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Modified-filtered-u LMS algorithm for active noise control and its application to a short acoustic duct

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ABSTRACT

This paper presents a new adaptive algorithm for active noise control (ANC) that can be effectively applicable to a short acoustic duct, such as the intake system of an automobile engine, where the stability and fast convergence of the ANC system is particularly important. The new algorithm, called the modified-filtered-u LMS algorithm (MFU-LMS), is developed based on the recursive filtered-u LMS algorithm (FU-LMS) incorporating the simple hyper-stable adaptive recursive filter (SHARF) to ensure the control stability and the variable step size to enhance the convergence rate. The MFU-LMS algorithm is implemented by purely experimental ways, and is applied to active control of noise in a short acoustic duct, and is validated using two experimental cases of which the primary noise sources are a sinusoidal signal embedded in white noise and a chirp signal. The experimental results demonstrate that the proposed MFU-LMS algorithm gives a considerably better performance than other conventional algorithms, such as the filtered-x LMS (FX-LMS) and the FU-LMS algorithms.

1. Introduction

The design of an appropriate adaptive algorithm is the key element for an active noise control system. In general, the purpose of the adaptive algorithm may be considered to be the system identification of a dynamic plant, as depicted in Fig. 1. For this approach to the system identification, the Least Mean Square (LMS) algorithm has been widely used because of its robustness and simplicity. The dynamic plant is thus usually modeled by the LMS algorithm with an FIR filter, which has a large number of all-zero-coefficients, or with an IIR filter which has a small number of pole-zero-coefficients [1].

In this paper, we consider the active noise control (ANC) system for a relatively short acoustic duct, such as the intake or exhaust system of an automobile engine. In such a case, the convergence time of the ANC system becomes especially important. If the dynamic plant is modeled based on the FIR filter structure, it requires a long convergence time due to the large number of filter coefficients. Also, for the short acoustic duct, the canceling loudspeaker is inevitably very close to the reference microphone, because of the geometrical restriction. This causes a high acoustic feedback in the ANC system, which contaminates the reference input signal. To resolve these issues, the recursive LMS algorithm based on the IIR filter is usually required [2,3]. Thus, the dynamic plant needs modeling by an adaptive IIR filter structure [3], with its ability to directly model the transfer functions with poles $B(z)$ and zeros $A(z)$, as depicted in Fig. 2.

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