A SINGLE SENSOR BASED ACTIVE CONTROL OF ACOUSTIC REFLECTIONS

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In this study, we propose an adaptive algorithm to achieve an acoustic absorption by a vibrating wall. The conventional two-sensor-based approach often exhibits poor performance when there is beamforming error. In this study, we propose a single-sensor-based algorithm, in which the reference and error signals of the adaptive filter are synthesized from the error sensor output. We also propose a method of estimating the impulse response of the secondary path in a single-sensor-based control system. The control filter is updated using the filtered-x LMS algorithm and computer simulations were conducted in a modelled sensor-actuator module. Simulation results confirmed superiority of the proposed algorithm over the conventional method.

1. Introduction

The active noise control (ANC) technique has been widely used to achieve high attenuation of the primary noise filed over a wide range of frequency band [1]. The principle of ANC is the superposition of the two acoustical waves with opposite phases from the primary and secondary sources [2]. In this paper, we develop an adaptive algorithm that can achieve acoustic absorption with a thin vibrating wall. The main strategy is to eliminate the reflective wave from the wall surface, among the signal components collected by the error sensor.

To achieve the same objective, various algorithms have been proposed. The earliest work related to the active absorbing system can be found in [3], where an analog control circuit, whose input was provided by a two-microphone probe, was designed to allow arbitrary reflection coefficients between 0 and 1.5. However, near-field effect caused major accuracy problem and the control was feasible only under 1kHz. Another adaptive reflection control algorithm based on a beamforming technique was also proposed in [4], where the incident and reflected waves were separated from the two closely spaced microphones by simply realizing judicious subtraction of the time delayed microphone outputs [4]. This two-sensor approach is very convenient to implement. However, its performance is heavily depends on the beamformer performance, and any error of the beamformer can result in serious degradation of the control performance. In addition, when a thin panel is considered for a noise transmission block [4] or an echo reduction [5], the sensors need to be encapsulated within a thin wall, which makes it even more difficult to achieve an accurate beamforming especially at low frequencies. Furthermore, the measure of the acoustic impulse responses of the encapsulated sensor-actuator paths is not typical.
The control algorithm presented in this paper obtained signal components for control from the single sensor output, so that the sensor-actuator module can be conveniently encapsulated within a thin panel. The paper includes detailed description of the algorithm and computer simulation results obtained in a modelled system.

2. Beamforming-based active reflective sound control system

A typical arrangement of the beamforming-based active sound reflection control system is illustrated in Fig. 1. The incident wave \( u(n) \) and reflected wave \( r(n) \) are separated utilizing the phase relationship between the two sensor signals \( s_1(n) \) and \( s_2(n) \).

\[
\hat{u}(n) = s_1(n) - s_2(n - \tau),
\]

\[
\hat{r}(n) = s_2(n) - s_1(n - \tau),
\]

where \( \tau \) is a sample delay between the sensor signals. If we further assume that the beamforming process produces no error, the estimated incident and reflected waves can be expressed in z-transform domain using as

\[
\hat{U}(z) = U(z)(1 - e^{-2\mu \tau}),
\]

\[
\hat{R}(z) = U(z)H_s(z)e^{-\mu \tau}(1 - e^{-2\mu \tau}) + C(z)(1 - e^{-2\mu \tau}).
\]

where \( H_s(z) \) is the transfer function of the reflective acoustic path between the second sensor and the wall surface. The control signal \( C(z) \) can be written as \( C(z) = H_c(z)Y(z) \) where \( H_c(z) \) is the transfer function between the control source and the second sensor, and \( Y(z) \) is the control filter output. In order to eliminate the reflective wave \( \hat{R}(z) \), the control filter output can be determined as

\[
Y(z) = -U(z)H_c^{-1}(z)H_s(z)e^{-\mu \tau}.
\]

An adaptive system for the sound reflection control can be implemented using the filtered-x normalized least-mean-square (FxNLMS) algorithm [1]. Fig. 2 shows a block diagram of the FxNLMS algorithm for the adaptive implementation of the control system shown in Fig. 1, where \( \hat{u}(n) \) is the estimated reference signal, \( y(n) \) is the control filter output, \( y'(n) \) is the control sound at the error sensor, \( e(n) \) and \( \hat{r}(n) \) are the estimated residual error and reflected signals, respectively. \( H_c(z) \) is the frequency response of the secondary path and its estimate is denoted by \( \hat{H}_c(z) \). \( P(z) \) is the unknown path between the incident and reflected waves, and \( W(z) \) is the digital filter that is adapted to generate the control signal in order to reduce primary noise.
The weight vector is updated in order to minimize the mean squared error (MSE) $J = E\{|e(n)|^2\}$, and if an $L$th-order control filter is used, the FxNLMS algorithm is given by

$$w(n + 1) = w(n) + \frac{\mu}{\sigma_x^2(n)} x'(n) e(n). \tag{9}$$

where $\mu$ is the step-size parameter and $\sigma_x^2(n)$ is the power estimate of the filtered reference input. $x'(n)$ is the filtered reference input vector with elements given by $x'(n - i) = b^T x(n - i), i = 0, 1, ..., L - 1$ where $x(n - i)$ and $b = [b_0, b_1, \cdots, b_{N-1}]^T$ represent the filtered reference input vector and the impulse response vector of the secondary path, respectively.

In order to control reflective wave, the estimated $\hat{u}(n)$ and $\hat{r}(n)$ are used as an acoustic reference and error signals for the FxNLMS algorithm. Thus, any error in estimating those signals can results in serious performance degradation. The beamforming technique used to estimate $\hat{u}(n)$ and $\hat{r}(n)$ is quite simple, but its performance can be problematic at low-frequencies [6], and the problem becomes more serious as the distance between the two sensors decreases.

3. Single Sensor based active reflective wave control System

3.1 Single sensor-based active reflective signal control algorithm

Without the control source, the single sensor signal is given by a sum of the primary noise $u(n)$ and the reflected wave $r(n)$. In addition, the reflected wave $r(n)$ can be represented as a filtered version of $u(n)$ since the latter is a reflection of the former. Thus, the reflected wave is estimated by filtering the incident wave using the impulse response $h_s(n)$ representing the reflective path between the sensor and the wall.
From Fig. 3, the single sensor signal can be expressed in z-domain as
\[ S(z) = U(z) + R(z) = U(z) + U(z)H_s(z). \] (10)
where \( H_s(z) \) is a z-transform of \( h_s(n) \). Thus, for known \( H_s(z) \), the incident and reflected waves can be estimated from the sensor signal as
\[ U(z) = \frac{S(z)}{1 + H_s(z)} \] (11)
\[ R(z) = S(z) - U(z) = S(z) \frac{H_s(z)}{1 + H_s(z)}. \] (12)
In time-domain, the signals can be obtained as
\[ u(n) = s(n) - \hat{h}_s^T u(n), \] (13)
\[ r(n) = h_s^T s(n) - \hat{h}_s^T r(n), \] (14)
where \( h_s = [h_{s,0}, h_{s,1}, ..., h_{s,M-1}]^T \) is the impulse response vector of the acoustic path corresponding to \( H_s(z) \). \( u(n) = [u(n), u(n-1), ..., u(n-M+1)]^T, s(n) = [s(n), s(n-1), ..., s(n-M+1)]^T \), and \( r(n) = [r(n), r(n-1), ..., r(n-M+1)]^T \) are the incident, sensor and reflected signal vectors, respectively. In Eqs. (13) and (14), we assumed the acoustic path \( H_s(z) \) is modelled using an \( M \)th-order FIR filter.

However, Eqs. (13) and (14) are non-causal because the unknown \( u(n) \) and \( r(n) \) are required to obtain the corresponding signals at the time index \( n \). But the sensor and the wall are separated by a certain distance, \( h_{s,0} \) should comprise an initial delay (zero values) corresponding to the traveling time of the path \( H_s(z) \). Thus, it is reasonable to assume that \( h_{s,0} = 0 \). Then, we obtain causal equations for estimating the incident and reflected signals:
\[ u(n) = s(n) - \hat{h}_s^T u(n-1), \] (15)
\[ r(n) = h_s^T s(n) - \hat{h}_s^T r(n-1), \] (16)
where \( \hat{h}_s = [h_{s,1}, h_{s,2}, ..., h_{s,N-1}, 0]^T \) is obtained by removing the first coefficient of \( h_s \).

Now, by adding a secondary control source, the sensor signal is given by
\[ S(z) = U(z) + H_s(z)U(z) + C(z). \] (17)
Similarly, for known \( H_s(z) \) and \( H_c(z) \), we can estimate the incident and reflected signals directly from the sensor signal as
\[ \hat{u}(n) = \hat{s}(n) - \hat{h}_s^T u(n-1), \] (18)
\[ \hat{r}(n) = h_s^T \hat{s}(n) - \hat{h}_s^T u(n-1), \] (19)
where \( h_c \) is the time-domain impulse response vector corresponding to \( H_c(z) \), and \( \hat{s}(n) = [\hat{s}(n), \hat{s}(n-1), ..., \hat{s}(n-L+1)]^T \) with \( \hat{s}(n) = s(n) - h_c^T y(n) \).

For the adaptive implementation of the reflection control using a single-sensor, the conventional FxLMS algorithm can be used, where the reference and reflected signals are estimated from the single sensor signal using Eqs. (18) and (19). The cost function is then given by \( J = E[|\epsilon(n)|^2] \) where \( \epsilon(n) = \hat{r}(n) - h_c^T y(n) \). Thus, The FxLMS algorithm is summarized as
\[ w(n+1) = w(n) - \frac{\mu}{\sigma_x^2(n)} x'(n)\epsilon(n), \] (22)
\[ x'(n) = [x'(n), x'(n-1), ..., x'(n-L+1)]^T, \ x'(n) = h_c^T u(n). \] (23)
Fig. 3 shows a schematic diagram of an adaptive reflection control system being implemented using the proposed single-sensor technique.
Estimation of the impulse responses of the acoustic paths

The impulse response $h_c$ can be estimated from the sensor output in response to white noise being applied to the secondary control source. However, the impulse response $h_s$ cannot be directly obtained from the sensor signal.

To measure $h_s$, we can use a sound source (speaker) facing the sensor-actuator module. In response to white noise sound generated through the sound source, the sensor output is given by

$$s(n) = \vartheta(n) + h_s^T \varphi(n)$$  \hspace{1cm} (24)

where $\vartheta(n) = h_p^T \varphi(n)$, $h_p$ and $h_s$ are the impulse response vectors corresponding to the acoustic paths shown in Fig. 3. If white noise has zero-mean and unit-variance, the correlation function between $\varphi(n)$ and $s(n)$ is calculated as

$$r(n-i) = E[s(n) \varphi(n-i)] = h_{p,i} + h_{p,i} \otimes h_{s,i}, \quad i = 0, 1, ..., M - 1,$$  \hspace{1cm} (25)

where $\otimes$ represents the convolution operation. If the acoustic paths denoted by the impulse responses $\{h_{p,i}\}$ and $\{h_{s,i}\}$ are lossless or near lossless, the two impulse responses do not overlap in time due to the distance between the sensor and the wall surface. If so, we can visually separate the two impulse responses corresponding to $\{h_{p,i}\}$ and $\{h_{p,i} \otimes h_{s,i}\}$ from the calculated correlation function. The early part of the impulse response will be $\{h_{p,i}\}$, and we can estimate $\{h_{s,i}\}$ from the late part of the impulse response as

$$h_{s,i} = \mathcal{F}^{-1}\{\mathcal{F}\{h_{p,i}\} \otimes \mathcal{F}\{h_{s,i}\}\} / \mathcal{F}\{h_{p,i}\}, \quad i = 0, 1, ..., M - 1$$  \hspace{1cm} (26)

where $\mathcal{F}\{}$ and $\mathcal{F}^{-1}\{}$ represent Fourier transform and inverse Fourier transform, respectively.

Even if the paths are lossy, it is still possible to separately estimate the impulse responses $\{h_{p,i}\}$ and $\{h_{s,i}\}$ from the correlation function, only if the effective duration of the impulse response $\{h_{p,i}\}$ is shorter than the initial time delay of the reflective acoustic path $\{h_{s,i}\}$.
4. Simulation

4.1 Simulation setup

The simulation setup for the reflected sound control is shown in Fig 1. A one-dimensional simulation environment was assumed. Two sensors were located in front of the secondary source, and the distance between the second sensor and the secondary source was 25.5 cm which corresponds to a 6 sample time-delay at an 8 kHz sampling rate. The distance between the sensors was 4.25 cm corresponding to a 1 sample time delay. In the proposed algorithm, only the second sensor output signal was used. Acoustic paths were modelled as lossless and the reflection coefficient of the wall surface was 0.7. The impulse responses were estimated using 32-tap FIR filters prior to the on-line adaptation. The adaptive filter had 16 taps and the step-size was 0.001. The background noise was added to incident wave at a 30 dB SNR.

4.2 Simulation result

Fig. 5 shows the result of the signal estimation, when a 200 Hz sinusoid signal was used as an incident wave. The results clearly show that the proposed algorithm accurately estimated the signal components from the single sensor signal.

![Figure 5](image)

Figure 5. Separation results: (a) 200 Hz incident signal, (b) the estimated incident signal, (c) the reflective signal, and (d) the estimated reflective signal using the proposed method.

Fig. 6 shows the sensor signals with ANC on and off. Fig. 6-(b) shows that the reflected signal actually decreased the peak value of the sensor output due to the phase difference. But after the ANC was on, the level of the incident wave was recovered at the sensor output. The difference between the actual incident signal and the sensor signal after ANC on is very small (Fig.6-(d)) after the algorithm converged, which implies that the proposed algorithm successfully reduced the reflective signal only.
Figure 6. FxLMS performance: (a) incident signal, (b) sensor signal with ANC off: sum of the incident and reflective signals, (c) sensor signal with ANC on, and (d) the difference between actual incident signal (a) and the sensor signal with ANC on (c).

Fig. 7 illustrates the rate of convergence of the proposed algorithm being measured with $\text{E}(|u(n) - s(n)|^2)$, and it is compared with the result of the beamforming-based algorithm in [4]. As can be seen, the proposed algorithm converged lower mean-square-error than the algorithm in [4] because the beamforming accuracy was not high enough to prevent erroneous estimation of the signal components, especially at low-frequencies.

Figure 7. The mean square errors of the beamforming-based (solid) and the proposed single-sensor-based (dashed) FxLMS algorithms.
5. Conclusion

In this paper, an effective single-sensor-based active reflective sound control algorithm was presented. The proposed algorithm estimates the incident and reflective signals directly from the sensor signal, and the control filter is updated using the filtered-x LMS algorithm. We also proposed a method of measuring the acoustic impulse responses in a single-sensor situation. Through the simulation result, we showed that the proposed algorithm can accurately estimate the signal components and it can reduce the reflective sound efficiently. Thus, the proposed algorithm can used to build an acoustic wall with arbitrary reflection coefficients.

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