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An adaptive algorithm for nonstationary active sound-profiling

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ABSTRACT

The use of active sound profiling (ASP) in the automobile industry has been under investigation for several years, and the applications have taken advantage of such techniques, balancing amplitudes instead of simply minimizing the sound pressure level (SPL). This paper presents a novel adaptive algorithm to profile nonstationary disturbances such as the noise generated by a gasoline engine. The new algorithm provides profiling capabilities for nonstationary disturbances and stability properties of the system, whilst expending minimum control effort. Mainly assisted through the use of a reshaping signal, necessary phase information is extracted from nonstationary disturbance signals. To deal with changes in sound as the operating conditions of the engine change, the short-time Fourier transform (STFT) filtered-x least mean square (FXLMS) scheme is introduced to improve the convergence rate. The stability properties are based on the command FXLMS approach, which prevents instabilities caused by magnitude errors in the estimated plant model. Moreover, modification of the STFT-FXLMS scheme improves stability and performance when phase error does exist. In this paper, the performance of the proposed algorithm is demonstrated through a series of simulations configured with either simulated noises or noises from a real engine. The results revealed the effectiveness of the proposed algorithm in profiling nonstationary harmonic noise, and enhancement of stability and performance due to the modification of the STFT-FXLMS scheme.

1. Introduction

The interior sound quality of a vehicle is becoming an important indicator for customer assessing driving experience [1–6]. In particular, engine related noise is of major importance in driving and riding perception, which involves vehicle sound appreciation [7,8]. Human perception is considered to expect that the sound mirror the vehicle running status linearly rather than be cancelled. Therefore, designing for interior sound comfort focuses on not only being quiet, but achieving an individual desired level of expected noise. The active noise control (ANC) system provides the possibility of dealing with this problem of controlling the noise by the action of a secondary noise depending on an introduced counter-noise source [9–13].

Conventional ANC systems aim to attenuate the sound pressure level as much as possible; however, human perceptions of sound in vehicle invoke the task of cancelling some unwanted noise at selected frequencies, while enhancing noise at other frequencies, or even controlling noise to a predetermined level [14–16]. For this purpose, Kuo et al. provided an active noise equalizer (ANE) that derived from the commonly used filtered-X least mean square (FXLMS) algorithm [17,18]. The main feature of the ANE algorithm is the insertion of gain factors β and $(1-\beta)$ into the split outputs of the control filter. As a result, the error noise is tuned by β, and the adaptive algorithm is fed with a pseudo error signal. When the pseudo error signal converges to zero, the real residual noise converges to a desirable value, tuned by β. Considering about the practical applications in sound field, Gonzalez et al. developed a multichannel case algorithmic variant of ANE, named common-error multiple-frequency ANE algorithm [19]. The algorithm independently controls some given frequencies noise and performs better in saving computational complexity and easier to be implemented in real controller than conventional ANE. Following on from this, Kuo et al. developed an active sound-quality control (ASQC) system based on the ANE technique [20,21]. The system was optimized using the filtered-error least mean square (FELMS) scheme and a normalized reference signal generator, which led to faster convergence rate and an increase in system stability. Then Kuo et al. presented a frequency-domain delayless ASQC structure which was efficient and provided faster convergence and reduction in computational complexity comparing with time-domain algorithms [22]. Based on this, Mosquera-Sanchez et al. introduced a frequency-domain multichannel scheme for independently controlling sound of multiple locations, and the performance of the control scheme was demonstrated by simulated and

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experimental results [23]. The cross-channel interferences of controlling units were addressed by using compensation schemes that can also assure implementation stability when the sensors/actuators were found as defective at given frequencies, but at the expense of losing control performance. Inspired by the research of Kuo et al., Oliveira et al. presented a NEX-LMS scheme ASQC algorithm for the equalization of engine noise based on the introduction of an adaptive gain factor β(ω) [8]. The performance of the NEX-LMS scheme was experimentally validated, and the controller worked effectively when the frequency was slowly run up (33 Hz to 100 Hz, > 15 s). All these aforementioned ANE-based ASQC algorithms equalize sound with the use of a gain factor; however, they are extremely sensitive to misestimation of the secondary path plant model at high system gains [24]. To weaken the influence of secondary model misestimation, Rees et al. proposed an alternative active sound profiling (ASP) scheme, which makes residual noise to converge to a created command value signal, instead of being tuned by gain factors [24]. The command value signal is created from the reference signal, which ensures that the frequency of command signal is same with frequency of the primary disturbance signal. Similar to the ANE algorithm proposed by Kuo et al., a pseudo error signal is used to update the adaptive filter weights, and the real residual noise signal converges to the command signal as the pseudo error converges to zero. Then, by setting the command signal in phase with the primary disturbance signal, the phase scheduled command-FXLMS (PSC-FXLMS) sound profiling algorithm can be derived, which significantly limits the required control effort. This algorithm is highly robust to plant model amplitude error, but is also sensitive to phase error of plant model. To deal with phase instability, an automatic phase command technique was incorporated into PSC-FXLMS; however, the modification results in an increase in control effort when the command signal amplitude is greater than the disturbance amplitude. To address this, a modified phase-scheduled-command FXLMS (MPSC-FXLMS) algorithm was proposed, which an intelligent adaptive hysteresis switching scheme was used to directly detect the phase of the disturbance signal instead of extract the phase from the estimated disturbance [25]. The MPSC-FXLMS algorithm shown robustness against both amplitude and phase errors of the plant model. While the ASP algorithms in [24,25] are always effective for stationary noise conditions, they perform badly and may even lead to instability when the disturbance is nonstationary or frequency modulation rate (chirp rate) signal. These algorithms use fast Fourier transform (FFT) to estimate the phase of the estimated disturbance signals, but FFT is unable to extract phase information when the disturbance signal is nonstationary. Therefore, it is difficult to use the ASP algorithms to control nonstationary disturbances.

In this paper, a novel adaptive algorithm for nonstationary noise profiling is proposed. The algorithm inherits the advantages of the PSC-FXLMS algorithm, which is characterized by minimum control effort and insensitivity to the magnitude error of plant model. The proposed algorithm applies a reshaping signal to provide a feasible estimate of the phase information of nonstationary disturbance signal, and minimum control effort can be achieved by setting the phase of the command signal to the same as the estimated phase of the disturbance signal. The time-frequency-domain FXLMS [26], based on the short-time Fourier transform (STFT), is introduced, as the STFT has the advantage of dealing nonstationary signals than FFT. Therefore, the STFT-FXLMS structure shows efficiency both in controlling stationary or nonstationary noise, but frequency-domain structures are merely effective in tailoring stationary (or very slow changing) sound. The STFT structure has higher computational cost than that of frequency-domain structure as using data windows, but it substantially saves computations than time-domain algorithms by replacing linear convolution to multiplication. Furthermore, the STFT-FXLMS shows relatively faster convergence than time-domain algorithms [26]. In addition, a modification of the STFT-FXLMS scheme is proposed, to mitigate the influence of phase error of the secondary path estimation and/or out-of-phase between the command signal and disturbance signal.

The rest of this paper is organized as follows. The development of the proposed algorithm is introduced in Section 2, and the modification of STFT-FXLMS based control scheme. Section 3 presents simulations for three disturbance cases, accompanied by results and analyses. Finally, some conclusions are provided in Section 4.

2. Proposed ASP algorithm

Engine sound in a vehicle is related to the revolving speed of the engine. In other words, the engine sound varies depending on the engine operating conditions; for example, the sound frequency of the engine increases or decreases as the engine revolving speed increases or decreases. In practice, this nonstationary type of engine noise is pervasive and difficult to control. In this paper, the aim of the proposed algorithm is to achieve a prescribed profile for this nonstationary sound in terms of sound order-level vs. revolving speed of the engine.

A schematic of the proposed adaptive algorithm scheme is shown in Fig. 1(a). The algorithm is based on the conventional feedforward FXLMS approach. The command value signal c(n) is created as the target signal for the residual error signal e(n) to converge to. Setting the command signal in phase with the disturbance signal achieves the goal of minimum control effort, which is of great importance and significance in practice. When the pseudo error e'(n) converges to zero, the residual error signal converges to the command signal, i.e., the residual

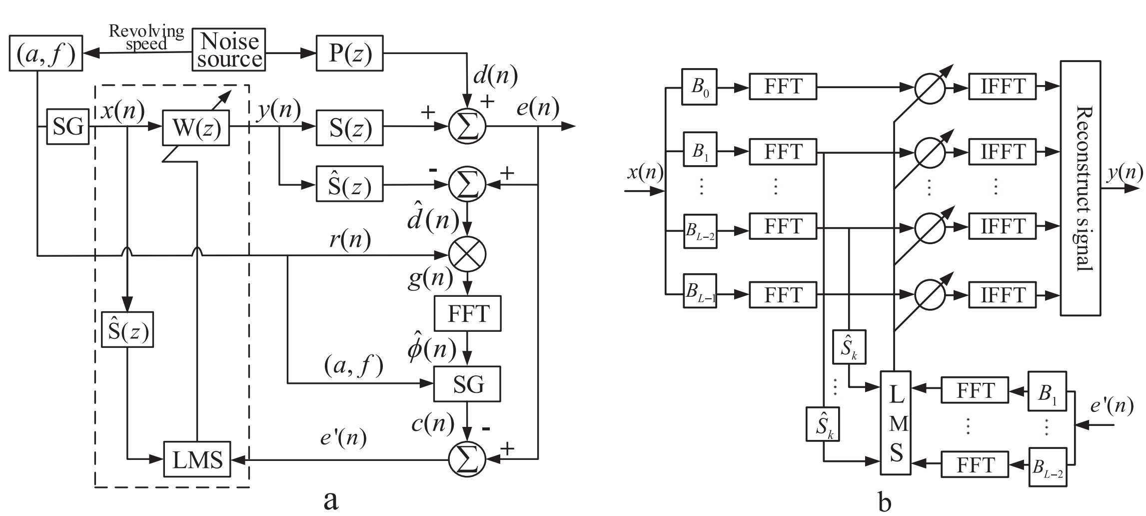

Fig. 1. (a) Block diagram of proposed algorithm, (b) STFT-FXLMS scheme.