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1pSPc22. Adaptive active control of free space acoustic noise

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This paper concerns adaptive active control of acoustic noise in free space. Conventional adaptive active noise control algorithms are efficient in acoustic ducts or headphones; however, they are very sensitive when being used in free space. An efficient adaptive active noise control algorithm for free space noise is developed based on analyzing noise field in a volumetric zone around the error microphone. The traditional algorithm and the proposed algorithm are fully implemented by using a high performance embedded controller. The implemented setup is then used for active control of acoustic noise in free space. Different experiments show that the traditional active noise control algorithm is not stable when being used for free space noise; however, the proposed algorithm is stable and attenuates noise in free space by about 15 dB.

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INTRODUCTION

Active noise control (ANC) systems control or cancel acoustic noise by generating another noise with the same magnitude and opposite phase [1]. This simple theory makes the creation of noise-free zones possible; however, its applications are currently restricted to simple cases. This is because the majority of ANC designs have been developed based on general signal processing models and without considering sound behaviors in the acoustic domain.

Signal processing theory looks at ANC only as a problem of adaptive filtering [2]. The solutions obtained by this approach, called adaptive ANC algorithms [3], have been widely used for different ANC applications. In these algorithms, an adaptive digital filter (ADF) drives a loudspeaker which generates a control sound field. An adaptive algorithm is responsible for adjusting the ADF while taking into account all the acoustical behaviors and uncertainties associated with the physical plant. Since this task is performed adaptively during the operation of the system, no a prior adjustment of the ADF is needed. Accordingly, the implementation of such systems does not require any accurate theoretical analysis on the physical plant in the acoustic domain. This is the beauty of adaptive signal processing (or adaptive ANC) that it enables us to control an unknown acoustic plant without having deep knowledge of the associated physical phenomena. However, the understanding of these phenomena is essential for surpassing available constraints in practical ANC, one of which is discussed in the following.

Adaptive ANC algorithms can practically attenuate noise in a long duct by about −20dB [4–7]. However, they are very sensitive when being used in free space. Also, they cannot significantly attenuate the noise in free space. The implementation of adaptive ANC algorithms in ducts or headphones is successful because the sound propagation model in them is relatively simple. Accordingly, most of available ANC algorithms are efficient in the acoustic media similar to ducts [8–10] or headphones [11–13]. Since the sound propagation in free space is more complicated, it requires a more advanced adaptive algorithm for coping with changes in medium or noise characteristics. Otherwise, the adaptation process diverges. The main motivation for this paper is to develop an adaptive ANC algorithm for free-space noise through modeling the sound propagation phenomenon associated with ANC mechanisms.

MODELING ADAPTIVE ANC CONSIDERING SOUND PROPAGATION

Fig. 1 shows a general block diagram for adaptive ANC with single-channel feed-forward structure [14, 15]. This model has been derived only based on general signal processing models and without considering sound propagation models. Linear systems P and S, called the primary path and secondary path respectively, are electro-acoustic signal channels which model the acoustic media. Discrete signal $x(n)$, called the reference signal, is measured by a microphone located close to the noise source (primary source). Also, $e(n)$, called the error signal, is measured by another microphone located at the desired zone of silence, Ω. W is an ADF which generates $y(n)$ for driving a loudspeaker (secondary source). The anti-noise generated by this loudspeaker propagates through S to reach Ω. On the other hand, the noise generated by the noise source, propagates through P to reach Ω. These two signals combine with each other across the medium; however, $e(n)$ that is the net sound pressure at the center of Ω is only taken as a measure of the residual noise. From the block diagram given, one can find that for having $e(n) = 0$, the ADF should be set to $W = PS^{-1}$.

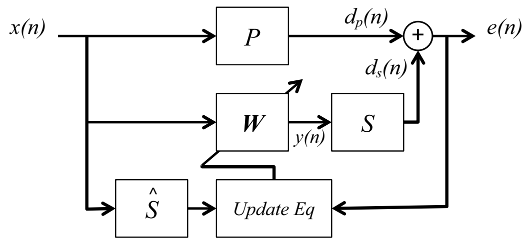

FIGURE 1: Functional block diagram of conventional single-channel adaptive ANC

However, reaching this ideal solution is not possible in practice because S^{-1} may be unstable and/or non-causal. For solving this problem, adaptive ANC algorithms are designated to find an stable and causal estimate for W in mean-square-error sense. Usually, the update equation of an adaptive ANC algorithm requires an estimate of the secondary path for filtering the reference signal. For this reason, these algorithms are called Filtered-x (Fx) adaptive algorithms. In Fig. 1, this estimate systems is shown by $\hat{S}$.

Now, let us look at the ANC block diagram in view point of acoustics. It is assumed that the noise source (primary source) and anti-noise source (secondary source) are harmonic monopoles located at the positions $\mathbf{r}_p=(r_p,\varphi_p,0)$ and $\mathbf{r}_s=(r_s,\varphi_s,0)$ in a spherical coordinate system. Also, it is assumed that the error microphone is located at the origin of the coordinate system. In this case, the reference signal $x(n)$ can be formulated as

$$x(n)=A_x e^{i\omega nT} \tag{1}$$

where $A_x \in \mathbb{C}$ is the temporal value of the monopole, ω is the normalized angular frequency of the monopole, and T is the sampling time. In this case, the acoustic field caused by this monopole can be modeled in an arbitrary position $\mathbf{r}=(r,\varphi,\theta)$ by [16]

$$d_p(n,\mathbf{r})=A_x\frac{e^{i\omega nT-ik|\mathbf{r}-\mathbf{r}_p|}}{|\mathbf{r}-\mathbf{r}_p|} \tag{2}$$

where $k=\frac{\omega}{c}$ is the wave number. Assuming that the primary monopole is located far from the origin, it can be shown for the positions located around the origin that $(r \ll r_p)$

$$d_p(t,\mathbf{r})=A_x\frac{e^{-ikr_p}}{r_p}\left(1-\frac{r}{r_p}\sin\theta\cos(\varphi-\varphi_p)\right)e^{i\omega nT-ikr\sin\theta\cos(\varphi-\varphi_p)} \tag{3}$$

On the other hand, $y(n)$ can be expressed as $y(n)=WA_x e^{i\omega nT}$. This signal is broadcasted by the secondary source. Accordingly, the sound field generated by the secondary monopole can be formulated by using Eq. (3) as

$$d_s(n,\mathbf{r})=WA_x\frac{e^{-ikr_s}}{r_s}\left(1-\frac{r}{r_s}\sin\theta\cos(\varphi-\varphi_s)\right)e^{i\omega nT-ikr\sin\theta\cos(\varphi-\varphi_s)} \tag{4}$$

In this case the net acoustic field around the origin can be expressed as

$$\varepsilon(n,\mathbf{r})=d_p(n,\mathbf{r})+d_s(n,\mathbf{r}) \tag{5}$$

Since the error microphone is located at the origin, $e(n)$ can be formulated as

$$e(n)=\varepsilon(n,\mathbf{0}) \tag{6}$$

By substituting Eqs. (3), (4) and (5) into (6) and setting the result to zero, the optimal ADF can be found as

$$\tilde{W}_{opt} = -\frac{r_s}{r_p} e^{-ik(r_p - r_s)} \tag{7}$$

This optimal solution corresponds to $W = PS^{-1}$ which is estimated by conventional adaptive ANC algorithms. As seen, this optimal solution is derived only based on the minimization of the sound field at a single point for which the error signal is measured. This causes the sensitivity of the adaptation process when the sound propagation model around this point is complicated (e.g. free space noise). This is the main reason that conventional ANC algorithms are very sensitive when being used for active control of noise in free space. In the next section, an alternative optimal solution, which is obtained through the minimization of the sound field in a volumetric zone around the error microphone, is found. Also, an adaptive algorithm for searching this optimal solution will be proposed.

Derivation of an Adaptive ANC Algorithm for Free Space Noise

Let us define a cost function as

$$J(a) = \frac{3}{4T\pi a^3} \iiint_{\mathbf{r} \le a} \int_0^T |\varepsilon(t, \mathbf{r})|^2 \, dt dV \tag{8}$$

$J(a)$ represents the average of the acoustic pressure over a sphere centered at the origin with the radius of a. Unlike the cost function in the conventional ANC algorithms, $J(a)$ considers the volumetric propagation of sound around the desired zone of silence. Now, the optimal solution for W can be found by minimizing $J(a)$:

$$W_{opt} = \arg_{W} \nabla \{J(a)\} = 0 \tag{9}$$

where ∇ denotes the gradient vector. Substituting Eq. (5) into (8), calculating the integral given in Eq. (8) and combining the results with Eq. (9) result in

$$W_{opt} = \rho \tilde{W}_{opt} \tag{10}$$

where complex parameter ρ is given by

$$\rho = \frac{3}{2\beta k a} j_1(2\beta k a) - \frac{3i}{4 r_s \beta k^2 a} \left[3 j_1(2\beta k a) - 2\beta k a j_0(2\beta k a)\right] \tag{11}$$

In Eq. (11), $j_v(.)$ represents the spherical Bessel function of the first kind and $\beta = -\sin(\varphi_p - \varphi_s)$. Note that in the derivation of Eq. (11), it is assumed that the secondary source is much closer to the origin than the noise source ($r_p \gg r_s$). Eq. (10) shows that the new optimal solution for W is proportionally related to the conventional optimal solution and the proportion parameter is a function of the noise frequency and the plant geometry. Now, it is desired to develop an adaptive algorithm for estimating the new optimal solution.

For the adaptive estimation of the optimal solution given in Eq. (10), it is proposed to modify the conventional adaptive ANC system as shown in Figure 2. According to this figure, W must converge to $\tilde{W}_{opt}$ again; however, as it is followed by a static gain of ρ; it can be concluded that $\rho\tilde{W}_{opt}$ is the ADF filter which estimates the anti-noise signal. Accordingly, the adaptive algorithm needs an error signal identical to that of the conventional adaptive ANC algorithms because it has to estimate $\tilde{W}_{opt}$ again. Therefore, the effects of ρ on $e(n)$ should be removed. From block diagrams

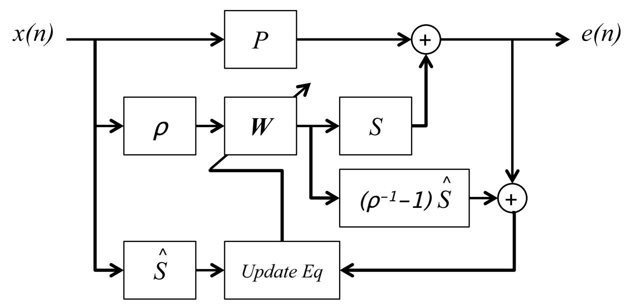

FIGURE 2: Proposed adaptive ANC system (which consider sound propagation around the desired zone of silence)

given in Figures 1 and 2, it can be seen that the difference between the two error signals can be expressed as $(\rho-1)WSX$ in the z-domain. Therefore, compensating for this difference requires that $(\rho-1)WSX$ to be subtracted from the error signal. This term can be constructed from $y(n)$, and it can be then subtracted from the error signal, as shown in Figure 2. (note that $y(n)$ is an available signal which can be expressed as ρWX in the z-domain.)

EMBEDDED IMPLEMENTATION AND EXPERIMENTAL RESULTS

Fig. 3 shows the schematic diagram of the fully implemented adaptive ANC system used in this research. The FxLMS algorithm [17] is used as the update equation of the system. The noise source is a computer-driven loudspeaker which generates a tonal interference. The microphone located close to the noise source is the reference microphone and the one located in Ω is the error microphone. The outputs of these microphones are fed to two A/D modules to produce $x(n)$ and $e(n)$. The A/D modules are connected to an FPGA chassis, where signal acquisition and conditioning algorithms can be performed. Also, the FPGA chassis produces $y(n)$ which is then transferred to a D/A module connected to the FPGA chassis. The output of the D/A module is then fed to a loudspeaker. Another hardware component of the proposed scheme is a real-time embedded controller which configures the FPGA chassis. Also, this controller is responsible for transferring data between the FPGA chassis and a PC, where FPGA codes are compiled and the measured data are monitored, analyzed and recorded. The acoustic signal produced by the loudspeaker propagates through the medium to reach Ω. The effects of the microphones, loudspeaker and other electro-acoustical components can be included in the primary and secondary paths [6]. In this case, the system shown in Fig. 3 completely matches the model given in Fig. 1.

As shown in Fig. 3, the implemented setup includes an acoustic head which has an external hearing structure similar to that of human. This acoustic head does not contribute to ANC operations; but it is used for monitoring the ANC system performance. For the verification of this paper theoretical results, the following experiments with the experimental setup are performed.

The system is placed in a large anechoic room, where sound propagates in three spatial dimensions without any reflections (similar to free space). The loudspeaker used as the primary source is driven by a tonal noise of the frequency of 200 Hz. In the first experiment, the radius of the zone is set to $a = 0$. Therefore, the adaptive algorithm considers only the error signal at the measured point (conventional FxLMS algorithm). The adaptation step-size of the FxLMS algorithm is set to the half of the step-size upper bound [5], resulting in the highest possible convergence rate. In this case, the adaptive algorithm diverges shortly after activating the ANC system. Therefore, the power of the error signal becomes unstable, as shown in Figure 4.

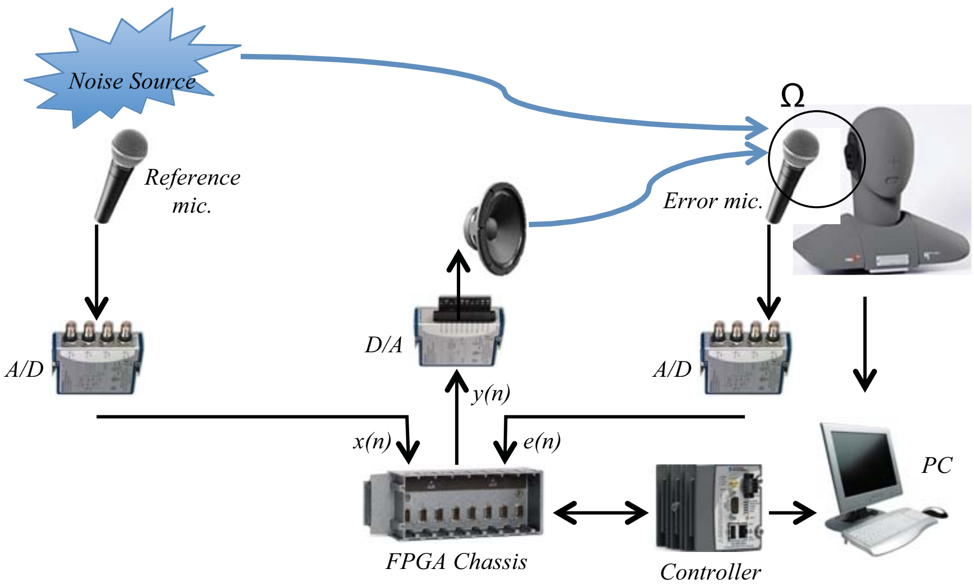

FIGURE 3: Layout of the implemented system

In the second experiment, a is set to 2 cm; therefore, the adaptive algorithm considers a spherical zone with the radious of 2 cm around the error point. As shown in Figure 4, the adaptive algorithm is stable in this case and it is able to attenuate the noise by about 15 dB after 250 mSec. In the third experiment, the sphere radius is set to $a = 5$cm. In this case, the adaptive algorithms is still stable but as the zone of interest is larger than the previous case, the adaptive algorithm cannot attenuated the noise more than 10 dB.

CONCLUSION

The optimal solution for the ADF used in conventional ANC is calculated only based on the minimization of the sound field at a single point(s) for which the error signal is measured. All the conventional adaptive ANC algorithms are designated to search for this optimal solution. This causes the sensitivity of the adaptation process when the sound propagation model around the error microphone is complicated. For example, this situation happens when the ANC system is used for the attenuation of noise in free space. This paper proposes an alternative optimal solution, which considers the sound field in a volumetric zone around the error microphone. This solution is obtained by considering the sound propagation in three diminutional space. Also, an adaptive algorithm for searching this optimal solution is proposed. This algorithm is found very efficient in practice.

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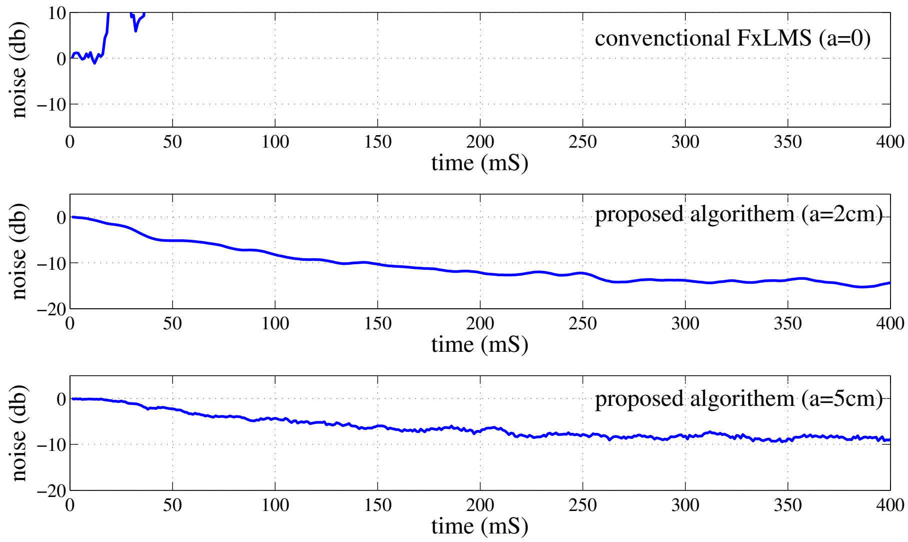

FIGURE 4: Experimental results

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