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Directional cancellation of acoustic noise for home window applications

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ABSTRACT

This paper focuses on an active noise cancellation system for a home window using a transparent acoustic transducer. In a traditional active noise cancellation system, direct microphone measurements are used for reference and error signals. In the case of the window application, both external and internal sound would be picked up by such microphones. This leads to adverse effects on the performance of the active noise cancellation system and also to distortion of the internal sound. To address this problem, a wave separation technique is proposed to separate the internal and external components of sound. The wave separation algorithm is based on the use of two microphones and an algorithm that separates components based on their direction of travel. An active noise cancellation system is implemented using wave separation for both the error and reference signal measurements. The performance of the resulting ANC system is experimentally tested in a cabin equipped with a window and results are presented. Experimental results show that the new system is able to accurately preserve desired internal sound while cancelling uncorrelated external noise.

1. Introduction

For homes close to airports and highways, windows constitute the primary path through which noise enters the home. For example, a typical frame wall may weigh 70 kg/m^2, while a typical single glazed window weighs about 7 kg/m^2. Consequently, the window can transmit roughly 10 times as much sound energy as is transmitted through the same area of wall [1]. In studying the effectiveness of various measures in improving building insulation against traffic noise, Utley et al. [2] concluded that window improvements provide the most satisfaction to home dwellers. The traditional passive method to reduce noise transmitted through windows is using sealed double-glazed windows in which two panes of glass are separated by a few millimeters of air cavity. The performance of double-glazed window is poor for low frequencies (<1 kHz), such as traffic noise [3]. The sound transmission loss of a double-glazed window is greater than 40 dB at 2 kHz but decreases to 20 dB at 100 Hz. Active noise cancellation (ANC) is a better solution for reduction of low frequency noise.

Two main approaches of actively controlling sound transmission into a room have been proposed in literature. The first approach, loudspeaker-based active noise control, uses loudspeakers as the secondary source. The loudspeakers are usually placed inside the room (room control) [3,4]. The complex acoustical field in a room makes the room control approach difficult and less effective. The other approach is active structural acoustic control (ASAC). In this approach thin panels are placed in the sound transmission path. They are used as loudspeakers and their vibration is actively controlled by vibration inputs, yielding the so-called "panel speakers" [5–7]. In this way the transmission of noise is controlled before noise enters the acoustically complex room. ASAC is used in this research because it has better potential for the window ANC application than room control.

In ANC, a secondary source creates an "anti-noise" to superimpose on the unwanted primary noise. Since the "anti-noise" has the same amplitude but a 180° phase difference with the primary noise, the superposition results in a reduced decibel level of noise in the environment. Feedforward algorithms are the most common and effective algorithms used in active noise control. In this research, the noise cancellation algorithm used to produce the "anti-noise" is a feedforward technique called the filtered-X least mean squares (FXLMS) algorithm [8,9].

A critical feature of the feedforward FXLMS algorithm is the need for a reference signal $x(n)$. The ANC system uses an upstream microphone to pick up the primary noise as a reference signal and a downstream microphone to measure the error signal (superposition of the noise and "anti-noise"). Unfortunately, the reference microphone will also pick up other sound that co-exists with the reference signal in the acoustic field, resulting in deterioration of the noise control results. For example, picking up the "anti-noise" generated by the secondary source that travels upstream will result in an adverse effect called "feedback effect" [8]. Picking up

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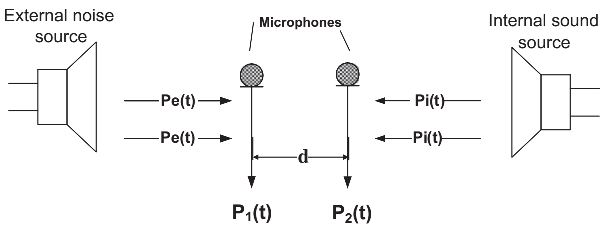

Fig. 1. The integration method for wave separation.

the primary noise reflected by the window to upstream can lead to instability during FXLMS adaptation. Picking up the internal sound (such as music and speech) radiated through the window to upstream will result in cancellation of the music and speech in homes. The error microphone placed inside the homes will also pick up the internal sound, which will cause a slow down in the convergence rate of the FXLMS algorithm. To mitigate the adverse effects caused by the "polluted" reference and error signals, this paper proposes the use of an acoustic wave separation algorithm. Wave separation can separate the sound at any point into components based on their direction of travel. It is used in this paper to separate the external noise from the internal sound picked up by the reference and the error microphones. Several wave separation algorithms have been proposed in literature. De Sanctis and van Walstijn [10] placed two microphones in a duct for wave separation. They adaptively estimated the frequency domain transfer function between the two microphones using a third microphone. Then the forward and backward going sounds were estimated using the transfer function and inverse Fast Fourier Transform. This frequency domain wave separation algorithm consumes a lot of computational power. Thus it is not suitable to be used with a real-time active noise control system. Time domain wave separation is a better option for real-time application. Kemp et al. [11] placed M ($M > 2$) microphones in the acoustic field to estimate the intermicrophone time domain transfer functions for forward and backward direction. Then forward and backward going sound waves are estimated by time domain convolution using the appropriate transfer functions. Because of the significant delay introduced by multiple convolutions, it is possible that this method also will fail to keep up with the requirements for a real-time ANC. Thus a more time-efficient wave separation in time domain is needed for this research.

2. Methods

2.1. Wave separation algorithm

In this research, a time-domain wave separation algorithm based on the momentum equation and continuity equation was used [12]. The algorithm utilizes two microphones placed a few centimeters apart, as shown in Fig. 1 below, in order to separate the external and internal components of sound at that point.

Let the acoustic pressure signals picked up by the two microphones be p_1 and p_2. If the distance d between the microphones is small relative to the smallest wavelength of the sound, the pressure at the midpoint is approximately:

$$p = \frac{p_1 + p_2}{2} \tag{1}$$

For a plane wave, the momentum equation [12] yields

$$\rho \frac{\partial u}{\partial t} + \frac{\partial p}{\partial x} = 0 \tag{2}$$

Since the distance between the two microphones is small, the spatial derivative can be approximated by

$$\frac{\partial p}{\partial x} = \frac{p_2 - p_1}{d} \tag{3}$$

Substituting into Eq. (2), the particle velocity is calculated as

$$u(t) = \frac{1}{\rho d} \int_0^t (p_1 - p_2) d\tau \tag{4}$$

The incident wave can be expressed as [13]

$$p_i = \sum_n A_n e^{j(w_n t - k_n x)} \tag{5}$$

where the wave number k_n is related to the frequency ω_n and the speed of sound c by the relation $k_n = \omega_n/c$. Substituting into the momentum Eq. (2), the particle velocity corresponding to the incident wave is

$$u_i = \frac{1}{\rho_0 c} p_i \tag{6}$$

Similarly the particle velocity caused by the reflected wave can be obtained as

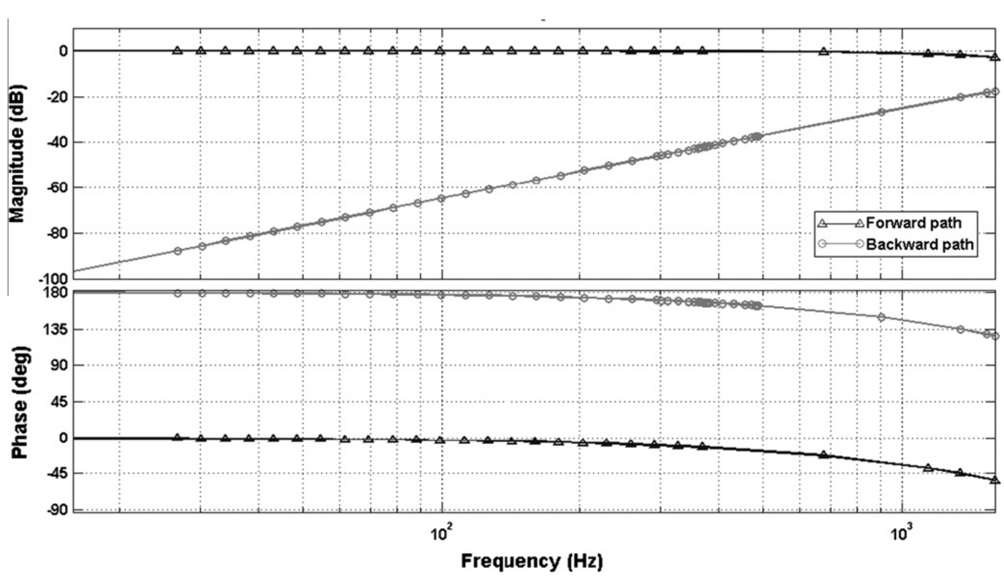

Fig. 2. The forward path and backward path transfer functions' frequency responses.