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STABILIZED MULTI-CHANNEL IIR FILTERS FOR ACTIVE CONTROL OF NOISE IN A DUCT

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Abstract

An adaptive IIR filter in Active Noise Control systems is more effective than an adaptive FIR filter when acoustic feedback exists, in which case an order of an adaptive FIR filter must be very large if some of poles of the ideal control filter are near the unit circle. But the IIR filters may have stability problems especially when the adaptive algorithm for adaptive filters is not yet converged. In this paper, a stabilized multi-channel recursive LMS (MCRLMS) algorithm for an adaptive multi-channel IIR filter is presented. RLMS algorithms could diverge before the algorithm is not yet converged. So, in the beginning of the ANC system, the stability of the RLMS algorithms could be improved by pulling the poles of the IIR filter to the center of the unit circle, and returning the poles to their original positions after the filter converges. Computer simulations and experiments for dipole ducts using a TMS320C32 digital signal processor have performed to show the effectiveness of a proposed algorithm.

INTRODUCTION

Active Noise Control (ANC) with additional noise sources to cancel noise is effective at low frequency. Using the advances in control systems technology and improved understanding of the physics of acoustic systems, many applications of ANC were presented[1,2].

There have been a considerable amount of effort devoted to duct noise problem since Lueg's early work[3]. This system used a single loudspeaker, and so is sometimes called the acoustic monopole system. The acoustic dipole and tripole were developed to provide a cancelling signal in the duct that propagated only in the direction of the noise propagation[1]. Multiple loudspeaker systems have been used to overcome the frequency dependent problems of control design.

Most adaptive control filters have used FIR structures based on filtered-x LMS (FXLMS) algorithms. Eriksson et al.[4] showed that IIR structures are more desirable for the active control of duct noise in order to remove the poles introduced by the acoustic feedback and presented an algorithm to adjust the coefficients of an IIR filter using the recursive least mean square (RLMS) algorithm of Feintuch [5]. If the FIR structure is used in these cases, the length of the FIR filter should be very large when the poles of that should be removed are near the unit circle. Since both of these approaches require knowledge of the secondary path transfer function, some adaptive algorithms which simultaneously estimate the transfer function of a secondary path have been presented [4]. Such adaptive techniques have a tendency to diverge when the parameters vary rapidly and it is difficult to apply them to the multiple sensor multiple speaker cases[6] because there are too many parameters to be estimated in each step.

Even though the IIR structure is more efficient than the FIR structure, the FIR filters have been widely used for the control filter due to the complexity of the coefficient fitting algorithm of the IIR filter and the stability problems especially when the adaptive algorithm for adaptive filters is not yet converged.

In this paper, a stabilizing procedure of RLMS algorithms for adaptive multi-channel IIR filters is proposed. RLMS algorithms usually diverge before the algorithm is not yet converged. So, in the beginning of the ANC system, the stability of the RLMS algorithms could be improved by pulling the poles of the IIR filter to the center of the unit circle, and returning the poles to their original positions after the filter converges. Experimental ducts for dipole ANC systems are modeled under the assumption that only plane waves propagate in the duct and also that the sources and the sensor are sufficiently far apart that near field effects can be ignored.

It is shown by computer simulations and experiments that the proposed algorithm is more stable even if the convergence parameters are larger than those of conventional RLMS algorithms.

MULTI-CHANNEL RLMS ALGORITHMS

The control signal $y_k(n)$ is generated by adaptive filtering of the reference signals as illustrated in figure 1.

$$y_k(n) = \sum_{j=i}^{J} A_{kj}^T(n) X_j(n) + \sum_{i=1}^{K} B_{kj}^T(n) Y_j(n-1), \qquad k = 1, 2, \cdots, k \tag{1}$$

Where

$$X_j(n) \equiv [x_j(n)\ x_j(n-1) \cdots x_j(n-L+1)]^T \qquad j = 1,\ 2,\ \cdots,\ J \tag{2}$$

$$Y_j(n) \equiv [y_k(n-1)\ y_k(n-2)\ \cdots\ y_k(n-I)]^T \quad k = 1, 2, \cdots, K \tag{3}$$

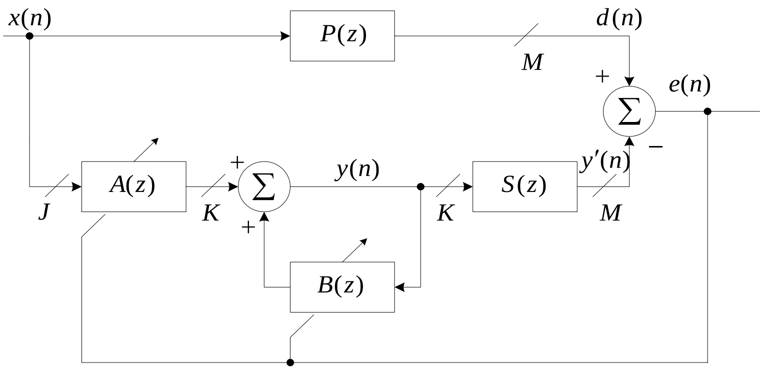


Figure 1 - A block diagram of multi-channel RLMS algorithms

Multi-channel adaptive filter can be given by using the recursive LMS algorithm as[2]

$$A_{kj}(n+1) = A_{kj}(n) + \mu_a \sum_{m=1}^{M} X'_{jkm}(n) e_m(n) \tag{4}$$

$$B_{kj}(n+1) = B_{kj}(n) + \mu_a \sum_{m=1}^{M} Y'_{jkm}(n) e_m(n) \tag{5}$$

where

$$A_{kj}(n) \equiv [a_{kj,0}(n)\ a_{kj,1}(n)\ \cdots\ a_{kj,L-1}(n)]^T \tag{6}$$

$$B_{kj}(n) \equiv [b_{kk,0}(n)\ b_{kk,1}(n)\ \cdots\ b_{kk,I}(n)]^T \tag{7}$$

$$X'_{jkm}(n) \equiv \hat{S}_{mk}(n) * X_j(n) \tag{8}$$

$$Y'_{jkm}(n) \equiv \hat{S}_{mk}(n) * Y_i(n) \tag{9}$$

$\hat{S}_{mk}(n)$ in equation (8) and (9) is the estimate of the secondary transfer function $S_{mk}(n)$.

A STABILIZED PROCEDURE

Feintuch[5] introduced an adaptive RLMS algorithm which has the computational simplicity of updating coefficients of adaptive filters, but there have been some doubts about its convergence property[7]. A stabilizing procedure to improve the convergence properties is presented. Basic idea of this procedure is based on the assumption that RLMS algorithms could diverge before the algorithm is not yet converged. Before the RLMS algorithms converge, poles of the IIR filter are pulled to the center of the unit circle, and the poles are returned to their original positions after the filter converges[8].

The denominator of control filter F(z) is

$$\mathrm{F(z)} = \begin{bmatrix} f_{11}(z) & f_{11}(z) & \cdots & f_{1\mathrm{K}}(z) \\ f_{21}(z) & \cdots & \cdots & f_{2\mathrm{K}}(z) \\ \vdots & \cdots & \cdots & \vdots \\ f_{\mathrm{K}1}(z) & \cdots & \cdots & f_{\mathrm{KK}}(z) \end{bmatrix} = \mathrm{I}_K - m\mathrm{B}(z) \qquad (10)$$

where

$$f_{kk}(z) = 1 + mb_{kk,1}(n)z^{-1} + m^2 b_{kk,2}(n)z^{-2} + \cdots + m^I b_{kk,I}(n)z^{-I}, 0 \le k \le K \qquad (11)$$

If z_1 is the root of $1 - b_{kk}(z)$, then mz_1 is also the root of $f_{kk}(mz)$. where

$$f_{kk}(mz_1) = 1 + mb_{kk,1}(mz_1)z^{-1} + m^2 b_{kk,2}(mz_1)z^{-2} + \cdots + m^I b_{kk,I}(mz_1)z^{-I}, 0 \le k \le K \qquad (12)$$

Since m is less than or equal to 1, the pole of IIR filters can be pulled to the center of z plane by replacing $1 - b_{kk}(z)$ with $f_{kk}(mz)$. The stability of adaptive filters could be improved by pulling the poles of the filter to the center of the unit circle, but the performance of adaptive filters become worse. So, the poles of the filter should be returned to their original positions after the filter converges[8]. The time varying gain can be used in place of m such that [9].

$$m(n+1) = \lambda m(n) + (1-\lambda), m(0) = 0, 0 \le \lambda \le 1 \qquad (13)$$

MODELING OF A DIPOLE DUCT

A dipole duct, as shown in figure 2, is modelled under the assumption that only plane waves propagate in the duct and also that the sources and the sensor are sufficiently far apart that near field effects can be ignored[10].

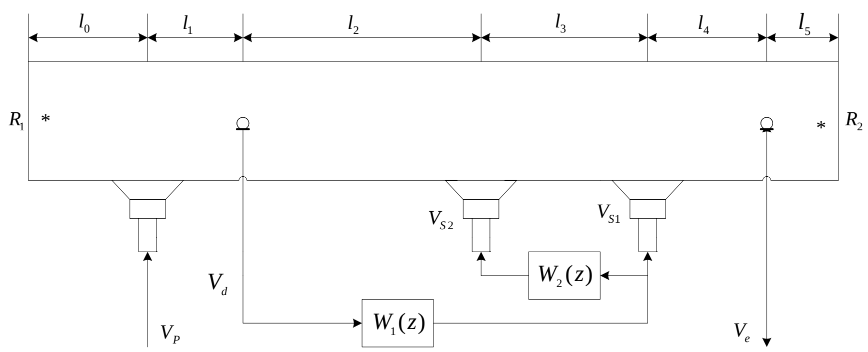


Figure 2 - Representation of a dipole duct

The primary and second sources and detection sensors are located as shown in figure 2. The left and right ends of the duct have complex pressure reflection coefficients R_1 and R_2 respectively. This system has three electrical inputs; V_p, V_{s1} and V_{s2} which drive the primary and secondary sources, and one electrical output, V_e, from the error sensor, as shown in figure 3.

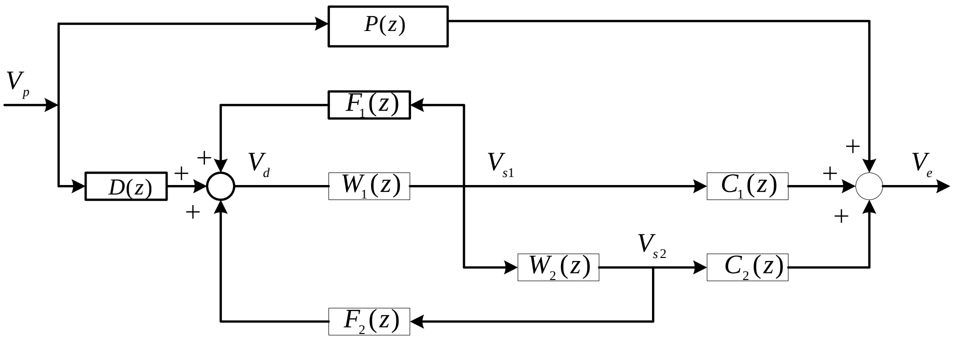


Figure 3 - A block diagram of a dipole duct ANC system

If all components in the system are linear, the output voltage V_e can be obtained as

$$V_e = A(z)V_p + C_1(z)V_{s1} + C_2(z)V_{s2} \tag{14}$$

If $C_1(z)$, $C_2(z)$ and A(z) are known, either from measurement or calculation, then the control filters $W_1(z)$ and $W_2(z)$ can be found. Using the standard steady state traveling wave theory to this system, the transfer functions in figure 2 can be derived by

$$P(z) = \frac{z^{-(n_1+n_2+n_3+n_4)} \times (1+R_1 z^{-2n_0}) \times (1+R_2 z^{-2n_5})}{(1-R_1R_2)z^{-2n})} \tag{15}$$

$$D(z) = \frac{z^{-n_1} \times (1+R_1 z^{-2n_0}) \times (1+R_2 z^{-2(n_2+n_3+n_4+n_5)})}{(1-R_1R_2)z^{-2n})} \tag{16}$$

$$C_1(z) = \frac{z^{-n_4} \times (1+R_1 z^{-2(n_0+n_1+n_2+n_3)}) \times (1+R_2 z^{-2n_5})}{(1-R_1R_2)z^{-2n})} \tag{17}$$

$$C_2(z) = \frac{z^{-(n_3+n_4)} \times (1+R_1 z^{-2(n_0+n_1+n_2)}) \times (1+R_2 z^{-2n_5})}{(1-R_1R_2)z^{-2n})} \tag{18}$$

$$F_1(z) = \frac{R_1R_2 z^{-(2n-n_2-n_3)} \times (1+(1/R_1) z^{-2(n_0+n_1)}) \times (1+(1/R_2) z^{-2(n_4+n_5)})}{(1-R_1R_2)z^{-2n})} \tag{19}$$

$$F_2(z) = \frac{R_1R_2 z^{-(2n-n_2)} \times (1+(1/R_1) z^{-2(n_0+n_1)}) \times (1+(1/R_2) z^{-2(n_3+n_4+n_5)})}{(1-R_1R_2)z^{-2n})} \tag{20}$$

Where n_i is the nearest integer to $l_i f_s / c$, n is the nearest integer to lf_s / c, and l, f_s and c are length of a duct, sampling frequency and sound speed respectively. We assume that all electro-acoustic transfer functions and directivity factors of the sources and the sensors are l.

COMPUTER SIMULATIONS AND EXPERIMENTS

Computer simulations of a dipole duct ANC system controlling gasoline engine noise in a duct have been performed to investigate the effectiveness of the proposed algorithm. The values of the duct parameters from l_0 to l_5 are 0.3, 0.5, 1.6, 0.3, 0.5 and 0.3 meters respectively. The reflection coefficients R_1 and R_2 are set to 0.2 and 0.1, and sampling frequency is set to 2 [kHz].

To investigate the convergence property of the proposed algorithm, the simulation results of the proposed algorithm were compared with those of conventional IIR filter algorithms and FXLMS algorithms as shown in figures 4 to 7. Computer simulations were done for monopole and dipole active noise control systems for duct noise attenuation.

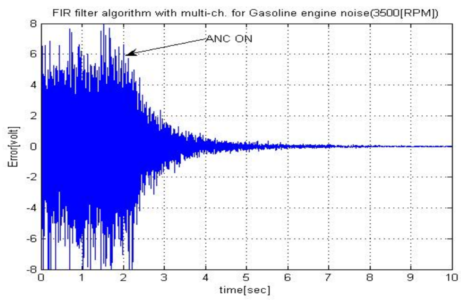


Figure 4 - Error (FXLMS, Dipole)

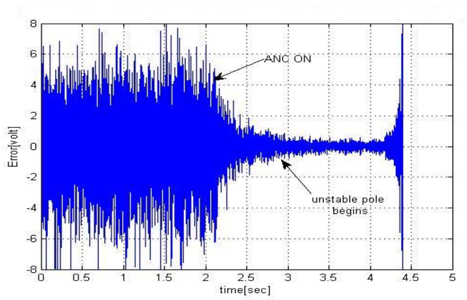


Figure 5 - Error (RLMS, Dipole)

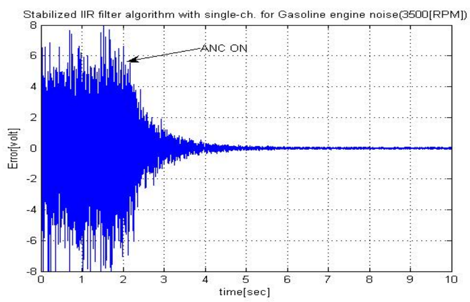


Figure 6 - Error (Proposed, Monopole)

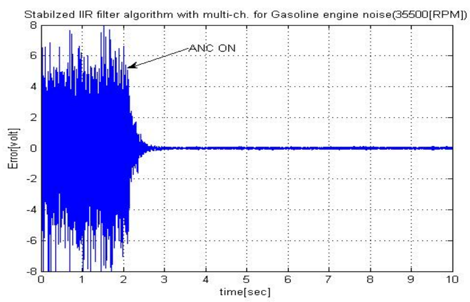


Figure 7 - Error (Proposed, Dipole)

The order of FXLMS is 256 and those of the proposed algorithm are 96 for a numerator of IIR filter and 4 for a denominator of IIR filter (say, 96+4). Simulation results show that the proposed algorithm converges better than FXLMS algorithms even when the computational burden of the proposed algorithm is smaller than that of FXLMS, and conventional RLMS algorithms diverge. Dipole systems show better results than monopole systems in all cases. The proposed stabilized RLMS algorithm converges well and all poles of IIR filters stay inside unit circle as shown in figures 8 and the poles of conventional IIR filters go to outside of unit circle as shown in figures 9.

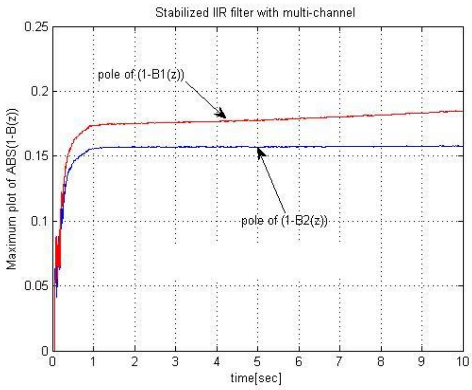


Figure 8 - Pole trajectory(Proposed)

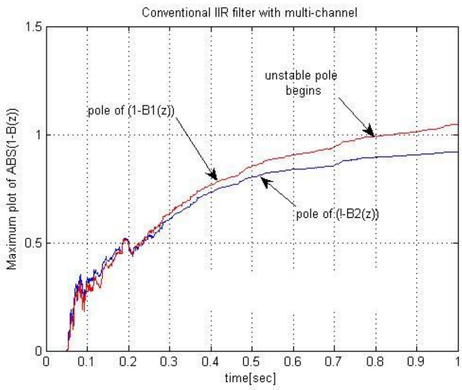
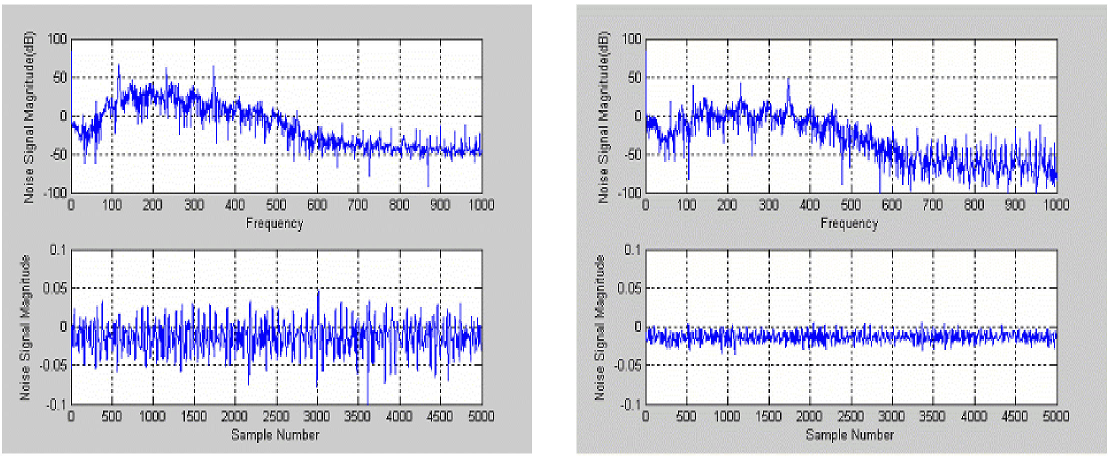


Figure 9 – Pole trajectory(RLMS)

Experimental results for multi-channel ANC systems for a dipole duct show the effectiveness of the proposed algorithm as shown in figures 10. Gasoline engine noise was used for reference signal and a TMS320VC33 DSP board was used for implementing adaptive filters. The proposed IIR filter of order 16+16 with 2 [kHz] sampling frequency was used. As shown in figure 10, we can get 25 [dB] tonal reduction and 20 [dB] overall reduction.



(a) Before control *(b) After control*

Figure 10 - FFT of noise and noise sequence

CONCLUSIONS

A stabilizing procedure of multi-channel RLMS algorithms for active noise control systems is proposed. This procedure is based on the facts that adaptive IIR filters usually diverge before the algorithm is not yet converged. So, in the beginning of the ANC system, the stability of the adaptive IIR filters could be improved by pulling the poles of the IIR filter to the center of the unit circle, and returning the poles to their original positions after the filter converges.

A dipole duct is modelled using the electro-acoustical theory for computer simulations. Computer simulations and experiments show that the proposed algorithm converges better than conventional recursive LMS algorithms and filtered-x LMS algorithms.

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