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Performance of Adaptive Hybrid System in Two Scenarios: Echo Phone and Acoustic Noise Reduction

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Additional information is available at the end of the chapter

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1. Introduction

The adaptive noise cancellation has proved being very efficient method in various practical applications such as voice clearance, recognition systems for voice, hands-free telephony, and medical applications such as hearing aids and fetal electrocardiography [1], etc. Figure 1 [1], depicts the basic principle of noise cancellation (understanding that noise is an unwanted signal, $d(n)$), which is described by main signals that feed the system.

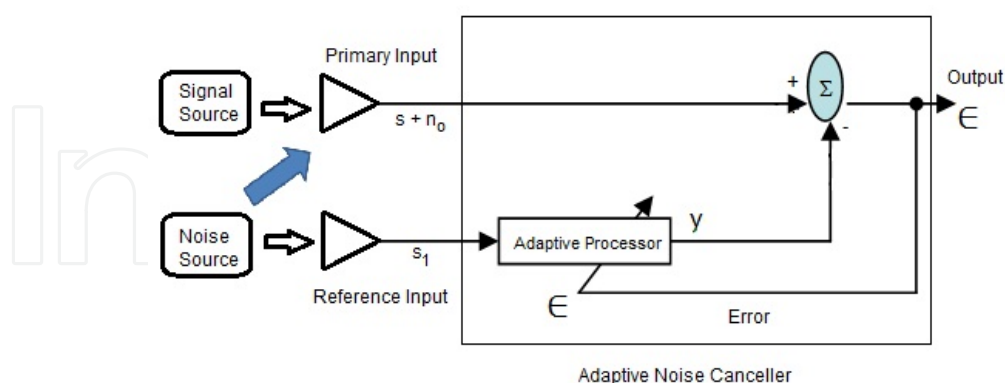


Figure 1. Adaptive noise cancelling approach

Acoustic noise has been studied in recent years due to growing interest in cancelling acoustic noise through active control, since it is increasingly common to find sources of noise in many industrial processes. Basic outlines of noise cancellation were based on the application

of passive attenuators that were used for many years without much success [2], however, development of digital signal processing has become increasingly feasible systems active noise cancellation. Active noise cancellation systems cancel unwanted acoustic noise based on the superposition principle: an acoustic noise of equal amplitude but opposite phase is generated in order to cancel out the unwanted noise.

This work discusses a scheme of active noise cancellation using adaptive algorithms of the digital filters required for the correct operation of the proposed system. The signal generation "anti-noise" to cancel the primary source of noise is a problem different from change of environment, since the signal is generated by electrical means and must be propagated acoustically to have the desired effect; this creates a delayed signal in the generation and propagation, so this change is necessary to calculate the required signal. This work considers the estimation of this modification done "offline" [2].

Hybrid ANC systems correspond to a combination of control structures from the feedback and feedforward systems, where the cancelling signal is generated based on the outputs of both the reference sensor and the error sensor. While the feedforward system attenuates the primary noise, which is correlated with the reference signal, the feedback system cancels the predictable components of the primary noise signal that are not observed by the reference sensor.

As an example of the efficiency of the adaptive hybrid systems, this work evaluates a Hybrid Active Noise Control (HANC) system under feedback acoustic situation. Proposed scheme objective is to compare the performance of HANC versus common references: feedback, feedforward and neutralization systems; the inner nature of HANC gives two main characteristics: on line modeling of secondary path and a good performance under acoustic feedback conditions. In the evaluated system, two least mean square (LMS) adaptive filters are used in the noise control process: one for the feedforward stage and the other for the feedback stage; both of them use the same error signal as used in the adaptation of the modeling filter. Then, the combination of the feedback and feedforward stages, results in a solid robustness for the system in acoustic feedback situation.

This chapter discusses a vital application in telecommunications processes, which is the echo in telephone line and the same time a new proposal: the hybrid structure proposed as a solution to this problem. Finally, the computer simulations are presented to show the success of the proposed system. So, this chapter presents an adaptive hybrid system to resolve the problems described: the noise cancellation using adaptive filtering and one proposal for echo cancellation system. Furthermore, we present a hybrid structure which consists of a feedforward structure, used to estimate the noise path, and a feedback structure, used to cancel the noise, i.e., the unwanted signal: echo in telephony systems or noise signals like conversations, snoring or engines. Hybrid active noise cancellation systems are a good solution to these two important problems, since they have the properties of both the feedforward and feedback systems.

2. Adaptive systems as a solution to problems of signal cancellation

2.1. Adaptive Filtering: Active Noise Control

An adaptive filter responds to changes in its parameters like its resonance frequency, input signal or transfer function that varies with time, for example. This behavior is possible since 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 composed of two parts: digital filter and adaptive algorithm.

One of the most important applications for this kind of system is active noise control (ANC). ANC systems must respond to changes in frequency of the primary noise they want to cancel out. In other words the primary non-stationary noise varies; hence we must use some kind of adaptive system, to get an acceptable cancellation that carried out many operations at a high speed. The ability of an adaptive filter to operate and respond satisfactorily to an unknown environment, and variations that may be involved in signal reference, to make a powerful adaptive filter for signal processing and control applications. There are several types of adaptive filters but generally all share the characteristic of working with an input signal (input vector), and a desired response (output vector). These two signals are used to compute an estimate of error (error signal), which allows control of the coefficients of the adjustable filter.

In other words, ANC is an approach to noise reduction and a secondary noise source that destructively interferes with the unwanted noise is introduced. In general, active noise control systems rely on multiple sensors to measure the unwanted noise field and the effect of the cancellation. The noise field is modeled as a stochastic process, and an adaptive algorithm is used to adaptively estimate the parameters of the process. Thus, active noise control involves an electroacoustic or electromechanical system that cancels the primary (unwanted) noise based on the principle of superposition; specifically, an anti-noise of equal amplitude and opposite phase is generated and combined with the primary noise, thus resulting in the cancellation of both noises. ANC is developing rapidly because it permits improvements in noise control, often with potential benefits in size, weight, volume, and cost. Thus, the active noise control has been object of an intense research and central subject in many scientific articles in the last 10 years.

On the other hand, unwanted acoustic noise is a by-product of many industrial processes and systems. This problem has become more and more evident as the applications of electronic communication systems increase, since their effects represent an important source of annoyances for the end user and they can reduce considerable the efficiency, the quality and the reliability of this type of systems. These ANC systems use an active form of noise control which includes the use of a second source of sound that generates a signal of the same characteristic as echo but with different phase. This allows to cancel this signal because the waves of sounds propagate linearly, which is known as superposition effect. Also, since the characteristics of the signal to cancel change constantly, in this case the echo, the system re-

quires a great capacity of adaptation. These adaptively systems, represent a feasible alternative for echo cancellation in telephone lines due to their processing, capacity and lower cost.

2.2. Cancelling Telephone Echo

Telephone echo, is a phenomenon produced by the mismatching impedance of the hybrid circuit used to couple the two lines with the four lines sections of long distance communication systems that considerably degrades the quality of telecommunication systems. Several systems have been proposed in the literature during the last several years, to solve these problems, such as adaptive echo cancelers. The figure 2 depicts the basic structure of described system.

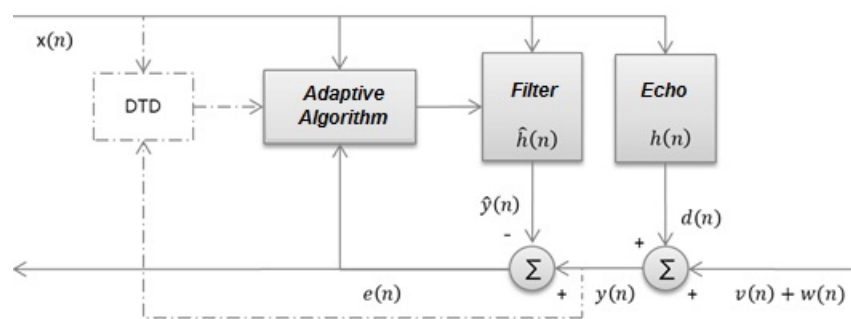


Figure 2. Echocancelling in long distance telephone systems

An echo canceler generates a replica of the echo signal and subtracts it from the signal to be transmitted generating the so-called pseudo echo, which is then used to update the echo canceler coefficients such that the mean square value of residual echo becomes a minimum. However the real time estimation of the hybrid impulse response is a difficult task for several reasons:

1. The echo path impulse response is non-stationary, and then the convergence of adaptation algorithm must be fast enough to track these changes.
2. The power spectral density of speech signals is not flat. This fact results in a slower convergence rate when gradient search based adaptive algorithms are used.
3. In most cases the echo canceler requires one hundred or more taps for an accurate estimation of a hybrid impulse response and several thousand of taps in the acoustic echo path case, which makes the use of efficient adaptation algorithms difficult.
4. The presence of both, near and far-end speakers simultaneously often occurs, which require some robust mechanisms or adaptation algorithms to handle it. Thus the development of low complexity and high convergence rate echo canceler structures has received a lot of attention, resulting in several efficient echo canceler structures and adaptation algorithms.

The most suitable tool for solving the two aforementioned problems is adaptive filtering which has been successfully applied in the solution of several practical problems [3].

3. ANC Systems: types and problematic

3.1. Types of ANC Systems

3.1.1. *A priori (Feedforward)*

Figure 4 shows, in a simplified way, an ANC Feedforward System, in which the digital filter $W(z)$ is used to estimate the unknown plant $P(z)$. It is assumed that both the plant and the filter have the same input signal $x(n)$. Moreover, a Filtered LMS (Filtered-X Least Mean Square, FXLMS) algorithm is introduced, which is a varying form of the LMS algorithm [2]. FXLMS algorithm solves the secondary path problem, described as the set of transformations that the filter signal and the adaptive error signal go through, on their way from an electric to an acoustic domain. During this electro-acoustic process, the signal may be delayed or altered in such a way that it is necessary to minimize such effects. The FXLMS algorithm technique consists of placing a filter, with the same properties as the secondary path, in the reference signal going towards the adaptive least mean square filter (LMS), as shown in figure 3.

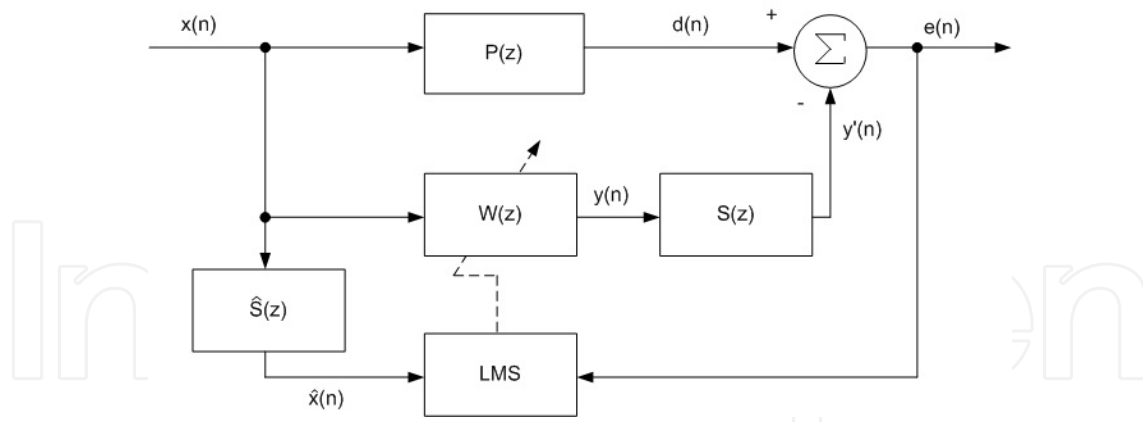


Figure 3. ANC Feedforward system with FXLMS algorithm

From Figure 3, filter $\hat{S}(z)$ is the model of the secondary path, defined by filter $S(z)$. Taking this into consideration, the update of filter $W(z)$ is given as follows:

$$\bar{w}(n+1) = \bar{w}(n) + \mu \hat{x}(n)e(n) \quad (1)$$

Where

$$\hat{x}(n) = \hat{s}(n) * \bar{x}(n) \quad (2)$$

3.1.2. *A posteriori* (Feedback)

There are some situations in which it is not possible to take into account the reference signal from the primary noise source in a Feedforward ANC system, due to difficult access to the source, or another reason that makes it hard to identify a specific signal through the reference microphone. A solution to this problem is to introduce a system, which will predict the behavior of the input signal; this system is known as *a posteriori* ANC (Feedback ANC), which is known for using only an error sensor and a secondary sound source to achieve noise control.

Figure 4 describes a Feedback ANC system with FXLMS algorithm, in which $d(n)$ is the noise signal, $e(n)$ is the error signal, defined as the difference between $d(n)$ and signal $y'(n)$, which is the adaptive filter's output once the secondary path has been crossed. Finally, the adaptive filter's input signal is generated by the sum of the error signal and the resulting signal from the convolution between the secondary path $\hat{S}(z)$ and the estimated output of the adaptive filter, $y(n)$.

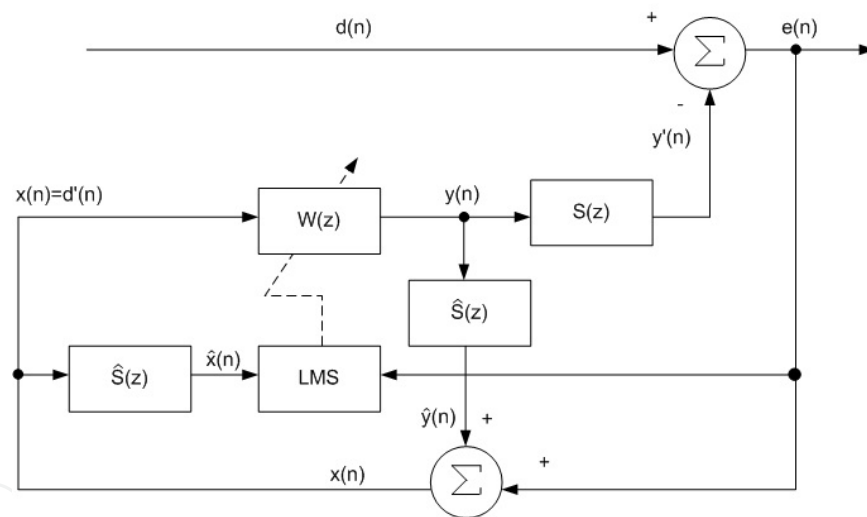


Figure 4. Feedback ANC system with FXLMS algorithm

3.1.3 What is a Hybrid system?

A hybrid ANC system is made up of an identification stage (*feedforward*) and a prediction stage (*feedback*). The combination of both stages needs two reference sensors: one close to the primary noise source and other with the residual error signal. Figure 6 shows the detailed block diagram of a hybrid ANC system, in which it is possible to observe the basic systems (*Feedforward*, *Feedback*) involved in the design. The attenuation signal, given by $y(n)$, results from the addition of both adaptive filter outputs, $W(z)$ and $M(z)$. Filter $M(z)$ represents the *Feedback* process of the adaptive filter, while filter $W(z)$ represents the *Feedforward* process.

The secondary path in the basic ANC system is also taken into consideration in the hybrid system, and is given by the transfer function $S(z)$.

Among the advantages of hybrid ANC systems we can mention:

1. The fact that lower order filters may be used to achieve the same performance;
2. The other two systems present much more significant plant noise than the hybrid system;
3. The combination of both systems allows for much more flexibility in regards of design; and,
4. Cancellation of both narrowband and broadband noise.

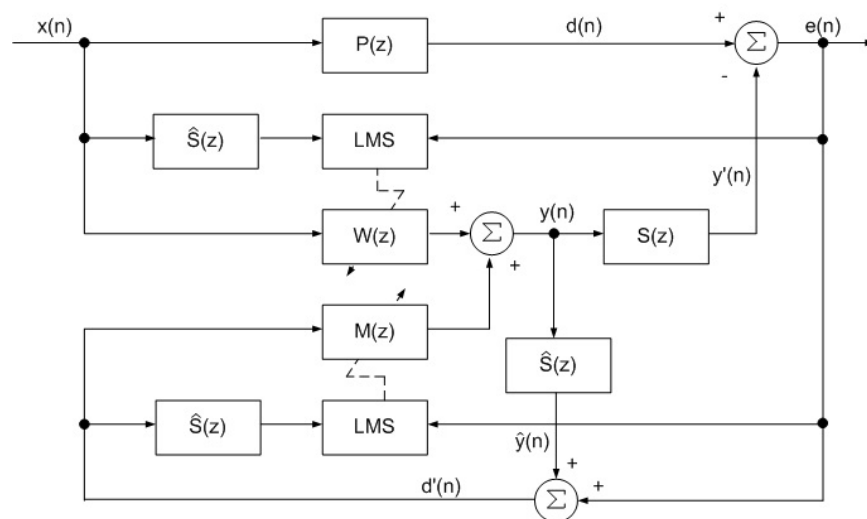


Figure 5. Hybrid ANC system with FXLMS Algorithm

The block diagram of the hybrid ANC system in Figure 5 also shows FXLMS algorithm to make up for the possible delays or problems induced by the secondary path [4].

3.2. Main problems in ANC Systems

3.2.1. Secondary Path Modeling

As mentioned previously, the process that transforms the resulting signal from the adaptive filter $y(n)$ into signal $e(n)$, is defined as secondary path. This characteristic takes into consideration the digital to analog converter, the reconstruction filter, the sound source, the amplifier, the acoustic path from the sound source to the error sensor, the error microphone, and the analog to digital converter. There are two techniques to estimate the secondary path, both with characteristics that make each method more comprehensive and sophisticated in certain ways; these techniques are: offline secondary path modeling and online secondary path modeling. The first method is performed with a *Feedforward* system, where the plant is

now $S(z)$ and the coefficients from the adaptive filter are the secondary path estimation, as shown in Figure 6 [4].

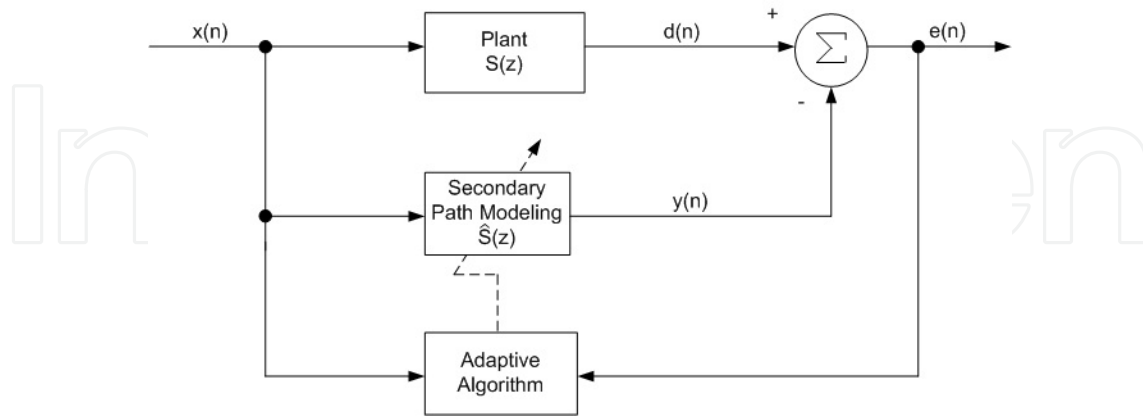


Figure 6. Offline Secondary Path Modeling

3.2.2. Acoustic Feedback

This property is typical of feedforward systems. Figure 7 shows the contribution of attenuation signal $y(n)$, which causes the system to degrade because of the signal present in the reference microphone.

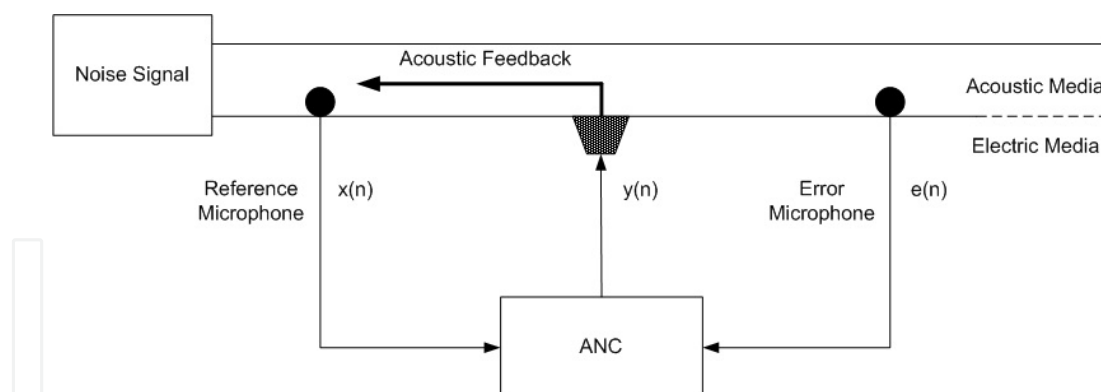


Figure 7. Feedforward ANC process with acoustic feedback

Two possible solutions for acoustic feedback problem are: acoustic feedback neutralization and the proposal of a hybrid system, which has a better performance in the frequency range and attenuation level of interest [4]. To evaluate this approach, we used a hybrid system as shown in Figure 8, where $F(z)$ is the transfer function of the feedback process.

The system proposed in [5] will be analyzed and this system, with a set of signals and experimental conditions, was completely evaluated in [6].

3.2.3. Online Acoustic Feedback Path Modeling

Most common way to eliminate acoustic feedback is to make an online path modeling, like indicated on [3] and, more recently, in relevant papers by [7] and [8]. However, one of the main characteristics of the hybrid system presented by the authors in [9] is that it does not take the secondary path modeling into consideration. Instead, the proposed hybrid system takes advantage of the inherent robustness of hybrid systems when it comes to acoustic feedback, figure 8.

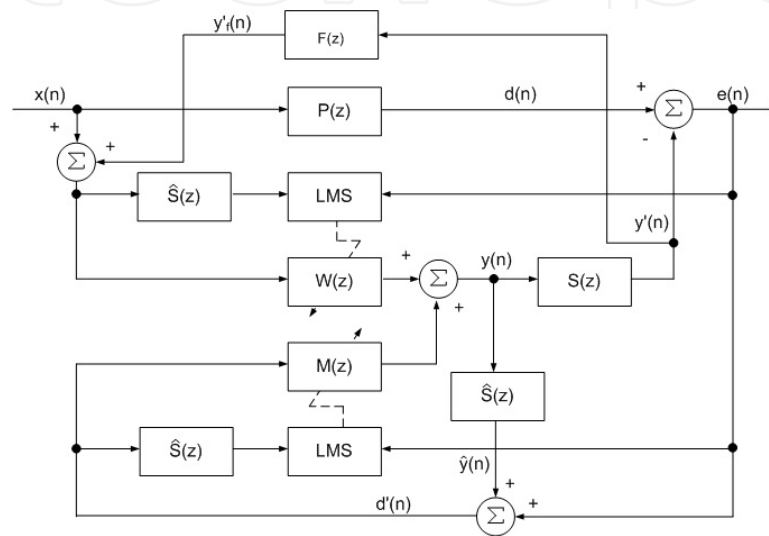


Figure 8. Hybrid ANC system with acoustic feedback

The system in Figure 9, proposed by [10], was used to compare the robustness of the HANC system against the neutralization system.

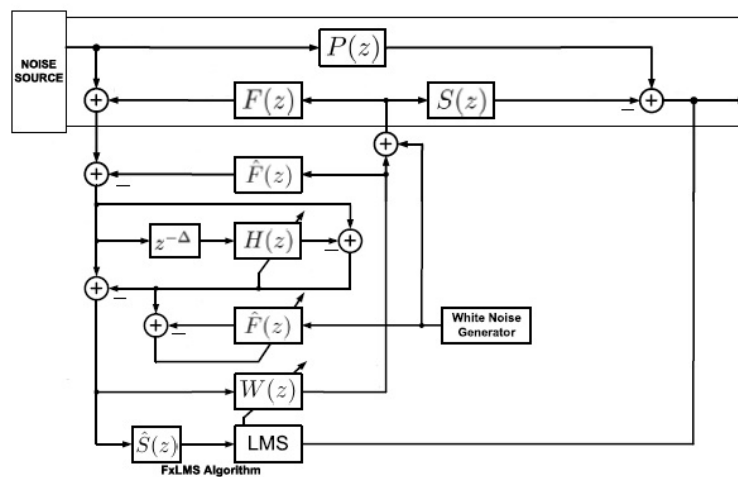


Figure 9. Kuo's Neutralization System

The details of the system in Figure 9 can be consulted in [10]. However, an important fact of this system is that it uses additive noise for modeling, also, as mentioned in [7], regarding predictable noise sources.

4. Echo Cancellation

4.1. Definition and general review

The echo is a problem that significantly degrades the quality of telecommunication systems. This occurs, in telephone line, due to the decoupling impedance hybrid which exists in the coils and are used to couple subscriber communication channel with the long distance channels. There is also the so-called acoustic echo which occurs in teleconferencing systems and hands free telephone systems. This type echo occurs due to acoustic coupling between loudspeakers and microphones used in these communication systems.

Several systems which try to solve this problem have appeared in the literature in recent years. Among these are: directional microphone arrangement [11], echo suppressors and adaptive echo cancellers [11, 12], etc. Among them, adaptive echo cancellation seems to be the best way to reduce the echo problem [13, 14]. An echo canceller generates an echo replica and subtracts the signal to be transmitted, generating a so-called residual echo. The echo residual is then used to adapt the coefficients of the system, using in most cases a gradient-based algorithm, in a way that the mean square value of the residual echo is progressively minimized [11, 12, 13, 14]. However, the real-time estimate of the impulse response of the hybrid or echo channel is a complex problem for several reasons:

1. The duration of the impulse response of a typical echo channels in teleconferencing systems in the order of several hundreds of milliseconds, which means that transversal filter coefficients of several thousand would be needed to reduce echo to acceptable levels. The impulse response of a typical acoustic echo channel is shown in Figure 10.
2. The impulse response of echo channel is non-stationary because it changes with the movement of the interlocutors, or the number of active subscribers on a given time. Thus the adaptive algorithm should be fast enough to track those changes.
3. The power density spectrum of the voice is not flat, and in many cases reduces speed of convergence of the adaptive algorithm. The correct estimate of the echo channel using structures with the least possible complexity and the relatively high speeds obtain convergence of the adaptation algorithm, as mentioned above, are non-trivial problems which have received considerable attention in recent years; among the different proposed have been proposed several echo cancellation systems, among we can mention: transverse echo cancellers, echo cancellers in the frequency domain, echo cancellers infinite impulse response subband echo cancellers, etc., [11, 12, 13, 14].

Besides the reduction in the complexity of the canceller, to allow correct estimation of the echo channel and the development of adaptive algorithms with rapid convergence, another

major problem is handle the simultaneous presence of echo near the speaker's voice. The situation we want to avoid is to interpret the speaker's voice echoing nearby, and make great changes in the echo channel estimated in an unsuccessful attempt to cancel this. A checked algorithm could operate incorrectly when the distant partner is present, so it is necessary to incorporate certain mechanisms within the system to avoid this effect [11, 12, 13, 14].

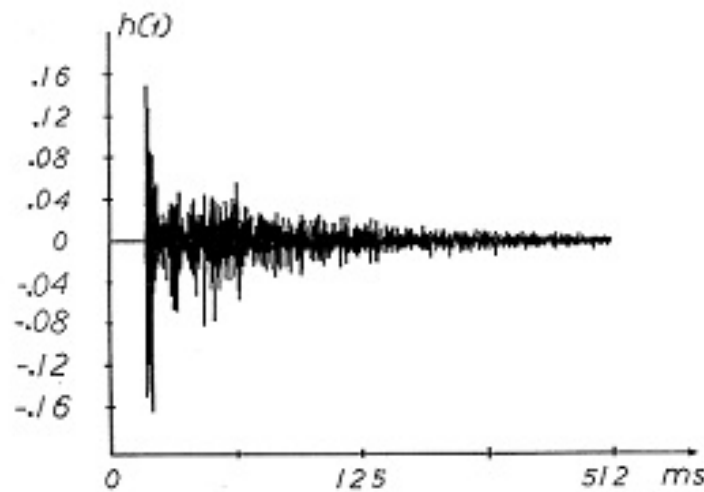


Figure 10. A typical impulse response of acoustic echo channel

There are few references about the convenience of using adaptive hybrid schemes for solving the problem of echo cancellation, and given the results obtained for applications for cancellation of acoustic noise [15], hybrid scheme is proposed for electrical noise cancellation, since it is on the phone lines where there is the problem described. Be detailed later about how to do and the results achieved.

4.2. Telephone Systems

A long distance telephone system basically consists of a 2-wire portion, known as the subscriber circuit, and connects the subscriber to the local exchange and long-distance circuits itself; this system consists of a transmission channel and another receiving, each of which consists of two wires. A hybrid transformer is used to couple circuits' long distance subscriber circuit and ideally isolate the transmission channels and reception of long-distance circuit. However due to the decoupling impedance, they are not completely isolated so that a portion of the received signal is delayed in the form of echoes. A similar problem arises in teleconferencing systems with so-called acoustic echo which occurs due to coupling between the microphone and speaker in the teleconference system. This result in a delayed and distorted replica of the signal produced by the loudspeaker is fed back into the microphone.

In both cases there is deterioration in the communication system, which resulted in the appearance of echo cancellers. These cancellers have proved to be the best way to solve this problem [11, 12]. The basic principle of echo cancellation, which is illustrated in Figure 11, is

to generate an echo replica, this is subtracted from the signal to be transmitted, resulting in the so-called residual echo that consists of part of the signal echo which could not be canceled more near the speaker's voice, if this is present [11, 12]. The residual echo is then used to adapt the parameters of the canceller in such a way that the residual echo power is progressively minimized.

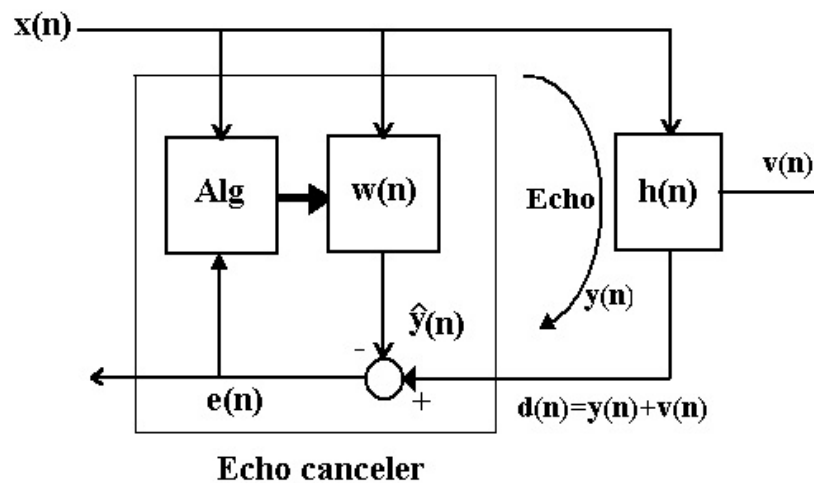


Figure 11. Echocancelling in long distance telephone systems

Echo canceller consists of two main parts. An adaptive filter, which generates an echo replica and is subtracted from the signal being transmitted, and a system commonly known as double-talk detector, this system prevents distortion due to the presence of the speaker's voice service or in the absence of the partner away. The first component is the structure of the adaptive filter along with its adaptation algorithm.

Some researchers have resulted in the appearance of various structures, such as transversal filters, subband structures, structures in the frequency domain, etc., and various adaptive algorithms, mostly based on gradient descent search. Second component, despite its importance, has received much less attention than the first component. Thus conducting research aimed at developing highly reliable mechanisms to avoid distortion due to the simultaneous presence of both parties, "double-talk detector," especially when using algorithms based on gradient descent search is of great importance.

5. Delimitation of Proposed ANC System and its Application

5.1. Evaluated ANC Structure

Figure 12 shows the block diagram of the evaluated hybrid ANC structure with online secondary path modeling. This 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, $F(z)$. Since the samples of feedback distortion are strongly correlated among them, they can be predicted [15].

As shown in Figure 12 signal, $a(n)$, is used simultaneously as:

1. The error signal to update the adaptive filter, $W(z)$, which corresponds to the feedforward stage used to identify the noise path, and,
2. 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 through the system $F(z)$; and,
3. To estimate $\hat{S}(z)$, which represents the online secondary path modeling adaptive filter.

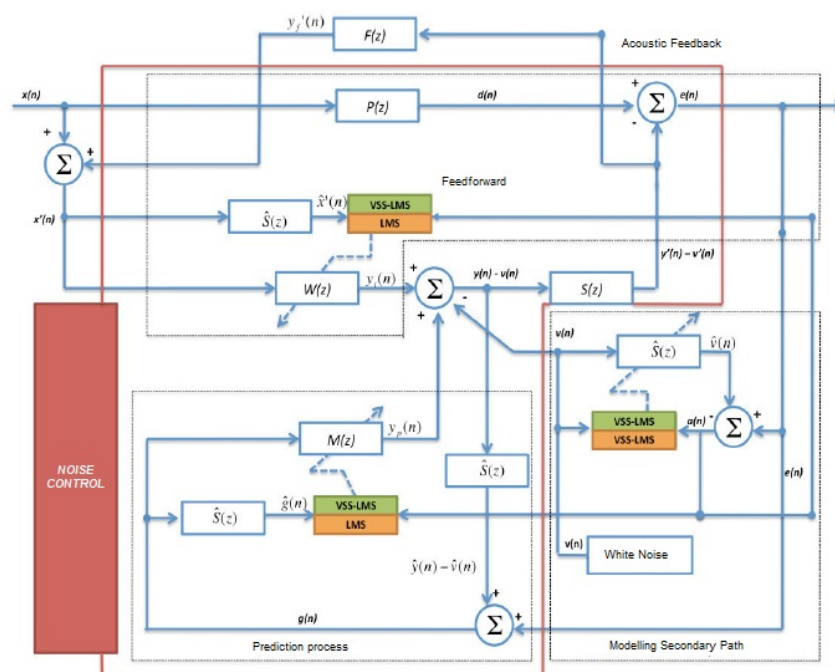


Figure 12. Evaluated hybrid ANC structure

The hybrid ANC contains the advantages of feedback and feedforward systems. The model presented by [16] was modified to adapt the system for a specific objective: reduce the residual echo. This system uses two input signal $x(n)$ and $d(n)$, one for each talker. The plant that models echo refers to the effect of mismatch of impedance present in the telephone circuit. The echo signal is $d(n)$ and the residual echo plus the far-end signal is represented by $e(n)$. This system incorporates the signal of the feedforward and the feedback effect that means both systems contribute to generate the cancelling signal, which approximates to the echo signal. Also this system includes a switch on the feedback system: when the echo signal and the far-end signal are highly correlated, the feedback system cancels part of the far-end signal even if the hybrid system already converged [17].

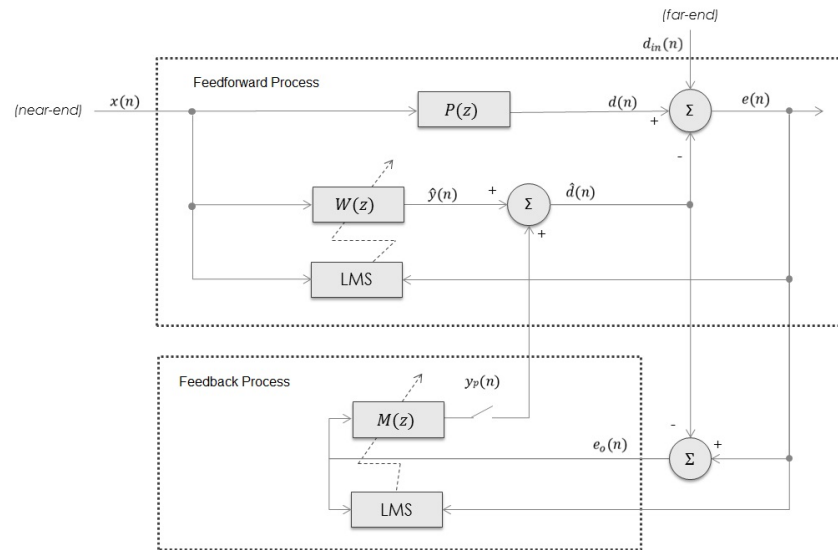


Figure 13. Adapted Hybrid ANC for Active Echo Cancellation

To analyze the system is necessary to consider the correlation between signals, as shown in the equation (3):

$$R = E[\bar{x}(n)\bar{x}^2(n)] \quad (3)$$

The cross correlation vector between the entrance and the echo is given by:

$$\bar{p} = E[d(n)x(n)] \quad (4)$$

and the correlation matrix can be written as follows:

$$\overline{Rw_0} = \bar{p} \quad (5)$$

where \bar{w}_0 is the optimum vector of the transversal filter. In the selected algorithm, LMS, the reference signal $x(n)$ is processed by an adaptive filter $W(z)$. In this case the coefficients of the filter are updated by the gradient of the error signal power obtained plus the previous coefficients and μ step size:

$$w(n+1) = w(n) + \mu x(n)e(n) \quad (6)$$

5.2. Active Echo Cancellation in Telephone Lines

There are two kinds of echo: electric and acoustic. The electric echo is present in traditional telephony lines because of the impedance mismatch of the conversion (from two to four

wires). The acoustic echo is the direct or indirect feedback of reflected signals to the microphone during a conversation. There are two controls applied to echo: suppressor and canceller systems. Echo Cancellation systems need to consider the disturbances in the far-end talker's signal and the superposition of the near-end talker's that generates double-talk. Two general approaches are the use of suppressors and the use of cancellers. The echo suppressor has a sensor that measures the voice signal power in each part of the circuit to decrease the impact of the echo. The echo suppressor changes the full duplex channel to a half-duplex channel [14, 18]. This characteristic is a disadvantage of this type of control because it cancels part of the speech. Echo cancellers use the superposition principle that means this system generates a similar signal with delay and attenuation similar to the transmitted signal. It is recommended to train the system to approach the characteristics of the echo signal. For this problem some authors [19, 20], offered different solutions based on Double-Talk Detector (DTD) [21]; this principle detects the presence of simultaneous speech of both talkers and pause the coefficient updating of the adaptive filter. It is known that the adaptive filter is the key to treat echo problems. It is necessary to consider the speed of convergence and robustness of the system. Most of echo cancellation systems use transversal filters and the LMS algorithm or variations of this to adjust the coefficients [22].

The result is an error signal named as residual echo signal due to estimation of the adaptive filter [21], this scenario, adapted to an ANC system is shown in Figure 14 [3].

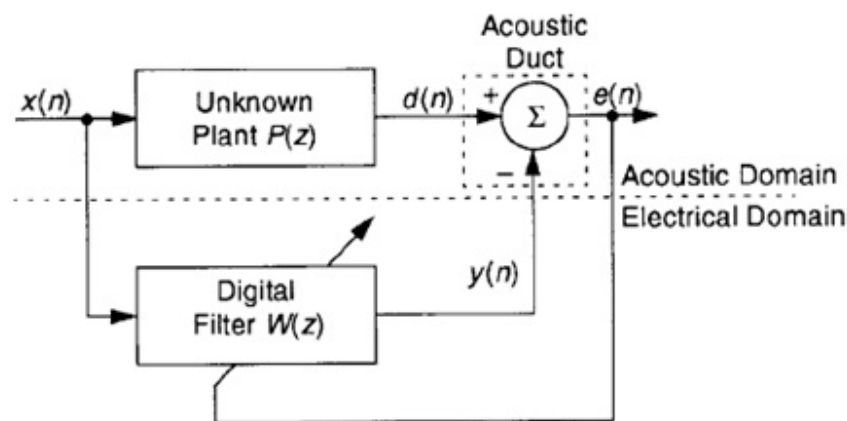


Figure 14. System identification viewpoint of ANC

From Figure 14, the residual echo $e(n)$ is defined as

$$e(n) = d(n) - y(n) \quad (7)$$

where $d(n)$ is the echo signal and $y(n)$ is the response generated by the adaptive filter after processing the algorithm. Also [3], presents the criteria of the Mean Square Error (MSE) to find the convergence point of the system. To analyze the performance of the Echo Cancell-

tion system Echo Return Loss Enhancement (ERLE) criteria was developed. The ERLE criterion is described in equation (8).

$$ERLE = 10 \log \left\{ \frac{E[d^2(n)]}{E[e^2(n)]} \right\} \quad (8)$$

ERLE parameter was used to evaluate the present proposed system.

6. Performance Parameters and Several Aspects Considered

6.1. Parameters and issues

The proposed system has different parameters to consider. These parameters determine whether the system converges or not.

1. *Step size (μ):* controls the system stability and speed of convergence, one for each part of the system (feedback and feedforward).
2. *Plant:* simulates the echo effect
3. *Adaptive filter $W(z)$:* length and values for established plants
4. *Number of blocks and iterations:* reflected in the number of samples observed
5. *Entrance signals:* including the near-end and the far-end

Step size values were taken by [16, 23]. The plant simulates the effect of echo that the near-end suffer because of the impedance mismatch, proposed by [24].

The input signals utilized are sorted into one of three types, considering the classification proposed by [3, 25], as well as companies such as [26].

1. *Continuous:* the level of sound remains constant or nearly constant with small fluctuations. For Echo cancellation, the selected signals were vacuum, four tones and silence.
2. *Intermittent:* the level of sound presents some fluctuations that can be periodic or random. The selected signals are real voices recorded in a computer for Echo considerations.
3. *Impulsive:* the level of noise presents impulses in a brief period of time.

For Acoustic Noise Reduction applications, the system was tested with several real sound signals taken from an Internet database [27]. The sound files were selected taking into account that the system is to be implemented in a duct-like environment. Also, six different types of signals were used for the analyzed system:

1. A sinusoidal reference signal with frequency of 300 Hz, and 30 dB SNR;

2. A reference signal composed of the sum of narrow band sinusoidal signals of 100, 200, 400, and 600 Hz; and,
3. The rest of the reference signals are .wav audio files with recordings of real noise sources, which are “motor” and “airplane”, as in [16].

The most important values are modeling error, as was defined by [28], and MSE, given by the ratio between the power of the error signal, and the power of the reference signal:

$$\Delta S(dB) = 10 \log_{10} \left[\frac{\sum_{i=0}^{M-1} [s_i(n) - \hat{s}_i(n)]^2}{\sum_{i=0}^{M-1} [s_i(n)]^2} \right] \quad (9)$$

$$MSE(dB) = 10 \log_{10} \left[\frac{\sum_{i=0}^{M-1} [e_i(n)]^2}{\sum_{i=0}^{M-1} [x_i(n)]^2} \right] \quad (10)$$

6.2. System Training

For experience, we need to train the system before to start to work [16]. So, we have two considerations:

1. For echo cancellation, we adapt the plant for 20 representative coefficients instead the 1000 given by [24]. The adaptive filter was a vector of 20 coefficients initialized in zero. The near-end voice was a female voice and silence for the far-end. The step size value were change until get the higher level of ERLE, after run the simulation of the system using Matlab®, with a software interface developed specifically for this purpose, the results of the adaptive filter were retaken to repeat the processing, when a 40dB of cancellation were achieve the training was stopped. The scenario for training work was single-talk with a single voice signal in the near-end.
2. For the situation for Acoustic Noise Reduction, secondary path is offline modeling stopped when the error is reduced -35dB similar to [15]. The excitation signal $v(n)$ used was white Gaussian noise with variance of 0.05.

7. Analysis of Results

7.1. Echo cancellation phone lines

To consider an approximation of a real system the results of processing echo of voice with the hybrid proposed system. We present the results using the female voice signal (Figure 15) in the near-end and two different masculine voice signals in far-end (Figure 16 and Figure 17).

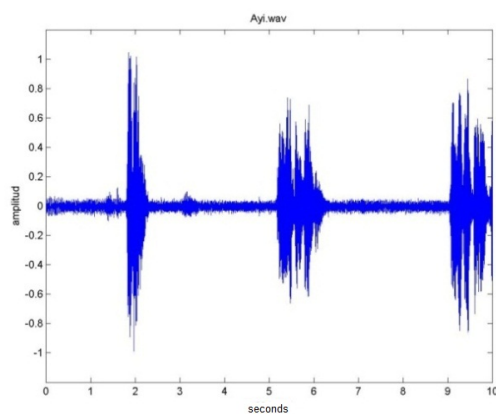


Figure 15. Female voice signal

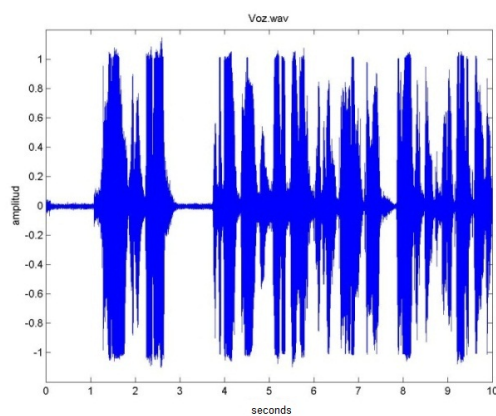


Figure 16. First masculine voice signal

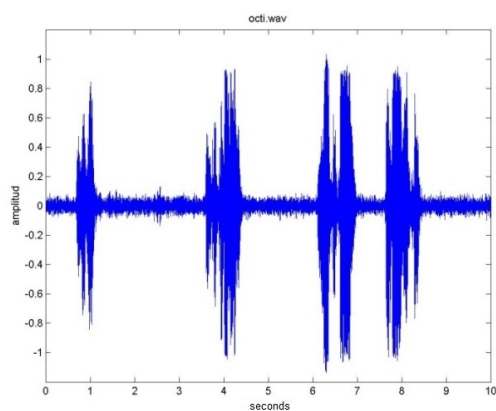


Figure 17. Second masculine voice signal

The echo signal generated by the adaptation of the plant is represented in the Fig 18.

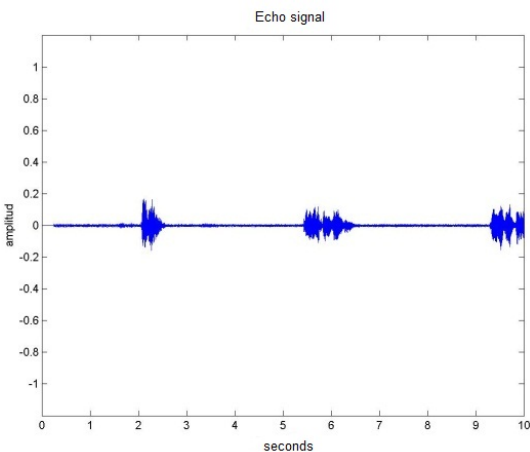


Figure 18. Echo of the female voice signal with adapted plant

Applying the function with the parameters of Table 1, the obtained results are shown in Figure 19 and Figure 20. Both figures show that system achieves cancellation of the echo signal.

Parameters	Value
Step size	0.1
Plant	From [24]
Blocks	1000
Iteration	80

Table 1. Analysis Parameters

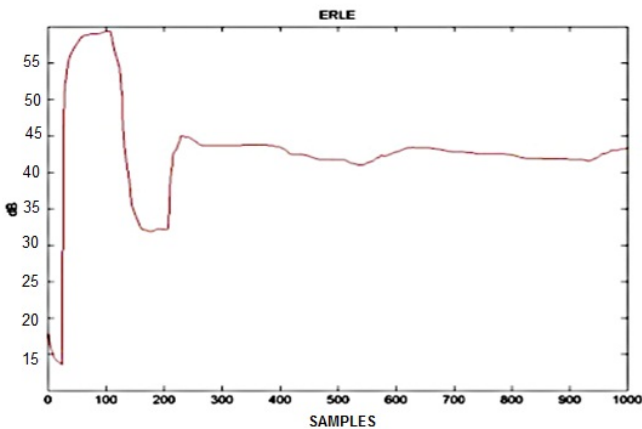


Figure 19. ERLE using female voice in the near-end and masculine voice 1 in far-end

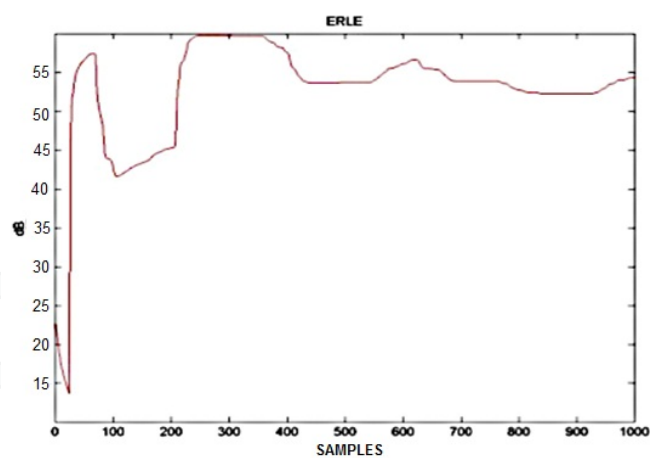


Figure 20. ERLE using female voice in the near-end and masculine voice 2 in far-end

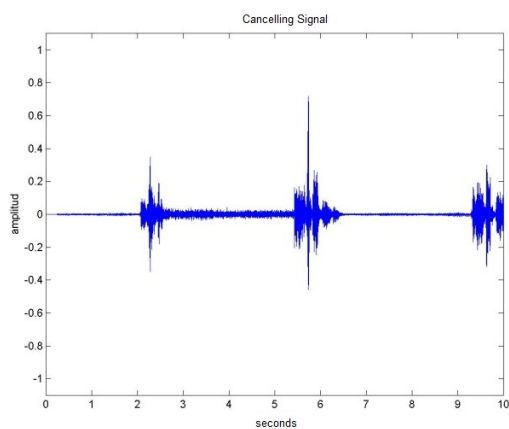


Figure 21. Cancelling voice signal, system with masculine voice 1

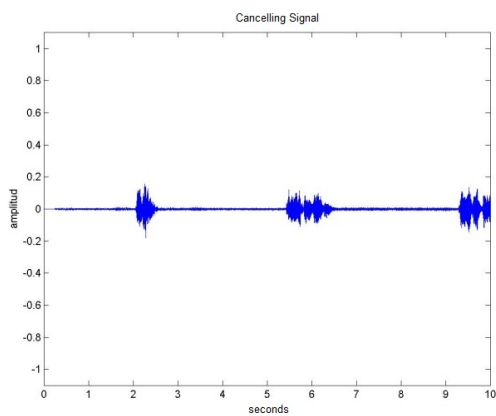


Figure 22. Cancelling voice signal, system with masculine voice 2

Looking for a detailed analysis in the cancelling signal (Figure 21), which imitates echo signal, for the first masculine signal, the system begins to diverge. This occurs because of the high correlation between the two entrances voices; this effect is given by the feedback because even when the system already converge starts to cancel the far-end signal [29].

Then, instead of the first male signal, another signal was used and the system converged better, this can be seen in Figure 22, this situation is because the correlation between this signal and the female is smaller.

As mentioned before, the step size factor has a major impact on the development of the system, and proved to be the main reason to make the system converge; additional simulations were performed using the parameters in Table 2; this means a smaller size step and the male voice first.

Parameters	Value
Step size	0.01
Plant	From [24]
Blocks	1000
Iteration	80

Table 2. Analysis Parameters for Additional Test

The system improves its performance using the parameters of Table 2. The generated cancelling signal (Figure 23), does not have impulsive periods.

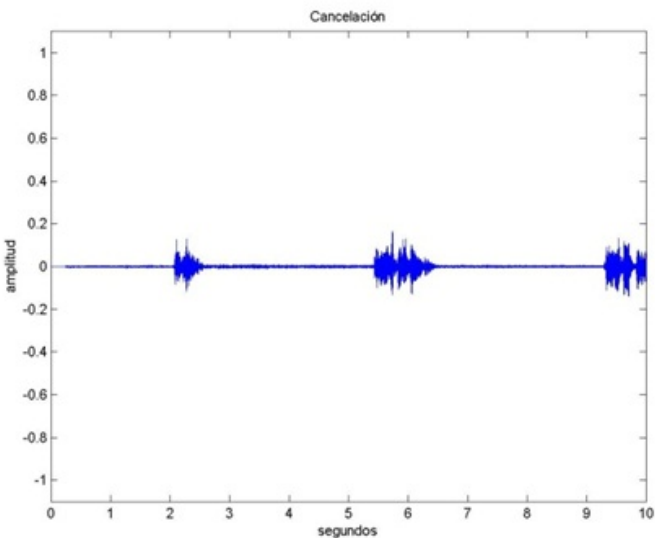


Figure 23. Cancelling voice signal, system with masculine voice 1 and adjusted step size

7.2. Active Noise Cancellation

7.2.1. General results (MSE and Modelling Error)

This section presents the simulation experiments performed for acoustic noise reduction. First, an offline modeling was used to obtain FIR representations of tap weight length 20 for $P_{\lambda}(z)$ and of tap weight length 20 for $S(z)$. The control filter $W(z)$ and the modeling filter $S(z)$ are FIR filters of tap weight length of $L = 20$ both of them. A null vector initializes the control filter $W(z)$. To initialize $S(z)$, 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.

Various articles on the subject of ANC were references taken into consideration before establishing main analysis parameters to determine the hybrid system's performance:

- a) Filter order; it is important to evaluate the system under filters of different orders. In this case, 20 coefficients were selected (we considered the fact that the distance between the noise source and the control system is not supposed to be very large).
- b) Nature of the filter coefficients; on a first stage, the coefficients were set according to real values taken from a previous study made on a specific air duct [2]. These coefficients were taken from the work done in [16] to determine the values of the primary and secondary path filters for an air duct.

The simulation results are presented according to the following parameters:

1. Mean Square Error (MSE); and
2. Modeling error from online secondary path modeling.

Table 3 shows the values used for the feedforward and feedback step sizes, as well as the range of step sizes used for the secondary path filter. The values were set by trial and error, starting with the values that were determined with the previous test.

Signal	Step size μ_w, μ_m	Step size μ_s
Continuous	0.000001	0.0001 – 0.001
Intermittent	0.000001	0.0001 – 0.001
Impulsive	0.000001	0.0001 – 0.001

Table 3. Filters Step Size Used in Proposed Analysis

Also, a white noise with zero mean and variance equal to 0.05 was used in the system. Since there were not enough resources to implement an abrupt secondary path change (which means there was only one set of values available for the secondary path filter from [15], a gradual change was made, given by the sum of a sinusoidal function to the secondary path

coefficients, from iteration 1000 to 1100. The best response was shown by the continuous signal; Figure 24 shows the Modeling error for this case, while Figure 25 shows the MSE.

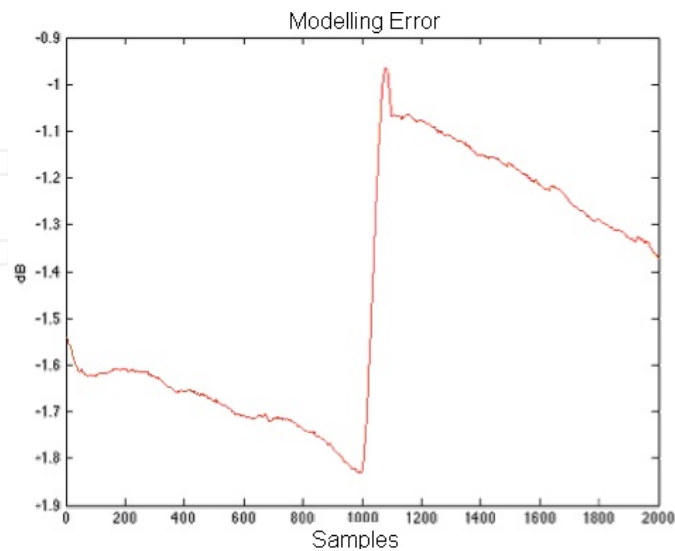


Figure 24. Relative modeling error for continuous signal

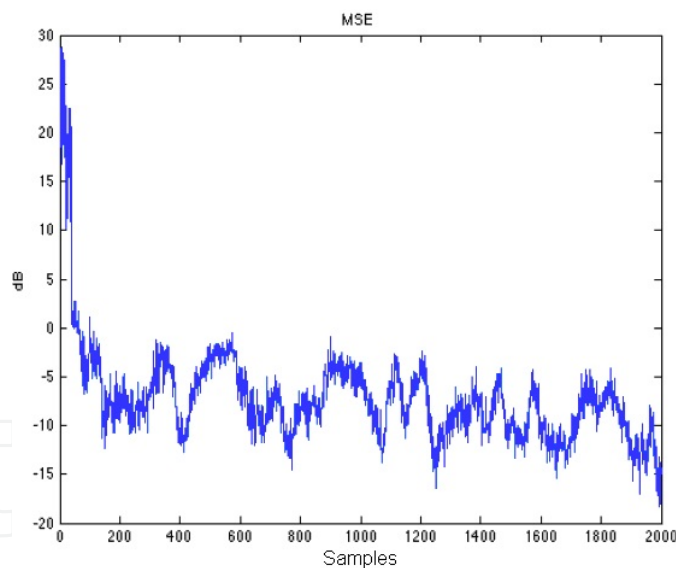


Figure 25. MSE for continuous signal

From Table 3, it can be noticed that the step sizes had to be considerably reduced, in the order of 1000, in comparison to the values established for the tests with Echo cancellation. This is due to the fact that the coefficient values are not necessarily within a range of -1 to 1, so the secondary path modeling needs a smaller step size to be able to achieve convergence.

For the intermittent signal, the effects of the small step sizes were similar: the system took more time to converge and the level of noise cancellation was reduced. Nonetheless, the re-

sponse achieved stability during the simulation. Figure 26 and Figure 27 correspond to the Modeling error and MSE for the intermittent signal, respectively.

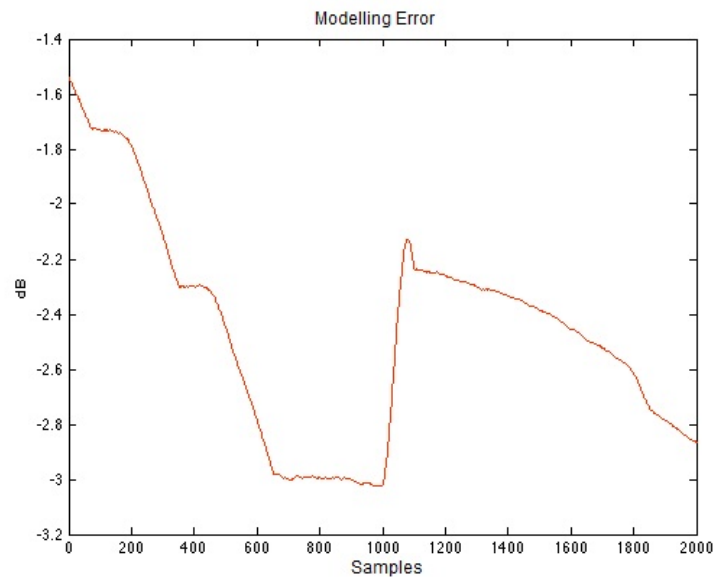


Figure 26. Relative modelling error for intermittent signal

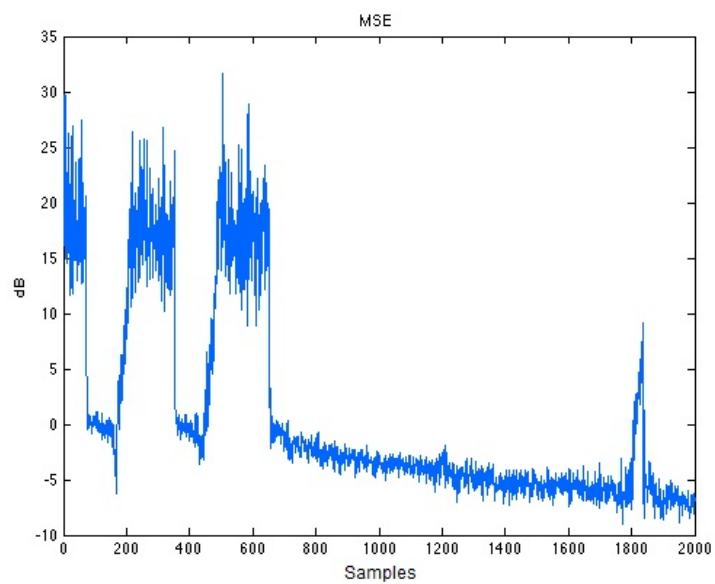


Figure 27. MSE for intermittent signal

Finally, for impulsive input signal the results were not as good as expected. The results can be explained since there are very abrupt changes in the signal amplitude, and the step size is very small. Hence, the values of the coefficients tend to infinity and the simulation stops abruptly.

7.2.2. Comparison versus Neutralization and Feedforward Systems

In this section, three paths were used: the main or primary path $P(s)$, the secondary path $S(s)$, and the acoustic feedback path $F(s)$. All the filters used in the evaluated proposals are finite response filters (FIR). The values of these paths are based on [2], and represent the experimental values of a given duct. A total of 25 coefficients will be used in all paths so as to report an extreme condition for a real duct under analysis.

To initialize $\hat{S}(z)$, the offline secondary path modeling is stopped when the modeling error has been reduced up to -35dB, similar to [15]. The excitation signal $v(n)$, is white Gaussian noise with variance equal to 0.05.

The values for the step size are adjusted by trial-and-error to achieve a faster convergence and stability, following the guidelines from previous work on Hybrid Active Noise Control [16], and the values selected in [7] for neutralization. A summary of the selected values for μ , is shown in Table 4.

System	Primary Path	Secondary Path	Feedback Path
	μ_p	μ_s	μ_f
Neutralization System	0.000001	0.00005	0.00005
Hybrid System	0.001	0.001	NA

Table 4. Filters Step Size Used in Proposed Analysis

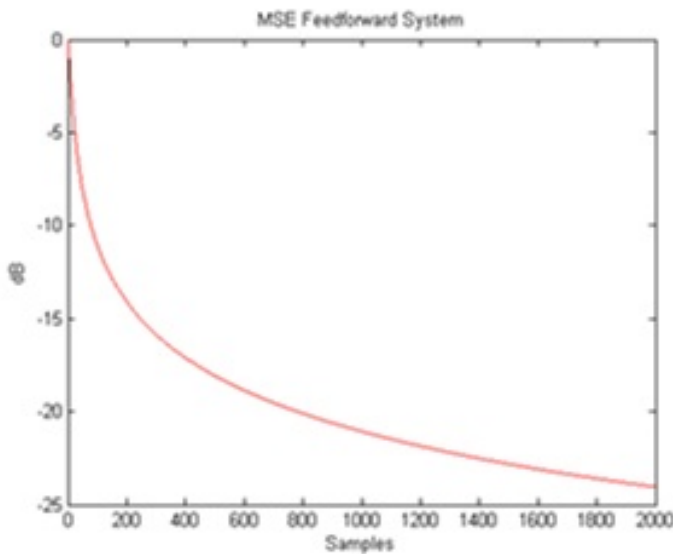


Figure 28. MSE with "sinusoidal" reference signal for Feedforward System

Figure 28 to Figure 41 show the result of the systems analysis with the previously mentioned set of signals. All results are shown in dBs, measuring the error power at the output (Mean Square Error).

First, we show the main signal for ANC systems, the sinusoidal signal. Figures 28 to 30 show the MSE value obtained.

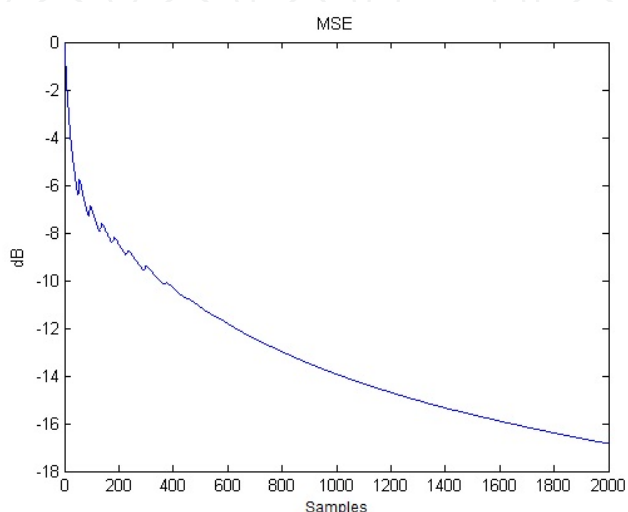


Figure 29. MSE with "sinusoidal" reference signal for Neutralization System

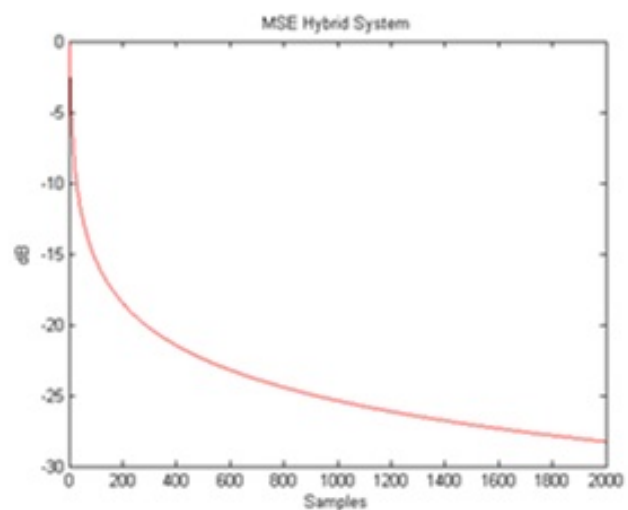


Figure 30. MSE with "sinusoidal" reference signal for Hybrid System

Another important is a narrow band signal, as explained before, is composed of the sum of narrow band sinusoidal signals of 100, 200, 400, and 600 Hz. Figures 31 to 33 show the results for this consideration

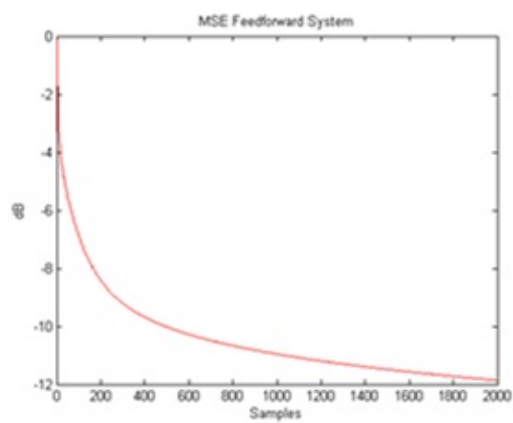


Figure 31. MSE with "4 tones" reference signal for Feedforward System

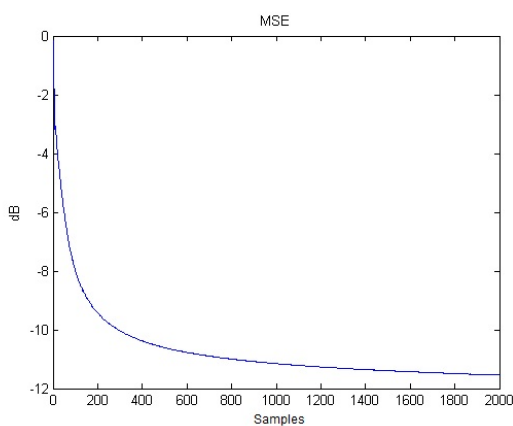


Figure 32. MSE with "4 tones" reference signal for Neutralization System

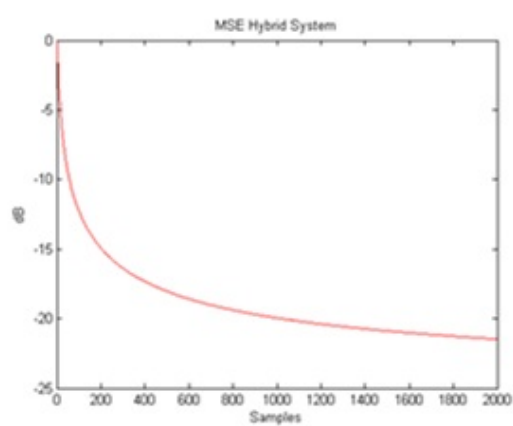


Figure 33. MSE with "4 tones" reference signal for Hybrid System

Finally, we use two recorded signals, corresponding to a "plane" and one to a "motor", meaning the evidence most relevant to our system. Of Figures 34 through 41, shows the convergence achieved with the proposed system.

Finally, it is important to consider that an ANC system should respond successfully to a change in the status of secondary path, which corresponds, for example, a possible movement of the microphone in a pipeline, or any vibration or change in of the system. Figures 24 and 25 show an abrupt change in secondary path at iteration 1000 [5]. We can observe that the behavior of both remain stable.

Figures 40 and 41 show selected results for the neutralization and hybrid systems, which are of greatest interest.

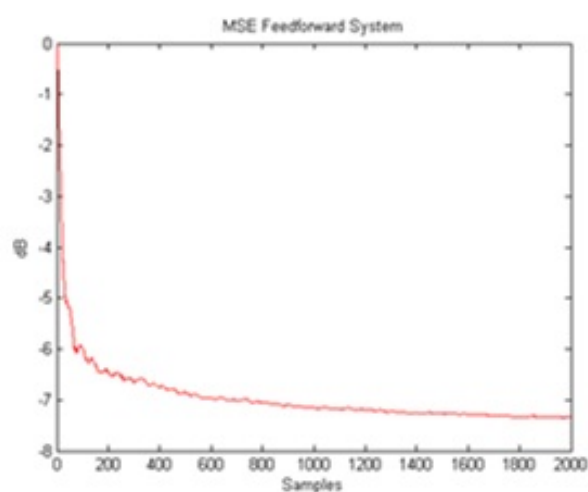


Figure 34. MSE with "Motor" reference signal for Feedforward System

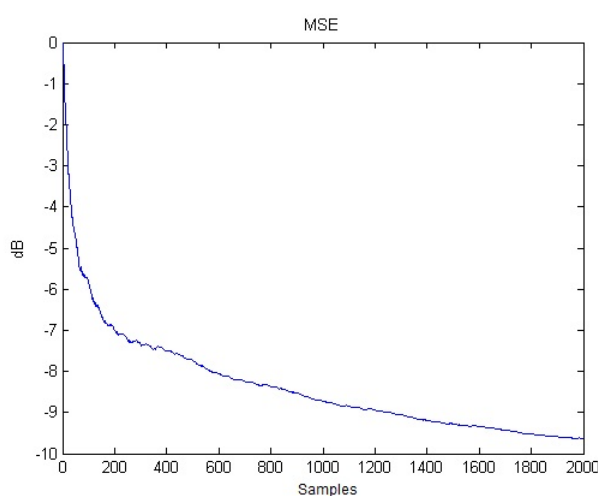


Figure 35. MSE with "Motor" reference signal for Neutralization System

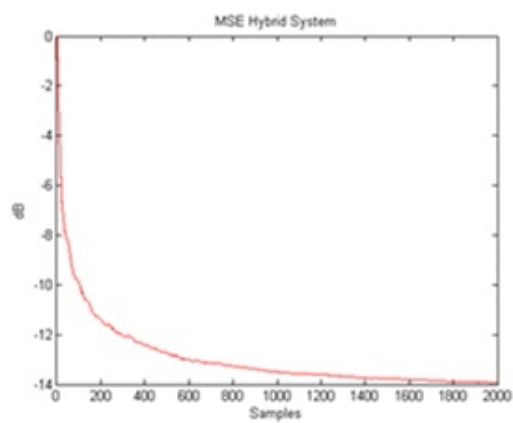


Figure 36. MSE with "Motor" reference signal for Hybrid System

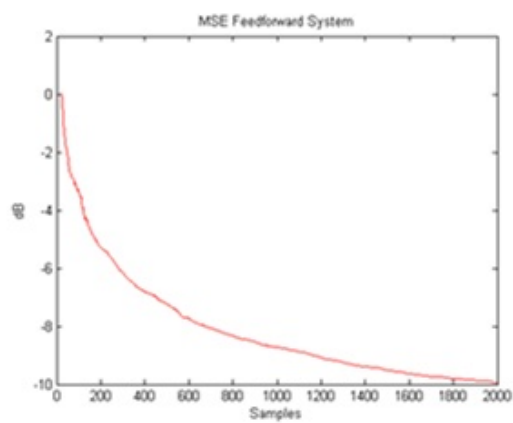


Figure 37. MSE with "Airplane" reference signal for Feedforward System

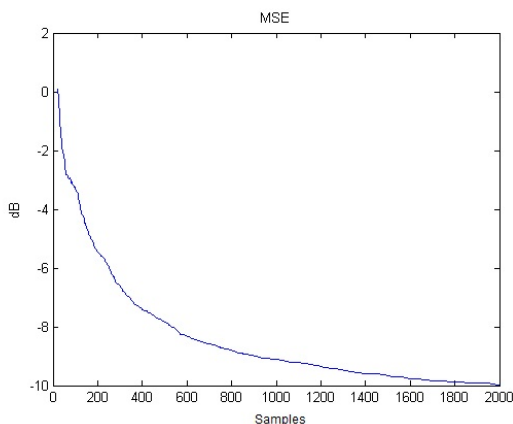


Figure 38. MSE with "Airplane" reference signal for Neutralization System

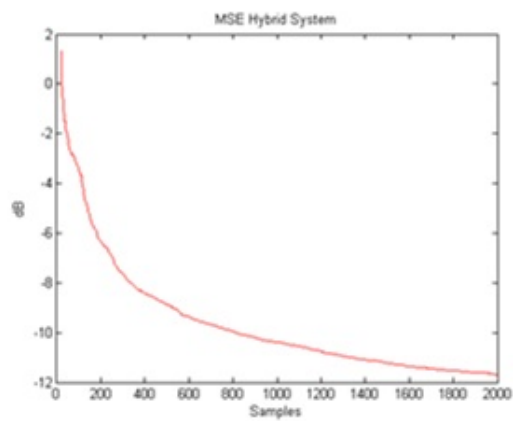


Figure 39. MSE with "Airplane" reference signal for Hybrid System

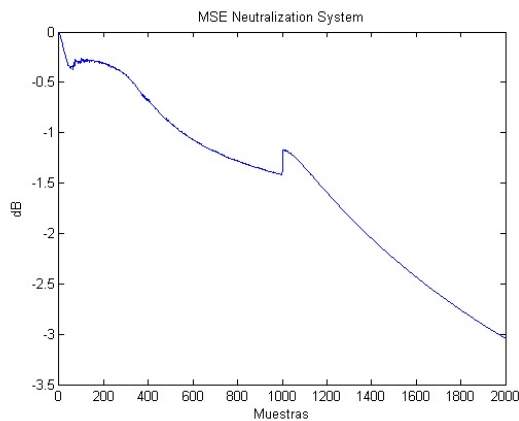


Figure 40. MSE with "4 tones" reference signal for Neutralization System, considering changing secondary path

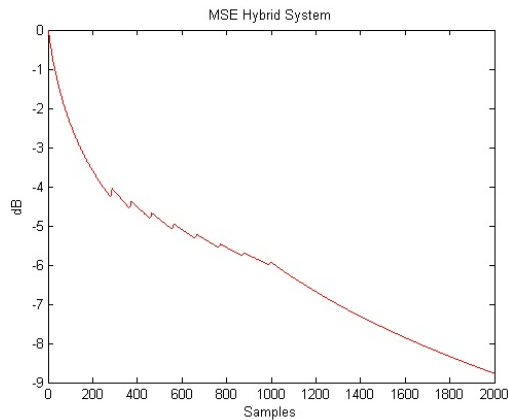


Figure 41. MSE with "4 tones" reference signal for Hybrid System, considering changing secondary path

8. Conclusions

The adaptive filtering is a powerful tool that offers various solutions to many fields of science today. This chapter shows the efficiency of the hybrid system in reducing electrical noise and noise currently present in conventional systems where noise becomes a significant cause of health problems, or a situation that can affect communications Internet or phone, to name a few.

Adaptive filtering, which has been successfully applied in the solution of several practical problems which main kinds are described some in this chapter, has relied mainly in the transversal filter structures. However, when the filter order becomes large, the transversal computational complexity and convergence rate may limit its capability for solving practical problems. This chapter presented an overview of the Hybrid System.

In particular, there are few references about hybrid systems, those conjoined feature more traditional patterns such as *a priori* and *a posteriori* systems. Of course they inherit the problems of these two, but the advantage they offer is based on the robustness of such systems for signals of different characteristics as continuous, intermittent and impulsive, and we tested a hybrid system in two interesting and relevant scenarios: unwanted signals in the fields of acoustics and telephony.

The proposed system works in an acceptable way for telephone echo problems, but it is necessary to consider and adjust the different parameters. The system is capable of cancelling echo of voice signals and can be applied to simulated scenarios of double talk without use the Double Talk Detector. Also it is necessary to evaluate the correlation between input signals since this correlation has a great impact of the performance of the system. If both signals are highly correlated, it is necessary to use a small step size for both feedback and feedforward systems. We established the double talk situation in telephony conversations as the test system for our Hybrid system including some talks simulating a real conversation.

With respect to Acoustic Noise Reduction, it must be notice that the results presented for a real-value filter coefficients refer to only one specific type of duct. This means that the response could probably improve in a different environment or in a duct with different properties. This situation represents a problem for the designer of a hybrid ANC, because for each environment where the system is to be applied would be no need to identify accurately the parameters to achieve the desired response. However difficult, this may not be impossible to do, so there is still a lot of work to be done with hybrid ANC systems.

This chapter discusses a new Hybrid Active Noise Control system and the impact adaptive filtering has on this field. The objective is to achieve improved performance at a reasonable computational cost in a Hybrid ANC system that considers two of the more important troubles of the ANC. We show two examples to prove the contribution of this system, one is a little generalist about cancelling several kinds of noise, and one very specific, which represents one persistent problem like telephone echo on telecommunications nowadays: networks have been modified by the use of new technologies and constant innovations have

led to automate the process of interconnection of subscribers, and the inclusion of forms of streaming media.

Therefore, he was a rigorous analysis of the results and their parameters under the above considerations. The results show the relevance of hybrid systems for consideration in removing acoustic noise or echo in telephony, with tools of adaptive systems. The advisability of this hybrid system is a matter that must be analyzed in depth.

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