A Hybrid Active Noise Canceling Structure

Andres Romero, Edgar Lopez, Mariko Nakano-Miyatake, Hector Perez-Meana

Abstract**—** This paper proposes a hybrid active noise canceling structure (ANC) which significantly reduces the distortion produced by the acoustic feedback present in most ANC systems with a system identification configuration. Here the ANC performance is improved by using two adaptive ANC systems: One with a system identification configuration, used to identify the acoustic path and another one with a predictive configuration which is used to reduce the feedback distortion. Besides the reduction of feedback distortion, an accurate secondary path is quite important, because a poor secondary path estimation will produce unacceptable degradation of the ANC performance. To this end an evaluation of the secondary path modeling algorithms proposed by Bao, Erickson and Zhang, modified to operate with a hybrid structure is also presented. Computer simulation results show that the hybrid structure using a modification of the method proposed by Zhang provides a fairly good noise cancellation performance even in the presence of acoustic feedback.

Keywords**—** Active noise canceling, Bao method, Erickson method, hybrid structure ANC, secondary path estimation, Zhang method.

I. INTRODUCTION

THE need to reduce unwanted environmental noise increases, as the number of industrial and domestic equipments also increases. Among these equipments we have some home appliances, air condition equipment, motors, power generators, car and airplane engines, mechanical vibrations produced by engines in operation, and digging machinery, etc. are just a few examples of processes or equipments that produce noise signals nuisance to human ear. While methods for mitigating these unwanted sounds already have been proposed, most of them based on passive elements, they offer a poor performance when they are used to cancel low frequency sounds. This drawback happens because in these situations the wavelength of the signal is longer than the size of the muffler liabilities. The relevance in the treatment of low-frequency sounds is that they produce fatigue and loss of concentration, affecting not only to the people performance, but also to the machinery and equipment present. That is because low-frequency sounds produce very intense vibrations that can fracture structures during very long periods of exposure [1]. Additionally the ANC Systems must respond to time environment and varying frequency characteristics of the primary noise which must be tracked to get an acceptable noise cancellation performance [1]. A suitable approach to solve this problem, appears to be the active noise canceling approach which is quite similar to the adaptive noise canceling approach proposed by Widrow and Stearns [2], although there are two important differences. Firstly in an ANC the cancellation process is carried out in the acoustic domain, while in adaptive noise canceling it is carried out in the electrical domain, doing it a more difficult task because the adaptive system must operate in a three dimensional space; and second in the ANC the output signal is filtered by the system S(z), as shown in Figs. 1 and 2, which produces a delay of the filter output with respect to its input, where S(z), known as secondary path, represents the effect of the filters, the A/D and D/A converters, loudspeakers, microphone and acoustic path between canceling loudspeaker and the microphone. To get an appropriate cancellation performance S(z) must be estimated in order to avoid performance degradation.

Two different configurations are widely used: The feed-forward and Feedback configurations, shown in Figs. 1 and 2 respectively, both of them with several advantage and disadvantages depending on the noise signal and environmental characteristics. The feed-forward ANC structure (Fig. 1) is able to handle both, narrow band and wide band noise, however in many cases the canceling signal produced by the ANC also reach the input microphone, producing a significant performance degradation. On the other hand the feedback ANC structure (Fig. 2) does not present this problem because this structure only uses the error microphone. However, because this structure generates its own input signal using a linear prediction operation, its performance degrades when the autocorrelation among its samples weakens [3]. Because the feed-forward ANC structure can be used to solve almost any active noise canceling problem, many efforts have been done to solve the feedback distortion problem [4]. Among them the hybrid structures which combine the properties of both realization forms appear to be a desirable alternative.

The filter output signal *y(n)*, as shown in Figs. 1 and 2, reach the canceling point to generate the output error, *e(n)*, through the so called secondary path, *S(z)*, which represents

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the effect of the digital to analog converter, the reconstruction filters, the loudspeaker, the amplifier and the acoustic path from the loudspeaker to the microphone error, as well as the effect of the error microphone and the analog to digital converter. Because the presence of *S(z)* causes that the input and the output error signals of adaptive algorithm be out of phase, to avoid distortion, *S(z)* must be estimated in order that both, the filter output error and input signals be filtered by, in theory, the same system. There are two techniques for estimating the secondary path: the offline and the online secondary path modeling. The first one is carried out using a system identification configuration, where the plant is $S(z)$ and the coefficients of the adaptive filter are the estimated secondary path. This approach performs very well when S(z) is time invariant. However this situation is seldom present in practice. Thus when the secondary path is time varying, an online modeling approach must be used. Because an accurate estimation of S(z) is very important, several approaches have appear in the literature in which a white noise sequence is used for S(z) estimation [5]-[9].

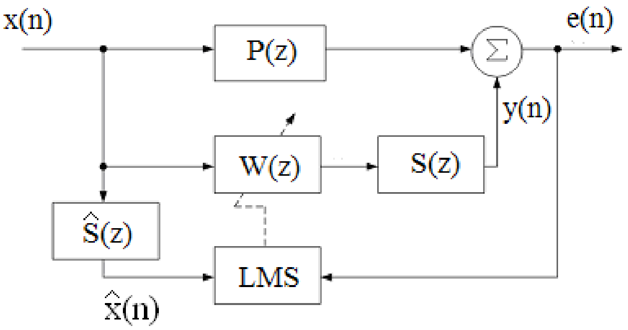

Fig. 1 ANC using a feedforward configuration

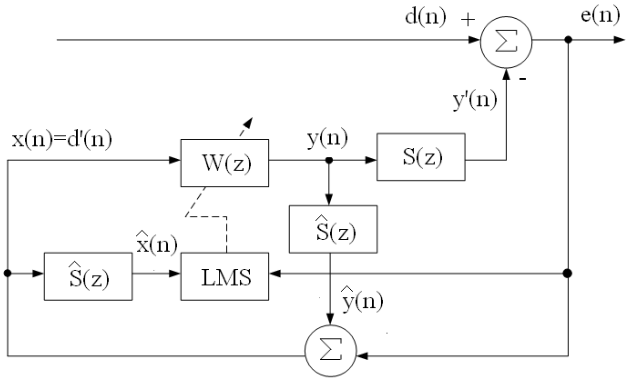

Fig. 2 ANC using a feedback configuration

This paper presents a hybrid structure which consists of a feed-forward structure operating with a system identification configuration, used to estimate the noise path, and a feedback structure with a predictive configuration, used to cancel the feedback acoustic noise. To avoid distortion due to the time varying conditions of, S(z), (Fig. 1) several secondary path estimation algorithms suitable for using in a hybrid structure, based on modification of previously proposed methods, are also presented.

II. HYBRID ACTIVE NOISE CANCELING STRUCTURE

Consider the hybrid ANC output error which is given by

$$E(z) = \left(P(z) + \left(W(z)X(z) + D_S(z)C(z)\right)S(z)\right) \quad , \tag{1}$$

where S(z) is the secondary path, unknown in advance, which must be estimated from the input data, W(z) and C(z) are the feed-forward and feedback sections which are updated using the FxLMS algorithm [1], [4], as shown in Fig. 3.

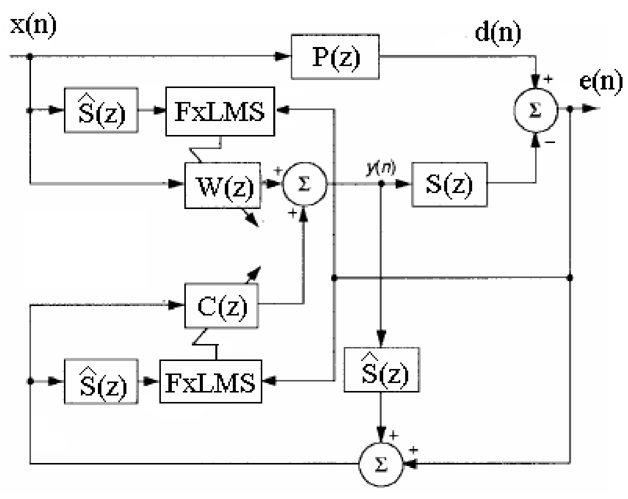

Fig. 3. Hybrid ANC structure

Fig. 3 shows the hybrid ANC structure output error is given by

$$E(z) = \left(P(z) + \left(W(z)X(z) + D_S(z)C(z)\right)S(z)\right) \tag{2}$$

Because an accurate estimation of W(z) and C(z) highly depends on S(z) several algorithms have been proposed to this end. Some of them are describe in the next section.

III. SECONDARY PATH ESTIMATION

The development of secondary path estimation algorithms has been a topic of active research during the last two decades, because the performance of almost all active noise cancelling systems strongly depends on accurate secondary path estimation algorithms. This fact has given as a results the development of several efficient algorithms, some of which are presented in the next subsections operating in a hybrid ANC structure.

A. *Modified Erickson Method*

To analyze the behavior of a modified Erickson method for secondary path estimation, suitable for using in a hybrid ANC, consider the output error given by [5], [9]

$$E(z) = D(z) + Y_S(z), \tag{3}$$

where

$$Y_S(z) = (Y(z) + R(z))S(z) \tag{4}$$

$$Y(z) = W(z)X(z) \tag{5}$$

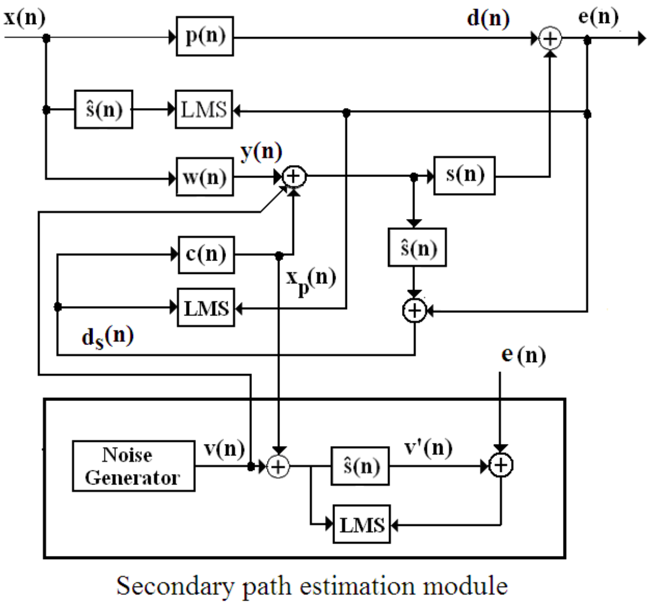

Fig. 4 Modified Ericson method for secondary path estimation

and $R(z)=X_p(z)+V(z)$. Here $v(n)$ is a white noise sequence. Thus, substituting (4) and (5) in (3), with D(z)=P(z)X(z), it follows that

$$E(z) = \big(P(z) + W(z)S(z)\big)X(z) + S(z)R(z) \tag{6}$$

From (6) it follows that the noise $S(z)R(z)$ is also present in the output error, and then the power of $R(z)$ must be small to avoid distortion on the ANC output [9]. On the other hand, as $S(z)R(z)$ increase respect to $X(z)$, the convergence factor should be smaller [9].

Next consider the output error used to estimate $S(z)$, which from Fig. 4 is given by

$$E_S(z) = E(z) - V'(z) \tag{7}$$

$$V'(z) = \hat{S}(z)R(z) \tag{8}$$

$R(z)=X_p(z)+V(z)$ and $V(Z)$ is a white noise sequence. Substituting (6) and (8) in (7) we get

$$E_S(z) = \big(P(z) + W(z)S(z)\big)X(z) + \big(S(z) - \hat{S}(z)\big)R(z) \tag{9}$$

From (9) it follows that the first term of the right side of this equation denotes the additive noise present during the secondary path estimation, which must be larger than $R(z)$. This fact do more difficult the secondary path estimation because, in this situation, the convergence factor must be very small to achieve the required performance [9]. Here $S(z)$ is updated using the LMS algorithm [2]. To solve this problem, which also degrades the noise path estimation, $W(z)$, several proposal have appeared in the literature. Some of them are described in the next subsections.

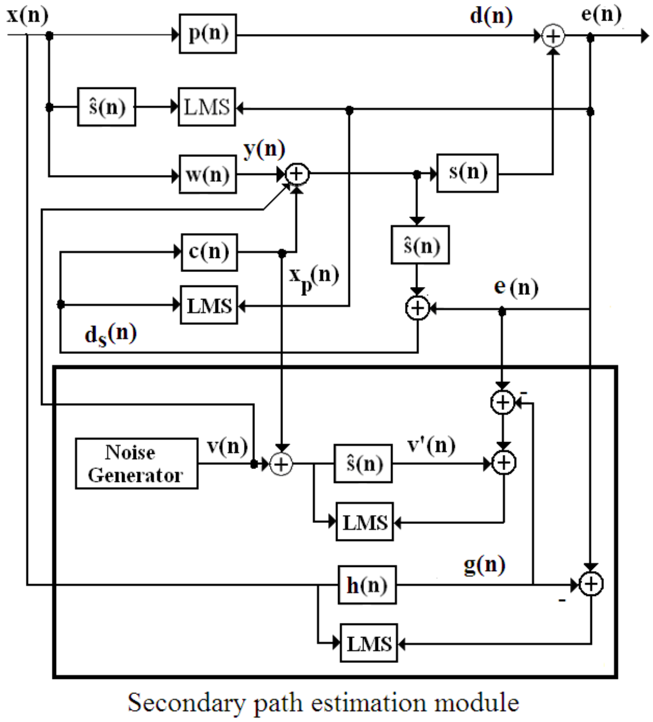

Fig.5 Modified method of Bao for secondary path estimation.

B. *Modified Bao Method*

One of the methods developed to overcome the problems still remaining in the method proposed by Erickson is the Bao secondary path estimation method.

Consider the ANC output error, $E(z)$, shown in Fig. 3 which is given

$$E(z) = D(z) + Y_S(z) \tag{10}$$

where

$$Y_S(z) = (Y(z) + R(z))S(z) \tag{11}$$

$$D(z) = P(z)X(z) \tag{12}$$

$$Y(z) = W(z)X(z) \tag{13}$$

and $R(z)=X_p(z)+V(z)$, where $V(z)$ is a white noise sequence. Then substituting (11)-(13) in (10) it follows that

$$E(z) = \big(P(z) + W(z)S(z)\big)X(z) + S(z)R(z) \tag{14}$$

From (14) it follows that the noise *S(z)R(z)*, is present in the output error and then the R(z) power must be small in order to avoid distortion at the system output. Thus as the power of *S(z)R(z)* becomes larger than *X(z)* the convergence factor must decrease. This condition is contradictory with the conditions required to estimate the secondary path, when a system identification configuration is used, as suggested by Erickson. To solve this problem consider *E(z)* given by [5], [8]

$$E'(z) = E(z) - G(z) \tag{15}$$

where

$$G(z) = H(z)X(z) \tag{16}$$

Substituting (14) and (16) in (15) we get

$$\begin{aligned} E'(z) = &\big(P(z) + W(z)S(z) - H(z)\big)X(z) \\ &+ S(z)R(z) \end{aligned} \tag{17}$$

Next consider the output error $E_s(z)$ which is given by [8]

$$E_S(z) = E(z) - V'(z) \tag{18}$$

where

$$V'(z) = E(z)\hat{S}(z) \tag{19}$$

$$\begin{aligned} E_S(z) = &\big(P(z) + W(z)S(z) - H(z)\big)X(z) \\ &+ \big(S(z) - \hat{S}(z)\big)R(z) \end{aligned} \tag{20}$$

From (17) it follows that, if *H(z)* provides a good aproach of *P(z)+W(z)S(z)*, the error $E_s(z)$ used for estimating *S(z)* becomes

$$E_S(z) = \big(S(z) - \hat{S}(z)\big)R(z) + \varepsilon(z) \tag{21}$$

where $\varepsilon(z)$ is the residual error produced during the *H(z)* estimation. Thus if the additive noise used for *S(z)* estimation is reduced, the power of *R(z)* required will be lower than the power of R(z) required by the Erickson method, producing in this situation less distortion at the ANC output *E(z)*. The adaptive filters used for secondary path estimation S(z) and H(z) are updated using the LMS algorithm.

C. *Modified Zhang Method*

Other approach used to solve some of the limitations existing in the Erickson method in the method proposed by Zhang.

Consider the output error *E(z)* given by

$$E(z) = D(z) + Y_S(z) \tag{22}$$

where

$$Y_S(z) = (Y(z) + R(z))S(z) \tag{23}$$

$$D(z) = P(z)X(z) \tag{24}$$

$$Y(z) = W(z)X(z) \tag{25}$$

$R(z)=X_p(z)+V(z)$, and *V(z)* is a white noise sequence. Substituting (23)-(25) in (22) it follows that

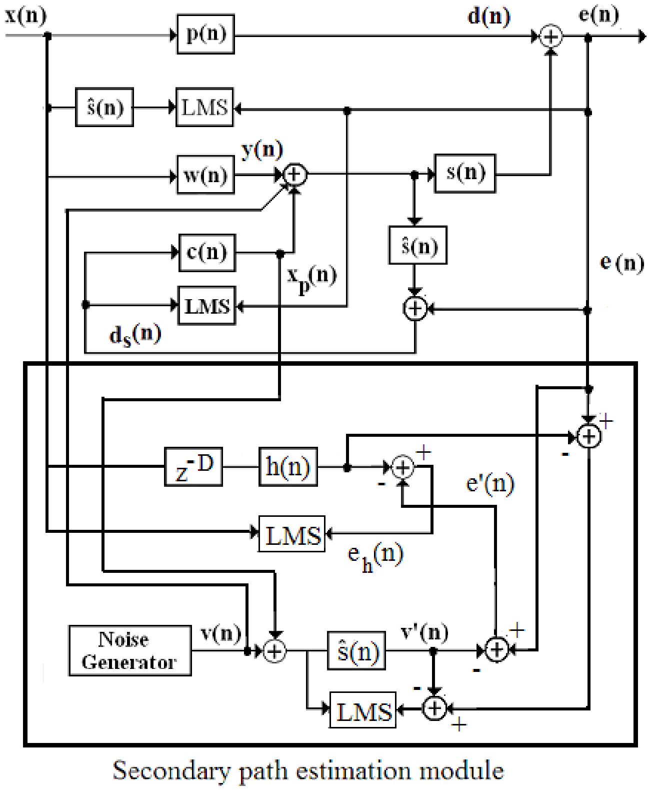

Fig. 6 Modified Zhang Method for secondary path estimation

$$E(z) = \big(P(z) + W(z)S(z)\big)X(z) + S(z)R(z) \tag{26}$$

From (26) it follows that the noise *S(z)R(z)*, is present in the output error and *R(z)* should be small to not distort the output signal. Additionally as *S(z)R(z)* becomes larger than *X(z)* the convergence factor should be smaller. This is opposite to the conditions required for accurate secondary path estimation using system identification configuration. To avoid this problem, consider *E'(z)* which is given by

$$E'(z) = E(z) - \hat{V}(z) \tag{27}$$

$$\hat{V}(z) = \hat{S}(z)R(z) \tag{28}$$

Substituting (28) and (29) in (26) it follows that

$$E'(z) = \big(P(z) + W(z)S(z)\big)X(z) + \big(S(z) - \hat{S}(z)\big)R(z) \tag{29}$$

Next consider $E_h(z)$ which is given by

$$E_h(z) = E'(z) - Y_h(z) \tag{30}$$

where

$$Y_h(z) = z^D H(z)X(z) \tag{31}$$

Substituting (29) and (31) in (30) it follows that

$$E_h(z) = \big(P(z) + A(z)S(z) - z^{-D}H(z)\big)X(z) + \big(S(z) - \hat{S}(z)\big)R(z) \tag{32}$$

Equation (32) shows that the additive noise present in the estimation of H(z) is reduced as compared with Bao method, given by (17). Next consider the output error $E_s(z)$ which is given by [5], [8]

$$E_s(z) = G(z) - \hat{V}(z) \tag{33}$$

where

$$G(z) = E(z) - Y_h(z) \tag{34}$$

Sustituting (26) and (28) in (31) obtenemos

$$G(z) = \big(P(z) + A(z)S(z) - z^{-D}H(z)\big)X(z) + S(z)R(z) \tag{35}$$

Finally substituting (28) and (35) in (33) it follows that

$$E_s(z) = \big(P(z) + A(z)S(z) - z^{-D}H(z)\big)X(z) + \big(S(z) - \hat{S}(z)\big)R(z) \tag{36}$$

From (36) it follows that if $z^{-D}H(z)$ provides a good estimation of $P(z)+W(z)S(z)$, the error $E_s(z)$ used to estimate $S(z)$ I approximately given by

$$E_s(z) = \big(S(z) - \hat{S}(z)\big)R(z) + \varepsilon(z) \tag{37}$$

where

$$\varepsilon(z) = \big(P(z) + W(z)S(z) - z^{-D}H(z)\big)X(z) \tag{38}$$

is the residual error produced by the estimation of $z^{-D}H(z)$ [8]. Thus because the additive noise used for estimation of $S(z)$ can be reduced, the power $R(z)$ may be smaller than the required by the Erickson method, resulting in a smaller distortion in the system output $E(z)$. Again the filters involved in the secondary path estimation are updated using the LMS algorithm.

IV. SIMULATION RESULTS

To evaluate the performance of hybrid noise cancelling together with the online secondary path estimation process, several computer simulations were carried out using actual environmental noise signal.

Figure 7 shows the power spectrum of an airplane motor noise (a) together with the output error power spectrum obtained using a feed-forward system of order 40 (b) and the power spectrum of the output error obtained using the hybrid ANC structure. The noise primary unknown acoustic path is an FIR system of order 40 (c). In both cases (b) and (c) the feedback acoustic path was also of order 40. Here the secondary path is assumed to known.

Figures 8-10 show the convergence performance of the hybrid ANC structures shown in Figs. 4-6, respectively, when it is required to cancel an actual airplane noise. Here the acoustic noise path, P(z), the secondary path, S(z) and the acoustic feedback path are FIR systems of order 40.

Figures 11-13 show the spectrogram obtained when the hybrid structure is required to cancel an actual airplane noise using the methods proposed by Erickson, Bao and Zhang, respectively, for secondary path estimation. The spectrogram of the actual noise signal is shown for comparison in Fig. 14.

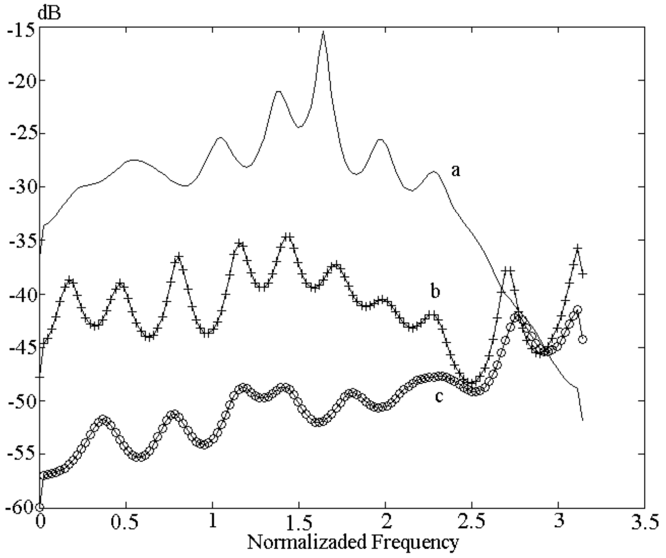

Fig. 7 Power spectral density of (a) airplane noise, (b) power spectral density obtained using a feed-forward structure, (c) Power spectral density obtained using the hybrid structure.

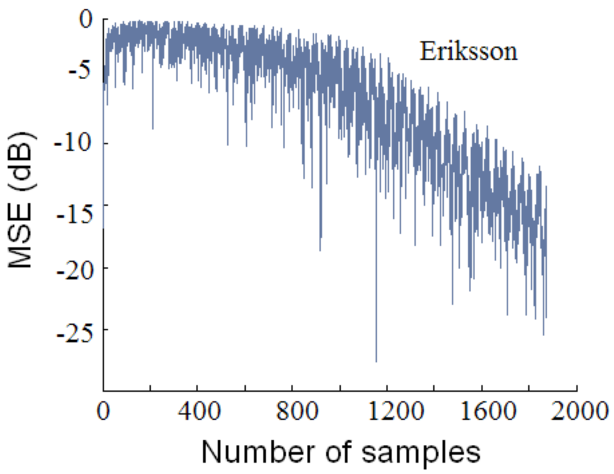

Fig. 8 Convergence performance of the hybrid ANC structure using the Erickson method for secondary path estimation.

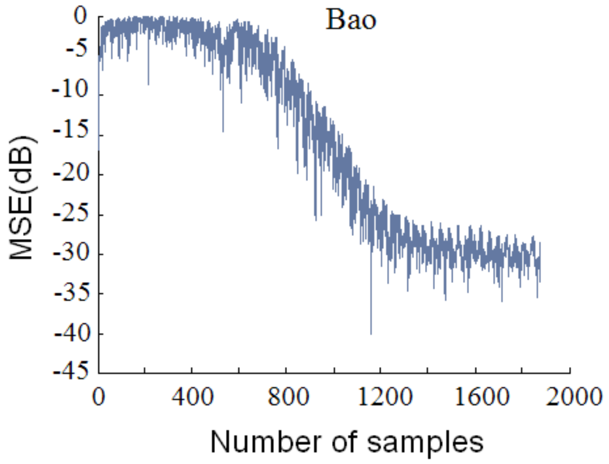

Fig 9 Convergence performance of the hybrid ANC structure using the Bao method for secondary path estimation.

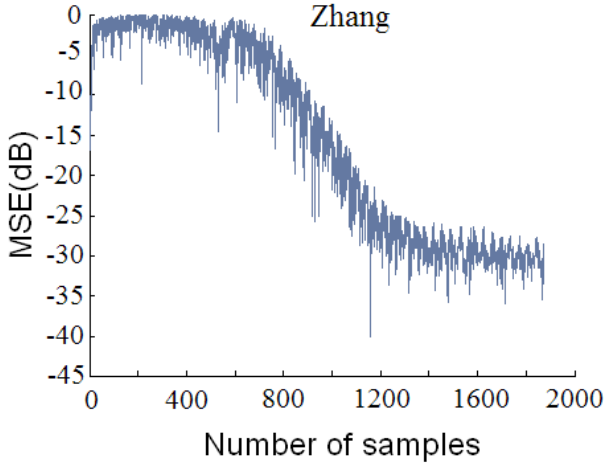

Fig 10 Convergence performance of the hybrid ANC structure using the Zhang method for secondary path estimation.

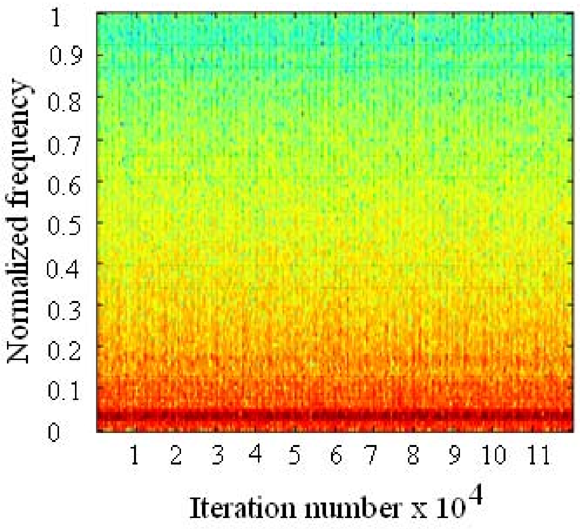

Fig 11 Spectrogram obtained using the Erickson method when required to cancel an actual airplane noise.

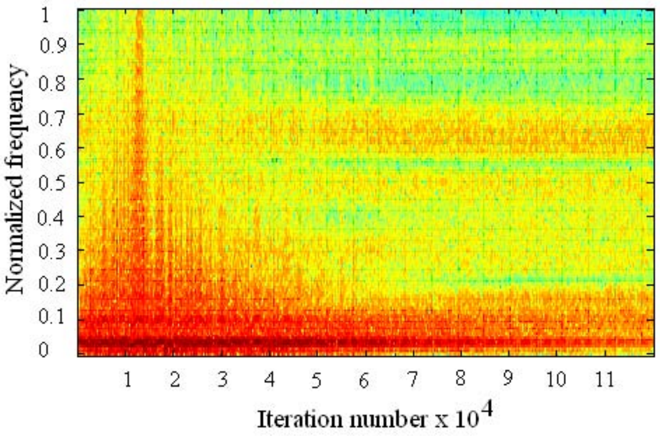

Fig 12 Spectrogram obtained using the Bao method when required to cancel an actual airplane noise.

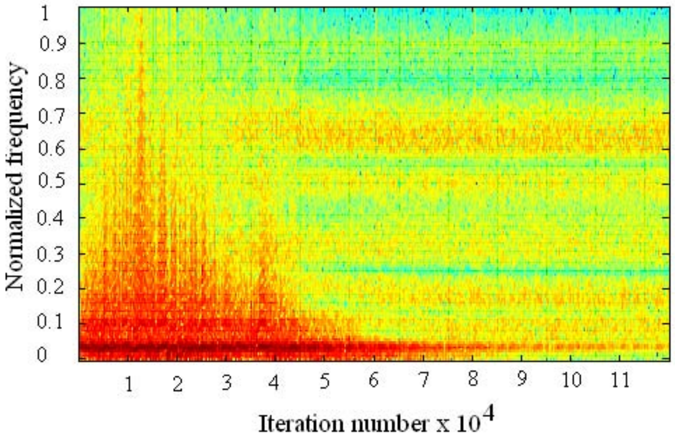

Fig 13 Spectrogram obtained using the Zhang method when required to cancel an actual airplane noise.

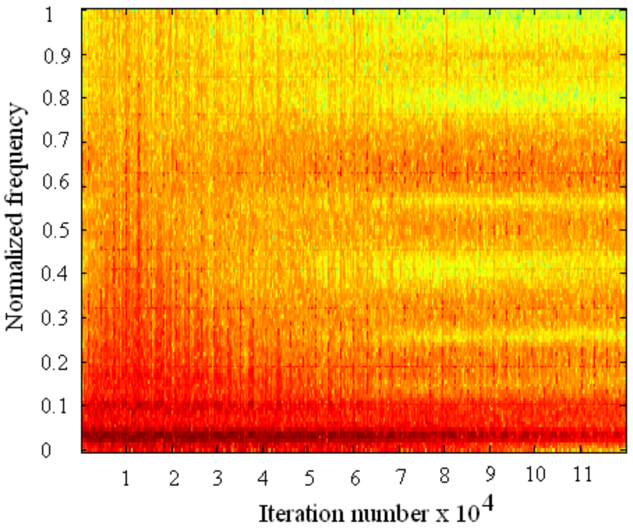

Fig 14 Spectrogram of an actual airplane noise.

V. CONCLUSIONS

This paper presented a hybrid ANC system that combines the feedforward and feedback structures, updated using the FxLMS, to improve the performance of ANC in presence of acoustic feedback distortion. This fact together with the online secondary path modeling allows the system to be adjustable for any kind of secondary path change. Three different secondary path algorithms were analyzed. Computer simulations show that the hybrid ANC system provides an improved performance when the Bao or Zhang methods are used. Finally computer simulation results support the effectiveness of hybrid ANC system

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