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Multichannel Filtered Inverse LMS Algorithm for ANC: Laboratory Implementation and Testing

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Summary
This paper describes the development and implementation of a filtered-inverse least-mean-square (FILMS) algorithm for a multichannel active noise control system (ANCS). The FILMS algorithm is derived from the well-known filtered-x LMS (FXLMS) algorithm with the aim of improving the rate of convergence and reducing the considerable computational load of the multichannel FXLMS algorithm. Mathematical development and equations describing the multichannel FILMS algorithm are presented in the first part of this paper and the laboratory setup and techniques used to implement and test FILMS-based ANCS are described in the second part. Results of the implemented system are presented and discussed.

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1. Introduction

Active noise control systems (ANCS) use an actuator (typically a loudspeaker) to inject a secondary sound into the environment, with the aim of cancelling or reducing the primary, unwanted sound. ANCS uses the reference sensor to sense the noise signal close to the noise source and the error sensor, usually a microphone, to monitor the actuator performance and adjust the control system parameters. The size of the zone of reduced sound pressure level (SPL) around the error microphone depends on the selected control algorithm, acoustic environment and wavelength of the acoustic noise – sound that needs to be cancelled. Using the multiple actuators and sensors, the multichannel ANCS can be formed in order to increase and control the size and shape of the generated zone of silence. The most popular control algorithm for ANC applications is the filtered-X LMS algorithm (FXLMS) [1]. FXLMS is the most frequently employed control algorithm in ANCSs mainly due to its simplicity, robustness and low computational load. Although the FXLMS algorithm is computationally simple, its convergence speed can be slow and signal dependent, while computational requirements can grow significantly with the number of control channels implemented in the system.

The need for a computationally simple ANC algorithm with a faster convergence rate is the main motivation for the many investigations conducted and reported in the literature. A modified FXLMS algorithm has been proposed by Bjarnason [2] to ameliorate the convergence problem. This modified algorithm behaves basically like an LMS algorithm. Its detailed stability analysis can be found in [3]. While this algorithm exhibits improved convergence performance over the conventional FXLMS, it achieves it at an increased computational cost of $3L_w$ computations per iteration, where L_w represents the length of the adaptive filter. This is still prohibitive in several applications since the value of L_w may be significant. In [4], Rupp presented a method to reduce the complexity of the FXLMS algorithm without changing the basic behaviour of the algorithms. If the filter order of the feedforward path is small in comparison to the adaptive filter length, the complexity of the algorithms is approximately $2L_w$.

Rupp and Sayed [5, 6, 7] presented a time-domain feedback analysis of the FXLMS algorithm, which has been receiving increasing attention in the literature due to its potential application in the ANC. In particular, authors introduced a generalised FXLMS variant and derived condition for its l_2-stability. The two modifications proposed by the authors still employ the same computational cost of the order of $2L_w$ computations per iteration. Both algorithms have improved convergence behaviour when compared with FXLMS but only one modification exhibits behaviour comparable with the original FXLMS algorithm. The same authors further introduced a generalised FXLMS recursion and established that the various recursions are of filtered-error form [5]. A special case of the generalised algorithm is the so-called modified FXLMS recursion, which refers to a recent modification of the standard FXLMS update. An optimal choice of the step-size parameter in order to guarantee faster convergence and conditions for robustness was also derived [5].

However, the performance of these and other algorithms only in certain cases approaches that of the modified

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FXLMS algorithm. Recently, an alternative method of the fast exact realisations of the FXLMS algorithm, which retains the characteristics of the original FXLMS algorithm but reduces the computational complexity and memory requirement, has been proposed by Douglas [8, 9, 10, 11]. In these papers, the author provides a computationally efficient method for removing the output delay within the error signal used to update the controller coefficients such that the standard LMS algorithm is effectively employed. As the filtered inputs are not computed directly, a significant reduction in computations and memory usage (about 80% for an 8-channel ANCS) is achieved using this approach. In most cases, fast realisations can also reduce the computational complexity of these algorithms for multichannel systems [12, 13].

A better understanding of the algorithm performance, which is useful for algorithm design, would follow from the knowledge of the transient and steady-state weight statistics for different implementations. Mean square estimation plays a crucial role in many problems of adaptive control and adaptive signal processing [14]. Thus, the question of how to design adaptive estimation system for optimised behaviour has met considerable interest.

In this paper a Multichannel Filtered-Inverse LMS (MFILMS) algorithm designed with the aim of reducing the computational load and at the same time increase the convergence speed of the multichannel FXLMS algorithm is derived and presented. This is achieved by adding an inverse structure to FXLMS, hence effectively splitting the FIR controller into two parts: a multichannel predictor and a multichannel delayed non-causal model of the inverse plant between error sensors and actuators. The approach of splitting the controller in two parts is not entirely new [15, 16, 17], and these two parts are sometimes referred to as the regulator and compensator. However, unlike the approaches taken in [16] and [17], the novel approach of this work is the use of multichannel equalisation techniques with and without cross-cancellation for ANC applications. Here, the techniques commonly used for sound reproduction systems [18, 19, 20, 21] are modified to be used in real-time ANCS. Mathematical analysis of both single and multichannel FILMS algorithm is performed in [22] and [23]. In [22] statistical analysis of the single-channel FILMS algorithm is performed without the use of independence theory (IT) assumptions, for the case of slow adaptation of filter weights. Simulation results confirmed the accuracy of the adopted approach showing very good agreement between the developed theory and the algorithm performance. Different variants of the MFILMS are detailed in [23] where some simulations have been performed to demonstrate the effectiveness of the adopted approach. In this paper the main steps in MFILMS algorithm derivation are given and followed by description and results of the laboratory implementation and testing of the MFILMS-based ANCS.

This paper is organised in the following way: The mathematical derivations and equations used to implement various variants of MFILMS algorithm are derived and presented in section 2. Section 3 describes the laboratory setup used to implement and test the multichannel ANCS performance. Methods used to estimate the multichannel secondary plant and its inverse, based on adaptive system identification techniques, are described in section 4. Performance of the multichannel plant and inverse plant estimation algorithms are discussed and compared to a better known and more often used single-channel estimation method. Finally, convergence rate and attenuation levels achieved at the error microphone positions and other points in the room are presented in section 5.

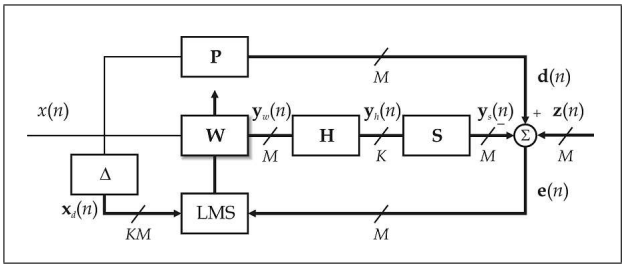


Figure 1. Multichannel realisation of the FILMS algorithm.

2. Multichannel FILMS algorithms

A block diagram of the single-reference/multiple-output FILMS algorithm is shown in Figure 1. This arrangement can be easily generalised for the case of multiple reference signals by combining them together into one.

The components of the error signal vector $\mathbf{e}(n)$ are formed by the M error microphone outputs. The vector $\mathbf{W}(n)$ represents the coefficients of all the M adaptive filters each one of length L_w. The inverse matrix, denoted as $\mathbf{H}$, consists of the $K \times M$ estimates of the secondary path's inverse filters $\mathbf{h}_{km}$ measured between the K secondary loudspeakers and the M error microphones each of length L_h. The secondary path matrix, denoted by $\mathbf{S}$, consists of the $M \times K$ secondary path's impulse responses $\mathbf{s}_{km}$ measured between the K secondary loudspeakers and the M error microphones each of length L_s. All system responses are modelled using linear FIR filters. Matrix $\mathbf{\Delta}$ in the update branch of the scheme consists of $M \times M$ filters whose coefficients depend on the chosen adaptive update technique applied in the filtering branches, discussed later in this section. The signal vector $\mathbf{y}_w$ consists of the M outputs of each adaptive filter obtained by filtering the reference signal $x(n)$ through the corresponding adaptive FIR filter, $\mathbf{w}_m(n)$. The signal vector $\mathbf{y}_h$ consists of the K input signals used to drive the K loudspeakers. The signal vector $\mathbf{y}_s$ consists of the M secondary path outputs. These signals combined with the corresponding set of desired signals $d_m(n)$ and the measurement output noise signals $z_m(n)$ form the error signals $e_m(n)$ used in adaptive process.

To formalise the approach and explicitly describe the multichannel FILMS algorithm, the following notation is introduced and used through the rest of this section:

L_p: the length of primary path impulse responses,
L_w: the length of adaptive filters,
L_s: the length of secondary path impulse responses,
L_h: the length of estimated secondary path inverse filter responses,
$\mathbf{x}(n) = \left[x(n),\ x(n-1)\ \ldots\ x(n-L_w+1)\right]^T$:
the reference signal vector of length L_w,
$\mathbf{x}'(n) = \left[x(n),\ x(n-1)\ \ldots\ x(n-L_p+1)\right]^T$:
the reference signal vector of length L_p,
$\mathbf{x}''(n) = \left[x(n),\ x(n-1)\ \ldots\ x(n-L_s-L_h)\right]^T$:
the reference signal vector of length $L_s + L_h - 1$,
$\mathbf{y}_w(n) = \left[y_{w1}(n),\ y_{w2}(n)\ \ldots\ y_{wM}(n)\right]^T$:
adaptive filter output signal vector,
$\mathbf{y}_h(n) = \left[y_{h1}(n),\ y_{h2}(n)\ \ldots\ y_{hK}(n)\right]^T$:
inverse filter output signal vector,
$\mathbf{y}_s(n) = \left[y_{s1}(n),\ y_{s2}(n)\ \ldots\ y_{sM}(n)\right]^T$:
secondary path output signal vector,
$\mathbf{d}(n) = \left[d_1(n),\ d_2(n)\ \ldots\ d_M(n)\right]^T$:
desired signal vector,
$\mathbf{e}(n) = \left[e_1(n),\ e_2(n)\ \ldots\ e_M(n)\right]^T$:
error signal vector,
$\mathbf{z}(n) = \left[z_1(n),\ z_2(n)\ \ldots\ z_M(n)\right]^T$:
measurement output noise vector.

Filter block matrices consisting of the multichannel filter vectors are defined as
$\mathbf{W}(n) = \left[\mathbf{w}_1^T(n),\ \mathbf{w}_2^T(n)\ \ldots\ \mathbf{w}_M^T(n)\right]^T$:
the adaptive filter coefficients vector of length ML_w,
$\mathbf{P} = \left[\mathbf{p}_1,\ \mathbf{p}_2\ \ldots\ \mathbf{p}_M\right]$:
primary path impulse response $L_p \times M$ matrix,
$\mathbf{S} = \left[\mathbf{s}_{km}\right]^T_{1\le k\le K, 1\le m\le M}$:
secondary path impulse response $M \times K$ block matrix,
$\mathbf{H} = \left[\hat{\mathbf{h}}_{km}\right]_{1\le k\le K, 1\le m\le M}$:
inverse filter $K \times M$ block matrix,
$\mathbf{\Delta} = \left[\Delta_{mm'}\right]_{1\le m\le M, 1\le m'\le M}$:
update filter $M \times M$ block matrix,
where
$\mathbf{w}_m(n) = \left[w_{m,0}(n),\ w_{m,1}(n)\ \ldots\ w_{m,L_w-1}(n)\right]^T$:
the m-th adaptive filter,
$\mathbf{p}_m = \left[p_0,\ p_1\ \ldots\ p_{L_p-1}\right]^T$: the primary path impulse response measured between the noise source, i.e. reference microphone, and the m-th error microphone,
$\mathbf{s}_{km} = \left[s_{km,0},\ s_{km,1}\ \ldots\ s_{km,L_s-1}\right]^T$: the secondary path impulse response measured between the k-th loudspeaker and the m-th error microphone,
$\hat{\mathbf{h}}_{km} = \left[\hat{h}_{km,0},\ \hat{h}_{km,1}\ \ldots\ \hat{h}_{km,L_k-1}\right]^T$: the estimated inverse filter response measured between the k-th loudspeaker and the m-th error microphone.

The $M \times M$ block-diagonal matrix $\mathbf{X}(n)$ that consists of the reference signal vector $\mathbf{x}(n)$ of length L_w is introduced as

$$\mathbf{X}(n) = \begin{bmatrix} \mathbf{x}(n) & 0 & \cdots & 0 \\ 0 & \mathbf{x}(n) & \cdots & 0 \\ \vdots & \vdots & \ddots & \vdots \\ 0 & 0 & \cdots & \mathbf{x}(n) \end{bmatrix} \tag{1}$$

and used to calculate the outputs of the adaptive filters. Multichannel FILMS algorithm can be described with the following set of equations:

$$\mathbf{d}(n) = \mathbf{P}^T\mathbf{x}'(n), \tag{2}$$

$$\mathbf{y}_w(n) = \mathbf{X}^T(n)\mathbf{W}(n), \tag{3}$$

$$\mathbf{y}_h(n) = \mathbf{H} * \mathbf{y}_w(n), \tag{4}$$

$$\mathbf{y}_s(n) = \left(\mathbf{S} * \mathbf{H}\right) * \mathbf{y}_w(n), \tag{5}$$

$$\mathbf{y}_{sm}(n) = \sum_{k=1}^{K}\sum_{m'=1}^{M} \mathbf{s}_{km} * \hat{\mathbf{h}}_{km'} * \left[\mathbf{x}^T(n)\mathbf{w}_{m'}(n)\right], \quad 1 \le m \le M, \tag{6}$$

$$\mathbf{e}(n) = \mathbf{d}(n) - \mathbf{y}_s(n) + \mathbf{z}(n). \tag{7}$$

Using these equations for the multichannel ANCS, a general adaptive filter update equation of the MFILMS algorithm and its modified versions are derived and discussed in the following sub-sections.

2.1. Complete multichannel FILMS algorithm

Combining (3) and (5) into (7), the instantaneous error signal vector is given by

$$\mathbf{e}(n) = \mathbf{d}(n) - \left(\mathbf{S} * \mathbf{H}\right) * \left[\mathbf{X}^T(n)\mathbf{W}(n)\right] + \mathbf{z}(n). \tag{8}$$

This can be expressed as M independent linear equations for $e_m(n)$, where $1 \le m \le M$,

$$e_m(n) = d_m(n) - \sum_{k=1}^{K}\sum_{m'=1}^{M} \mathbf{s}_{km} * \hat{\mathbf{h}}_{km'} * \left[\mathbf{x}^T(n)\mathbf{w}_{m'}(n)\right] + \mathbf{z}_m(n). \tag{9}$$

The update equation for all M adaptive filters can be written in the usual form as

$$\mathbf{W}(n+1) = \mathbf{W}(n) - \frac{\mu}{2}\nabla\hat{\xi}(n). \tag{10}$$

The cost function $\xi(n)$, defined as the sum of mean-square-error (MSE) signals, is estimated assuming that each MSE component can be approximated by its instantaneous squared error, i.e. $E[e_m^2(n)] \approx e_m^2(n)$. The gradient of $\hat{\xi}(n)$ is calculated with respect to the m'-th adaptive filter $\mathbf{w}_{m'}$ as

$$\begin{aligned} \nabla_{w_{m'}}\hat{\xi}(n) &= 2\sum_{m=1}^{M} e_m(n)\frac{\partial e_m(n)}{\partial \mathbf{w}_{m'}} \\ &= -2\sum_{m=1}^{M}\left[\sum_{k=1}^{K}\mathbf{s}_{km} * \hat{\mathbf{h}}_{km'} * \mathbf{x}''(n)\right]e_m(n). \end{aligned} \tag{11}$$

For further simplicity, a new $M \times M$ block matrix $\mathbf{X}_{sh}$ with the (m', m)-th block equal to

$$\left(\mathbf{X}_{sh}\right)_{m'm} = \sum_{k=1}^{K}\mathbf{s}_{km} * \hat{\mathbf{h}}_{km'} * \mathbf{x}''(n) \tag{12}$$

is introduced, such that the gradient vector $\nabla\hat{\xi}(n)$ can be written in matrix form as

$$\nabla\hat{\xi}(n) = -2\mathbf{X}_{sh}\mathbf{e}(n). \tag{13}$$

By substituting (13) into (10), the general adaptive update equation for the multichannel FILMS algorithm is given by

$$\mathbf{W}(n+1) = \mathbf{W}(n) + \mu \mathbf{X}_{sh} \mathbf{e}(n), \tag{14}$$

and for the m'-th adaptive filter, can be written as

$$\mathbf{w}_{m'}(n+1) = \mathbf{w}_{m'}(n) + \mu \sum_{m=1}^{M} \left[\sum_{k=1}^{K} \hat{\mathbf{s}}_{km} * \hat{\mathbf{h}}_{km'} * \mathbf{x}''(n) \right] e_m(n). \tag{15}$$

The $\mathbf{X}_{sh}$ matrix is a result of filtering the reference signal through the $(\mathbf{S} * \mathbf{H})$ combination. In practice, the $\mathbf{S}$ matrix is not available and needs to be replaced with the $\hat{\mathbf{S}}$ matrix which is an estimate of $\mathbf{S}$ usually obtained using an off-line system identification method. Moreover, a perfect equalisation of loudspeaker impulse response requires that $(\mathbf{s}_{km} * \hat{\mathbf{h}}_{km})$ has a flat magnitude response. For multichannel realisation, this condition translates into the requirement that $(\mathbf{S} * \mathbf{H})$ must be an identity matrix in the z-domain.

The adaptive algorithm given by (14) and (15) and denoted further as the Complete Multichannel FILMS (CMFILMS) algorithm, requires the use of signals from all M error microphones for the adjustment of the coefficients of all M adaptive filters at each algorithm iteration. To reduce the heavy computational load of this algorithm, convolutions between the filters $\hat{\mathbf{s}}_{km}$ and $\hat{\mathbf{h}}_{km'}$ can be computed off-line, i.e. before the adaptation process.

2.2. Complete single-error multichannel FILMS algorithm

Using the precomputed $\hat{\mathbf{s}}_{km} * \hat{\mathbf{h}}_{km'}$ convolutions, the computational load of the CMFILMS algorithm is somewhat reduced but the weights update process remains a computationally demanding task for the large number of implemented channels since the number of online convolutions is proportional to M^2. Therefore, it may be beneficial to build an ANCS where each adaptive filter is controlled with only one corresponding error signal. One way to assure the stability of that system in a free field would be by positioning each error microphone closer to its associated secondary source than to the other secondary sources [7] in order to make the distance from each secondary source to its uncontrolled microphones larger than the distance to its controlled microphone.

This modification, referred to further as Complete Single-Error Multichannel FILMS (CSEMFILMS), leads to a significantly improved and computationally less demanding adaptive algorithm, since the combinations of $\hat{\mathbf{s}}_{km}$ and $\hat{\mathbf{h}}_{km}$ can be replaced with pure delays (the same delays required to make $\hat{\mathbf{h}}_{km}$ causal). Moreover, this allows a considerable saving of memory resources.

The general update equation for all M adaptive filters in the case of CSEMFILMS algorithm is given by

$$\mathbf{W}(n+1) = \mathbf{W}(n) + \mu \mathbf{X}_{\delta} \mathbf{e}(n), \tag{16}$$

where $\mathbf{X}_{\delta}$ is the block diagonalised matrix with the m-th block equal to

$$\left(\mathbf{X}_{\delta}\right)_m = \sum_{k=1}^{K} \hat{\mathbf{s}}_{km} * \hat{\mathbf{h}}_{km} * \mathbf{x}''(n) = \sum_{k=1}^{K} \mathbf{x}''(n - D_{km}), \tag{17}$$

where D_{km}, $1 \le k \le K$, $1 \le m \le M$, are the delays of corresponding combinations between the (k, m)-th secondary path impulse response and its estimated inverse filter.

Using (17), the update equation for the m-th adaptive filter can be rewritten as

$$\mathbf{w}_m(n+1) = \mathbf{w}_m(n) + \mu \left[\sum_{k=1}^{K} \mathbf{x}''(n - D_{km}) \right] e_m(n). \tag{18}$$

2.3. Simplified multichannel FILMS algorithm

Assuming similar characteristics of electro-acoustic transducers in the ANCS, a simpler version of the single reference-multichannel FILMS algorithm can be obtained by estimating only the inverses of direct secondary paths and neglecting the inverses of cross paths in the inverse plant. This approach is applicable only for the ANCS with an equal number of loudspeakers and error microphones, $K = M$. Such modification is denoted as a Simplified Multichannel FILMS algorithm (SMFILMS). The inverse filter $M \times M$ block matrix $\mathbf{H}$ is now diagonalised consisting of only the direct path inverses.

The matrix $\mathbf{X}_{sh}$ is also significantly reduced using this approach with the (m', m)-th block equal to

$$\left(\mathbf{X}_{sh}^{S}\right)_{m'm} = \mathbf{s}_{m'm} * \hat{\mathbf{h}}_{m'm'} * \mathbf{x}''(n). \tag{19}$$

Since all inverses of the cross paths are omitted, the number of convolutions done off-line is reduced by the factor M. The adaptive update equation for the CMFILMS algorithm given by (14) can then be rewritten as

$$\mathbf{W}(n+1) = \mathbf{W}(n) + \mu \mathbf{X}_{sh}^{S} \mathbf{e}(n), \tag{20}$$

and for the m'-th adaptive filter as

$$\mathbf{w}_{m'}(n+1) = \mathbf{w}_{m'}(n) + \mu \sum_{m=1}^{M} \left[\hat{\mathbf{s}}_{m'm} * \hat{\mathbf{h}}_{m'm'} * \mathbf{x}''(n) \right] e_m(n). \tag{21}$$

2.4. Simplified single-error multichannel FILMS algorithm

The SMFILMS algorithm can be recast in an even simpler form for the case of single-error update by reducing the matrix $\mathbf{X}_{sh}^{S}$ to $\mathbf{X}_{\delta}^{S}$ with the m-th block equal to

$$\left(\mathbf{X}_{\delta}^{S}\right)_m = \mathbf{x}''(n - D_{mm}), \tag{22}$$

where D_{mm}, $1 \le m \le M$, are delays resulting from the corresponding combinations between the (m, m)-th secondary path impulse response and its estimated inverse filter.

Finally, the update equations for the Simplified Single-Error Multichannel FILMS (SSEMFILMS) algorithm are given by

$$\mathbf{W}(n+1) = \mathbf{W}(n) + \mu \mathbf{X}_{\delta}^{S} \mathbf{e}(n), \tag{23}$$

and for the m-th adaptive filter:

$$\mathbf{w}_m(n+1) = \mathbf{w}_m(n) + \mu \mathbf{x}''(n - D_{mm}) e_m(n). \tag{24}$$

2.5. Update filter matrices for the MFILMS algorithms

In previously derived update equations (14), (16), (20), and (23), the matrices $\mathbf{X}_{sh}$, $\mathbf{X}_{\delta}$, $\mathbf{X}_{sh}^{S}$ and $\mathbf{X}_{\delta}^{S}$ represent the results of filtering the reference signal $x(n)$ through the $M \times M$ update filter block matrix $\mathbf{\Delta}$ shown in Figure 1. This filter matrix depends on the structure of the inverse and secondary plant that follows the adaptive filters. For the purpose of simplicity, the single-error signal approach is also incorporated into this matrix.

The filters $\Delta_{m'm}$ in the update filter matrix $\mathbf{\Delta}$ ($1 \leq m \leq M$, $1 \leq m' \leq M$), are defined for the four proposed versions of the MFILMS algorithm such as

$$\Delta_{m'm} = \begin{cases} \sum_{k=1}^{K} \hat{\mathbf{s}}_{km} * \hat{\mathbf{h}}_{km'} & \text{CMFILMS} \\ \begin{cases} \sum_{k=1}^{K} \delta(n - D_{km}) & \text{if } m' = m \\ 0 & \text{otherwise} \end{cases} & \text{CSEMFILMS} \\ \hat{\mathbf{s}}_{m'm} * \hat{\mathbf{h}}_{m'm'} & \text{SMFILMS} \\ \begin{cases} \delta(n - D_{mm}) & \text{if } m' = m \\ 0 & \text{otherwise} \end{cases} & \text{SSEMFILMS} \end{cases} \tag{25}$$

The length of those filters in the case of CMFILMS and SMFILMS necessary to adequately represent the result of convolution $\hat{\mathbf{s}} * \hat{\mathbf{h}}$ is $L_s + L_h - 1$, and for the single-error approach depends merely on the maximum delay D_{km}.

2.6. Computational complexity of the MFILMS algorithms

The computational loads of the proposed MFILMS algorithms given in Table I along with the multichannel FXLMS algorithm are estimated by counting the number of multiplications at each algorithm iteration. For both CMFILMS and SMFILMS algorithms, the complexities calculated in the case of the single-error update approach are reduced by $M^2(L_w + L_s + L_h - 1) - ML_w$ multiplications. The use of delay lines in filtering branches makes CSEMFILMS and SSEMFILMS algorithms far less computationally demanding and memory intensive than their full-update versions, CMFILMS and SMFILMS.

For the chosen set of system parameters ($L_s = 200$, $L_h = 100$, $L_w = 40$), the number of required computations for the various number of channels implemented, is calculated according to Table I and plotted in Figure 2 for all four versions of the multichannel FILMS. It can be seen that the complexity of the SMFILMS is comparable to the CMFILMS for the low numbers of channels but SMFILMS becomes more efficient for the channel numbers greater than 8.

Table I. Computational complexities of the proposed multichannel algorithms.

Algorithms	Number of multiplies
MFXLMS	$KML_s + K(M+1)L_w + M$
CMFILMS	$KML_h + M(M+1)L_w + M^2(L_s + L_h - 1) + M$
CSEMFILMS	$KML_h + 2ML_w + M$
SMFILMS	$ML_h + M(M+1)L_w + M^2(L_s + L_h - 1) + M$
SSEMFILMS	$ML_h + 2ML_w + M$

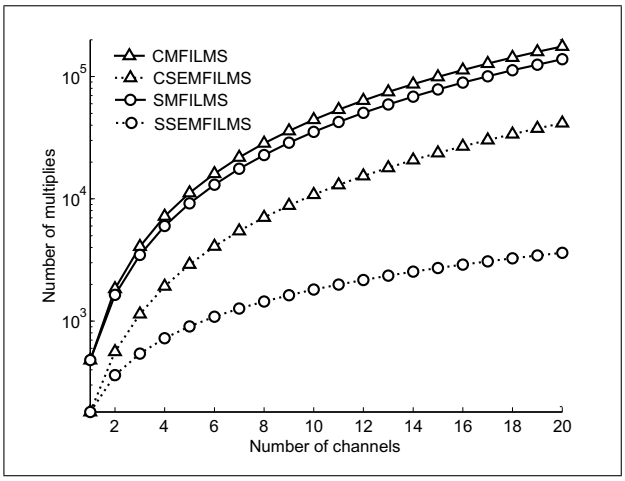


Figure 2. Relationship between computational complexity and the number of implemented channels for MFILMS with adaptive filter length $L_w = 40$.

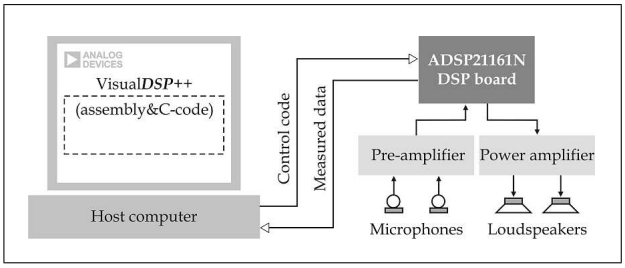
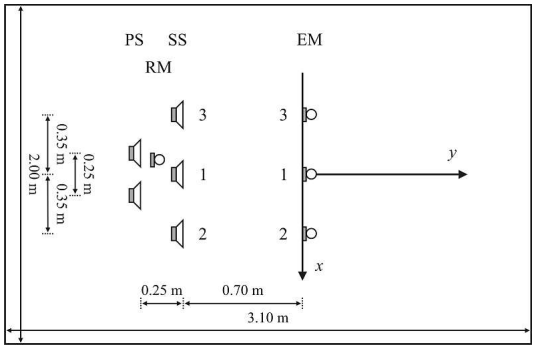


Figure 3. Hardware used for ANC experiment.

3. Laboratory setup and system implementation

A schematic of the equipment used to implement the prototype ANCS is shown in Figure 3. Data acquisition and signal processing in the multichannel ANCS is performed using the Analog Device's SHARC processor ADSP21161N.

This is a 32-bit floating-point DSP processor with a 100 MHz core clock. The audio interface on the 21161N EZ-KIT Lite board used in the experiments is realised through the AD1836 multichannel codec with six DACs and four ADCs limiting the maximum number of con-



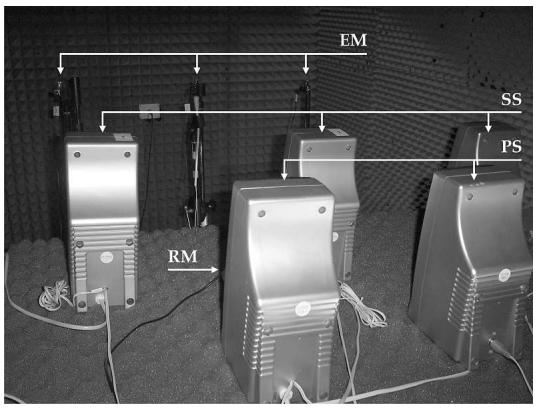


Figure 4. Laboratory setup for the three-channel ANCS experiment (PS: primary speakers, SS: secondary speakers, RM: reference microphone, EM: error microphones).

trol channels implemented in the laboratory to three. Additional daughter cards based around the same AD1836 codec are available on the market and could be used to extend the number of channels.

Since the operating range of the ANC is 150 Hz to 2 kHz, the sampling rate selected for this application was $f_s = 4$ kHz resulting in a sampling time of $T_s = 0.25$ ms. The number of instructions required for the implementation of the multichannel FXLMS and FILMS algorithms increases with the increase in the filter length which calls for highly optimised code to satisfy the real time constraint of a 0.25 ms computation time. It is important to note that the electrical delay from the reference input to the loudspeaker should be less than the acoustic delay, i.e. the computation time for the "anti-noise" should be a small fraction of the sampling interval.

All the experiments were performed in the small rectangular non-anechoic room with dimensions 3.10 m × 2.00 m × 1.85 m. The three-channel ANCS was arranged in the parallel-walls geometry arrangement as illustrated in Figure 4. The system was placed at 1.20 m distance away from the floor.

To simulate the primary source, a pair of standard active PC stereo speakers was used and positioned 0.25 m from each other. A multi-sinusoidal test signal (to be cancelled), containing five frequencies 200, 300, 400, 500 and 600 Hz was generated by the RealTime Analyzer software package and broadcast to the primary speakers using a standard PC.

The same type of active PC stereo speakers was used for the three cancelling secondary sources providing satisfactory low frequency response in the range of interest. These speakers were placed 0.25 m in front of the primary speakers and connected to the output channels of the DSP board. The signals used to drive the speakers were mono signals so that each mono speaker was connected to one audio output on the DSP board.

Reference and error signals were captured using unidirectional condenser microphones with a flat frequency response between 20 Hz to 20 kHz, connected to the inputs of the DSP board. A custom-built, battery-powered preamplifier circuit was used to condition the microphone signals. The preamplifier output was connected to an amplifier and filter circuit. The error microphones were placed at 0.95 m from the primary speakers. Separation distance between the error microphones as well as distance between the secondary speakers was 0.35 m. In order to avoid potential feedback from the secondary sources, the reference microphone was optimally placed in front of the primary source at a distance 0.08 m.

The sampling frequency used during the experiments was 4 kHz. This was achieved by down-sampling signals acquired from the I/O of the DSP board since the original sampling frequency of the DSP board was fixed to 48 kHz. To avoid signal aliasing, low-pass filters with 2 kHz cut-off frequency were used before the Analog-to-Digital Converters (ADCs) and after the Digital-to-Analog Converters (DACs).

The error microphone data was recorded using the VisualDSP++ software. SPLs at other positions in the room were monitored and recorded using RealTime Analyzer software and a standard PC microphone. A digital SPL meter (Lutron SL-4001) was also used to monitor the performance of the ANCS elsewhere in the room.

4. Inverse system estimation

In order to implement the MFILMS-based system described with the equations given in section 2, it is necessary to estimate a complete system plant and its inverse.

An adaptive system identification technique based on the LMS update of the adaptive filter weights is used to estimate the response of the secondary paths in the proposed ANCS. A single-channel adaptive system estimation technique using an LMS estimator is discussed in detail in [1]. To speed up the plant estimation process, the multichannel system estimation approach [1] can be adopted. Both single and multichannel estimation schemes have been implemented in the laboratory resulting in almost identically estimated multichannel system responses.

In addition to multichannel plant estimation, FILMS-based ANCS also requires the estimation of the inverse plant – inverted versions of the measured secondary path responses. The inversion of the system transfer function is the key point of the equalising procedure in many DSP

applications, and a number of advanced techniques were developed to this end [18, 19, 24, 25]. However, almost all of those techniques produce a large number of FIR filter coefficients, and thus cannot be implemented in real-time on a low-cost DSP.

The calculation of the inverse of a large convolution matrix is a complex and time consuming operation with computational requirements far too high to be implemented in real-time on a low-cost DSP. Furthermore, due to the accumulation of truncation and rounding-off errors the result can become worthless. An alternative approach employing the LMS adaptive algorithm to obtain optimal (Wiener) solution [26] has been used instead. Single and multichannel versions of this approach are described in the rest of this section.

Figure 5 shows the system setup for the estimation of the secondary path inverse in the single-channel configuration. An adaptive filter $W(z)$, drives the error between its output $y_w(n)$ and delayed input signal $d(n)$ to zero forming the inverse model of the secondary path $S(z)$ between the loudspeaker and error microphone. Delay D, used in order to obtain a stable non-causal inverse filter, needs to be large enough to account for the propagation delay of the secondary path $S(z)$, D_s, and the adaptive (inverse) filter $W(z)$, D_h, i.e. $D \geq D_s + D_h$. If delay D is smaller than the overall delay $D_s + D_h$, the system will be unstable and the algorithm would diverge without ever finding the optimal solution for the inverse plant. Providing a suitable choice of delay D and the length of the adaptive filter $W(z)$, L_w, a good delayed inverse of the secondary path can be estimated using this setup [27, 28].

To speed up the process of the inverse plant estimation a multichannel inverse plant estimation scheme can be considered. Two potential schemes for the estimation of the multichannel inverse plant have been implemented and tested.

The two-channel pre-equaliser structure combined with cross-cancellation technique proposed by Nelson [20] estimates four inverse filters that can be employed in the CMFILMS algorithm. This scheme is shown in Figure 6a. Another approach shown in Figure 6b uses the two-channel setup based on the FXLMS configuration in order to estimate only two equalising filters that can be used in the SMFILMS algorithm. This scheme avoids the individual inverse estimation for each secondary path as obtained filters include the influence of both direct and cross paths for each actuator used in the system. Results achieved using the described inverse plant estimation schemes are presented in the next section of this paper.

5. Experimental results

Two and three-channel FXLMS, CSEMFILMS and SSEMFILMS algorithms were implemented in the laboratory and the results of those experiments are presented in this section. The direct and inverse single-channel estimation approaches described in section 4 were used to obtain the

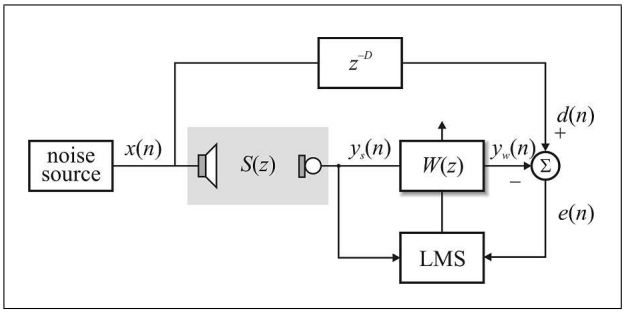


Figure 5. Block diagram of direct single-channel inverse estimation.

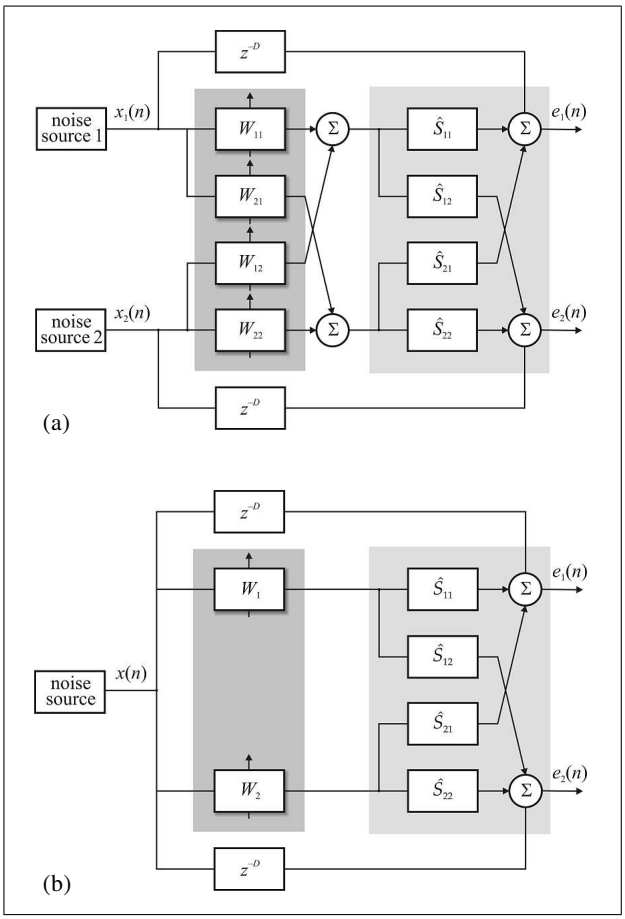


Figure 6. Block diagram of the two-channel indirect inverse secondary path estimation (a) scheme with four inverse filters, (b) scheme with two inverse filters.

impulse responses of the secondary paths and their inverses for use in FXLMS and FILMS algorithms. Multichannel estimation approaches were also tested, resulting in a similar set of results. The lengths of the impulse and inverse responses were $L_s = 200$ and $L_h = 100$ taps respectively. The responses of the first three cancelling paths in the three-channel ANCS setup are shown in Figure 7. Those are the responses between the loudspeaker 1 and the error microphones 1, 2 and 3 from Figure 4. Corresponding inverses are shown in Figure 8.

Time evolution of the signals measured at the error microphone 2 from the system and learning curves for two-

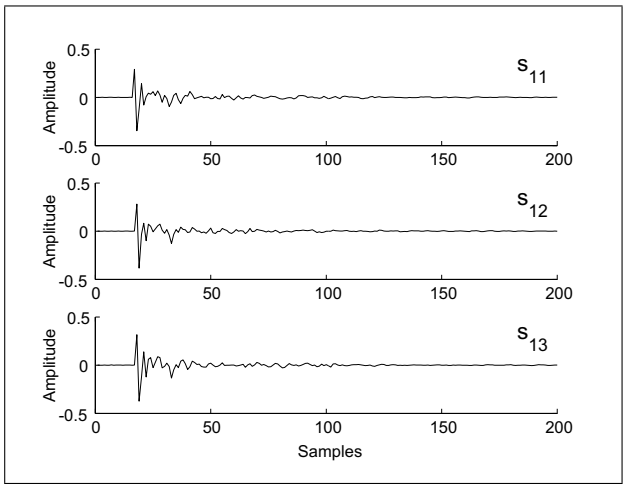


Figure 7. Estimated secondary path impulse responses between loudspeaker 1 and microphones 1, 2 and 3 of the three-channel ANCS.

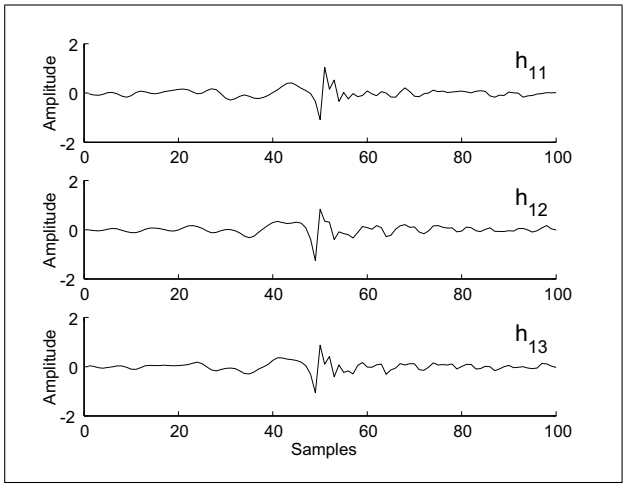


Figure 8. Estimated inverse filter responses between loudspeaker 1 and microphones 1, 2 and 3 of the three-channel ANCS.

channel FXLMS, CSEMFILMS and SSEMFILMS algorithms are presented in Figure 9. Faster convergence of two tested variants of the FILMS algorithm, compared to FXLMS algorithm can be clearly identified on this figure where the error signal is plotted in the time domain. The CSEMFILMS algorithm reaches its steady state after approximately 5000 iterations (1.25 s), SSEMFILMS after 15000 iterations (3.75 s) and FXLMS after 25000 iterations (6.25 s).

To assess the steady state performance of those algorithms and the level of residual signal on the error microphones, the error signal measured before and after cancellation for the three-channel ANCS is plotted in Figure 10 in both time and frequency domains. In addition to the higher convergence rate compared to that of the FXLMS algorithm, two implemented variants of FILMS algorithm can also achieve lower error levels at the microphones after the convergence phase. The two higher frequencies in the primary signal (500 Hz and 600 Hz) are not cancelled using the FXLMS. Increasing the number of adaptive filter

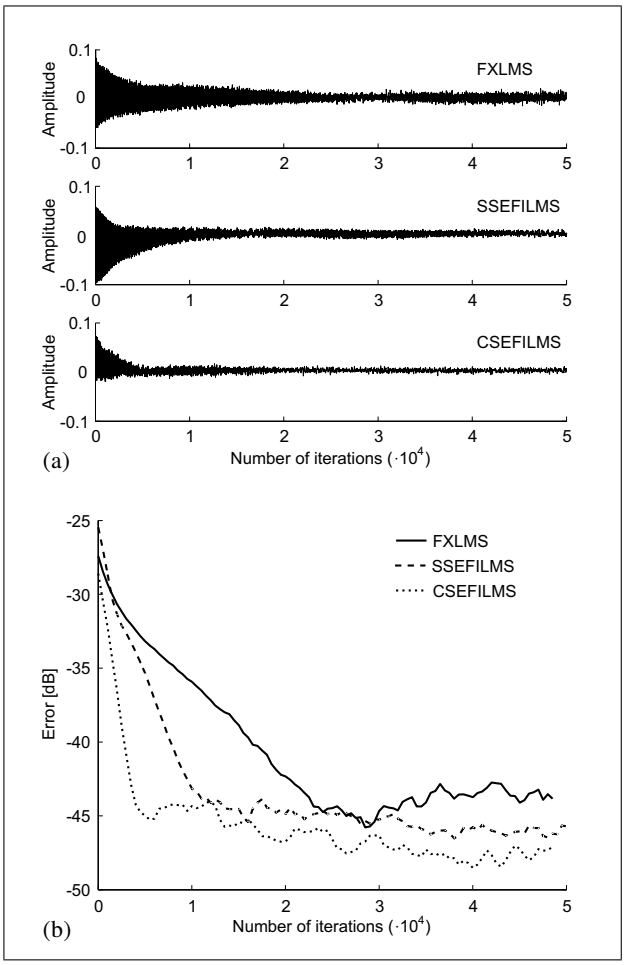


Figure 9. Comparison of the two-channel FXLMS, SSEMFILMS and CSEMFILMS algorithms: (a) time evolution of the error signal and (b) learning curves.

taps in the FXLMS configuration to $L_w = 100$ taps did not improve its performance, resulting only in a further deterioration of the convergence rate. The CSEMFILMS, for both configurations (two- and three-channel), completely cancelled all five frequencies in the primary signal at the measurement position of the second error microphone. A similar performance is achieved for an even smaller number of filter taps ($L_w = 40$ and 60) than used in this algorithm. In order to compare the convergence rates of those algorithms, the same number of adaptive filter taps was used for each algorithm to ensure that the convergence is not affected by this parameter. Although the SSEMFILMS algorithm results in the similar levels of attenuation for each frequency as the FXLMS, it is much less computationally demanding and faster than FXLMS.

Finally, the average attenuation at each frequency, measured at the successive control points in the room separated by a distance of 0.17 m across the x-axes, is shown in Figure 11 for the single-channel FILMS and the three-channel CSEMFILMS algorithms. The measurement was performed using the standard PC microphone and RealTime Analyzer software. The attenuation is measured without weighting factors, i.e. for the flat frequency re-

Table II. Attenuation in dB measured at the positions of error microphones for the single- and three-channel ANCSs.

Channel no.	Position	Frequency [Hz] 200	300	400	500	600	ALL
1-channel	EM3	42.6	14.0	11.2	-0.8	10.5	10.3
	EM1	30.9	31.5	34.0	22.5	13.6	19.9
	EM2	17.6	14.4	9.0	-0.9	3.3	6.4
3-channel	EM3	23.9	30.4	23.0	15.4	20.5	17.0
	EM1	26.0	46.8	23.8	19.2	12.9	17.0
	EM2	27.2	35.1	22.7	12.2	20.4	13.1

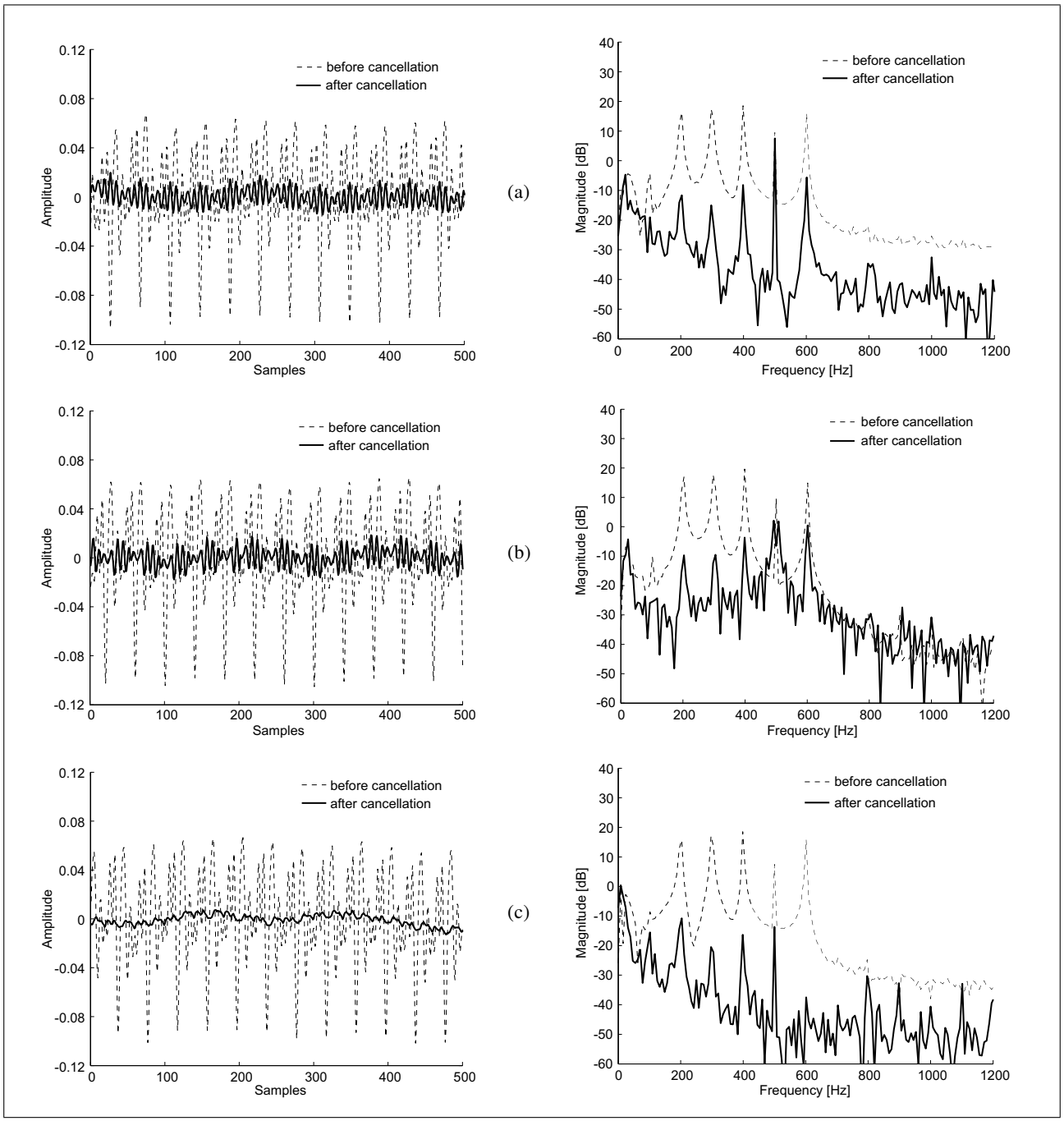


Figure 10. Error signal before and after cancellation in time (left) and frequency (right) domain for the three-channel. (a) FXLMS, (b) SSEMFILMS and (c) CSEMFILMS algorithms.

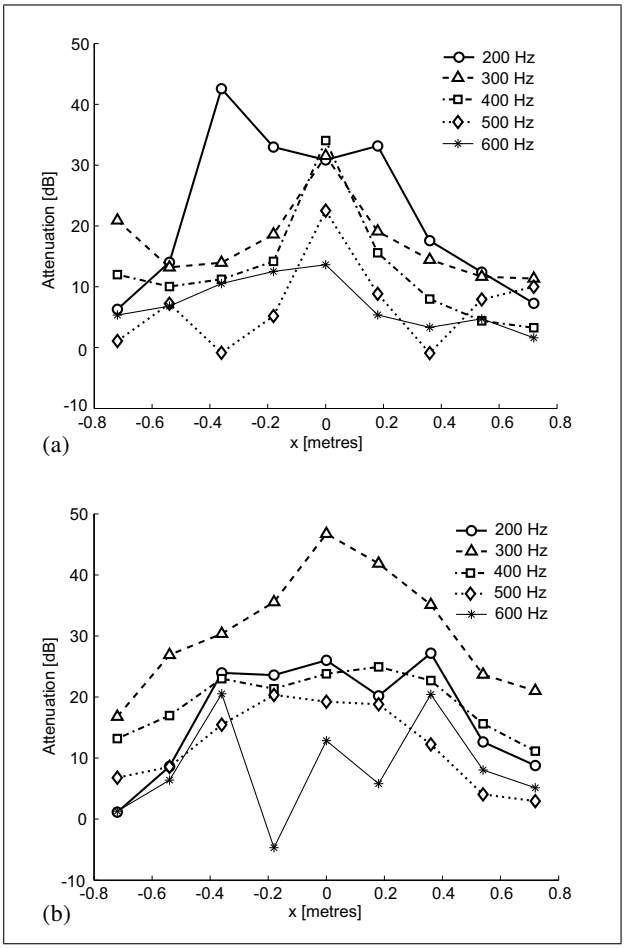


Figure 11. Attenuation at each frequency measured along the axis with error microphones (a) single-channel FILMS and (b) three-channel CSEMFILMS.

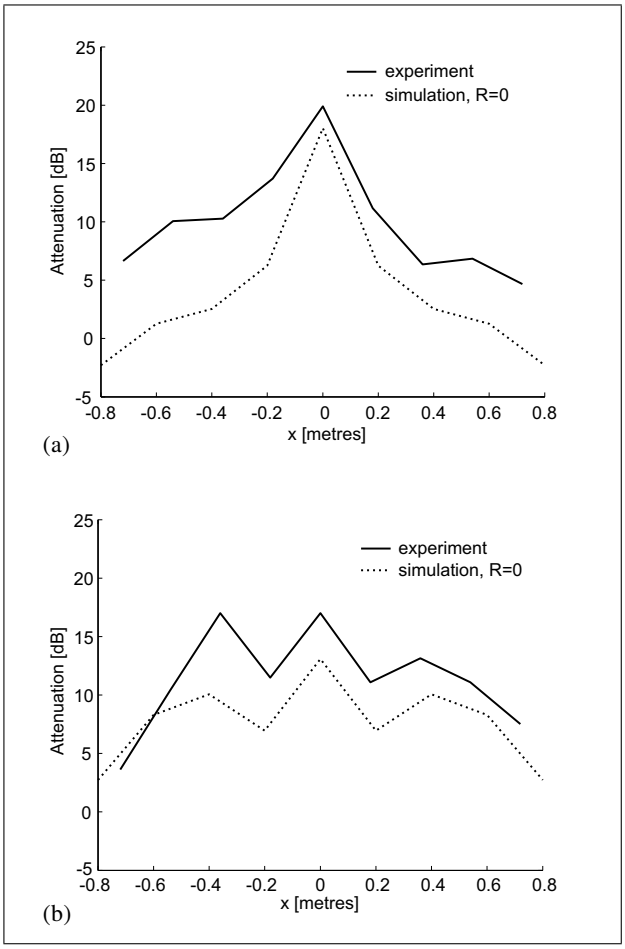


Figure 12. Total attenuation measured along the axis with error microphones (a) single-channel FILMS and (b) three-channel CSEMFILMS.

sponse of the PC microphone. In the single-channel case, the levels of attenuation are higher in the vicinity of the error microphone 1 but drop rapidly further away from this position. This is especially noticeable for the frequency of 500 Hz where a small increase of SPL after cancellation occurred at the positions of the other two microphones (2 and 3), not used in the single-channel experiment. Therefore, destructive interference at the position of microphone 1 results in constructive interference at the positions of the other two microphones, since the separation distance between the error microphones is equal to half of the wavelength of this sound ($0.5\lambda = 0.34\,\mathrm{m} \approx 0.35\,\mathrm{m}$). For the three-channel case, similar levels of attenuation are measured at the positions of all three microphones making the shadow zone even wider for the higher frequencies. The attenuation levels measured at the positions of each microphone are given in Table II. The total attenuation achieved with single- and three-channel systems is plotted in Figure 12.

It can be seen that the attenuation results obtained for one, two and three-channel systems agree with the corresponding results obtained with the mathematical modelling approach described in [29]. Better agreement with theoretical results could be obtained for the measurements performed in a large anechoic chamber where attenuation levels in a wider area around the ANCS could also be measured. This facility was not available for the experimentation.

6. Summary

In the first part of the paper, derivation of multichannel FILMS algorithm is described in detail. Starting with the most general algorithm, CMFILMS, computationally less demanding algorithms have been developed using the partial updates technique (CSEMFILS) and simplified multichannel inverse plants containing only inverses of direct secondary paths (SMFILMS). Both approaches have been combined to produce even more efficient algorithms (SEMFILMS) with inferior performance regarding the convergence rate and minimum mean-square-error.

In this phase of research it was estimated that the benefits of computational savings achieved with modified MFILMS algorithm schemes can outweigh the potential deterioration in system performance and result in an efficient practical system where a large number of control channels is required in an ANC application. To prove this

assumption MFILMS had to be investigated in laboratory conditions.

In order to test the performance of the derived FILMS algorithm and its simplified variants, multichannel FILMS as well as FXLMS-based ANCSs were designed and implemented in the laboratory using the Analog Devices ADSP 21161N EZ-Kit. The system was initially tested and implemented on the ANCS with a low number of channels (2 and 3) due to hardware limitations with the potential to implement more complex systems using MFILMS algorithm in the future.

Single and multichannel methods can be used to estimate the secondary plants in the ANCS. These approaches are well known and investigated in the ANC field and are not discussed in this paper. The multichannel approach although more complex to implement is less time consuming and can therefore be a more convenient solution for systems with high number of channels.

Difficulties in obtaining accurate estimates of inverse plant functions are discussed in more detail since the use of accurate inverse estimates represent a crucial step in achieving both high convergence rates and attenuation levels using MFILMS-based ANCS. Single and multichannel inverse estimation schemes are compared and discussed on the basis of the equalisation error, similar responses in terms of magnitude, flatness and filter order.

Finally, noise cancellation performance of MFILMS-based ANCS was measured and compared with that of the multichannel FXLMS algorithm. The rate of convergence of the FILMS algorithm is comparable with the well-known FXLMS algorithm and the computational load is reduced, providing good estimates of secondary and inverse plants can be obtained during the system and inverse system estimation processes. Various multichannel FILMS configurations are implemented and tested. Results show improved convergence rates and steady state performance of the ANCS which, combined with reduced computational requirements for simplified MFILMS variants raises hopes that a practical ANCS based on the MFILMS algorithm can be designed and applied to noise control problems in the near future.

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