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# Frequency-domain delayless active sound quality control algorithm

Sen M. Kuo[a], Ravi K. Yenduri[a], Abhijit Gupta[b,*]

[a]*Department of Electrical Engineering, Acoustics and Vibration Center, Northern Illinois University, DeKalb, IL 60115, USA*
[b]*Department of Mechanical Engineering, Acoustics and Vibration Center, Northern Illinois University, DeKalb, IL 60115, USA*

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**Abstract**

This paper presents several broadband active sound quality control algorithms based on delayless frequency-domain techniques and subband adaptive filters. This efficient algorithm provides faster convergence and reduced computational complexity as compared to a time-domain active noise equalizer. An equal-loudness compensation method is also introduced for designing the shaping filter to achieve the desired sound quality. Computer simulations validate this algorithm in applications requiring high-order adaptive filters.


## 1. Introduction

Active noise control is based on the principle of superposition, i.e., an unwanted primary noise is attenuated by a secondary noise of equal amplitude and opposite phase [1,2]. The design of an active noise control system pursues maximum attenuation of the primary noise [3]. However, some applications need to consider psychoacoustics, which concerns the characteristics of the residual noise to match human preference [4]. This demand leads to the extension of active noise control to include acoustic noise shaping for sound quality control. Sound quality is the perceptual reaction to the sound of a product that may affect overall evaluation of that product [5].

Active sound quality control can be realized by a time-domain broadband active noise equalizer introduced in Ref. [6]. This algorithm controls the residual noise spectrum defined by the shaping filter $C(z)$. This filter is designed such that its magnitude spectrum $|C(\omega)|$ determines the desired shape of the residual acoustic noise. The filtered-X least-mean-square algorithm [7] is used to update the coefficients of the adaptive filter $W(z)$ for minimizing a pseudo-error signal instead of minimizing the residual noise in conventional active noise control systems. The time-domain active noise equalizer is optimized to reduce a passband disturbance caused by uncorrelated noise for improved stability [8].

*Corresponding author. Tel.: +1 815 753 9379; fax: +1 815 753 0416.
*E-mail address:* gupta@ceet.niu.edu (A. Gupta).

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**Nomenclature**

| | | | |
|---|---|---|---|
| $C(z)$ | shaping filter | $M_1$ | length of secondary-path estimate filter |
| $d(n)$ | primary noise | $P$ | number of weights for each subband filter |
| $D$ | decimation factor | $P(z)$ | primary path |
| $e(n)$ | residual noise | $s(n)$ | impulse response of the secondary path |
| $e'(n)$ | pseudo-error | $S(z)$ | secondary path |
| $e'_m(n)$ | subband pseudo-error | $\hat{S}(z)$ | estimate of secondary path |
| $E'(k)$ | fast Fourier transform of $e'(k)$ | $W(z)$ | adaptive filter |
| $F(z)$ | prototype filter | $x(n)$ | reference input |
| FFT | fast Fourier transform | $y(n)$ | output of adaptive filter |
| $L$ | length of the primary adaptive filter | $\mu$ | step size |
| LMS | least-mean-square | $\mu_m(n)$ | normalized step size |
| $m$ | frequency-bin index | * | linear convolution |

In some broadband active sound quality control applications, the length of the adaptive filter can be very long, which results in high computational complexity. This requirement increases the cost of the system and also reduces convergence speed of the LMS algorithm. The computational efficiency resulting from the fast Fourier transform (FFT) has led to the implementation of adaptive LMS algorithms in the frequency domain [9]. Unfortunately, the FFT introduces undesired delay due to block processing of the signal. For active noise control applications, delay critically limits the bandwidth over which noise cancellation can be achieved [2]. In this paper, a delayless frequency-domain broadband active sound quality control system is developed. This system is also modified to further reduce the computational complexity.

In addition, subband methods [10] have also been developed to improve the convergence speed and reduce the computational complexity of high-order adaptive filters. The processing of signals in subbands not only reduces the computational burden because adaptive filtering is performed at a lower decimation rate, but also results in faster convergence because the spectral dynamic range is greatly reduced in each subband. However, similar to frequency-domain adaptive filters, the disadvantage of subband adaptive filters is the introduction of delay into the signal path by the filter bank. To overcome this problem, a delayless subband adaptive filter was developed for active noise control applications [11]. Signal path delay is avoided while retaining the advantages of subband processing. The technique utilized in this paper develops a broadband active sound quality control system based on the delayless subband adaptive filters.

The remainder of the paper is organized as follows. The proposed delayless frequency-domain active sound quality control system is presented in Section 2, and a version using delayless subband adaptive filters is presented in Section 3. Section 4 analyzes the computational complexity of these broadband systems and compares them with the time-domain algorithm introduced in Ref. [6]. The design of shaping filter $C(z)$ with equal-loudness compensation for filter design is presented in Section 5.

## 2. Delayless frequency-domain active sound quality control systems

The structure of the delayless frequency-domain broadband active sound quality control system is illustrated in Fig. 1 [12]. To avoid the delay caused by collecting $N$ samples before applying a discrete Fourier transform to perform convolution via multiplication in the frequency domain, the adaptive filter $W(z)$, which is a finite impulse response filter, is implemented directly in time domain. However, the filter updating is performed in the frequency domain and the modified spectrum transformed to update the filter coefficients in the time domain. This delayless structure also prevents an undesired circular convolution in frequency-domain filtering. The canceling signal used to drive the secondary source is computed as

$$y'(n) = y(n) - c(n) * y(n) = y(n) - \sum_{m=0}^{N} c(m)y(n-m), \tag{1}$$