An Active Noise Cancelling Algorithm with Secondary Path Modeling

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Abstract: - This paper presents a hybrid active noise canceling (HANC) algorithm to overcome the acoustic feedback present in most ANC system, together with an efficient secondary path estimation scheme. The HANC system provides a solution of two fundamental problems present in these kind of ANC systems: The first consists in a reduction of the acoustic feedback from the cancellation loudspeaker to the input microphone, using two FIR adaptive filters, one with a feedforward configuration an the other with a feedback adaptive filter configuration. To overcome the secondary path modeling problem, a modification of the method proposed by Akhtar is used. Computer simulation results are provide to show the noise cancellation and secondary path estimation performance of presented scheme.

Key-Words.- Active noise canceling, secondary path estimation, feed-forward ANC, feedback ANC, FxLMS, hybrid structure, Akhtar method.

1 Introduction

The need to eliminate unwanted noise is greater, as it is an expression of the limited tolerance that we have as individuals to the perception of sounds generated by industrial equipment, appliances and some general properties that are unpleasant for most people. Generators are just a few examples of processes or equipments that produce signals nuisance to human ear. Mechanical vibrations produced by engines in operation, digging machinery and electricity. While methods for mitigating these unwanted sounds already have been proposed, most of them based on passive elements, they offer a poor response to low frequency sounds. This drawback happens [1], when the wavelength of the signal is long compared to the size of the muffler liabilities. The relevance in the treatment of low-frequency sounds is that they produce fatigue and loss of concentration, thus affecting the people performance, machinery and equipment present. That is because low-frequency sounds produce very intense vibrations that can fracture structures during very long periods of exposure.

An adaptive filter responds to changes in its parameters, like for example: its resonance frequency, input signal or transfer function that varies with time. This behavior is possible due to the adaptive filter coefficients vary over time and are updated automatically by an adaptive algorithm.

Therefore, these filters can be used in applications where the input signal is unknown or not necessarily stationary. An adaptive filter is made up of two parts: a digital filter and an adaptive algorithm. The block diagram of an adaptive filter is shown in Fig. 1, we can see that the adaptive algorithm needs, two inputs signals, $x(n)$ and $e(n)$ as its references to set the parameters of the digital filter and update its coefficients.

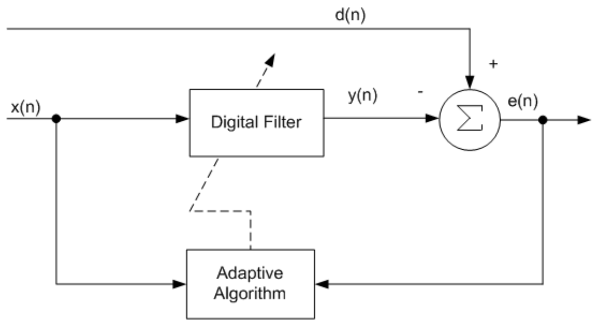

Figure 1 Adaptive System

On other hand, 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:

Consider the ANC shown in Figs. 2 and 3 which are similar to the well known adaptive noise canceling 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; and second the ANC output is filtered by a system S(z), as shown in Figs. 2 and 3, which produces a delay of the filter output with respect to its input. S(z), known as secondary path, which represent the effect of filters, A/D and D/A converters, loudspeakers, microphone and acoustic path between canceling loudspeaker and microphone must be estimated in order to avoid performance degradation.

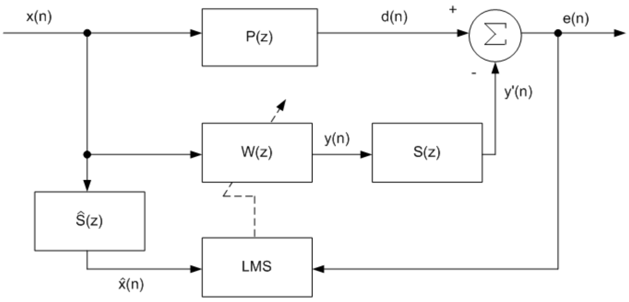

Figure 2 Feedforward ANC System with FXLMS Algorithm

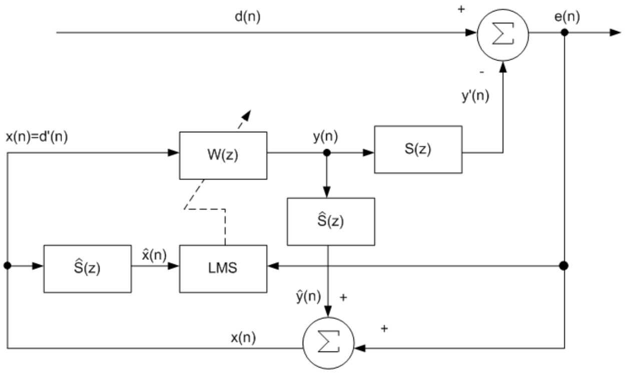

Figure 3 Feedback ANC System with FXLMS Algorithm

Two configurations are widely used: the feedforward and feedback configurations, shown in Figs. 2 and 3 respectively, both of them with advantage and disadvantages depending on the noise signal and environment characteristics. The feedforward ANC structure 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. This fact produces significant performance degradation. On the other hand the feedback ANC structure does not present this kind of distortion because this structure does not use any input microphone, as shown in Fig 2. However, because this structure generates its own input signal though a linear prediction operation, its performance degrades when the correlation among its samples weakens [3]. Because the feedforward ANC structure can be used to solve almost any noise cancelling problem, important 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.

As mentioned before, the filter output signal y(n) reach the cancelling point to generate the output error, e(n), through the so called secondary path, S(z), which takes in consideration the digital to analog converter, reconstruction filters, the loudspeaker, amplifier, the acoustic from the loudspeaker to the microphone error, the error microphone, and 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 which are: offline and the online secondary path modeling. The first one is carried out using a Feedforward system where the plant now 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 in practice this situation is seldom present in practice. 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, among them the Eriksson [5] and Akhtar [6], in which a white noise sequence is used for S(z) estimation, are two of the more widely used.

This paper presents a hybrid structure which consists of a feedforward structure, used to estimate the noise path, and a feedback structure, used to cancel the feedback acoustic noise, as shown in Fig. 4. To avoid distortion due to the time varying conditions of S(z), a secondary path estimation algorithm based on the Akhtar method [6] is proposed.

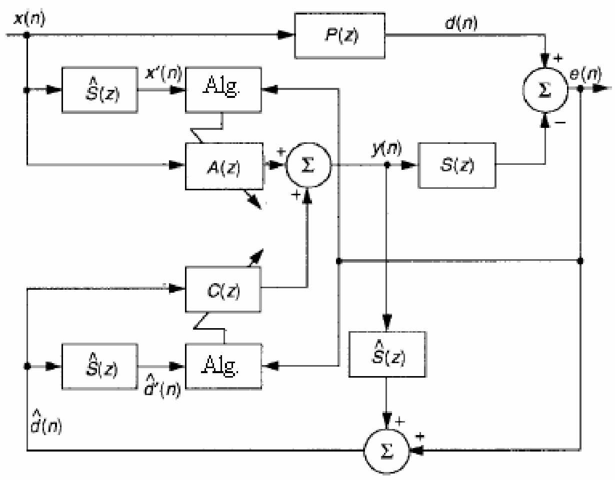

Fig. 4 Hybrid ANC structure

ISBN: 978-960-6766-85-5

ISSN: 1790-5109

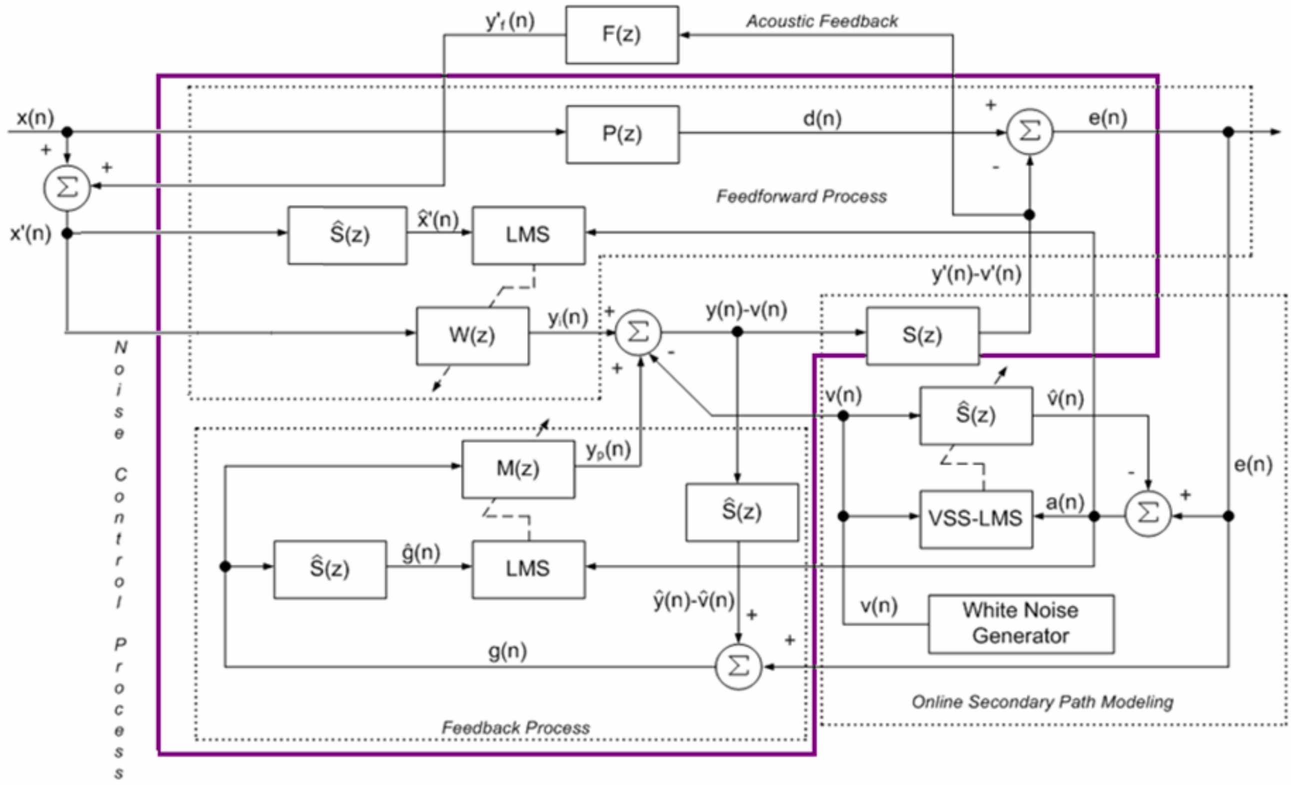

Figure 5 Proposed hybrid ANC structure

2 Proposed ANC Structure

Figure 4 shows the bock diagram of proposed hybrid ANC structure with online on-line secondary path modeling. Proposed hybrid ANC structure consists of a feedforward stage, W(z), which is used to estimate the noise path, P(z), and a predictive structure, M(z), which is used to cancel the distortion due to the acoustic feedback path, F(z). The main idea is that, because the samples of feedback distortion are strongly correlated among them, they can be predicted.

As shown in Fig. 5 the same signal, *a(n),* is used as the error signal to update the adaptive filter, *W(z),* which corresponds to the feedforward stage used to identify the noise path, as well as to update the linear predictive filter *M(z)*, which intends to cancel the distortion produced by the feedback propagation from the canceling loudspeaker to the input microphone thorough the system *F(z),* as shown in Fig. 3; and to estimate $\hat{S}(z)$ that represents the online secondary path modeling adaptive filter. From Fig. 3, we can see that the error signal of all the ANC system is given by:

$$e(n) = d(n) + [v(n) - y(n)] * s(n) \tag{1}$$

where *d(n)* is the desired response, *v(n)* is a white noise signal, *s(n)* is the finite impulse response of the secondary path and *y(n)* is the secondary path input signal used to produce the acoustic noise that achieves the attenuation of the primary noise. Thus from Fig. 3 it follows that

$$y(n) = y_i(n) + y_p(n) \tag{2}$$

$$y_i(n) = \overline{\mathrm{w}}^{\mathrm{T}}(n)\overline{\mathrm{x}}'(n) \tag{3}$$

$$\overline{\mathrm{w}}(n) \equiv [w_0(n), w_1(n), ..., w_{L-1}(n)]^{\mathrm{T}}, \tag{4}$$

$$\overline{\mathrm{x}}'(n) \equiv [x'(n), x'(n-1), ..., x'(n-L+1)]^{\mathrm{T}} \tag{5}$$

$$x'(n) = x(n) + y'_f(n) - v'_f(n) \tag{6}$$

Here (6) denotes the reference signal that already considers the effects of the acoustic feedback. Thus taking in account the acoustic feedback consideration it follows that:

$$y'_f(n) = y'(n) * f(n) \tag{7}$$

$$v'_f(n) = v'(n) * f(n) \tag{8}$$

Both equations (7) and (8) contains $f(n)$, the finite impulse response of the acoustic feedback filter; moreover $y'(n)$ and $v'(n)$ are the signals that have already been filtered by *S(z)*. Next consider the predictive stage whose output signal is give by

ISBN: 978-960-6766-85-5

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$$y_p(n) = \overline{\text{m}}^{\text{T}}(n)\overline{\text{g}}(n) \quad (9)$$

$$\overline{\text{m}}(n) \equiv [m_0(n), m_1(n), ..., m_{M-1}(n)]^{\text{T}} \quad (10)$$

$$\overline{\text{g}}(n) \equiv [g(n), g(n-1), ..., g(n-M+1)]^{\text{T}} \quad (11)$$

$$g(n) = e(n) + \hat{y}(n) - \hat{v}(n) \quad (12)$$

$$\hat{v}(n) = v(n) * \hat{s}(n) \quad (13)$$

$$\hat{y}(n) = y(n) * \hat{s}(n) \quad (14)$$

From (13) and (14) it follows that during the estimation of ANC coefficients vectors it is necessary to used the signals *y(n)* and *v(n)* once both of them already have been filtered by the estimated secondary path $\hat{S}(z)$. Thus an accurate estimation of $\hat{S}(z)$ is required. To this end Akhtar method will be used.

The advantages of using the Akhtar's method [6], [7] for the secondary path modeling in proposed ANC system are reflected in the VSS-LMS algorithm that allows the modeling process to selects initially a larger step size, $\mu_s(n)$ which decreases to a minimum value according with the decreasing of $[d(n) - y'(n)]$. If the filter $W(z)$ is slow in reducing $[d(n) - y'(n)]$, then the step size may remain to small value for more time. Furthermore, the signal $a(n) = e(n) - \hat{v}(n)$ is the same error signal use to update all the adaptive filters involved in the ANC system, $W(z)$, $M(z)$ and $\hat{S}(z)$. The reason to use this signal is that for $W(z)$, $[v'(n) - v(n)] < v'(n)$ compared with the Eriksson's method. So when $\hat{S}(z)$ converges, that is

$$\hat{S}(z) \approx S(z), \quad (15)$$

ideally

$$v'(n) \approx v(n) \Rightarrow v'(n) - v(n) \to 0. \quad (16)$$

Thus, using the Akhtar method [6] for secondary path estimation, the proposed hybrid structure in updated as follows:

$$\begin{aligned}\overline{\text{w}}(n+1) = \overline{\text{w}}(n) + \mu_w \overline{\hat{\text{x}}}(n)[d(n) - y'(n)] \\ + \mu_w \overline{\hat{\text{x}}}(n)[v'(n) - \hat{v}(n)]\end{aligned} \quad (17)$$

$$\begin{aligned}\overline{\text{m}}(n+1) = \overline{\text{m}}(n) + \mu_m \overline{\hat{\text{g}}}(n)[d(n) - y'(n)] \\ + \mu_m \overline{\hat{\text{g}}}(n)[v'(n) - \hat{v}(n)]\end{aligned} \quad (18)$$

$$\begin{aligned}\overline{\hat{\text{s}}}(n+1) = \overline{\hat{\text{s}}}(n) + \mu_s \overline{\text{v}}(n)\left[v'(n) - \hat{v}(n)\right] \\ + \mu_s \overline{\text{v}}(n)[d(n) - y'(n)]\end{aligned} \quad (19)$$

Although (13) shows that when $\hat{S}(z)$ converges, the whole control noise process of the system is not perturbed by the estimation process of $\hat{S}(z)$, it is significant to identify that the online secondary path modeling is degraded by the perturbation of $\eta(n) = \mu_s \overline{\text{v}}(n)[d(n) - y'(n)]$.

3 Computer Simulations

This section presents the evaluation of proposed system by computer simulation using synthetic as well as actual acoustic noise signals. The modeling error for secondary path estimation, was defined as:

$$\Delta S(dB) = 10\log_{10}\left[\frac{\sum_{i=0}^{M-1}[s_i(n) - \hat{s}_i(n)]^2}{\sum_{i=0}^{M-1}[s_i(n)]^2}\right] \quad (20)$$

An offline modeling was used to obtain FIR representations of tap weight length 20 for $P(z)$ and of tap weight length 20 for $S(z)$. The control filter $W(z)$ and the modeling filter $\hat{S}(z)$ are FIR filters of tap weight length of $L = 20$ both of them. A null vector initializes the control filter $W(z)$. To initializes $\hat{S}(z)$, offline secondary path modeling is performed which is stopped when the modeling error has been reduced to -5dB. The step size parameters are adjusted by trial and error for fast and stable convergence.

Figure 6 shows the performance oh proposed system, when the reference signal is a sinusoidal signal of 200Hz. A zero mean uniform white noise is added with SNR of 20dB, and a zero mean uniform white noise of variance 0.005 is used in the modeling process. Figure 7 shows the corresponding curves for the cancellation process. The order of P(z), W(z) and S(z) and F(z) was in all cases equal to 20.

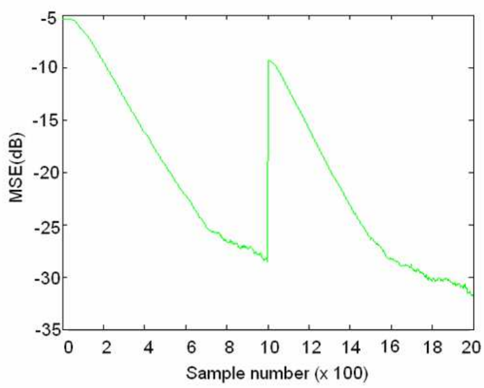

Figure 6 Convergence of performance of secondary path estimation algorithm when the noise to be cancelled is a sinusoidal signal. In iteration 1000 there is an abrupt change on S(z).

Figure 8 and 9 shows the convergence performance of proposed ANC structure when it is required to cancel a narrow band noise signal that consists of 4 sinusoidal signals of frequencies 100, 200, 400, 600 Hz. A zero mean uniform white noise is added with SNR of 20dB, and a zero mean uniform white noise of variance 0.005 is used in the modeling process. Here in the iteration 1000 it is performed an abrupt change with the secondary path. The order of P(z), W(z) and S(z) and F(z) was in all cases equal to 20.

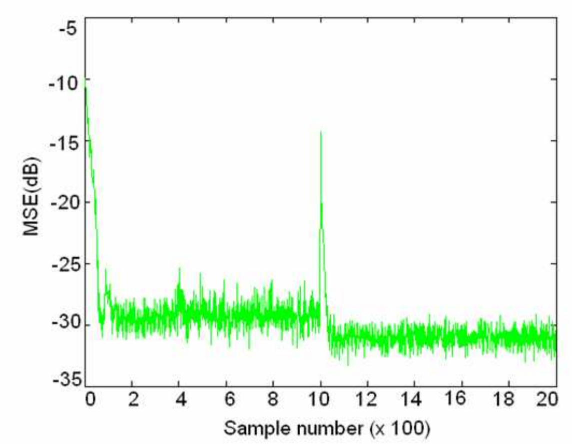

Figure 7 Convergence performance of proposed ANC when the noise to be cancelled is a sinusoidal signal. In iteration 1000 there is an abrupt change on S(z).

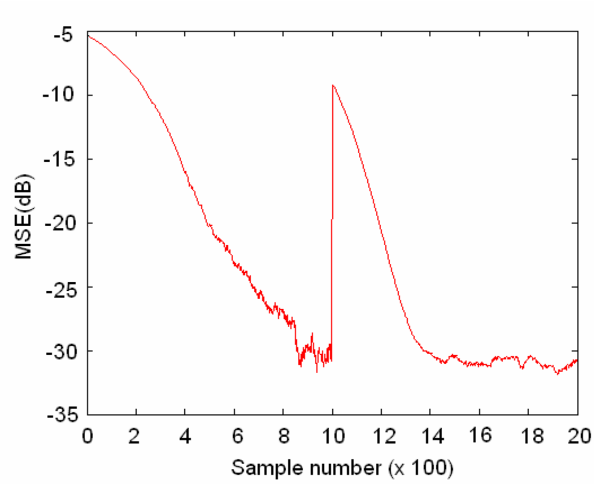

Figure 8 Convergence performance of secondary path estimation algorithm, when the noise to be cancelled is a narrow band noise. In iteration 1000 there is an abrupt change on S(z).

Figures 10 and 11 shows the convergence performance of proposed structure when is required to cancel an actual noise motor signal (mixed band noise). A zero mean uniform white noise of variance 0.005 is used in the modeling process. In iteration 600 it is performed an abrupt change on the secondary path. Simulation results show that proposed ANC performs fairly well when required to cancel both synthetic and actual noise signals.

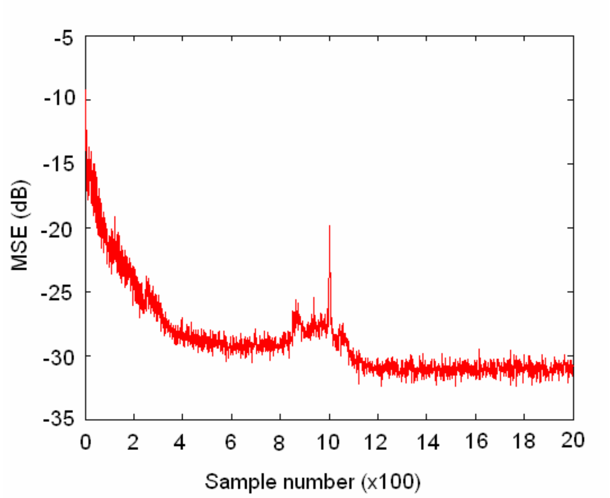

Figure 9 Convergence performance of proposed ANC when the noise to be cancelled is a narrow band noise. In iteration 1000 there is an abrupt change on S(z).

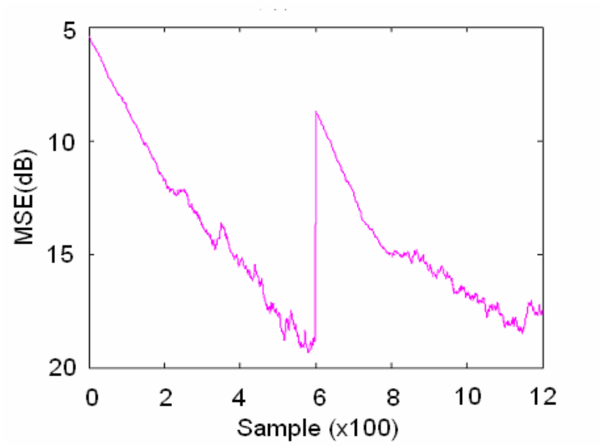

Figure 10 Convergence performance of secondary path estimation algorithm, when the noise to be cancelled is a motor noise signal. In iteration 600 there is an abrupt change on S(z).

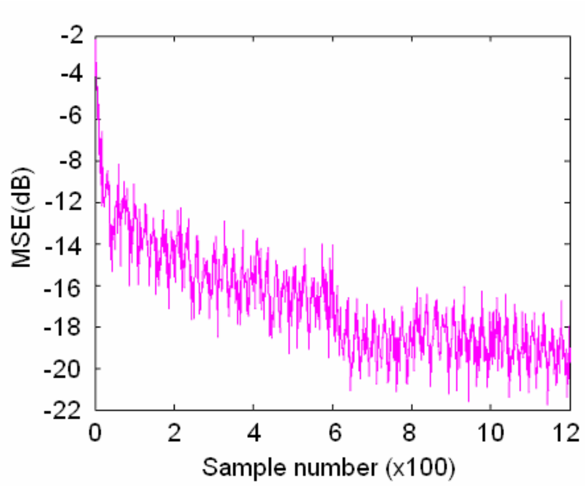

Figure 10 Convergence performance of proposed ANC when the noise to be cancelled is a motor noise signal. In iteration 600 there is an abrupt change on S(z).

4 Conclusions

This paper proposed 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 (gradual ideally). Computer simulations show that proposed system provides an improved performance, at somewhat increased computational cost because the Akhtar's online secondary path method, but this method compensates the noise control process for the feedforward and feedback stages.

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