Control Engineering Practice 18 (2010) 1045–1052

Contents lists available at ScienceDirect

Control Engineering Practice

journal homepage: www.elsevier.com/locate/conengprac

Active noise control in a forced-air cooling system

R.A. de Callafon, J. Zeng, C.E. Kinney *

University of California, San Diego, Department of Mechanical and Aerospace Engineering, 9500 Gilman Drive, La Jolla, CA 92093-0411, USA

ARTICLE INFO

Article history:
Received 5 September 2008
Accepted 13 May 2010
Available online 3 June 2010

Keywords:
Active noise control
Feedforward control
Parameter estimation
Digital filters
Control applications

ABSTRACT

This paper discusses the design methodology for the active noise control of sound disturbances in a forced-air cooling system. The active sound cancellation algorithm uses the framework of output-error based optimization of a linearly parametrized filter for feedforward sound compensation to select microphone location and demonstrate the effectiveness of active noise cancellation in a small portable data projector. Successful implementation of the feedforward based active noise controller on a NEC LT170 data projector shows a 20–40 dB reduction per frequency point in the spectrum of external noise of the forced-air cooling system can be obtained over a broad frequency range from 1 to 5 kHz. A total noise reduction (unweighted) of 9.3 dB is achieved.

1. Introduction

Small electronic systems often require forced air-cooling to control the temperature of large power sensitive components in the system. Moving air through a fan provides an effective resource for cooling, but suffers from the drawbacks of noise due to turbulence and vibrations. In applications where external sound disturbances interfere with the environment, passive or active attenuation can be used to control sound emission of forced-air cooling systems (Gee & Sommerfeldt, 2003). Passive noise control is effective at reducing high frequency sound components but requires large amounts of absorption material to reduce low frequent noise signals (Bernhard, 2000; Gentry, Guigou, & Fuller, 1997; Hu & Lin, 2000). Especially for small electronic systems, this is not a viable solution.

Active noise control (ANC) can be used for sound reduction and can be particularly effective at lower frequency sound components. ANC allows for much smaller design constraints and has received attention in recent years (O'Brien, Pratt, & Downing, 2002; Sano & Terai, 2003) and many applications of active noise techniques can be found in the literature (e.g., Berkman & Bender, 1997; Cabell & Fuller, 1999; Esmailzadeh, Alasty, & Ohadi, 2002; Fuller & Von Flotow, 1995). The basic principle and idea behind ANC is to cancel sound by a controlled emission of a secondary opposite (out-of-phase) sound signal (Denenberg, 1992; Wang, Tse, & Wen, 1997).

In active noise cancellation systems with a relatively small amount of feedback from the control speaker to the pick-up microphone, called acoustic coupling, feedforward compensation provides an effective resource to create a controlled emission for sound attenuation. Algorithms based on recursive (filtered) least mean squares (LMS) minimization (Haykin, 2002) can be quite effective for the estimation and adaptation of feedforward based sound cancellation (Cartes, Ray, & Collier, 2002). To facilitate an output-error based optimization of the feedforward compensation, a linearly parametrized finite impulse response (FIR) filter has been used for the recursive estimation and adaptation.

Many other viable ANC structures exist, such as hybrid ANC (Yuan, 2004) which combines feedforward and feedback control. In this application, hybrid ANC is not practical, as an error microphone externally located on the device will pick up a significant amount of noise, limiting the ANC to feedforward. In the current work, the error microphone is used to evaluate the reference or pick-up microphone location only. The final noise control system uses one reference microphone and one control speaker. This was done to keep the number of sensors low and the signal-to-noise ratio high, which is desirable for product development.

In this paper the framework of output-error based optimization of a linearly parametrized filter for feedforward sound compensation is used to demonstrate the effectiveness of active noise cancellation in a small portable data projector depicted in Fig. 1. The projector in Fig. 1 is equipped with a shielded internal directional pick-up microphone to measure the sound created by the forced-air cooling of the projector's light bulb. Non-invasive small directional speakers located at the inlet and outlet grill of the data projector are used to minimize acoustic coupling and complete the active noise control system.

* Corresponding author. Tel.: +1 858 822 1763; fax: +1 858 822 3107.
E-mail addresses: callafon@ucsd.edu (R.A. de Callafon), jzeng@ucsd.edu (J. Zeng), cekinney@gmail.com (C.E. Kinney).
URL: http://www.mae.ucsd.edu/research/callafon/ (R.A. de Callafon).

0967-0661/$ - see front matter © 2010 Elsevier Ltd. All rights reserved.
doi:10.1016/j.conengprac.2010.05.007

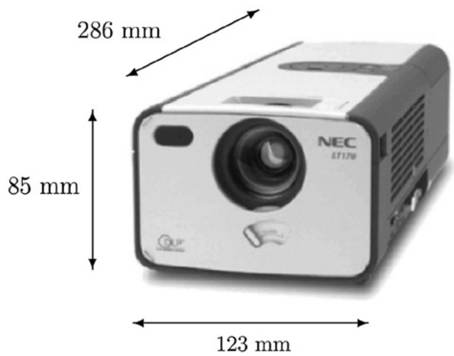

Fig. 1. Dimensions of NEC LT170 data projector with inlet side grill for forced-air cooling system.

The feedforward control algorithm for active noise control presented in this paper is based on the estimation of a generalized finite impulse response filter (Zeng & de Callafon, 2003). Generalized or orthogonal FIR models have been proposed in Heuberger, Van Den Hof, and Bosgra (1995) and exhibit the same linear parametrization as a standard FIR filter. Combined with an affine optimization of the filter coefficients, an optimal feedforward compensator for noise cancellation can be obtained for the data projector. Implementation of the feedforward based ANC on the data projector shows a 20–40 dB reduction of external noise of the forced-air cooling system over the frequency range from 1 to 5 kHz.

2. Active noise control

2.1. Analysis of feedforward compensation

In a scalar feedforward noise compensation, an (amplified) signal $u(t)$ from an internal pick-up microphone is fed into a feedforward compensator F that controls the signal $u_c(t)$ to a control speaker for sound compensation. In order to analyze the design of a feedforward compensator F for active noise cancellation, consider the signal $e(t)$ that reflects the combined effect of sound due to external disturbance and control speaker.

The objective of the ANC system is to minimize the signal $e(t)$ and an error microphone can be used to measure $e(t)$ and monitor the performance of the ANC system. The dynamical relationship between the discrete time sampled signals in the ANC system can be characterized by difference equations, where the operator q is used to denote a unit sample delay $qu(t) = u(t+1)$. The dynamic relationship between the control input $u_c(t)$, the sound disturbance $u(t)$, and the error signal $e(t)$ is shown in Fig. 2. From this figure, it is clear that the error signal can be characterized by

$$e(t) = H(q)u(t) + G(q)u_c(t) \tag{1}$$

where $H(q)$ is a stable filter in the 'primary path' and $G(q)$ is a stable filter in the 'secondary path' of the ANC system. $H(q)$ and $G(q)$ characterize the discrete time dynamic aspects of the sound propagation from the sound disturbance and control input, respectively, to the error signal $e(t)$.

The sound disturbance $u(t)$ of the forced-air cooling system measured at the pick-up microphone is characterized by

$$u(t) = W(q)n(t)$$

where $n(t)$ is a zero-mean white noise signal with variance $E\{n(t)^2\} = \lambda$ and $W(q)$ is a (unknown) stable and stably invertible noise filter. The combination of a zero-mean and filtered white noise signal $u(t)$ allows for a characterization of a rich class

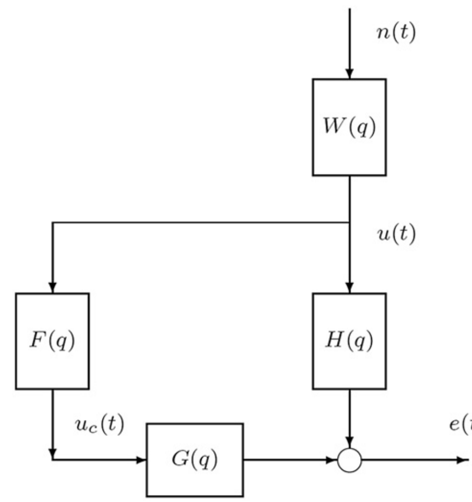

Fig. 2. Block diagram of the dynamic relationship between the control input $u_c(t)$, the sound disturbance $u(t)$, and the error signal $e(t)$.

of random sound disturbances for which the discrete time spectrum $\lambda|W(e^{j\omega})|^2$ can be modeled by a spectral decomposition (Ljung, 1999). In case the sound disturbance $u(t)$ itself is not influenced by the feedforward sound compensation $u_c(t) = F(q)u(t)$, the performance signal $e(t)$ can easily be described by

$$e(t) = W(q)[H(q) + G(q)F(q)]n(t) \tag{2}$$

and is a stable map, provided the feedforward compensator $F(q)$ is stable. Absence of a correlation between $u_c(t)$ and $u(t)$ given by $u(t) = G_c(q)u_c(t)$ requires a well-designed ANC system that minimizes the acoustic coupling[1] $G_c(q)$. For the ANC application of the data projector discussed in this paper, acoustic coupling is minimized by small directional speakers located at the outside of the data projector. As a result, the effect of acoustic coupling is not considered in the remaining part of the paper. However, if acoustic coupling is present then the method presented in de Callafon and Zeng (2006) can be used to eliminate the feedback.

In case the transfer functions in (2) are known, an ideal feedforward compensator $F(q) = F_i(q)$ can be obtained in case

$$F_i(q) = -\frac{H(q)}{G(q)} \tag{3}$$

is a stable and causal transfer function. The solution of $F_i(q)$ in (3) assumes full knowledge of $G(q)$ and $H(q)$. Moreover, the filter $F_i(q)$ may not be a causal or stable filter due to the dynamics of $G(q)$ and $H(q)$ that dictate the solution of the feedforward compensator $F_i(q)$. An approximation of the feedforward filter $F_i(q)$ can be made by an output-error based optimization that aims at finding the best causal and stable approximation $F(q)$ of the ideal feedforward compensator in $F_i(q)$ in (3).

The output-error based approximation can be characterized by examining the variance of the discrete time error signal $e(t)$ in (2) that is given by

$$\frac{\lambda}{2\pi}\int_{-\pi}^{\pi} |W(e^{j\omega})|^2 |H(e^{j\omega}) + G(e^{j\omega})F(e^{j\omega})|^2 \, d\omega$$

where λ denotes the variance of $n(t)$. In case variance minimization of the error microphone signal $e(t)$ is required for ANC, the

[1] Acoustic coupling is a feedback term from the control speaker to the pick-up microphone that can cause poor performance and instabilities if present and not accounted for in the design of the feedforward compensator.