Comparison of Crosstalk Cancellation Filtering Algorithms in Vehicle Audio Systems

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Abstract: Two adaptive filtering algorithms for acoustic crosstalk cancellation in vehicle audio systems are developed and compared in the present study. Acoustic crosstalk always occurs in a multiple-loudspeaker system when sound transmits from the right loudspeaker to left ear and vice versa. The purpose of crosstalk cancellation system is to deliver the desired signals exactly at the listener's ears and eliminate undesired sound. Most of the conventional algorithms for acoustic crosstalk cancellation in an audio system involve an adaptive filter with the least-mean-square (LMS) error algorithm. However, convergence speed and performance are often limited when the audio source varies. In the present study, an adaptive variable step-size least-mean-square (VSS LMS) error algorithm and an adaptive Kalman filtering algorithm are proposed for improving the crosstalk cancellation performance in vehicle audio systems. The comparison in performance and analysis of the proposed algorithms and traditional LMS error algorithm are also described.

Keywords: Acoustic crosstalk cancellation; adaptive filter; vehicle audio system.

I. INTRODUCTION

With the rapid growth of signal processing technology, sound reproduction techniques are employed to improve sound quality in audio systems. These techniques include crosstalk cancellation, virtual source synthesis and the reproduction of three-dimensional sound field that attempt to reproduce precisely sound signals at the listener's ears. The direct method for realizing perfect sound reproduction makes use of inverse filters to compensate for the responses of the loudspeakers. The primary objective of the sound reproduction system is to reduce frequency response distortion and undesirable acoustic crosstalk effect.

Research interest in acoustic crosstalk cancellation for a two-loudspeaker system has developed since Atal filed his patent in 1960, as shown in Fig. 1 [1], in which and are the left and right channel sound signals, respectively. The transfer functions are the head-related transfer functions (HRTF's) that represent the transmission paths from the loudspeakers to the ears of the listener. In order to achieve identical reproduction of signals and at the left and right ears, respectively, it is necessary to design inverse filters to convolve with HRTF's for compensating for the responses of the loudspeakers. In 1988, Miyoshi and Kaneda proposed inverse filtering techniques for room acoustics [2]. In 1992, Nelson et al. proposed an adaptive LMS error algorithm for designing inverse filters of crosstalk cancellation network [3]. Meanwhile, a fast deconvolution method was used for multi-channel deconvolution [4]. In 2001, Gonzalez et al. used a Kalman filtering technique for the inverse filter design in a sound reproduction system [5]. A robustness analysis of crosstalk cancellation and effect of loudspeaker positions were presented by Ward and Elko [6, 7]. The immersive audio system was also designed to implement a head-related transfer function (HRTF) filter for loudspeakers [8, 9]. Except for an audio system, a weighted least-mean-square error method was also proposed for calculating the inverse functions with proper delays of binaural room impulse responses [10].

In the present study, the effect of acoustic crosstalk is considered for designing the inverse filter in audio system inside the vehicle cabin. When vehicle loudspeakers play audio signals from audio sources, the driver's ears may receive crosstalk signals from the opposite loudspeaker. However, vehicle loudspeaker systems deliver the binaural sound signals to each ear of the driver. Hence, it is necessary to eliminate the crosstalk effects. Such effect exists in all loudspeaker-based systems. While each ear of the listener receives the desired sound from the same-side loudspeaker, it also receives the unwanted sound from the opposite-side loudspeaker. The sound analysis within a vehicle cabin is aimed at the driver in a seated position, which is an asymmetric location relative to the vehicle loudspeaker system. This situation will produce the time and level differences for drivers in sound localization. However, the most popular approach to overcoming this problem is to obtain the transfer functions of each listener's ear for designing adaptive inverse filters to compensate the responses of each ear. Two algorithms in inverse filter design for crosstalk cancellation are developed and examined in this paper. Apart from the adaptive filter with the LMS error algorithm, a VSS LMS error algorithm and a Kalman filtering algorithm for improving convergence speed and good performance are compared in the present study. The principles of the proposed filtering algorithms are described in the following section.

II. ADAPTIVE ALGORITHMS OF INVERSE FILTER DESIGN

A. VSS LMS Error Algorithm for Crosstalk Cancellation

A general block diagram of crosstalk cancellation system that includes an adaptive filter using a VSS LMS error algorithm is shown in Fig. 2 [3, 11]. As illustrated, the signals $x_1(n)$ and

$x_2(n)$ are programmed sound source from the audio system. The transfer functions $C_{ij}(z)$ represent the transmission paths from the loudspeakers to the microphone. For ideal crosstalk cancellation, the loudspeakers are used to deliver desired signals to each of the driver's ear. In order to achieve that, each input signal is filtered through a set of filters denoted by $W_{ij}(z)$. The system can be described as a two-input-two-output system

$$\begin{bmatrix} d_1 \\ d_2 \end{bmatrix} = \begin{bmatrix} C_{11} & C_{12} \\ C_{21} & C_{22} \end{bmatrix} \cdot \begin{bmatrix} W_{11} & W_{12} \\ W_{21} & W_{22} \end{bmatrix} \cdot \begin{bmatrix} x_1 \\ x_2 \end{bmatrix} \tag{1}$$

By expending the product, Eq. (1) can be modified as follows

$$\begin{bmatrix} d_1 \\ d_2 \end{bmatrix} = \begin{bmatrix} C_{11}W_{11} + C_{12}W_{21} & C_{11}W_{12} + C_{12}W_{22} \\ C_{21}W_{11} + C_{22}W_{21} & C_{21}W_{12} + C_{22}W_{22} \end{bmatrix} \cdot \begin{bmatrix} x_1 \\ x_2 \end{bmatrix} \tag{2}$$

Equation (2) can then be written as follows

$$\begin{bmatrix} d_1 \\ d_2 \end{bmatrix} = \begin{bmatrix} C_{11}x_1 & C_{12}x_1 & C_{11}x_2 & C_{12}x_2 \\ C_{21}x_1 & C_{22}x_2 & C_{21}x_2 & C_{22}x_2 \end{bmatrix} \cdot \begin{bmatrix} W_{11} \\ W_{21} \\ W_{12} \\ W_{22} \end{bmatrix} \tag{3}$$

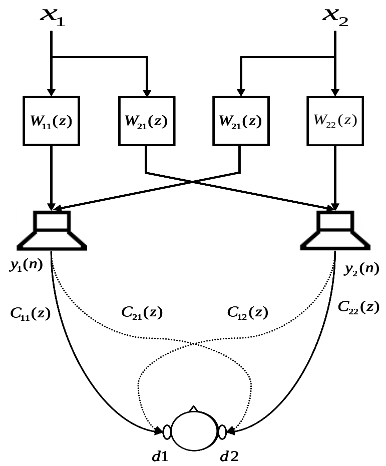


Fig. 1. Crosstalk cancellation system using the inverse filtering method.

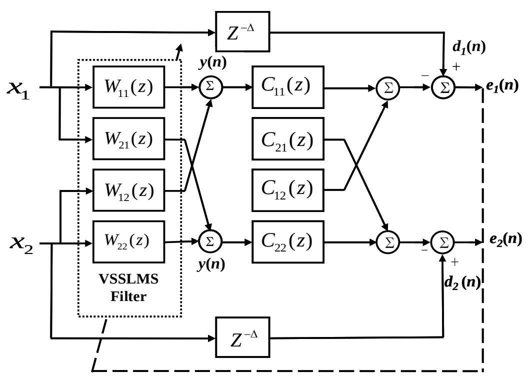


Fig. 2. Structure of crosstalk cancellation system using the VSS LMS error algorithm.

For a traditional LMS error algorithm, the tap weight equations are updated as follows

$$\mathbf{w}(n+1) = \mathbf{w}(n) + \mu \mathbf{R}^{\mathrm{T}}(n) e(n), \tag{4}$$

where $\mathbf{w}(n)$ is the coefficient vector at time n, μ is the convergence coefficient, $\mathbf{R}(n) = \mathbf{C}_{ij}(n) x_k(n)$, $e(n)$ is the adaptation error defined by

$$e(n) = d(n) - \mathbf{R}(n)\mathbf{w}. \tag{5}$$

Here, the convergence coefficient μ is chosen as a constant. Alternatively, in the proposed VSS LMS error algorithms, the step size is updated by

$$\mu(n+1) = \rho\mu(n) + \gamma e^2(n) \tag{6}$$

where $0 < \rho < 1,\ \gamma > 0$, and $\mu(n+1)$ is set to be $\mu_{\max}$ or $\mu_{\min}$ when it falls below or above these lower and upper bounds, respectively. Unfortunately , the

$$\mu(n+1) = \rho\,\mu(n) + \gamma\, p(n)^2, \tag{7}$$

where

$$p(n) = \beta\, p(n-1) + (1-\beta)\, e(n) + e(n-1). \tag{8}$$

The limitations of $\rho,\ \gamma$ are the same as those of VSS LMS: β is an integer and $0 < \beta < 1$. The restrictive condition of the modified algorithm is the assumption that the state error $e(n)$ is uncorrelated and, hence, results in a small value for the step-size after convergence.

B. *Kalman Filtering Algorithm for Crosstalk Cancellation*

In this section, an adaptive inverse filtering technique with the Kalman filtering algorithm is presented. Consider the crosstalk cancellation structure as shown in Fig. 1, assume that each FIR filter $C_{ij}(n)$ has M coefficients and each inverse filter has K coefficients. In this case, we first consider the left signal in order to simplify the complexity of the computation. The system can be expressed in matrix form as

$$\begin{bmatrix} d_1 \\ d_2 \end{bmatrix} = \begin{bmatrix} C_{11} & C_{12} \\ C_{21} & C_{22} \end{bmatrix} \begin{bmatrix} W_{11} \\ W_{21} \end{bmatrix} \tag{9}$$

where $d_1 = \left[d_1(0), \ldots, d_1(M+K-2) \right]^T$,

$d_2 = \left[d_2(0), \ldots, d_2(M+K-2) \right]^T$ represents the desired responses that have $(M+K-1)$ coefficients.

$$C_{ij} = \begin{bmatrix} C_{ij}(0) & & 0 \\ \vdots & \ddots & C_{ij}(0) \\ C_{ij}(M-1) & \ddots & \vdots \\ 0 & & C_{ij}(M-1) \end{bmatrix}$$ is a $(M+K-1) \times K$ matrix,

$W_{11} = \left[w_{11}(0), \ldots, w_{11}(K-1) \right]^T$ is a $K \times 1$ vector,

$W_{12} = \left[w_{12}(0), \ldots, w_{12}(K-1) \right]^T$ is a $K \times 1$ vector.

According to the theory of the Kalman filter, the inverse filtering problem can be expressed as the state space equation. The process equation is

$$s(n+1)=F(n+1,n)s(n)+v_1(n) \tag{10}$$

The measurement equation, which describes the observation vector, is

$$y(n)=D(n)s(n)+v_2(n) \tag{11}$$

The measurement equation relates the observable output of the system $y(n)$ to the state $s(n)$, as depicted in Fig. 3, in which $s(n+1)$ is the state vector, and $y(n)$ is the output vector. The terms $v_1(n)$ and $v_2(n)$ are uncorrelated zero-mean white-noise process representing the process and measurement noise, respectively. Their correlation matrix is defined by

$$E\left[v_1(n)v_1^{\mathrm{H}}(n)\right]=\begin{cases}\mathrm{Q}_1(n), & \mathrm{n}=\mathrm{k}\\ 0, & \mathrm{n}\neq\mathrm{k}\end{cases} \tag{12}$$

$$E\left[v_2(n)v_2^{H}(n)\right]=\begin{cases}\mathrm{Q}_2(n), & \mathrm{n}=\mathrm{k}\\ 0, & \mathrm{n}\neq\mathrm{k}\end{cases} \tag{13}$$

It is assumed that $s(0)$, the initial value of the state, is uncorrelated with both $v_1(n)$ and $v_2(n)$ for $n\geq 0$. The noise vector $v_1(n)$ and $v_2(n)$ are statistically independent.

$$E\left[v_1(n)v_2^{H}(n)\right]=\mathbf{0} \tag{14}$$

In order to solve the inverse filtering problem, Eq.(9) can be expressed as Eq.(11). Hence, the state vector $s(n+1)$ must be estimated recursively. The recursive Kalman filtering algorithm estimates the process state $s(n+1)$ by using the entire observed data, consisting of the observations $y(1),y(2),\ldots,y(n)$.

A block diagram representation of the Kalman filter is depicted in Fig. 4. The Kalman filtering procedure is summarized as follows [13]：

$$s(n+1|y_n)=F(n+1,n)\hat{s}(n|y_{n-1})+G(n)\alpha(n) \tag{15}$$

$$G(n)=F(n+1,n)K(n,n-1)D^{H}(n)\times\left[D(n)K(n,n-1)D^{H}(n)+Q_2(n)\right]^{-1} \tag{16}$$

$$\alpha(n)=y(n)-D(n)\hat{s}(n|y_{n-1}) \tag{17}$$

$$K(n+1,n)=F(n+1,n)K(n)F^{H}(n+1,n)+Q_1(n) \tag{18}$$

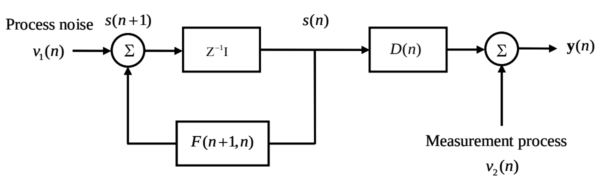


Fig. 3. Signal-flow graph representation of a linear equation system.

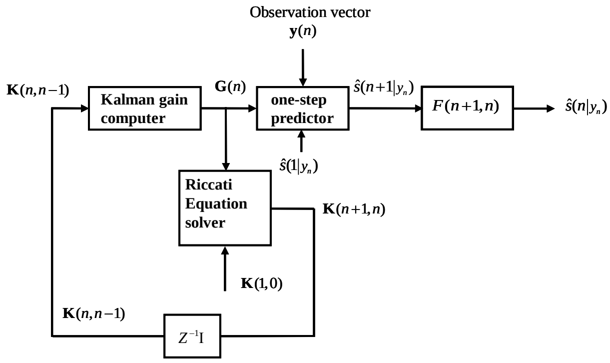


Fig. 4. Block diagram of the Kalman filter based on one-step prediction.

III. MEASUREMENT AND SIMULATION RESULTS

The HRTF in frequency domain is characterized to model the acoustical properties of the sound field. This section shows the process of measurement of the individual HRTFs. The measurement was taken inside a vehicle cabin, which was designed as an anechoic chamber having the dimensions of $1.5\times1.35\times0.65$ m, as shown in Fig. 5. Two loudspeakers were mounted at the back of the vehicle compartment, and the microphones PCB (130D20) were located in the ears of the driver in seated position. The HRTFs were measured ~~by~~ using a dynamic signal analyzer and the frequency range is set to be a 3200 Hz frequency bandwidth. The frequency response functions can be documented as finite impulse response functions shown in Fig. 6. The FIR functions are usually provided with a non-minimum phase. Therefore, the inverse of the transfer functions cannot be computed exactly.

In order to verify the proposed method, a crosstalk cancellation simulation has been conducted. First, the FIR filters $W(z)$ are calculated recursively by the proposed algorithm. For the VSS LMS error algorithm, each impulse response inverse filter $W(z)$ selected 128 samples, and a modeling delay had 64 samples. The modeling delays are used to form a stable causal inverse filter. Figure 7 shows the impulse response of the inverse filters calculated by Kalman filtering algorithm. Figures 8 and 9 illustrate the performance of crosstalk cancellation using VSS LMS error algorithm and Kalman filtering algorithm in the frequency domain, respectively. The direct path has a flat frequency response, and the crosstalk path is attenuated by more than 15 dB. Comparing the VSS LMS error and Kalman filtering algorithm used for crosstalk cancellation shows that the Kalman filtering algorithm provided 3-5 dB more cancellation than the VSS LMS error algorithm. Figure 10 shows the comparison of convergence speeds and mean square error (MSE) in various adaptive filters. As can be seen, the Kalman filtering algorithm demonstrates the best convergence speed than the VSS LMS error and traditional LMS error algorithms. From the above results, the performance of acoustic crosstalk cancellation in the Kalman filtering inverse algorithm is better and faster than the VSS LMS error adaptive inverse algorithm in the application of crosstalk cancellation systems.

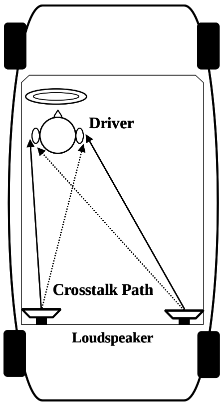


Fig. 5. Crosstalk effect in a vehicle cabin. Solid line depicts direct path; dashed line depicts crosstalk path.

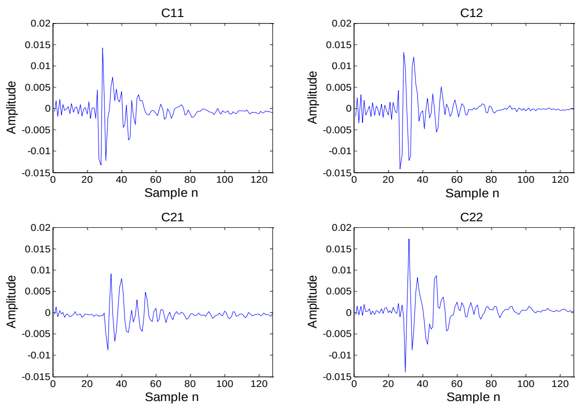


Fig. 6. Measured impulse responses in a vehicle cabin.

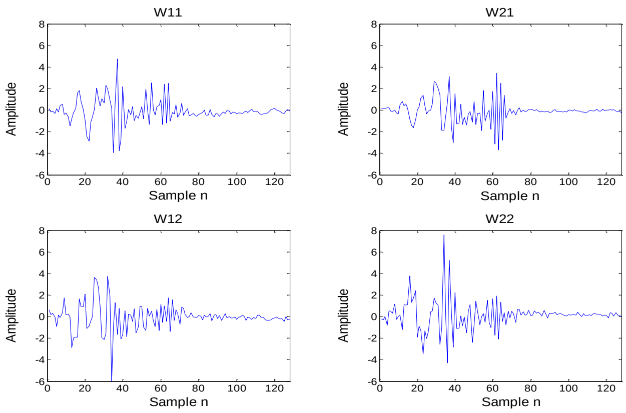
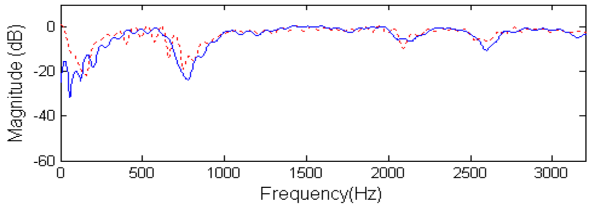
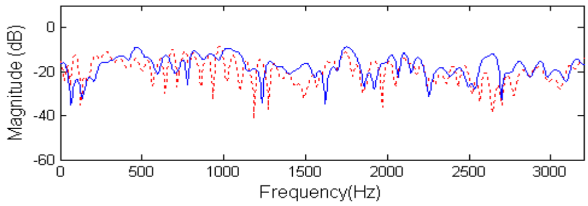


Fig. 7. Inverse filters are calculated by the Kalman filtering algorithm. (a) W_{11}; (b) W_{21}; (c) W_{12}; (d) W_{22}.

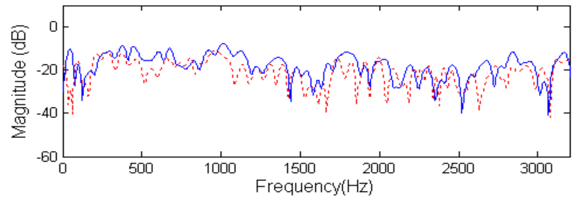


(a)

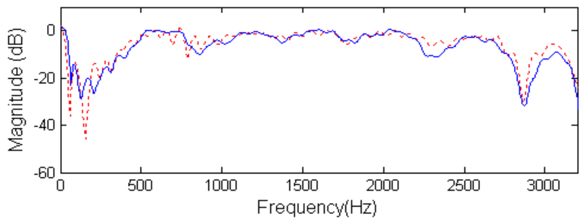


(b)

Fig. 8. Performance of acoustic crosstalk cancellation. (a) Frequency-response function between the left signal $x_1(n)$ and the left output $d_1(n)$; (b) Frequency-response function between the left signal $x_1(n)$ and the right output $d_2(n)$. Solid line depicts the simulation results using VSS LMS error algorithm; dotted line depicts the simulation results using the Kalman filtering algorithm.



(a)



(b)

Fig. 9. Performance of acoustic crosstalk cancellation. (a) Frequency-response function between the right signal $x_2(n)$ and the left output $d_1(n)$; (b) Frequency-response function between the input signal $x_2(n)$ and the right output $d_2(n)$. Solid line depicts the simulation results using VSS LMS error algorithm; dotted line depicts the simulation results using the Kalman filtering algorithm.

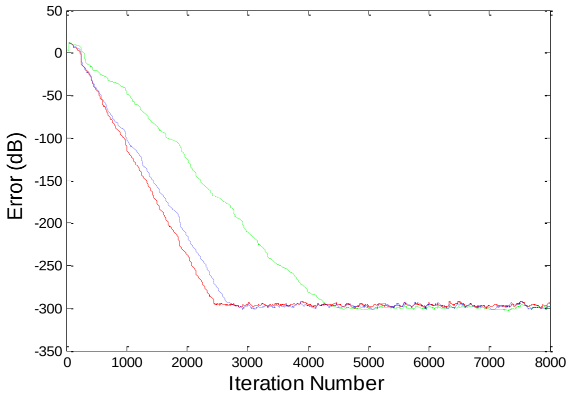


Fig. 10. Comparison of convergence speed in various adaptive filters. A solid line depicts the Kalman filtering algorithm; a dotted line depicts the VSS LMS error algorithm; a dashed line depicts the traditional LMS error algorithm.

IV. CONCLUSION

The crosstalk cancellation system is meant to eliminate undesired sound in loudspeaker systems. In this paper, the crosstalk cancellation techniques making use of adaptive inverse filtering that involves VSS LMS error algorithm and Kalman filtering algorithm are proposed and developed for a vehicular audio system. A performance comparison of the Kalman filtering algorithm, the LMS error and VSS LMS error algorithm is presented. According to the simulation results, the Kalman filtering algorithm has the best performance and the fastest convergence speed in crosstalk cancellation. Although the Kalman filtering algorithm is relatively complex, it enables a more effective cancellation of crosstalk in vehicular audio systems.

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