LETTER

Active Noise Canceling for Headphones Using a Hybrid Structure with Wind Detection and Flexible Independent Component Analysis

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SUMMARY This letter presents a method for active noise cancelation (ANC) for headphone application. The method improves the performance of ANC by deriving a flexible independent component analysis (ICA) algorithm in a hybrid structure combining feedforward and feedback configurations with correlation-based wind detection. The effectiveness of the method is demonstrated through simulation.

key words: active noise cancelation, headphone, independent component analysis, flexible score function

1. Introduction

Recently, headphones have attracted significant consumer interest because personal mobile devices such as smartphones are widely used. Therefore, headphone manufacturers seek to improve their products by utilizing additional functionality such as noise reduction (NR) (e.g., [1]). Although passive NR performs well when blocking high frequency noise over several kHz with a half-wavelength shorter than the thickness of the shielding material in the headphone cup, active noise cancelation (ANC) has been used to successfully cancel the noise passing through the headphone cup by generating a destructive canceling signal.

As a basic ANC structure using adaptive filters, feedforward ANC suffers from performance degradation due to wind noise, since it distorts the external microphone’s reference signal that is used to cancel the penetrating noise [2]. On the other hand, feedback ANC introduces a feedback loop, and therefore, low-frequency noise can only be successfully canceled by retaining the stability of the loop, which may result in relatively less NR performance [2]. Therefore, a hybrid structure combining feedforward and feedback systems [3] is considered, and a method to switch their contributions according to correlation-based wind detection is presented to achieve improved NR performance by combining the advantages of both methods.

In addition, some methods of ANC based on the least-mean-squares (LMS) algorithm have been used for headphones (e.g., [4]). However, their performance can be degraded for powerful target playback signals or nonstationary noise because the error signal, directly proportional to filter weight updates in the LMS algorithm, may have large amplitudes, possibly leading to divergence. Unfortunately, these sounds occur in typical scenarios for headphone listening, such as listening to rock music in noisy streets. In order to overcome this drawback, this letter introduces independent component analysis (ICA) for ANC in the hybrid structure for headphone applications [5]. The filter weight updates in the ICA-based ANC method use a compressed version of the error signal to avoid spoiling filter adaptation. In particular, since the distribution of the target signal varies in headphone listening, a flexible ICA (FICA) algorithm [6] is derived by estimating an adaptive filter and the distribution of the target signal at the same time, further improving the NR performance.

2. Proposed Hybrid Structure for ANC with Wind Detection

ANC in headphones can be accomplished by canceling the noise signal penetrating into a headphone cup with its antiphase signal. The destructive canceling signal is estimated in the electrical domain through operation in an electric circuit, although the actual cancelation occurs in the acoustic domain by the superposition of acoustic signals after a headphone driver plays the destructive canceling signal superimposed on a desired target playback signal.

The feedforward structure utilizes the external microphone, in addition to the error microphone, which is placed on the external housing to capture environmental noise used as the reference signal [2]. For successful cancelation of the penetrating noise signal, the reference signal should be processed to provide the destructive canceling signal. Unfortunately, wind usually cannot pass through the headphone cup material but causes turbulence at the membrane of the external microphone. Consequently, the reference signal is distorted, and it may cause severe degradation of NR. On the other hand, the feedback structure obtains the reference signal from the error microphone located at the quiet zone, and therefore, the wind cannot distort the reference signal [2]. However, the acquisition of the reference signal from the error microphone introduces a feedback loop, and it is suitable for canceling low-frequency noise only while retaining a stable feedback loop. Thus, for effective NR in headphone applications, a hybrid structure combining the feedforward and feedback configurations may be considered [3].

The hybrid structure with switching mode according to wind detection is proposed, to prevent performance degra-
The proposed hybrid structure for ANC.

![Diagram of ANC system](image)

Fig. 1 The proposed hybrid structure for ANC.

...dation caused by the reference signal with wind noise corruption. For real-time applications, a simple correlation-based wind detection method is regarded in the proposed method [7]. Wind noise components acquired by two external microphones, placed outside left (L) and right (R) headphone cups covering the two ears, are nearly uncorrelated with each other. Thus, wind can be easily detected by measuring the absolute value of the correlation coefficient between the L and R external microphones defined as [7]

\[
\rho_{v_Lv_R} = \frac{\sum_n v_L(n)v_R(n)}{\sqrt{\sum_n v_L^2(n)} \sqrt{\sum_n v_R^2(n)}},
\]

where \(v_L(n)\) and \(v_R(n)\) represent the signals acquired by L and R microphones, respectively. Strong wind was detected with a threshold of 0.7 and a hysteresis window of \(\pm 0.05\).

In case strong wind exists, the reference signal acquired by the external microphone in the feedforward configuration can be significantly distorted, whereas the reference signal in the feedback configuration is not affected by the wind due to shielding by a headphone cup. Thus, feedforward and feedback parts for the proposed ANC structure shown in Fig. 1 are switched according to the detection of wind. External noise \(u(t)\) passes through the headphone cup to the quiet zone, modeled by the acoustic channel \(h(t)\). A destructive canceling signal \(y(n)\) is generated at the filter output, which is mixed with the target signal \(s(n)\) to drive the headphone driver, and it is superimposed at the quiet zone to cancel out the penetrating noise \(d(t) = h(t) * u(t)\). The residual noise and the target signal are acquired, giving an error signal \(e(n)\), which is fed to the FICA filter update process. The destructive canceling signal \(y(n)\) consists of \(w^T_{ff}(n)r_{ff}(n)\) with weighting factor 1 - \(\lambda_g\) and \(w^T_{fb}(n)r_{fb}(n)\) with weighting factor \(\lambda_g\), where \(w_{ff}(n)\) and \(r_{ff}(n)\) denote a filter coefficient vector and a reference signal vector, respectively.

\[
y(n) = (1 - \lambda_g)w^T_{ff}(n)r_{ff}(n) + \lambda_g w^T_{fb}(n)r_{fb}(n),
\]

\[
e(n) = d(n) + s(n) - y(n).
\]

3. Proposed FICA-Based ANC Algorithm

Most ANC systems widely used are based on the normalized LMS (NLMS) algorithm. In case of the feedforward ANC (corresponding to the case \(\lambda_g = 0\)), the filter update rule is

\[
\Delta w_{ff}(n) = \frac{e(n)r_{ff}(n)}{r^T_{ff}(n)r_{ff}(n)}.
\]

The NLMS algorithm attempts to minimize the power of the error signal \(e(n)\) that results in a minimization of the power of the residual noise. However, with a moderately large step size required to speed up the convergence, minimizing the power of the error signal does not necessarily result in cancelation of the residual noise. This is because uncorrelatedness between the target and noise signals may not hold for a relatively short time interval corresponding to a quasi-fixed adaptive filter with a large step size. If the noise is nonstationary or even impulsive, the error signal may have abrupt increase in amplitude due to unsuccessful cancelation of noise. In particular, since the NLMS update amount in Eq. (4) is directly proportional to the error signal \(e(n)\), a powerful target signal or nonstationary noise can upset the stable convergence of the adaptive filter.

In order to avoid the performance degradation of ANC in the presence of nonstationary or even impulsive noise and/or a powerful target playback signal, this letter describes an ANC algorithm based on ICA with a flexible score function in noise-canceling headphones. ICA, a signal processing method to express multivariate data as linear combinations of statistically independent random variables, has been developed to recover sources from mixtures [8]. In the context of conventional adaptive noise cancelation, ICA-based approaches have been developed to improve the performance by maximizing the independency among output signals that may incorporate higher order statistics [5]. Mutually independent signals composing a vector \(u\) can be separated from their mixtures, by maximizing the entropy of \(z = g(u)\) where \(g(\cdot)\) denotes a function approximating the cumulative density function [8]. In the hybrid structure for ANC shown in Fig. 1, an ICA algorithm can be derived by regarding the reference signals for feedforward and feedback parts, \(r_{ff}(n)\) and \(r_{fb}(n)\), as dummy outputs, in addition to a system output signal \(e(n)\) to obtain \(u = [e(n) r_{ff}(n) r_{fb}(n)]\). The entropy of \(z = [z_1(n) z_2(n) z_3(n)]\) is expressed in terms of the Jacobian \(J\) as

\[
H(z) = -E[\log p(z)] = -E[\log p(t)] + E[\log |J|],
\]

where \(p(\cdot)\) denotes the probability density function (PDF), the input vector \(t\) is equal to \([s(n) r_{ff}(n) r_{fb}(n)]\), and

\[
J = \begin{vmatrix}
\frac{\partial z_1}{\partial s} & \frac{\partial z_1}{\partial r_{ff}} & \frac{\partial z_1}{\partial r_{fb}} \\
\frac{\partial z_2}{\partial s} & \frac{\partial z_2}{\partial r_{ff}} & \frac{\partial z_2}{\partial r_{fb}} \\
\frac{\partial z_3}{\partial s} & \frac{\partial z_3}{\partial r_{ff}} & \frac{\partial z_3}{\partial r_{fb}}
\end{vmatrix} = \frac{\partial z_1}{\partial e} \frac{\partial z_2}{\partial r_{ff}} \frac{\partial z_3}{\partial r_{fb}} = p(e(n)) \frac{\partial z_2}{\partial r_{ff}} \frac{\partial z_3}{\partial r_{fb}}.
\]
Maximizing $H(z)$ with respect to filter coefficient vectors $w_{ff}(n)$ and $w_{fb}(n)$ provides the filter update rules for ICA-based ANC, whose normalized form can be represented as

$$
\Delta w_{ff}(n) \propto (1 - \lambda_{g}) \varphi(e(n))|r_{ff}(n)|/|r_{ff}(n)|,
$$

(7)

$$
\Delta w_{fb}(n) \propto \lambda_{g} \varphi(e(n))|r_{fb}(n)|/|r_{fb}(n)|,
$$

(8)

where $\varphi(e) = -d \log(p(e))/de$ represents the score function. Therefore, the update rule is similar to the NLMS-based ANC, except for existence of the score function [5]. After the adaptive filter has converged sufficiently, the error signal provides the target playback signal. In headphone listening scenarios, the target playback signal may vary, and therefore, its PDF should not be fixed. In this case, a flexible score function can be considered to reflect the distribution of the error signal during filter estimation [6]. As a parametric model for approximating the PDFs of various natural sounds, the generalized Gamma distribution (GGD) can be used [9]. For a random variable $x$, the GGD is defined as

$$
p(x|\alpha, \beta, \gamma) = \frac{\gamma \beta^{\gamma}}{\Gamma(\alpha)} |x|^\gamma e^{-\beta |x|^\alpha},
$$

(9)

where $\alpha$, $\beta$, and $\gamma$ are parameters determining the distribution shape, and $\Gamma(\alpha)$ denotes the Gamma function. The score function for $e(n)$ can be derived from the GGD as

$$
\varphi(e(n)|\alpha(n), \beta(n), \gamma(n)) = \frac{\text{sign}(e(n))}{|e(n)| + \epsilon} \frac{\gamma \beta^{\gamma}}{\Gamma(\alpha)} (\gamma |b(n) e(n)|^{\gamma(n)} - \alpha |n| \gamma(n) + 1),
$$

(10)

where $\epsilon$ is a small positive constant to avoid division by zero. $\alpha(n)$, $\beta(n)$, and $\gamma(n)$ can be estimated by using maximum likelihood (ML) estimation. Furthermore, an online estimation algorithm using a forgetting factor is also available [9].

4. Experimental Results

The performance was evaluated by the NR ratio (NRR) computed by the ratio of the penetrating noise power to the residual noise power in the system output $e(n)$, given by

$$
\text{NRR(dB)} = 10 \log_{10} \left( \frac{\sum_{n} d^{2}(n) - g(n)^{2}}{\sum_{n} d^{2}(n)} \right).
$$

(11)

For the target signal, an intro of approximately 5 s of various clips was used. The clips were chosen from various popular genres, including jazz, pop, vocal phrase, classical, and electronic dance music. For the noise source, 12 noise clips including stationary, nonstationary, and impulsive noises from the Noise-X database of Rice University and Soundjay [10] were used. All signals used in the simulation have a sampling rate of 44.1 kHz to match the most common audio sampling rate.

For computer simulation, a filter was modeled to approximate a headphone cup attenuation curve, with data obtained from audio analyzer measurements using pink noise in an anechoic chamber, as shown in Fig. 2. The filter was convolved with a noise source signal to simulate the penetrating noise, whereas adaptive filters to generate the destructive canceling signal were set to have 10 taps. To simulate the time elapsed during A/D conversion, a delay of 45 μs, corresponding to the sample group delay of high-speed sigma-delta conversion, was added to each conversion process. Extensive simulation was performed to choose the appropriate step sizes, which were 0.0005, 0.0005, and 0.0004 for NLMS, ICA, and FICA algorithms, respectively.

Figure 3 shows a comparison of score functions. By comparing the NLMS update rule with those of conventional ICA with a fixed score function and FICA, its score function can be regarded as the identical function. The ICA rule that assumes output signals to be Laplace-distributed has only $-1$, $0$, or $1$ as score values. On the other hand, the score function of the FICA update rule varies according to the estimated distribution of the error signal. The error signal corresponding to the score function in Fig. 3 (c) was more or less sparse than the Laplace distribution (corresponding to a more abrupt or smooth change at 0, respectively).

For illustrative purpose, Fig. 4 shows traffic noise spectrograms. For various noise profiles, NRRs are summarized in Fig. 5. ICA and FICA yielded better NR performance than NLMS. The filter update values of NLMS are proportional to $e(n)$ that still contains the target signal after convergence; thus they sometimes have large update amplitudes, and may thus spoil the filter convergence. Therefore, the step size should be set to a small value to avoid divergence, which may cause a conflict with fast convergence. In particular, when the noise was impulsive, such as door slam or machine gun noises, the performance gap between NLMS and ICA became significant, because ICA could provide bounded and compressed update amounts that were safe from divergence. Furthermore, FICA yielded better NR performance with faster convergence than ICA owing to the flexible score function that helps to generate appropriate weight updates, according to the error signal distribution.

To evaluate the robustness against wind noise for the proposed ANC structure, a sample of wind recorded by stereo microphones 10-cm apart from each other was added to four noise profiles, to generate the reference signal for the
noise, but the feedback configuration in the proposed structure helped to adapt the filter. Figure 7 shows NRRs for FICA-based ANC. The results demonstrated that the proposed method was effective for wind noise because it could successfully switch the contributions of the feedforward and feedback configurations accordingly.

5. Conclusion

In this letter, FICA-based ANC in a hybrid structure with wind detection was proposed, in order to improve NR performance in headphones. The experimental results showed that the ICA-based filter adaptation provided better performance than the NLMS method for various noise profiles. In particular, the FICA-based algorithm further improved NR performance with faster convergence. In addition, the hybrid structure combining feedforward and feedback configurations could significantly remove noise by switching between their contributions, using correlation-based wind detection.

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