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NEX-LMS: A novel adaptive control scheme for harmonic sound quality control

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ABSTRACT

This paper presents a novel adaptive control scheme, with improved convergence rate, for the equalization of harmonic disturbances such as engine noise. First, modifications for improving convergence speed of the standard filtered-X LMS control are described. Equalization capabilities are then implemented, allowing the independent tuning of harmonics. Eventually, by providing the desired order vs. engine speed profiles, the pursued sound quality attributes can be achieved. The proposed control scheme is first demonstrated with a simple secondary path model and, then, experimentally validated with the aid of a vehicle mockup which is excited with engine noise. The engine excitation is provided by a real-time sound quality equivalent engine simulator. Stationary and transient engine excitations are used to assess the control performance. The results reveal that the proposed controller is capable of large order-level reductions (up to 30 dB) for stationary excitation, which allows a comfortable margin for equalization. The same holds for slow run-ups (>15 s) thanks to the improved convergence rate. This margin, however, gets narrower with shorter run-ups (≤ 10 s).

1. Introduction

Interior noise in a vehicle is an important element in the customer perception of the overall vehicle's quality [1–9]. The interior noise is made up of contributions from many sources: some only contribute to the overall loudness and annoyance (e.g., uncorrelated road or wind noise), while others reveal important information on the operation of the vehicle and can invoke a desired emotional response [10], as it is the case with engine noise, targeted in this paper.

The engine-related interior sound quality design is of major importance for vehicle sound branding, as it underlines its image (e.g., sportiveness, refinement, luxury, etc.) [4,11]. In this context, sound branding brings an extra motivation for the use of active control, as it would enable easy and inexpensive adaptations to local markets, of products based on global platforms, meeting distinct customer expectations with the same hardware [4]. Moreover, such control systems can be implemented without compromising other engine/vehicle design attributes, e.g., emissions, fuel consumption, exterior noise, etc. [11]. These two aspects, easy local-marked adaptation and decoupled design parameters, would simplify the NVH (noise vibration and harshness) development process and make it a potential field of application for active control.

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Demonstrations of the viability of active control in cavity noise applications, including automotive interior noise reduction, have been described by several authors in the past few years [12–19].

Usually, the objective of these applications is to reduce the noise generated by the primary source(s) as much as possible. The novelty in this framework is to account for the human perception when defining performance criteria, either to evaluate or to drive the design of active solutions [11,20–23]. In this context, sound quality specialists could prescribe the ideal (or the brand signature) engine sound in terms of its order-level vs. RPM (revolutions per minute) profiles, which would be achieved by means of active sound quality control (ASQC).

This paper focuses on the use of ASQC to match prescribed engine order-level vs. RPM profiles. The control strategy is based on the feedforward filtered x-LMS algorithm (Fx-LMS) [24–27], to which modifications are proposed in order to improve the convergence rate and allow order-level equalization. This control strategy is described in Section 2. The test setup and the real-time implementation of the controller are described in Section 3. Section 4 presents the experimental results. Finally, some general conclusions are addressed in Section 5.

2. Novel adaptive algorithm for sound quality control: NEX-LMS

The aim of the proposed control scheme is to achieve a pre-defined order-level vs. RPM profile, thus achieving a desired sound quality target in an authentic ASQC manner. Besides converging to some pre-defined amplitudes, the controller must cope with varying engine speeds. Therefore, the controller has to be capable of tracking changes on the disturbance, converging as fast as possible to the desired output value. To this end, an adaptive feedforward strategy is proposed, which is a modified version of the standard Fx-LMS algorithm. The first modifications aim at improving the convergence rate of the adaptive scheme, in order to cope with changing RPMs during normal engine operation. Afterwards, equalization capabilities are implemented to allow the tuning of order-levels according to the RPM. In this section, each component of the proposed ASQC scheme is described, starting from the standard adaptation algorithm and the implemented improvements. Finally, the equalization feature is addressed.

The core of the control scheme is an adaptive digital filter. One of its simplest representations is depicted in Fig. 1(a). It consists of two distinct elements: an adaptive algorithm and a digital filter (W). The former adjusts the coefficients of the latter, in this case, a finite impulse response (FIR) filter as in Eq. (1), to perform the desired signal processing [27]:

$$\mathbf{w}(n) \equiv [w_0(n) \;\; w_1(n) \;\; \cdots \;\; w_{(L-1)}(n)]^T \tag{1}$$

where $\mathbf{w}(n)$ is the vectorial representation at the instant n of the filter W with L coefficients w.

Alternatively, infinite impulse response (IIR) filters can be used, as they might require less elements than FIR filters. On the other hand, although methods to ensure IIR filter stability are available, stability of FIR filters is guaranteed. Also, the performance surface of FIR implementations is quadratic, guaranteeing conversion to the global optimum, which does not hold for IIR implementations [28].

The most commonly used performance criteria for digital filter adaptation are those based on mean square error (MSE). The minimization of the MSE, $\xi(n) \equiv E[e^2(n)]$, is usually achieved by a steepest-decent method. Eventually, this results in the well-known least mean square (LMS) algorithm (Fig. 1 a) for updating the coefficients of W:

$$\mathbf{w}(n+1) = \mathbf{w}(n) + \mu \mathbf{x}(n) e(n) \tag{2}$$

where μ is the convergence coefficient, $\mathbf{x}(n)$ the reference signal and $e(n)$ the instantaneous error, i.e., the difference between the disturbance signal $d(n)$ and the controller output signal $y(n)$. In active control of noise, $d(n)$ is usually the unwanted outcome of the process, e.g., the noise coming from the engine (primary source) through the primary path.

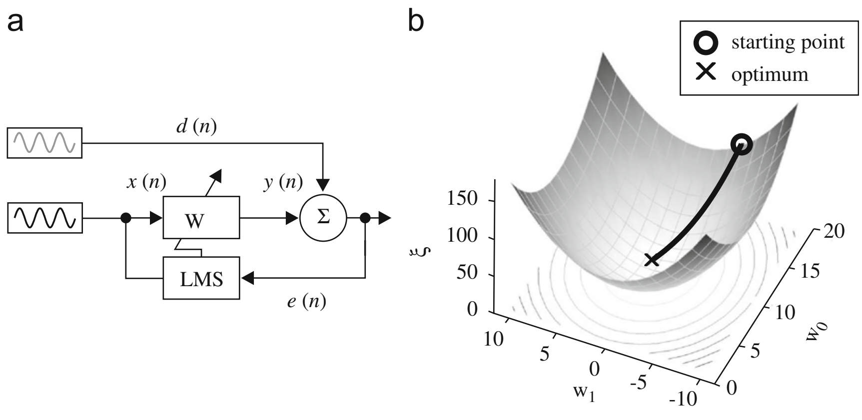

Fig. 1. LMS adaptive system: (a) block diagram and (b) performance surface with $L=2$.