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Technical Note

Data-reusing-based filtered-reference adaptive algorithms for active control of impulsive noise sources

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ABSTRACT

This paper deals with the adaptive algorithms for active noise control (ANC) systems being employed for the impulsive noise sources. The standard filtered-x least mean square (FxLMS) algorithm; based on the minimization of the variance of the error signal; is well suited for attenuation of Gaussian noise sources. For the impulsive noise; modeled as a stable non-Gaussian process; however, the second order moments do not exist and hence the FxLMS algorithm becomes unstable. The filtered-x least mean p-power (FxLMP) algorithm – based on minimizing the fractional lower order moment (FLOM) – gives robust performance for impulsive ANC; however, its convergence speed is very slow. This paper proposes two data-reusing (DR)-based adaptive algorithms for impulsive ANC. The Proposed-I DR algorithm is based on the normalized step-size FxLMS (NSS-FxLMS) algorithm, and the Proposed-II DR algorithm is based on the Author's recently proposed NSS generalized FxLMP (NSS-GFxLMP) algorithm. Extensive simulations are carried out, which demonstrate the effectiveness of the proposed algorithms in comparison with the existing algorithms.

1. Introduction

Active noise control (ANC) is based on the principle of destructive interference between acoustic waves [1]. Essentially, the primary noise is canceled around the location of the error microphone by generating and combining an antiphase canceling noise [2]. The block diagram for a single-channel feedforward ANC system is shown in Fig. 1, where $P(z)$ is the primary acoustic path between the reference noise source and the error microphone, and $S(z)$ denotes the secondary path between the canceling loudspeaker and the error microphone. The reference noise signal $x(n)$ is filtered through $P(z)$ and appears as a primary disturbance signal $d(n)$ at the error microphone. The objective of the adaptive filter $W(z)$ is to generate an appropriate antinoise signal $y(n)$ propagated by the secondary loudspeaker. The antinoise signal combines with the primary disturbance signal to create a zone of silence in the vicinity of the error microphone. In this paper we address ANC of impulsive noise sources. An impulsive noise can be modeled by stable non-Gaussian distribution [3]. We consider impulsive noise with standard symmetric α-stable (SαS) distribution $f(x)$ having a characteristic function of the form,

$$\varphi(t) = e^{-|t|^{\alpha}}, \tag{1}$$

where $0 < \alpha < 2$ is the characteristic exponent [3]. The characteristic exponent α is a shape parameter, and it measures the "thickness" of the tails of the density function. If a stable random variable has a small value for α, then the distribution has a very heavy tail. For $\alpha = 2$ the relevant stable distribution is Gaussian, and for $\alpha = 1$ it is the Cauchy distribution. The probability density distributions (PDFs) of standard SαS process for various values of α are shown in Fig. 2. It is evident that for a small value of α, the process has a peaky and heavy tailed distribution.

In practice, the impulsive noises are often due to the occurrence of the noise disturbances with a low probability but large amplitude. The impulsive noise exists in many real-world applications such as punching and stamping machines in manufacturing plants, presses, combustion engines, pile drivers, IV pump sounds in the hospitals [4,5]. Furthermore, it can be shown that a fairly general type of noise signals, for example road traffic noise, office noise, etc., can be more accurately described as SαS processes rather Gaussian ones [6]. Thus it is very important to study ANC for impulsive noise sources.

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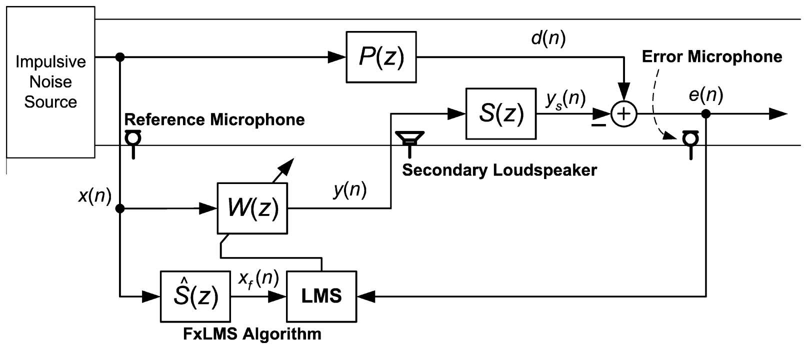

Fig. 1. Block diagram of FxLMS algorithm-based single-channel feedforward ANC system for impulsive noise source.

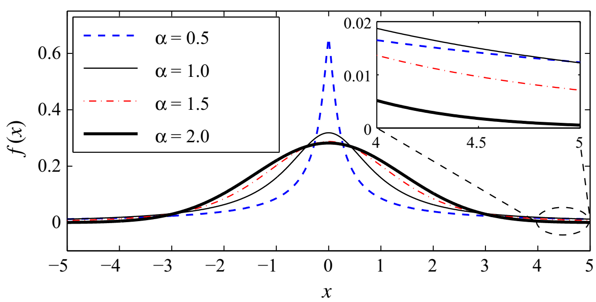

Fig. 2. The PDFs of standard symmetric α-stable (SαS) process for various values of α.

The most famous adaptation algorithm for ANC systems is the filtered-x LMS (FxLMS) algorithm [7,8], which is a modified version of the LMS algorithm [9]. For stable distributions, the moments only exist for the order less than the characteristic exponent [3], and hence the mean-square-error criterion, which is the basis for the FxLMS algorithm, is not an adequate optimization criterion. In [10], the filtered-x least mean p-power algorithm (FxLMP) has been proposed, which is based on minimizing a fractional lower order moment (FLOM) (p-power of error) that does exist for stable distributions. It has been shown that FxLMP algorithm with $0 < p < \alpha$ shows better robustness (as compared with the FxLMS algorithm) for impulsive ANC; however, the convergence speed is very slow especially when α is small. In order to improve the robustness of adaptive algorithms for processes having PDFs with heavy tails (i.e. signals with outliers), the large amplitude samples may be ignored or be replaced by an appropriate threshold value. The modified algorithms recently proposed combine these approaches as well as employ the concept of the normalized step-size (NSS). In this paper we investigate application of data-reusing (DR) type algorithm for ANC of impulsive noise sources. We propose two algorithms by utilizing the concept of DR with our recently proposed algorithms. The main idea is to improve the stability by efficiently normalizing the step-size, and improve the convergence speed by reusing the recent data.

The rest of the paper is organized is as follows. Section 2 gives a brief overview of the existing algorithms for ANC of impulsive noise sources. Section 3 details the proposed algorithms and Section 4 presents results of the computer simulations followed by the concluding remarks in Section 5.

2. Overview of existing algorithms

2.1. FxLMS algorithm and its variants

In Fig. 1, the error microphone measures the residual noise $e(n) = d(n) - y_s(n)$, where $y_s(n) = s(n) * y(n)$ is the secondary canceling signal, $*$ denotes linear convolution, and $s(n)$ is impulse response of the secondary path $S(z)$. Assuming that $W(z)$ is a finite impulse response (FIR) filter of tap-weight length L, the standard FxLMS algorithm is summarized below [7]:

$$y(n) = \mathbf{w}^T(n)\mathbf{x}(n), \tag{2}$$

$$\mathbf{x}_f(n) = \hat{s}(n) * \mathbf{x}(n), \tag{3}$$

$$\mathbf{w}(n+1) = \mathbf{w}(n) + \mu e(n)\mathbf{x}_f(n), \tag{4}$$

where $\mathbf{x}_f(n) = [x_f(n), x_f(n-1), \ldots, x_f(n-L+1)]^T$ is the filtered-reference signal vector, and $x_f(n) = \hat{s}(n) * x(n)$ is the reference signal $x(n)$ filtered through the secondary path modeling (SPM) filter $\widehat{S}(z)$ which accounts for the model of the secondary path $S(z)$, and where $\hat{s}(n)$ is the impulse response of $\widehat{S}(z)$. The tap-weight and reference signal vectors are, respectively, given as

$$\mathbf{w}(n) = [w_0(n), w_1(n), \ldots, w_{L-1}(n)]^T, \tag{5}$$

$$\mathbf{x}(n) = [x(n), x(n-1), \ldots, x(n-L+1)]^T, \tag{6}$$

where $[\cdot]^T$ denotes the transposition. The reference signal vector, used in the update equation of the FxLMS algorithm and in generating the canceling signal $y(n)$, is given in Eq. (6) which shows that the samples of the reference signal $x(n)$ at different times are treated "equally". This may cause the FxLMS algorithm to become unstable in the presence of impulsive noise, as rigorously shown in [11, Section 2.1]. To overcome this problem, the samples of the reference signal $x(n)$ are ignored in Sun's algorithm [11], if their magnitude is above a certain threshold set by statistics of the signal. Thus the reference signal is modified as

$$x\prime(n) = \begin{cases} x(n), & \text{if } x(n) \in [c_1, c_2] \\ 0, & \text{otherwise} \end{cases} \tag{7}$$

and the updated equation in Sun's algorithm [11] is given as

$$\mathbf{w}(n+1) = \mathbf{w}(n) + \mu e(n)\mathbf{x}'_f(n), \tag{8}$$

where $\mathbf{x}'_f(n) = [x'_f(n), x'_f(n-1), \ldots, x'_f(n-L+1)]^T$, and $x'_f(n) = \hat{s}(n) * x'(n)$ is modified-filtered-reference signal. Effectively, this algorithm assumes the same PDF for $x'(n)$ with in $[c_1, c_2]$ as that of $x(n)$, and simply neglects the tail beyond $[c_1, c_2]$. Noting that ignoring the samples removes the information completely, and the error signal $e(n)$ may also be peaky in nature (especially at the start-up of ANC), we have proposed thresholding the peaky samples in the reference and error signals as [12]

$$x''(n) = \begin{cases} c_1, & x(n) \leqslant c_1 \\ c_2, & x(n) \geqslant c_2 \\ x(n), & \text{otherwise} \end{cases} \tag{9}$$

$$e''(n) = \begin{cases} c_1, & e(n) \leqslant c_1 \\ c_2, & e(n) \geqslant c_2 \\ e(n), & \text{otherwise} \end{cases} \tag{10}$$