since the controller only has to implement the appropriate gain and phase shift characteristics at each frequency.



8. (a) Active noise system (block diagram) using feedforward control. (b) Equivalent electrical configuration.

Fig. 3- Principle of feedback control

Another difference between the broadband and harmonic controllers is that in the latter case, an electrical reference signal can often be obtained directly from the mechanical operation of the primary source. Such a reference signal is completely unaffected by the action of the secondary source and the control is purely feedforward, as illustrated in Fig.3. In the broadband case, such as random noise propagating in a duct, a detection microphone often has to be used "upstream" of the secondary source to provide the reference signal, A (diagram 1 of Fig. 1). In this case, the output of the detection microphone, as well as being influenced by the primary source, will also be affected by the operation of the secondary source.

The LMS Algorithm

The Least Mean Square (LMS) algorithm, introduced by Widrow and Hoff in 1959 is an adaptive algorithm, which uses a gradient-based method of steepest descent. 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.

The LMS algorithm consists of two basic processes:

- A filtering process, which involves (a) computing the output of a linear filter in response to an input signal and (b) generating an estimation error by comparing this output with a desired response.
- An adaptive process, which involves the automatic adjustment of the parameters of the filter in accordance with the estimation error.

The signal flow graph representation of the LMS algorithm is shown in the following figure.

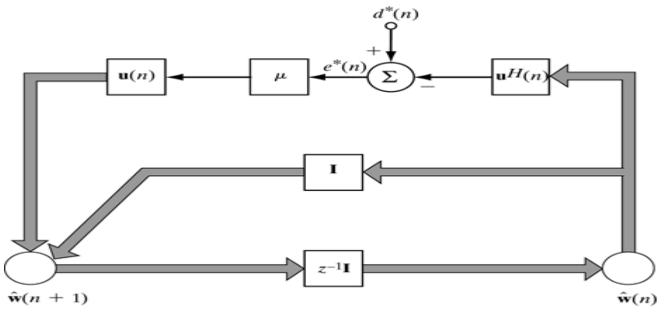


Fig. 4- Signal-flow graph representation of the LMS algorithm

From the method of steepest descent, the weight vector equation is given by

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

Where μ is the step-size parameter and controls the convergence characteristics of the LMS algorithm; e2(n) is the mean square error between the output y(n) and the reference signal which is given by,

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

The gradient vector in the above weight-update equation can be computed as:

$$\nabla_w(E\{e^2(n)\}) = -2r + 2Rw(n)$$

In the method of steepest descent, the biggest problem is the computation involved in finding the values r and R matrices in real time. The LMS algorithm, on the other hand, simplifies it by using the instantaneous values of covariance matrices r and R.

$$R(n) = x(n)x^h(n)$$

$$r(n) = d^*(n)x(n)$$

Therefore, the weight-update can be given by the following equation, The LMS algorithm is initiated with an arbitrary value w(0) for the weight vector at n=0. The successive corrections of the weight vector eventually leads to the minimum value of the mean squared error.

Therefore the LMS algorithm can be summarized in following equations:

Filter output: y(n)= wh(n).u(n)

Estimation error or error signal: e(n) = d*(n) – y(n)

Tap-weight adaptation: w(n+1) = w(n) + μx(n)e*(n)

Normalized LMS (NLMS) Algorithm

In the standard form of an LMS filter, the adjustment applied to the tap-weight vector of the filter at iteration (n+1) consists of the product of three terms:

- The step-size parameter μ, which is under the designer's control.
- The tap-input vector u(n), which is supplied by a source of information.

The estimation error e(n) for real-valued data, or its complex-conjugate e*(n) for complex valued data, which is calculated at iteration n.

The adjustment is directly proportional to the tap-input vector u(n). Therefore, when u(n) is large, the LMS filter suffers from a gradient noise amplification problem. To overcome this difficulty, we may use the normalized LMS filter. In particular, the adjustment applied to the tap-input vector u(n) at iteration (n+1) is "normalized" with respect to the squared Euclidean norm of the tap-input vector u(n) at iteration n-hence the term "normalized."

The weight update function, for the NLMS adaptive filter algorithm, is defined as

$$f(\mathbf{u}(n), e(n), \mu) = \mu e(n)\frac{\mathbf{u}^*(n)}{a + \mathbf{u}^H(n)\mathbf{u}(n)}$$

Performance comparison of various Adaptive filter algorithms

Performance of Normalized LMS algorithm in estimating a 32-order FIR filter

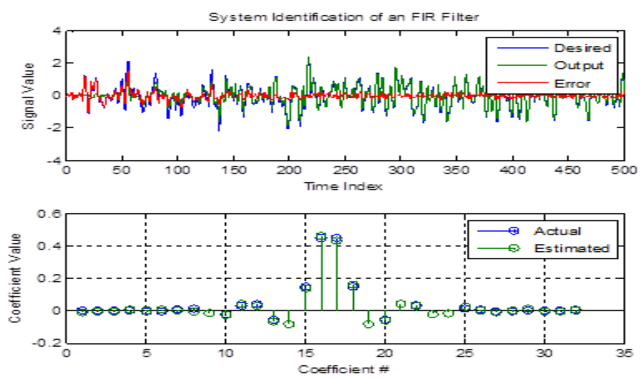


Fig. 5- output of filter using NLMS

Performance of LMS algorithm Adaptive Filter in estimating a 32 order FIR filter

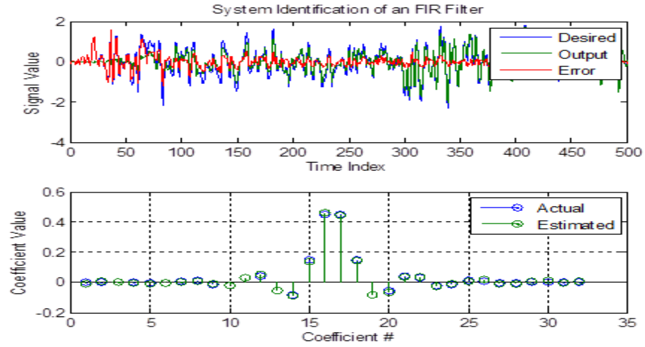


Fig. 6- output of filter using LMS

Conclusion

Conventional methods of suppressing noise do not work well at low frequencies. Active noise control exploits the long wave-lengths associated with low frequency sounds. It works on the principle of destructive interference between the sound fields generated by the original “primary” sound source and that due to other “secondary” sources, whose acoustic outputs can be electrically controlled.

Active noise control (ANC) system generates sound waves with opposite phase with respect to the background noise. This anti-noise causes active cancellation of acoustic pressure in a given zone – the quiet zone.

The reference sensors measure the noise signals coming from the Primary Source. This signal is then fed to the anti-noise transducers which produce an anti-noise 180 degree out of phase with noise through the Secondary Source.

Through the Principle of Superposition, the acoustic waves cancel each other in the zone of quiet while leaving the rest of the sound field relatively unchanged.

This paper explores the principles of Active Noise Control through Digital Signal Processing algorithms. Using computer simulations, the Adaptive filtering algorithms to reduce ambient noise are shown.

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