A Multi-Channel Feedforward ANC System Using a Novel FXLMS Algorithm in Solving Classroom Aviation-Noise Problems

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***Abstract*—Adaptive or Active Noise Control (ANC) system is now widely used in many applications. In this paper, a novel multi-channel feedforward ANC system is presented to lower the level of the aviation noises which occurred in the KMITL University's classrooms. A system is simulated in MATLAB/Simulink environment. The proposed multi-channel feedforward ANC system configurations including related equations are described in this paper. The modified Filtered-X Least-Mean-Square (FXLMS) method is used for updating the weight vector in ANC control block diagram in order to achieve the optimal weight vector resulting in minimal errors. How to find bounds on the step size and how to estimate the rate of convergence as well as the steady-state errors of the proposed system with FXLMS algorithm are also elaborated. The secondary-path effects are considered and compensated in this research. The simulation results are presented in terms of noise attenuation capabilities and the Mean-Square-Error (MSE) convergence time. Simulation results and conclusions presented in this paper will be used for further analysis to improve the system performance of the proposed ANC methodology before implementing in the actual classroom.**

***Index Terms*—aviation noise, FXLMS algorithm, MSE convergence time, multi-channel feedforward ANC system**

I. Introduction

Nowadays, noise-polluted problems become one of the critical health-concerning issues due to the expansion of the urban society. People health problems may be caused by noise pollutions depending on the noise levels (dBA), like annoying, causing painful and might lead to hearing impairment. Until now, a couple of passive acoustic noise control solutions have been proposed, for example, sound-absorbing materials, protective barriers and others. Unfortunately, the noise-suppression performances of those passive solutions are quite inefficient.

Noise caused by the aircrafts, or often called aviation noise, is the one of most commonly-found noise sources. King Mongkut's Institute of Technology Ladkrabang (KMITL) University has distinct aviation-noise problems due to the location which located right next to Bangkok Suvannabhumi international airport. Such an airport has aircrafts taking off and landing twenty-four hours daily. In consequences, professors and students always have to deal with aviation-noise problems when having classes caused by aircraft flying-over the university buildings for landing and taking off.

In this paper, we proposed a multi-channel ANC system with FXLMS algorithm to resolve an aviation-noise problem in KMITL's classrooms. All of the actual aircraft/aviation noises used in this research had been gathered from the real KMITL classroom. They will also be graphically analyzed using some advanced Digital Signal Processing (DSP) techniques in this paper.

A. Aircraft Noise Source (Aviation Noise Source)

Today, Suvannabhumi international airport has only two runways, which is capable of serving seventy-six aircrafts per hour. In the year 2039, the plan is to scale-up into four runways in order to expand aviation capacity. Consequently, the more runways it has, the more aircraft noises will become. Fig. 1 in this paper presents Suvannabhumi international airport Noise Exposure Forecast (NEF) information [1] predicting the noise-level contour when operating in full capacity of four runways. The certain airport NEF can be expressed as in equation (1).

$$\mathrm{NEF}_{ij} = \mathrm{EPNL}_{ij} + 10\log_{10}(N_d + 16.67N_n) - 88 \qquad (1)$$

where NEF_{ij} is an effectively-perceived noise level for the aircraft type i and the flight path j, N_d is the number of aircrafts in the day time and N_n is the number of aircrafts in the night time. Then, sound exposure level L_{dn} (in terms of dBA) can be estimated as $L_{dn} \approx \mathrm{NEF}+35$. Fig. 1 illustrates three contours representing three critical areas near the airport affected by aircraft-noise problems. The blue-contour area was predicted to have NEF > 40, which L_{dn} is 75 dBA, the green-contour area was predicted to have *NEF* 35-40, which L_{dn} is 70-75 dBA and the red-contour area was predicted to have *NEF* 30-35, which L_{dn} is 65-70 dBA. KMITL University is located in the green-contour area which means it is quite a critical area affected by aircraft noises.

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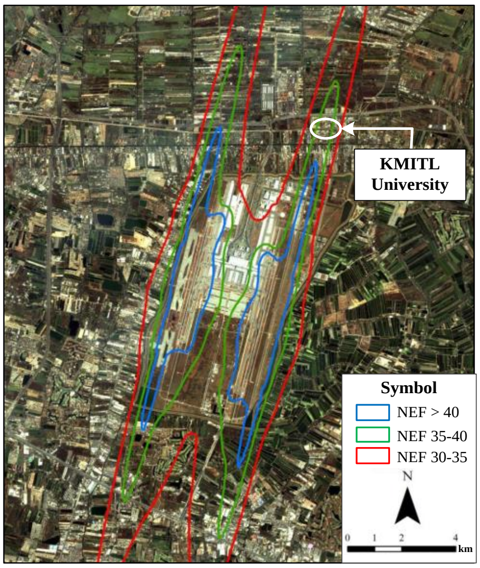

Fig. 1. Suvannabhumi Airport NEF level in 2039 [1].

Fig. 2. KMITL classrooms affected by aircraft noise

B. KMITL Classroom Affected by Aircraft Noises

Fig. 2 presents that KMITL University is located in the radius of four kilometers from the second runway (1R-19L) of Suvannabhumi international airport. Since all the buildings in KMITL University are sitting on the same direction as all the airport runways, it is unavoidable to be directly affected by aviation-noise problems when the airplanes taking off and landing all the times.

Thailand is located in a tropical area which average temperature is above thirty degree Celsius all year round, the installations of in-house air conditions are commons. Therefore, all the buildings in KMITL University have air conditioners installed, which luckily are able to reduce some noise pollutions from outside of KMITL buildings. These circumstances can be counted as indirect passive noise control methodologies. However, when the noise pollution level is too high e.g., aviation or aircraft noises, the indirect passive noise control methodology is insufficient. The more-advanced active noise control (ANC) strategy shall be brought into play in order to control the unwanted noise pollutions to the acceptable level.

Many types of ANC system and research have been widely used to mitigate the noise pollution since the advancement of digital signal processing disciplines.

For example, the active noise control method for an aircraft flying-over sound transmission through an open window, which presented an experimental work on active control of sound transmission through a restricted opening bottom hinged window [2]. Another work is to exploit the underdetermined system in multichannel active noise control for open windows, which is separated the multichannel ANC problem into two subproblems to simplify the ANC system without degrading the noise reduction performance [3]. The active noise control of sound through full-sized open window, which presented the active noise control utilizing acoustic transducers arranged around the open window to generate a secondary incidence noise that destructively interferes with the real noise [4]. A multi-channel feedforward active noise control system with optimal reference microphone selector based on time difference of arrival, which proposed the system with an optimal reference microphone selector to solve the problem that the unwanted noise propagates to the control point faster than the anti-noise [5]. Active sound radiation control with secondary sources at the edge of the opening, which is proposed the implementation of secondary sources at the edge of cavity opening and investigated the active sound reduction performance of the system numerically and experimentally [6].

II. Active Noise Control (ANC)

The ANC system, which is responsible to create a local silent zone, has recently received considerable interest. The first patent on an active noise control was granted to German engineer, Paul Leug, in 1936 [7]. He described a technique for controlling sound by introduction additional sound, as illustrated in Fig 3. When the advent of digital technology did the realization, adaptive ANC systems become possible. The theory of adaptive ANC, which an adaptive algorithm automatically adjusts the ANC device, was established by Widrow in 1975 [8]. After that, ANC systems are widely used in term of acoustic domain.

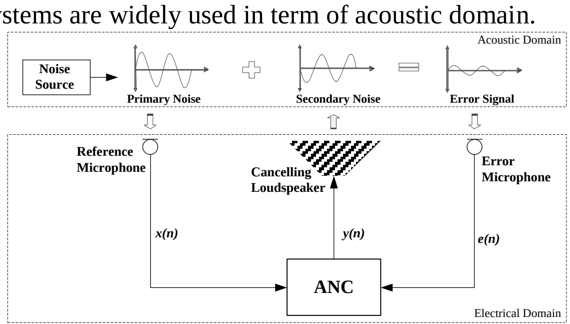

Fig. 3. The principle ANC system by Paul Leug and Widrow [7], [8]

A. Physical Principles of ANC

Sound waves are described by variations in the acoustic pressure through space and time. The evolution of the acoustic pressure as a function of position and time can be described by the wave propagation equation in three-dimensional space as

$$\nabla^2 p(x,y,z,t) - \frac{1}{c}\frac{\partial^2}{\partial t^2} p(x,y,z,t) = 0 \qquad (2)$$

where $p(x, y, z, t)$ denotes the acoustic pressure at position x, y, z and continuous time t, operator ∇^2 is the Laplacian and c is the propagation speed of the sound in the ambient. ANC system is based on the superposition property of acoustic wave, when two sound waves interfere with each other, the sound pressure at time t can be simplified to

$$p(x,y,z,t) = p_1(x,y,z,t) + p_2(x,y,z,t) \tag{3}$$

where $p_1(x, y, z, t)$, $p_2(x, y, z, t)$ are the sound pressure of primary and secondary source, respectively. The sound pressure summation of the two sound waves can be made zero (silent zone), if the two sound waves are equal in magnitude and opposite in phase [9].

B. Feedforward ANC System

A feedforward ANC can be categorized into two types. The first one is a single-channel feedforward ANC system, which consists of a single reference sensor (reference microphone), a single cancelling loudspeaker (secondary source), a single error sensor (error microphone) and the ANC itself. The reference input is picked up by a reference sensor. The ANC functions as a smart controller to compute and generate the estimated signal to the cancelling loudspeaker. The error microphone is then used to monitor the performance of the system. The main objective of the ANC is to minimize the measured error signal and thus the residual acoustic noise.

Another one is a multi-channel feedforward ANC system, which consists of multiple reference sensors, multiple cancelling loudspeakers, multiple error sensors and multiple ANCs. The system acts to reduce noises (to create silence zones) as a single-channel feedforward ANC system. But they can have more silence zone areas because of a greater number of cancelling loudspeakers.

In this paper, a multi-channel feedforward ANC system is presented to increase silence zone areas in one demonstrated KMITL classroom.

III. Multi-Channel Feedforward ANC

From the problem aforementioned in section I and II, we proposed the design and simulation of a multi-channel feedforward ANC system with FXLMS algorithm to perform adaptive coefficient to reduce the aircraft noise flying-over our KMITL classroom.

A. Adative Filter and Its Algorithm

As mentioned, A multi-channel feedforward ANC system consists of multiple references/multiple outputs/multiple errors and multiple ANCs. ANCs are the significant parts of the system that compute the inputs (from reference sensors) to perform the outputs (to drive cancelling loudspeakers). Generally, an ANC composes of adaptive filter and its algorithm. In this paper, we present Finite Impulse Response (FIR) filter and FXLMS algorithm to perform the ANCs.

Fig. 4 shows a block diagram of a multiple-reference/multiple-output feedforward ANC using FXLMS algorithm, where the ANC filter $\mathbf{W}$ has J reference input signals $x_j(n)$ that are elements of signal vector $\mathbf{x}(n)$. The controller generates K secondary signals that are elements of vector $\mathbf{y}(n)$. Therefore, the controller ANC is represented by a $K\times J$ matrix $\mathbf{W}$ (K rows and J columns) with each element of an adaptive FIR filter $W_{kj}(n)$ [10] [11].

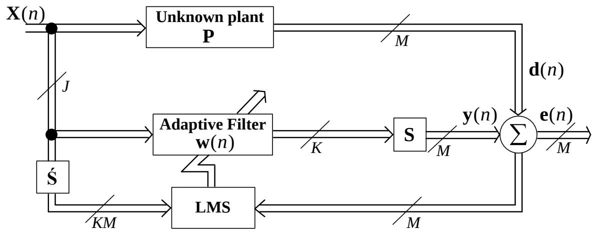

Fig. 4. Block diagram of a multiple-reference/multiple-output feedforward ANC using FXLMS algorithm.

The matrix $\mathbf{S}$, secondary path effects, contains $M\times K$ secondary paths defined in eq. (4), [12], [13].

$$\mathbf{S}(n) = \begin{bmatrix} s_{11}(n) & s_{21}(n) & \cdots & s_{M1}(n) \\ s_{12}(n) & s_{22}(n) & \cdots & s_{M2}(n) \\ \vdots & \vdots & \ddots & \vdots \\ s_{1K}(n) & s_{2K}(n) & \cdots & s_{MK}(n) \end{bmatrix} \tag{4}$$

Typical ANC systems, the $\mathbf{S}(n)$ matrix is not available and will be replaced by the $\acute{\mathbf{S}}(n)$ matrix, which is an estimate of $\mathbf{S}(n)$ using off-line modeling method. Therefore, a matrix of filtered reference signal vectors is defined as

$$\acute{\mathbf{S}}(n) = \begin{bmatrix} \acute{s}_{11}(n) & \acute{s}_{21}(n) & \cdots & \acute{s}_{M1}(n) \\ \acute{s}_{12}(n) & \acute{s}_{22}(n) & \cdots & \acute{s}_{M2}(n) \\ \vdots & \vdots & \ddots & \vdots \\ \acute{s}_{1K}(n) & \acute{s}_{2K}(n) & \cdots & \acute{s}_{MK}(n) \end{bmatrix} \tag{5}$$

Each controller in the matrix $\mathbf{W}$ is represented by $\mathbf{w}_{kj}(z)$, where j is the reference input index and k is the secondary source index. The secondary signal output to the k^{th} secondary source is

$$y(n) = \sum_{j=1}^{J} y_{kj}(n), \; k = 1, 2, \cdots, K \tag{6}$$

$$y_{kj}(n) = \mathbf{w}_{kj}^{T}(n)\mathbf{x}_j(n) \tag{7}$$

$$\mathbf{x}_j(n) \equiv \left[x_j(n), x_j(n-1), \cdots, x_j(n-L+1) \right]^T \tag{8}$$

where $j = 1, 2, \cdots, j$ and L is the length of FIR filter. There are $M\times K$ different secondary paths $S_{mk}(z)$ between the secondary sources and error microphones, which are modeled by $S_{mk}(z)$ to generate an array of filtered reference signals $x'_{jkm}(n)$ for multiple-channel FXLMS algorithm. This algorithm adjusts the coefficients of the $K\times J$ adaptive filters $w_{kj}(z)$ in the controller, which is expressed as

$$\mathbf{w}_{kj}(n+1) = \mathbf{w}_{kj}(n) + \mu \sum_{m=1}^{M} \mathbf{x}'_{jkm}(n) e_m(n) \tag{9}$$

where

$$\mathbf{x}'_{jkm}(n) \equiv \acute{s}_{mk}(n) * \mathbf{x}_j(n) \tag{10}$$

and μ is the step size that determines the stability and the convergence rate of the FXLMS algorithm.

In terms of acoustic domain at error sensor *m*th, determine $d_m(n)$ is the desired signal (noise signal in a classroom), $y_m(n)$ is the estimated signal (cancelling signal) computed from the adaptive filter (ANC system) and $e_m(n)$ is an error signal between $d_m(n)$ and $y_m(n)$ computed as shown in Eq. (11)

$$e_m(n) = y_m(n) - d_m(n) \tag{11}$$

If the adaptive filter output $y_m(n)$ is identical to the desired signal $d_m(n)$. Therefore, when $d_m(n)$ and $y_m(n)$ are acoustically combined, the residual error is

$$e_m(n) = y_m(n) - d_m(n) = 0 \tag{12}$$

which results in perfect cancellation of both sounds based on the principle superposition.

B. *Acoustic Design for a Classroom*

We have a 12-stories classroom building in the KMITL university area, which is high enough to reach the aircraft noise and is located near the second runway of the airport (approximately 4 km).

KMITL classroom in the building is simulated as illustrated in Fig 5.

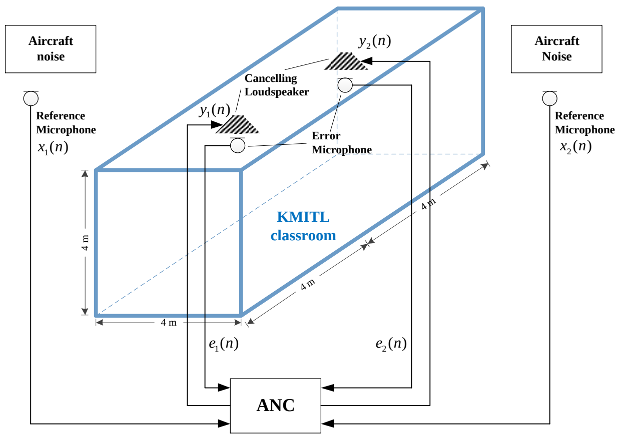

Fig. 5. KMITL classroom with a multi-channel ANC system.

The classroom is a rectangular enclosure of dimension 4m×4m×8m. A multi-channel feedforward ANC system is designed in the classroom by using two reference microphones to measure the aircraft noises, two cancellation loudspeakers to perform secondary sources, two error microphones to measure the residual signals, and an ANC system to compute the input signals from reference microphones for driving secondary sources. We called a 2×2×2 multi-channel ANC system.

In terms of acoustic design, which are identical to development and implementation of a multi-channel active control system for the reduction of road induced vehicle interior noise [14], acoustic devices that will be used in the system to perform silence zone areas must be optimized. Consequently, we will use reference microphones and error microphones whose specification is compatible for frequency response and sensitivity of the noise sources. For the cancelling loudspeakers, they must be suitable for an acoustic design and the dimension of our classroom such as the reverberation, the direction of sound from the cancelling loudspeakers, the location of loudspeakers and the power of loudspeakers.

C. *2×2×2 Multi-Channel Feedforward ANC*

From section A and B, a 2×2×2 multi-channel feedforward ANC system is modeled in the classroom, two-reference signal, two-cancelling loudspeaker and two- error microphone. From (6) and (7), we can compute both of cancelling signals, which drive cancelling loudspeakers, from the ANC system as

$$y_1(n) = y_{11}(n) + y_{12}(n) = \mathbf{w}_{11}^T(n)\mathbf{x}_1(n) + \mathbf{w}_{12}^T(n)\mathbf{x}_2(n) \tag{13}$$

$$y_2(n) = y_{21}(n) + y_{22}(n) = \mathbf{w}_{21}^T(n)\mathbf{x}_1(n) + \mathbf{w}_{22}^T(n)\mathbf{x}_2(n) \tag{14}$$

where the $\mathbf{w}_{11}(n)$, $\mathbf{w}_{12}(n)$, $\mathbf{w}_{21}(n)$ and $\mathbf{w}_{22}(n)$ are the impulse responses of the adaptive filters $\mathbf{w}_{11}(z)$, $\mathbf{w}_{12}(z)$, $\mathbf{w}_{21}(z)$ and $\mathbf{w}_{22}(z)$, respectively.

From (9) and (10), these four adaptive filters, which are performed the ANC system, are updated by the FXLMS algorithm as

$$\mathbf{w}_{11}(n+1) = \mathbf{w}_{11}(n) + \mu \begin{Bmatrix} [\acute{s}_{11}(n)\mathbf{x}_1(n)]e_1(n) \\ +[\acute{s}_{21}(n)\mathbf{x}_1(n)]e_2(n) \end{Bmatrix} \tag{15}$$

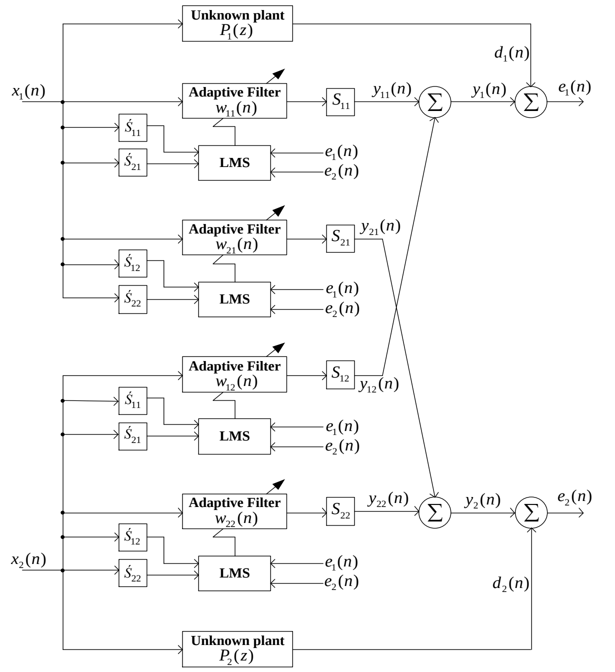

Fig. 6. Diagram of a proposed 2×2×2 multi-channel feedforward ANC

$$\mathbf{w}_{21}(n+1) = \mathbf{w}_{21}(n) + \mu \begin{Bmatrix} [\acute{s}_{12}(n)\mathbf{x}_1(n)]e_1(n) \\ +[\acute{s}_{22}(n)\mathbf{x}_1(n)]e_2(n) \end{Bmatrix} \tag{16}$$

$$\mathbf{w}_{21}(n+1) = \mathbf{w}_{21}(n) + \mu \begin{Bmatrix} [\acute{s}_{11}(n)\mathbf{x}_2(n)]e_1(n) \\ +[\acute{s}_{21}(n)\mathbf{x}_2(n)]e_2(n) \end{Bmatrix} \tag{17}$$

$$\mathbf{w}_{22}(n+1) = \mathbf{w}_{22}(n) + \mu \begin{Bmatrix} [\acute{s}_{12}(n)\mathbf{x}_2(n)]e_1(n) \\ +[\acute{s}_{22}(n)\mathbf{x}_2(n)]e_2(n) \end{Bmatrix} \tag{18}$$

From (13) to (18), we can illustrate a block diagram of the $2\times2\times2$ multi-channel feedforward ANC system as shown in Fig. 6, which is demonstrated how the ANC system is complicated and performed.

IV. Simulation Results

For the simulations, we present results of applying the collected noise through the $2\times2\times2$ multi-channel feedforward ANC system using the FXLMS algorithm to minimize the actual noise gathered from the actual demonstrating classroom that occurred when the time of aircraft taking off and landing.

A. Aircraft Noise

The actual noise used in this simulation, which is the same as we used to simulate a single-channel ANC in our paper [15], is a random aircraft noise recorded from the outside of university classrooms near Suvarnabhumi airport for 5 seconds at the sampling rate 16 kHz. The key characteristic of this noise is shown in Fig. 7. As follows: Fig. 7 (a) illustrates the time-domain plot of the collected noise, Fig. 7 (b) illustrates the frequency-domain plot of such a signal, Fig. 7 (c) presents the power spectrum of the noise in dB/Hz, and Fig. 7 (d) plots the short time Fourier transform (STFT) of such a noise to illustrate the complexity of the interested noise. It can be seen clearly from Fig. 7 that our noise here is highly non-linear with multiple frequencies.

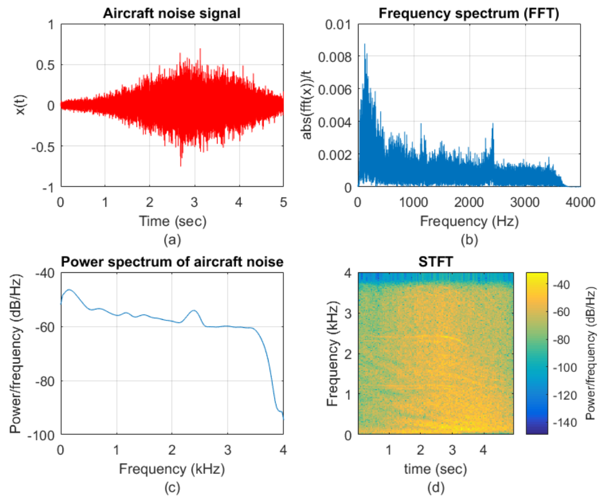

Fig. 7. Aircraft noise characteristics recorded outside our classroom.

B. Multi-Channel Feedforward ANC Simulation Result

The simulation results in this case were done by passing the collected noise through the $2\times2\times2$ multi-channel feedforward ANC system previously shown in Fig. 7.

The aircraft noise acts as input signal $x_1(n)$ and $x_2(n)$ of the ANC system, which the difference of the two input signals is amplitude that is $x_2(n)=0.8x_1(n)$. The error signal $e_1(n)$ and $e_2(n)$ are obtained by summation of $d_1(n)$, $y_1(n)$ and $d_2(n)$, $y_2(n)$ respectively. Fig. 8 shows the simulation results in this case by plotting $x_1(n)$, $d_1(n)$, $y_1(n)$ and $e_1(n)$ in the same time axis. Fig. 9 identically shows $x_2(n)$, $d_2(n)$, $y_2(n)$ and $e_2(n)$.

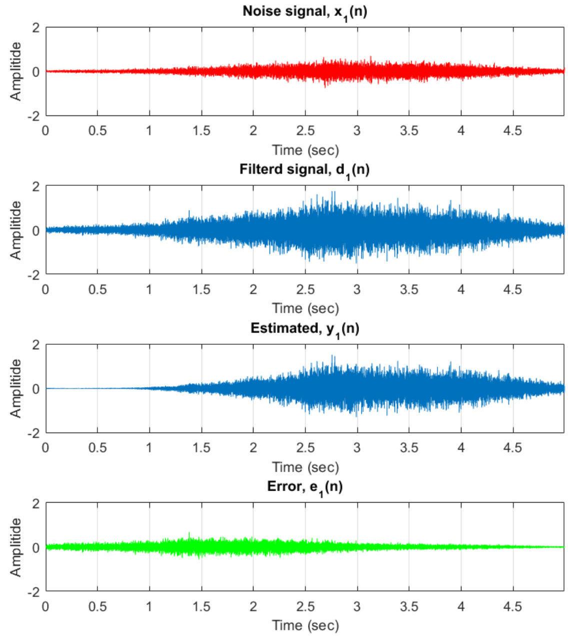

Fig. 8. Simulation results $x_1(n)$, $d_1(n)$, $y_1(n)$, $e_1(n)$ of the ANC system

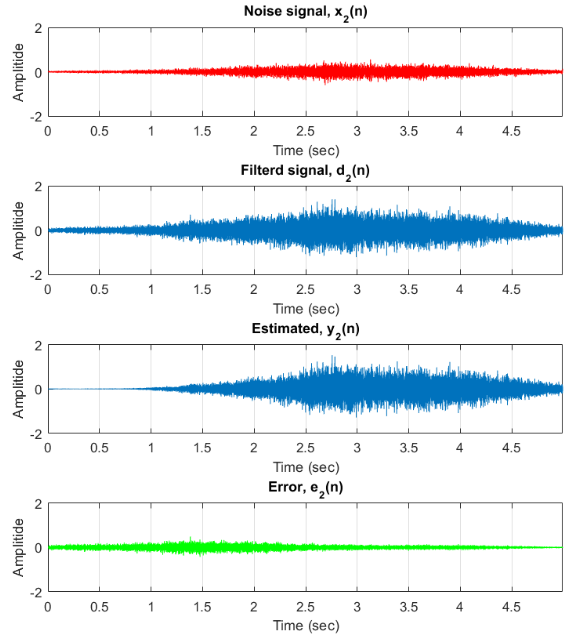

Fig. 9. Simulation results $x_2(n)$, $d_2(n)$, $y_2(n)$, $e_2(n)$ of the ANC system

A variation of the relationships between the length of the adaptive filter (L) and the FXLMS step size (μ) can be concluded that the best practice for this scenario is when using the adaptive filter length 512 and the step size 0.001, respectively.

C. Noise Attenuation Capabilities

Noise attenuation in this paper is analyzed using frequency-domain analysis of the residual signal $e_1(n)$ and $e_2(n)$, defined as

$$C_{e_m x}(\omega) = \sum_{j=1}^{J} C_{e_m x_j}(\omega) \tag{19}$$

where $C_{e_m x}(\omega)$ is the magnitude-squared coherence function between the error signal $e_m(n)$ and reference signal $x_j(n)$. It follows that maximum possible noise reduction by an adaptive feedforward multi-channel ANC system in decibels.

Fig. 10 shows estimated noise attenuation at error microphone $e_1(n)$ and Fig. 11 shows estimated noise attenuation at error microphone $e_2(n)$. The attenuation plots show how the system can attenuate the aircraft noise in each of frequency. For example, in Fig 10, the noise attenuation at $e_1(n)$, at frequency 500 Hz that the multi-channel ANC system can attenuate the noise level about 15 dB, at frequency 1000 Hz that the multi-channel ANC system can also attenuate the noise level about 15 dB. Both of Fig. 10 and Fig. 11 are the final noise attenuation capabilities of the multi-channel feedforward ANC system.

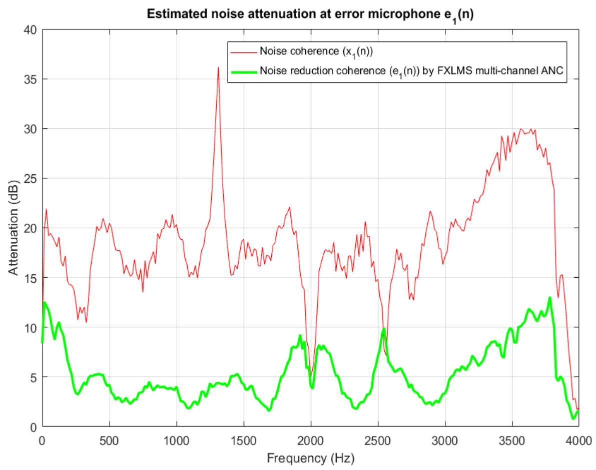

Fig. 10. Noise attenuation $e_1(n)$ from the ANC system

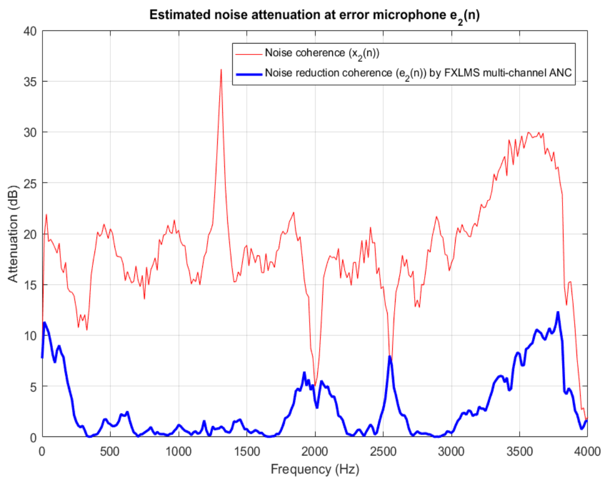

Fig. 11. Noise attenuation $e_2(n)$ from the ANC system

D. MSE Convergence Time

The MSE convergence time of this ANC system is considered when the error signal $e_1(n)$ and $e_2(n)$ are equal to or less than 10^{-4}. As shown in Fig. 12, the convergence time of the multi-channel feedforward ANC system is about in iteration 350000th for both of $e_1(n)$ and $e_2(n)$. For the reason, we can conclude that a multi-channel feedforward ANC system generally requires longer convergence than do a single-channel feedforward ANC system, which we have done in [15].

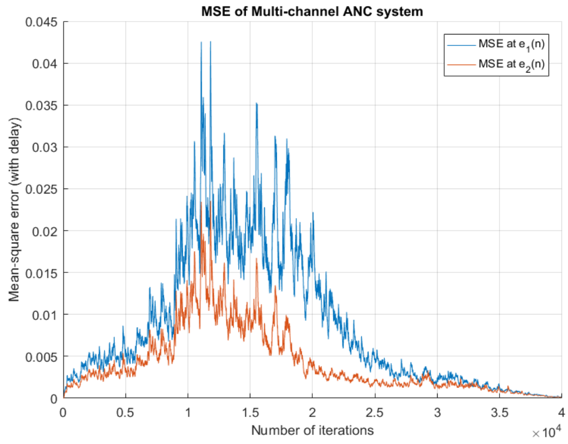

Fig. 12. MSE convergent time of a multi-channel feedforward ANC

V. Conclusion

The conclusion can be made here that for this particular highly-nonlinear aviation-noise problem in KMITL classroom mentioned in this paper, by choosing the optimal key parameters like FXLMS step size and adaptive filter length, the proposed multi-channel feedforward ANC system is far superior to the single feedforward ANC system mentioned in [15] in terms of noise attenuation capabilities. Unfortunately, the MSE convergence time performance of this proposed strategy is slower than the simpler one mentioned in [15] due to the more complicate in FXLMS algorithm versus the previous LMS algorithm, and also the consideration of the secondary-path effects makes the overall system more complex. The simulation results from this paper should be a good guideline for any relevant decision making. However, there are always some rooms for improvement in any circumstance. For further research, different kinds of FXLMS algorithms as well as different configurations of a multi-channel feedforward ANC system could be considered. It is also should be noted here that, in this paper, we neglected some surrounding factors that might affect the system performance like the electromagnetic interference (EMC) and the effect of other electronics devices.

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