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Development of adaptive algorithm for active sound quality control

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Abstract

This paper develops an active sound-quality control (ASQC) system based on the active noise equalization (ANE) technique, and optimizes it with the filtered-error least mean square (FELMS) algorithm and normalized reference signal generator. The ASQC system controls the sound quality of products such as engines by changing the amplitudes of harmonics. This optimized system uses the FELMS algorithm to limit disturbances in the passband caused by uncorrelated interferences with high gains in the secondary path, thereby increasing the system stability. It achieves fast convergence by normalizing the amplitudes of internally generated sinusoids in the reference signal according to the magnitude response of the secondary path at the corresponding frequencies. Computer simulations demonstrate the desired spectral shaping capability with faster convergence and reduced passband disturbance.

1. Introduction

Active noise control (ANC) [1,2] uses the principle of superposition, whereby a secondary noise of equal amplitude and opposite phase cancels an unwanted primary noise. In many practical ANC applications, the primary noise produced by rotating machines (such as engines) is periodic and contains multiple harmonic-related narrowband components. In the periodic ANC system, a nonacoustic sensor such as a tachometer or an accelerometer [3] can replace the reference microphone. The sensor output synchronizes an internally generated reference signal, thus preventing feedback from the secondary source to the reference sensor. This periodic ANC system was analyzed in Ref. [4] using the filtered-X least mean square (FXLMS) algorithm [5], where the reference signal is filtered by an estimate of the secondary path for updating the adaptive filter's coefficients.

The design of an ANC system usually pursues maximal attenuation of the primary noise. However, in some applications, it is desirable to retain a small residual noise with a specified spectral shape. For example, in an automobile, or earth-moving machine, the driver needs some audible information about engine speed to be able to control the vehicle safely. Active sound-quality control is another potential use of ANC, which changes

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Nomenclature

ASQC	active sound quality control
ANE	active noise equalization
FELMS	filtered-error least mean square
$S(z)$	secondary Path
$\hat{S}(z)$	estimate of secondary path
$W(z)$	output of the adaptive filter
β	gain
$e(n)$	residual noise
$e'(n)$	pseudo error
$x(n)$	input (single-frequency sine wave)
μ	step size
$d(n)$	primary noise
$H(z)$	closed-loop transfer function
$u(n)$	sinusoidal components
$v(n)$	uncorrelated components
$s(n)$	impulse response
*	linear convolution
C	amplitude of reference signal
ϕ	initial phase
ψ	phase shift
A	gain
$C(z)$	adaptive filter
$e_c(n)$	instantaneous squared error signal
$H_c(z)$	transfer function of adaptive comb filter
r	pole radius
S_k	gain of $S(z)$ at frequency ω_k
R	input autocorrelation matrix
λ_{2K}	smallest nonzero eigenvalue of the matrix **R**
$A_k = 1/S_k$	amplitude at ω_k

(amplifies or attenuates with predetermined values) the amplitudes of noise components to improve sound quality or change noise signature. These demands lead to an extension of the ANC concept to include ASQC.

The equalization system for periodic noise with multiple harmonics is called a narrowband ANE, which has been developed in Refs. [6,7]. This algorithm also has been implemented and analyzed in the frequency domain [8], and extended to equalize broadband noise [9]. The fundamental difference between the narrowband ANC and ANE is the system transfer functions. The magnitude response of the narrowband ANC system presents multiple notches with infinite nulls at controlled frequencies, so that the system attenuates those noise components completely. However, the depths of individual notches of the narrowband ANE system are independently adjustable and can even be changed to peaks with predetermined amplitudes without affecting the characteristics of other notches.

Active sound-quality control using the narrowband ANE algorithm inherits the problems of ANC systems with the FXLMS algorithm. These include passband disturbances due to uncorrelated interference at frequencies where the magnitude response of the secondary path has high gain [10], and slow convergence due to the eigenvalue spread of the input autocorrelation matrix determined by the magnitude response of the secondary path [11]. This paper modifies the solutions [10,11] for solving these two critical problems in ANC systems, integrates them with the narrowband ANE [6,7], and presents the optimized ASQC algorithm [12].

2. Narrowband active sound-quality control

A block diagram of the general narrowband ANE system for controlling periodic noise is shown in Fig. 1, where $\hat{S}(z)$ is an estimate of $S(z)$, which is the secondary path from the output of the adaptive filter $W(z)$ to the output of error microphone. The output $y(n)$ of the two-weight filter is split into two branches, the canceling branch and the balancing branch. The gains β and $(1-\beta)$ are inserted in these two branches to adjust the residual noise. Instead of minimizing the residual noise $e(n)$ measured by the error microphone, the ANE system minimizes the computed pseudo error $e'(n)$.

For analysis purpose, we assume a single-frequency sine wave as the input $x(n)$, although this system can be extended to equalize multiple sine waves by connecting several adaptive filters in parallel. For controlling a single-frequency sinusoidal noise, a pure sinewave is internally generated as the reference signal, and the adaptive filter contains two coefficients. This adaptive filter minimizes the instantaneous squared pseudo-error signal using the least mean square (LMS) algorithm [13] expressed as

$$w_i(n+1) = w_i(n) + 2\mu e'(n)x_i(n), \quad i = 1, 2, \tag{1}$$