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ESCUELA DE POSGRADO

Analysis and implementation of active noise control strategies using piezo and EAP actuators

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Lima, March 2015

PONTIFICIA UNIVERSIDAD CATÓLICA DEL PERÚ

TECHNISCHE UNIVERSITÄT ILMENAU

Analysis and Implementation of Active Noise Control Strategies using Piezo and EAP Actuators

Masterarbeit

von

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Platzhalter für die Aufgabenstellung vom Prüfungsamt

Selbstständigkeitserklärung

Hiermit versichere ich, die vorliegende Arbeit selbstständig und unter ausschließlicher Verwendung der angegebenen Literatur und Hilfsmittel erstellt zu haben.

Die Arbeit wurde bisher in gleicher oder ähnlicher Form keiner anderen Prüfungsbehörde vorgelegt und auch nicht veröffentlicht.

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Unterschrift

Acknowledgements

God, thanks to guiding me in this stage...

This little work was made with help of:

Parents and sister, thanks for your love and support.

PUCP, DAAD and TU Ilmenau, thanks for the scholarship and thanks to create this cooperation programs, in order to motivate students for research.

Mr. Tafur, Mr Barriga thank you for your knowledge support in Control Theory and Mechanic Design, during the Master studies.

Mr. Sattel and Mr Ströhla, thank you to let me participate in the develop of this work.

Mr Chernukhin, thank you for all your academic advise. Your innate passion for engineering and science had motivate me to continue researching.

Mr. Medina, thank you for your guide in the beauty compression of Science Physics to be applied in Engineering.

Kurzfassung

Der Lärm beeinträchtigt die Lebensqualität der Menschen sowie an der Arbeit als auch Zuhause. Heutzutage wird die Lärmunterdruckung meistens durch die passiven oder semiaktiven Schallreduktionsmaßnahmen durchgeführt. Dabei sind diese mit hohen Platzbedarf und Kosten bei der Dämmung vom tieffrequenten Lärm verbunden. Darüber hinaus ist die Benutzung von passiven und semiaktiven Systemen wegen strengen Platzanforderungen oft gar nicht möglich. Deswegen werden die auf dem Konzept vom phasenverschobenen Signal basierenden aktiven Schalldämmungsmaßnahmen eingesetzt.

In der vorliegenden Arbeit wurden die Strategien und Algorithmen zur aktiven Schallreduktion in den Lüftungsanlagen analysiert, implementiert und experimentell getestet. Simulationen der Algorithmen wurden mit Hilfe von Matlab Software durchgeführt. Während der experimentellen Untersuchungen wurden der Piezoaktor als aktiver Element des Schallreduktionssystems eingesetzt.

41 Abbildungen
4 Tabellen
75 Seiten

Abstract

Noise, affecting people's life quality on the workplace and at home, is nowadays being reduced mostly using passive or semi active noise cancellation. A drawback of such systems is their size and costs by cancellation of low frequency noise. Moreover for space-critical applications the using of such systems is often impossible due to strictly space requirements. That's why the active noise control systems using a signal shifted in phase related to the original noise signal are seems to be a very promising concept for noise cancellation.

In the current work the strategies and algorithms for active noise reduction were analyzed, implemented and experimentally tested for application to the duct ventilation systems. Simulations of the algorithms were implemented using Matlab software. By experimental testing the piezo actuator was used as an active element of the Active Noise Control (ANC) system.

41 figures
4 tables
75 pages

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List of Abbreviation

ANC Active Noise Control
LMS Least Mean Square
FXLMS Filter X Least Mean Square
FIR Finite Impulse Response
IIR Infinite Impulse Response
AWGN Additive White Gaussian Noise
SNR Signal to Noise Ratio
SPL Sound Pressure Level

List of Symbols

$S_{ee}(\omega)$ auto power spectrum of error
$C_{dx}(\omega)$ coherence function s
$S_{dd}(\omega)$ spectrum of desired signal

CHAPTER 1

Introduction

1.1. Motivation

In order to provide an appropriate temperature inside of the buildings the ventilation system are being used. Due to the fact, that the heat transfer is being achieved using convection the fans and extractors are being used to produce wind flows.

Fans and extractors are loud in operation, therefore passive or active mechanisms are used in order to reduce the noise inside the building. However, passive mechanisms are costly and need big space to be fixed around the ventilation system to absorb the noise.

In order to avoid big costs and to meet space requirements for space critical applications ANC system are being used. Another big advantage of such systems is the possibility to program it for individual requirements of users regarding frequency band that should be damped.

1.2. Objectives

In this thesis the algorithms and strategies for active noise control should be analyzed, implemented and simulated in Matlab as well as experimentally tested in the real-time computation system for application in duct ventilation systems.

In order to achieve an active noise control the speakers based on piezo and EAP actuators should be tested.

1.3. Conclusions and future work

The ventilation systems produce sound by mechanical consequences of operation [Lar11]. For this reason it is necessary to design the devices that may attenuate and delete undesired sound waves. Furthermore, they can provide satisfactory performance for users. In this thesis it is an overview of some existing as well as design and implementation of new algorithm for ANC applicable for ventilation systems is given.

Simulated and experimental tests of an adaptive controller designed for a prototyped ventilation system are shown. The characteristics of the adaptive controller achieves attenuation of sound signals produced by the ventilation system and external disturbances which affect the system control. The system designed has a quickly response in finding the appropriate weights for the filter representing a system behavior.

A general review of many systems in the actual laboratories and industry, that have a good performance in reducing of the sound disturbances produced by mechanical effects in ventilation system is shown.

It is shown the mathematical equations to describe the process of attenuation and deletion of undesired noise [Lar11]. Furthermore, an analytical description for testing of the designed algorithm as well as a strategy for comparison between simulation and experimental results are proposed.

For testing of the designed algorithms a ventilation system prototype described in [Che15] was used. In order to design the algorithms for ANC some characteristic of this prototype were measured. The algorithms were simulated using Matlab and compared with experimental responses of designed prototype.

CHAPTER 2

Active Noise Control

2.1. State of the art

In many offices and buildings an adequate ventilation system is needed, in order to get a pleasant environment for people who are working or living inside. Nevertheless, the disadvantage of ventilation systems using is noise pollution, which is produced by fan motors. Therefore, it is necessary to control the noise. In order to achieve this a concept of the opposite wave is being used. In doing so a A wave sound is canceled by another wave with the same frequency but opposite amplitude. This property is widely used by industry. In many transport systems [Dab13], ANC is applied to prevent external noise signals affecting the performance of the driver and crew. This objective is also found in several intelligent headphones [AuT15]. The mining industry also requires ANC [Bar94] for hearing health care of their workers, who work with noisy engines. Currently, ANC is widely investigated in smartphone applications, to increase their functionality in noise control[N.12].

Figure 2.1 shows a summary of ANC applications. Different kinds of mechanisms have been designed in order to attenuate unwanted sound. For example, in ventilation systems noise is reduced by passive resistive silencers which are special materials placed around the duct ventilations [Gui07], [Lar11].

Figure 2.2 shows the components setup, which was used by author [Lar11] to evaluate the performance of ANC in duct systems.

Figure 2.3 shows the Cabin designed at the TU Ilmenau by Mr. Chernukhin, in which the ANC algorithms were tested.

Noise control of ventilation systems can be achieved by passive silencers, which are absorbent materials of noisy wave. However, these materials are expensive, if they are being used for damping of the low frequency noise.

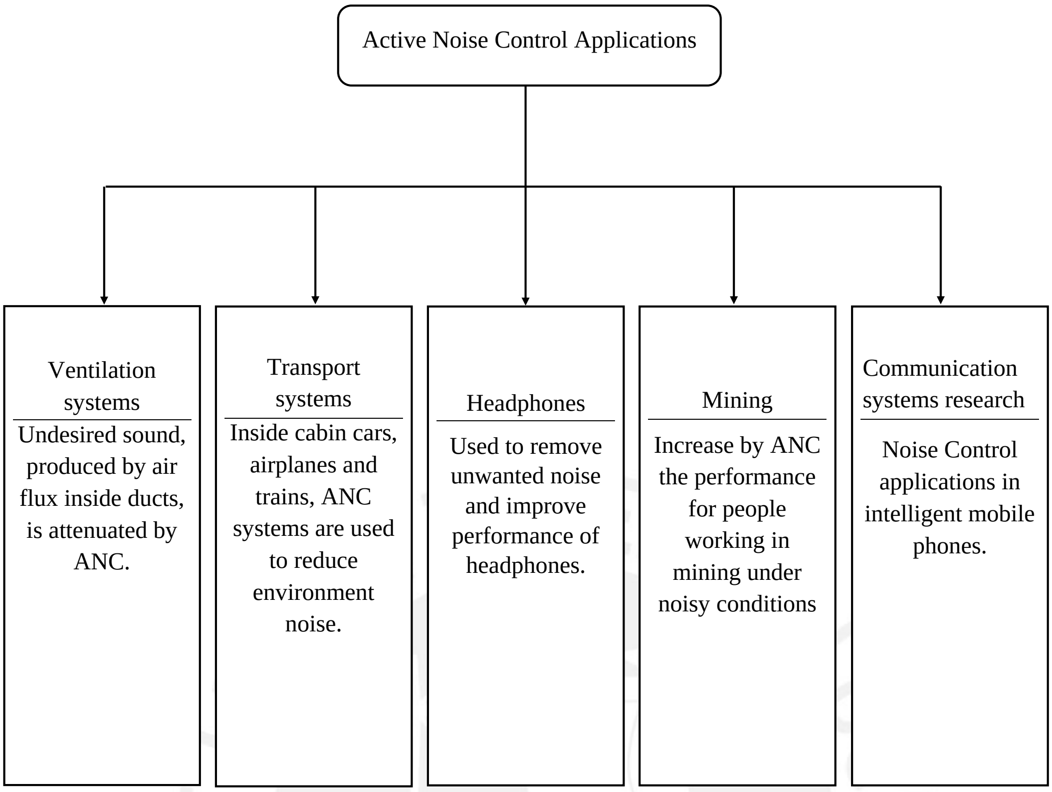

Figure 2.1.: ANC applications

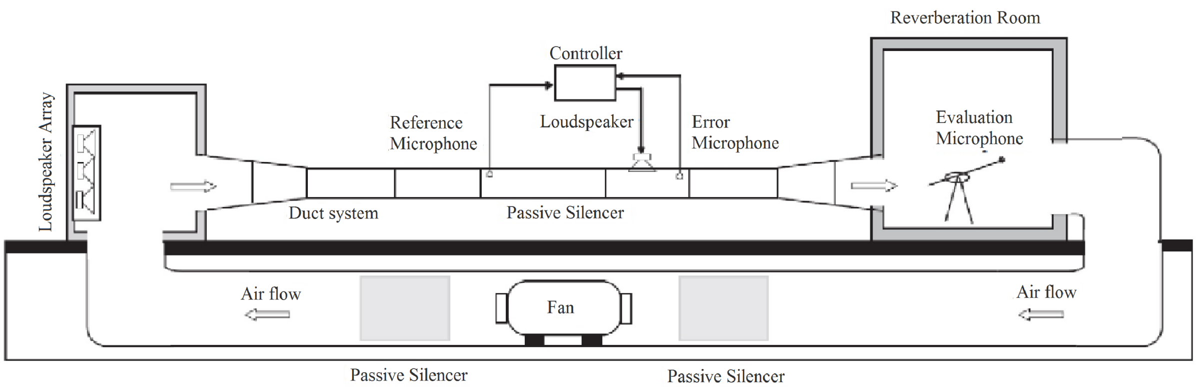

Figure 2.2.: ANC in duct systems proposed by author [Lar11].

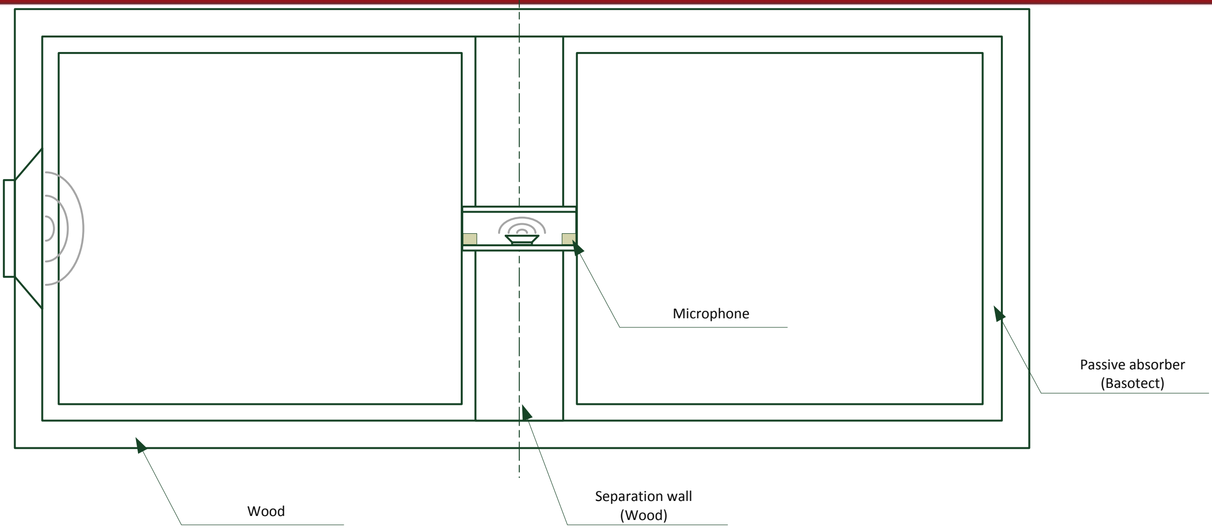

Figure 2.3.: Cabin designed in TU Ilmenau [Che15].

Nevertheless, the noise control can also be obtained by ANC, using control algorithms. In this case a noise signal coming to the first loudspeaker (most left in figure 2.3) is being called a "primary noise". The signal from the second loudspeaker is being called "secondary noise". The aim is to minimize a signal coming to the second microphone due to proper adjustion of the second loudspeaker output (see figure 2.3). In order to achieve this a control algorithm is being used, that tries to minimize an error between the signal being received by the second microphone and an estimation of this signal provided as an output of the control algorithm. The error analysis can be performed by feedback or feedforward controllers.

2.2. Feedforward and feedback controller

An adaptive ANC algorithm has been designed in order to achieve a better performance when the system is being affected by external conditions. The disturbances in the system are canceled due to automatic adjustment of the weights constants during the adaptation.

Figure 2.4, taken from [KM99], represents the general system identification for broad-band noise in ANC, where X(n), W(n), P(z) are respectively input signal, digital filter and plant transfer function.

The system designed should have the characteristic to measure the unwanted sound signal and be able to generate a signal capable to cancel the undesired signal. The equation 2.1, as is described by authors [KM99], shows the relation for auto power spectrum of error ($S_{ee}(\omega)$), which means the difference between the spectrum of desired signal ($S_{dd}(\omega)$) in the system with the coherence function ($C_{dx}(\omega)$). If this difference is equal to 1, then the auto power spectrum of residual error is null, that means no error in the canceling signal, because no power spectrum is saved. Nevertheless, if the coherence is equal to null, then the power spectral error is equal to power spectral of the desired signal, that is not the aim.

$$S_{ee}(\omega) = [1 - C_{dx}(\omega)]S_{dd}(\omega) \tag{2.1}$$

The estimation of an unknown plant $P(z)$ is achieved using an adaptive filter W(z) which is shown in figure 2.4 as proposed by [KM99]. The adaptive filter is flexible according to the change of the weights to be assigned.

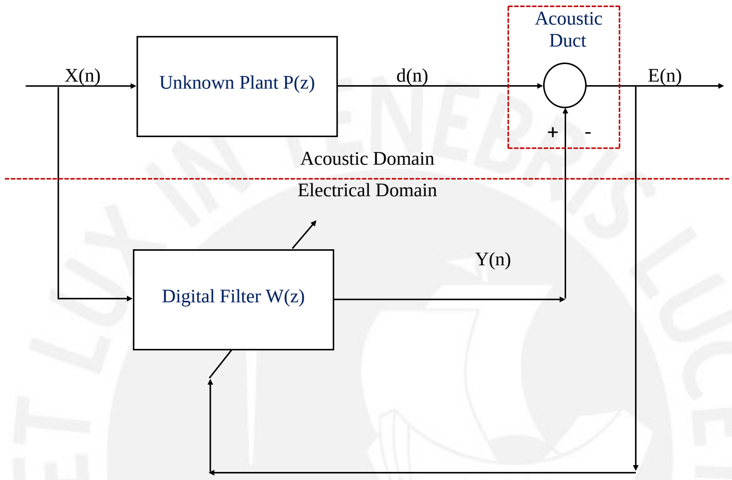

Figure 2.4.: System identification model proposed by authors [KM99].

After system identification, it is possible to design some kind of adaptive controller. Figure 2.5 shows the system model identification proposed by [KM99]. In order to get the attenuation of the undesired noise signal, LMS is defined as a procedure to obtain the error between the output and reference signal by a successive correction of the weights of the filter.

Figure 2.5 shows a simple ANC system with a filter, LMS and a secondary path. which was proposed by authors [KM99].

The error is given by the equation (2.2). That equation is achieved from the figure 2.5, which means a feedforward analysis in the system with the plant transfer function $P(z)$ through the secondary path $S(z)$ and the filter $W(z)$. The system is analyzed in Z transform, because it will be processed by a computer. The primary path is the acoustic response calculated from the reference sensor to the error sensor, that means transfer function $P(z)$. On the other hand, the secondary path $S(z)$ is the transfer function calculated from the filter output to the error sensor.

$$E(z) = [P(z) - S(z)W(z)]X(z) \tag{2.2}$$

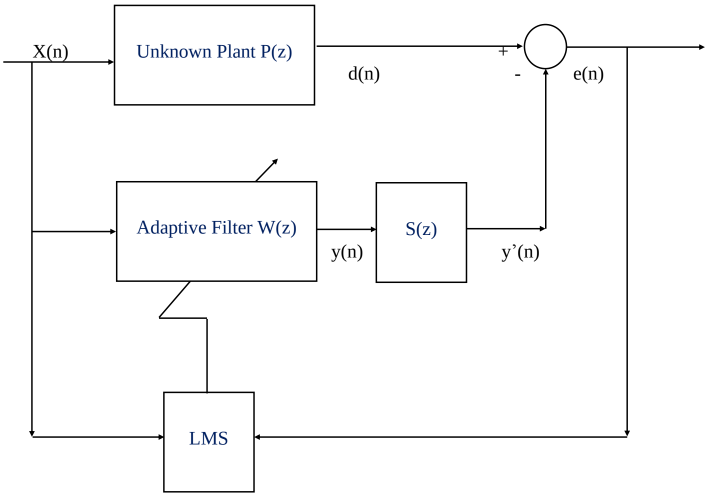

Figure 2.5.: Simple ANC scheme proposed by authors [KM99].

The digital transfer function of the filter is being analyzed by an ideal error equal to zero, is shown in the equation (2.3)

$$W(z) = \frac{P(z)}{S(z)} \tag{2.3}$$

2.2.1. FXLMS algorithm analysis

The filter that properly adjust the error with the reference signal $X(Z)$, is known as FXLMS filter. This filter avoids instabilities caused by the presence of the transfer function $S(Z)$ of the secondary path FXLMS filter [KM99] and [Mor13]. Equation 2.4 shows the error signal through the FXLMS algorithm: In which $d(n)$ is the desired signal, $S(n)$ is the impulse response of the secondary path transfer function applied at time instant n, $\mathbf{w}(n)$ and $\mathbf{X}(n)$ are the coefficient and signal vectors of the filter $W(z)$, $\xi(n)$ is the mean square cost function, * denotes linear convolution [KM99].

$$e(n) = d(n) - S(n) * [\mathbf{w}^T(n)\mathbf{X}(n)] \tag{2.4}$$

Then the error output of adaptive filter is given by equation 2.5 [KM99].

$$\hat{\xi}(n) = e^2(n) \tag{2.5}$$

$$\mathbf{w}(n+1) = \mathbf{w} - \frac{\mu}{2}\nabla\hat{\xi}(n) \tag{2.6}$$

from equations 2.4 and 2.5, as was analyzed by the authors [KM99]:

$$\nabla e(n) = -S(n) * X(n) = -X'(n) \tag{2.7}$$

With which is defined:
$$\nabla\hat{\xi}(n) = -2X^{'}(n)e(n) \quad (2.8)$$

replacing equation 2.8 in equation 2.6, as was proposed by authors [KM99]:

$$\mathbf{w}(n+1) = \mathbf{w}(n) - \mu X^{'}(n)e(n) \quad (2.9)$$

Then the secondary path is achieved [KM99]:

$$X^{'}(n) = \hat{S}(n) * X(n) \quad (2.10)$$

For the adaptive filter to carry through directly to the error signal, the FXLMS ANC algorithm should be changed by the assumption: The estimated secondary path signal is the same as the secondary path signal. This process is represented by the figure 2.6 and 2.7, as was analyzed by authors [KM99].

It is necessary to know the maximal step size μ as a function of the input signal power P and sampling time Δ , with target to design the control filter. The process described is verified by the equations 2.11 and 2.12 and explained by the authors [KM99].

$$\mu_{max} = \frac{1}{(P_{x'}(L+\Delta))} \quad (2.11)$$

$$P_{x'} = E[x^{'2}(n)] \quad (2.12)$$

Figure 2.6 shows the ANC using FXLMS algorithm designed, proposed by authors [KM99].

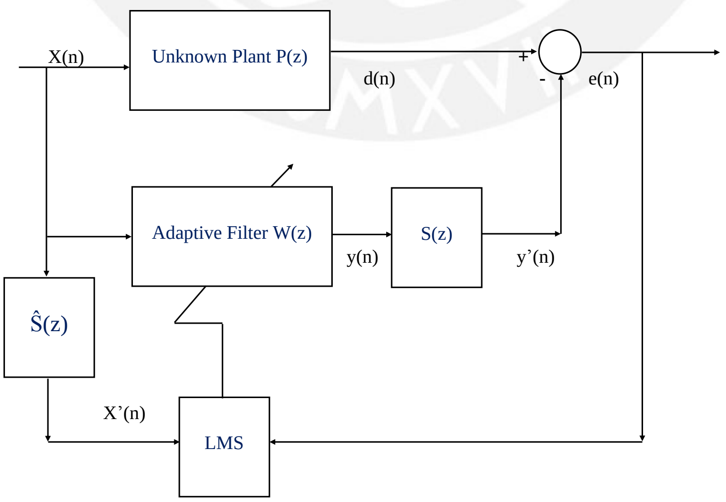

Figure 2.6.: ANC using FXLMS algorithm [KM99].

The figure 2.7 shows the equivalent of the ANC using FXLMS algorithm designed, when $\hat{S}(z) = S(z)$

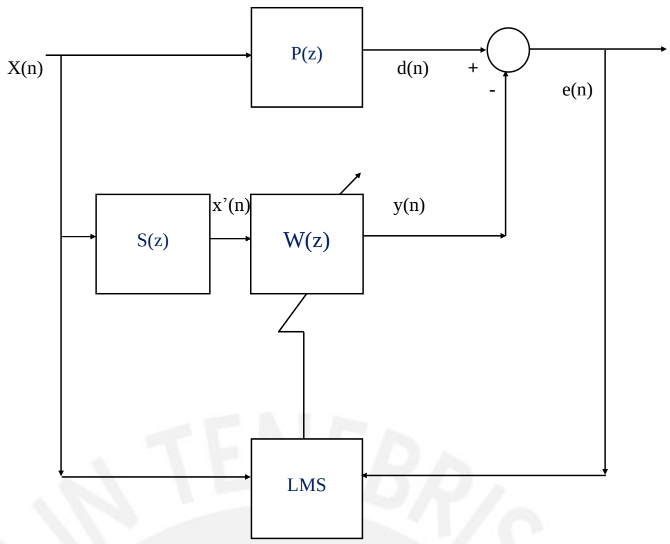

Figure 2.7.: ANC using FXLMS algorithm for the case $\hat{S}(z) = S(z)$

The Optimal Unconstrained Transfer Function is given by the equation 2.13 in which $W^0(z)$ is the Optimal Unconstrained Transfer Function, S_{xx} is auto power spectrum of primary signal X, S_{uu} is an auto power spectrum of measurement noise $u_(n)$ associated with sensors, P(z) is the transfer function of primary path, S(z) is the transfer function of secondary path. Authors [KM99] have analyzed from equation 2.13 that measurement noise will be in presence of input signal X.

$$W^0(z) = \frac{P(z)S_{xx}(z)}{[S_{xx}(z) + S_{uu}(z)]S(z)} \tag{2.13}$$

It is usual that the secondary path signal is distorted when high noise level is introduced in low frequencies. The solution is to modify the cost function in order to constrain the adaptive filter weights, as was proposed by the authors in [KM99] and shown in equation 2.14 that represents a leaky FXLMS algorithm, γ is weight in the control, $\omega(n)$ and $\mathbf{X}(n)$ are the coefficient and signal vectors of the filter $W(Z)$, $\xi(n)$ is the mean square cost function, $e(n)$ is the error, μ is step size, as is analyzed by authors [KM99].

$$\hat{\varepsilon}(n) = \epsilon^2(n) + \gamma\omega^T(n)\omega(n) \tag{2.14}$$

in which

$$\omega(n+1) = \nu\omega(n) + \mu x^{'}(n)e(n) \tag{2.15}$$

$$\nu = 1 - \mu\gamma \tag{2.16}$$

and the Leakage Factor is inside the range $0 < \nu < 1$. Leakage factor has information of stabilizing effect of the system.

2.2.2. Feedforward FXLMS ANC algorithm

This algorithm works with the estimated signal in order to get a good compensation with the secondary path signal. The feedback FXLMS ANC algorithm works with the error signal directly to achieve a better estimated signal. On other hand, the hybrid feedforward/feedback FXLMS ANC algorithm has the advantage of a quick response by the filter adaptation.

From analysis made by the authors [KM99]: The steady state transfer function is given by the equation 2.17, in which $P(z)$ is the transfer function of primary path, $S(z)$ is the transfer function of secondary path and $F(z)$ is the feedback path transfer function from the output of adaptive filter $W(z)$.

$$W^0(z) = \frac{P(z)}{S(z) + P(z)F(z)} \tag{2.17}$$

The open-loop transfer function associated with the feedback loop is shown by the equation 2.18. This equation is important to analyze stability of the system [KM99].

$$H^{ol}(z) = W(z)F(z) \tag{2.18}$$

which is the same to the equation 2.19 :

$$H^{ol}(z) = \frac{P(z)F(z)}{S(z) + P(z)F(z)} \tag{2.19}$$

Figures 2.8 shows the block diagram of FXLMS ANC algorithm with feedback from secondary source, analyzed by the authors [KM99].

Figure 2.9 shows the block diagram of the most simple Feedback FXLMS ANC algorithm designed by the authors [KM99]. The feedback neutralization is achieved between the difference of the secondary signal with the reference sensor signal:

Figure 2.10 shows the block diagram for Adaptive IIR filter, which for stability reasons can be approximated by an Finite Impulse Response (FIR) function with small step size [KM99].

The output signal from IIR filter is given by the equation 2.20 as was described by [KM99]:

$$y(n) = a^T(n)x(n) + b^T(n)y(n-1) \tag{2.20}$$

In which the components a(n) and b(n) are shown in the following equations 2.21 and 2.22 and L is the number of samples.

$$a(n) = \begin{pmatrix} a_0(n) & a_1(n) & \dots & a_{L-1}(n) \end{pmatrix}^T \tag{2.21}$$

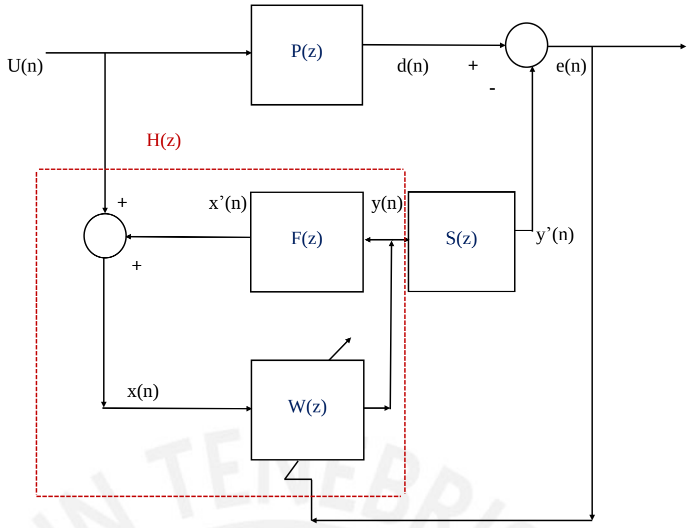

Figure 2.8.: ANC system wih feedback [KM99].

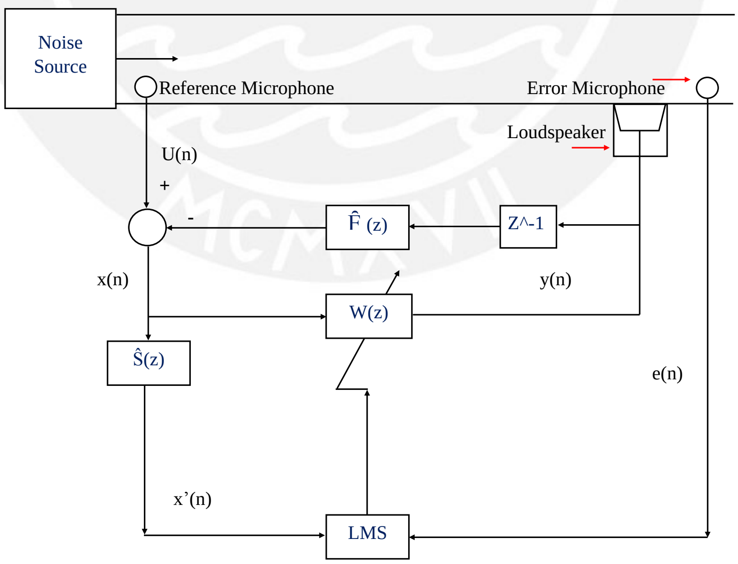

Figure 2.9.: ANC with acoustic feedback neutralization [KM99].

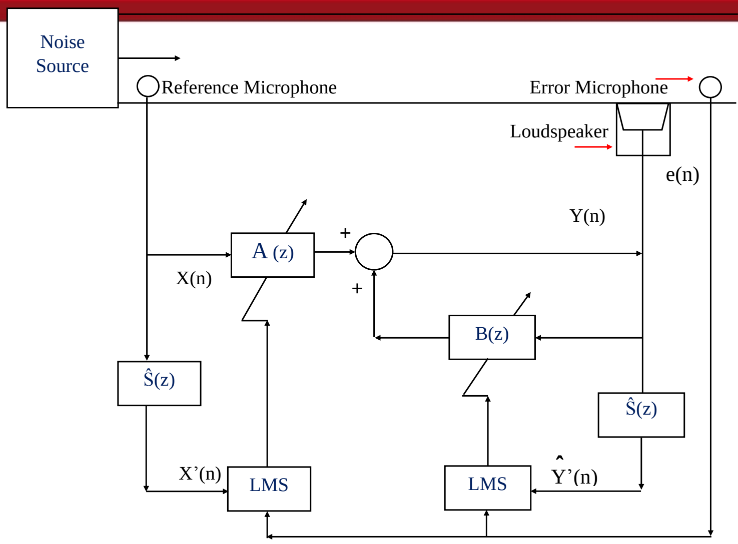

Figure 2.10.: Adaptive IIR filter ANC system [KM99]

$$b(n) = \begin{pmatrix} b_1(n) & b_2(n) & \dots & b_M(n) \end{pmatrix}^T \tag{2.22}$$

The optimal weights to minimize the error are given by equations 2.23 and 2.24, in which $e(n)$ is the error, μ is the step size, and $X(n)$ is the primary signal.

$$a(n+1) = a(n) + \mu x^{'}(n)e(n) \tag{2.23}$$

$$b(n+1) = b(n) + \mu \hat{y}^{'}(n-1)e(n) \tag{2.24}$$

The filtered signal, by the adaptive IIR filter, is shown in equation 2.25, in which $S(n)$ is the secondary path transfer function.

$$\hat{y}^{'}(n-1) = \hat{S}(n) * y(n-1) \tag{2.25}$$

2.2.3. Narrow Band Active Noise Control

Many systems as motors, fans, and engines produce periodic noise, that make it possible to find fundamental frequency and harmonics in order to get ANC in narrow band, that is represented in figure 2.11. Nevertheless, not every system produces periodic noise signals. [KM99]

Figure 2.11 shows block diagram for Narrow Band Feedforward controller.

The actual canceling noise waveform for the secondary loudspeaker is shown in the equation 2.26:

$$y(n) = \omega_{j(n)}(n) \tag{2.26}$$

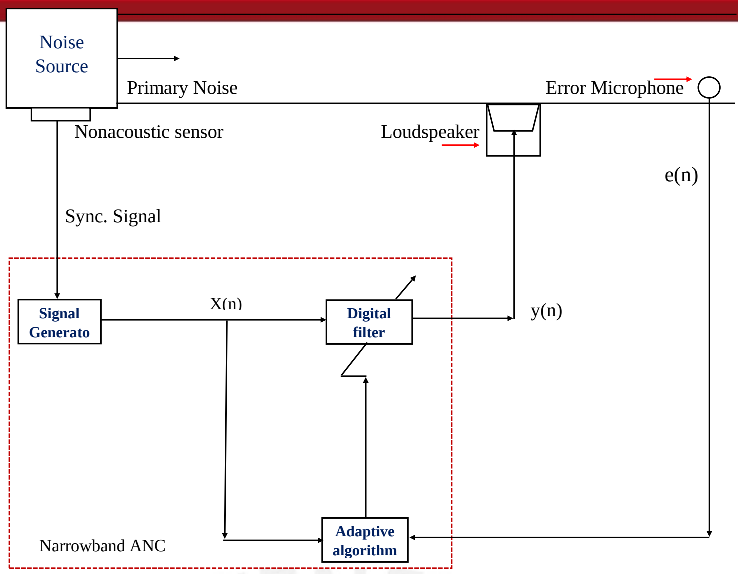

Figure 2.11.: Block diagram Narrow Band Feedforward controller [KM99].

where $j(n)$

$$j(n) = n \mod L \tag{2.27}$$

"It can be implemented as a pointer incremented in a circular fashion between zero and L - 1 for each sampling period, controlled by interrupts generated from the synchronization signal" [KM99]. Where L is number of samples, n is instant time, τ is time delay, T is sampling period, $N = \frac{T_0}{T}$, T_0 is period of time with fundamental frequency ω_0.

The adaptation unit in order to achieve the weights is calculated by LMS and shown in the equation [KM99]:

$$\omega_l(n+1) = \omega_l(n) + \mu e(n) \tag{2.28}$$

only when:

$$l = j(n - \delta) \tag{2.29}$$

also:

$$\delta = [\tau/T] \tag{2.30}$$

$$\omega_l(n+1) = \omega_l(n) \tag{2.31}$$

For other cases the waveform synthesis method is equivalent to an adaptive FIR excited by a Kronecker impulse, as was analyzed by the authors [KM99]:

$$\omega_l(n+1) = \omega_l(n) \tag{2.32}$$

$$x(n) = \sum_{k=-\infty}^{\infty} \delta(n - kN) \tag{2.33}$$

The figure 2.12 shows how the periodic noise is cancelled by the output of an adaptive filter, using the periodic impulse train as reference x(n).

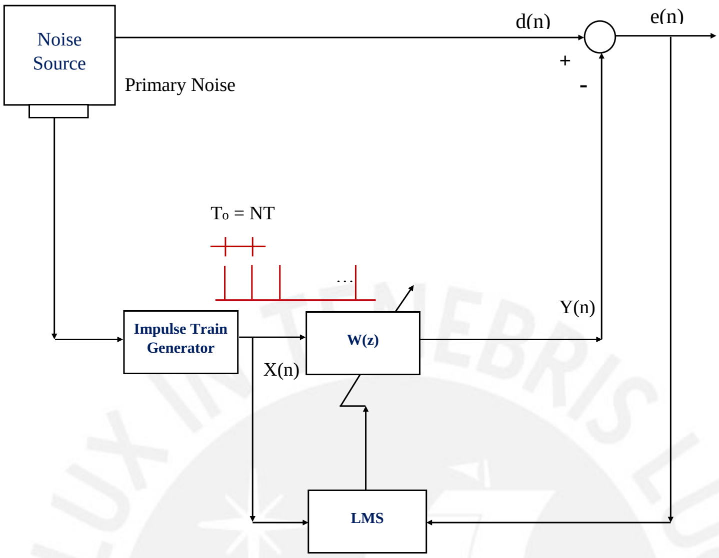

Figure 2.12.: Block diagram Impulse Response analyzed by the authors [KM99].

The transfer function H(z) between the primary input D(z) and the error output E(z) is shown in the equation 2.34, which was analyzed by the authors [KM99].

$$H(z) = \frac{E(z)}{D(z)} = \frac{1 - z^{-L}}{1 - (1 - \mu)z^{-L})} \tag{2.34}$$

The parameter μ must be inside of the range as suggested by authors [KM99] for stability considerations:

$$0 < \mu < 1 \tag{2.35}$$

The effects of secondary path S(z) must be compensated by using the FXLMS algorithm that calculates a secondary path estimate $\hat{S}(z)$.

For the order L = N, the output is computed with the equations 2.36 as was described by the authors [KM99]:

$$x^{'}(n) = \sum_{l=0}^{L-1} \hat{S}_l x(n - l) \tag{2.36}$$

Therefore, the FXLMS algorithm for synchronous waveform synthesis, the ANC systems is described by the equation 2.37:

$$\omega_l(n + 1) = \omega_l(n) + \mu e(n)\hat{s}_k(n, l) \tag{2.37}$$

in which

$$l = 0, 1, ..., L-1 \tag{2.38}$$

also:

$$k(n,l) \equiv (n-l) \mod L \tag{2.39}$$

The presence of S(z) modify the transfer function of the controller in the equation 2.40, as was described by the authors [KM99]:

$$H(z) = \frac{1 - z^{-L}}{1 - (1 - \mu s(z))z^{-L}} \tag{2.40}$$

The steady state transfer function for delayed LMS algorithm is given by the equation 2.41, proposed by the authors [KM99]:

$$H(z) = \frac{1 - z^{-L}}{1 - (1 - \mu z^{-[\Delta/L]L})z^{-L}} \tag{2.41}$$

where

$$\hat{s}(z) = z^{-\Delta} \tag{2.42}$$

and Δ is the number of samples of delay for y(n) to e(n)

2.2.4. Adaptive Notch Filter

“An adaptive Notch filter can be realized by using an adaptive Noise Canceller with a sinusoidal reference signal. The advantages of the Adaptive Notch filter are that it offers easy control of Bandwidth (this is important for ventilation duct systems), an infinity null, and the capability to adaptively track the exact frequency of the interference in order to get a good performance for ANC” [KM99]. The figure 2.13 shows a single frequency adaptive Notch filter proposed by the authors [KM99].

The steady state transfer function $H(z)$ from the primary input d(n) to the noise canceller output e(n) is given by the equation 2.43, where A is an amplitude of input signal $X(n)$, $z = e^{j\omega_0}$ or $z = e^{-j\omega_0}$, j is complex index, as was analyzed by the authors [KM99]:

$$H(z) = \frac{E(z)}{D(z)} = \frac{z^2 - 2zcos(\omega_0) + 1}{z^2 - (2 - \mu A^2)zcos(\omega_0) + 1 - \mu A^2} \tag{2.43}$$

For a general Lth order adaptive filter, the equation 2.43 becomes in the following equation:

$$H(z) = \frac{z^2 - 2zcos(\omega_0) + 1}{z^2 - (2 - \frac{\mu L A^2}{2})zcos(\omega_0) + 1 - \frac{\mu L A^2}{2}} \tag{2.44}$$

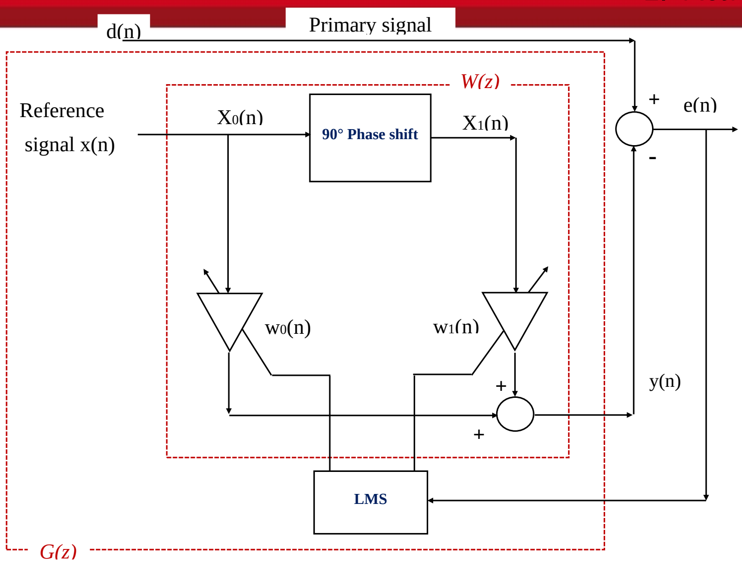

Figure 2.13.: Single frequency adaptive Notch filter [KM99].

The 3dB bandwidth of the Notch filter is estimated with the approximation, which was proposed by authors [KM99]:

$$B \approx \frac{\mu L A^2}{4\pi T} \tag{2.45}$$

The time constant of the adaptation is given by the following approximation, as was proposed by authors [KM99]:

$$\tau_{mse} \leq \frac{2T}{\mu A^2} \tag{2.46}$$

Single frequency ANC: The application of the Adaptive Notch Filter to periodic ANC has been developed by Ziegler. A recursive quadratic oscillator provides two orthogonal components, as was proposed by the authors [KM99]:

$x_0(n)$ and $x_1(n)$

which are used as reference inputs in the adaptive filter. The delayed LMS algorithm updates the filter weights to minimize the residual error, that is shown by the equation 2.47

$$\omega_l(n+1) = \omega_l(n) + \mu e(n) x_l(n-\Delta) \tag{2.47}$$

where $l = 0$ or $l = 1$ and Δ is the secondary path transfer function compensation, as was proposed by the authors [KM99].

The above delay unit can be replaced by a secondary path estimate of S(z) as in the figure 2.14, which was proposed by the authors [KM99]:

The adaptive weights are updated by equation 2.48:

$$\omega_l(n+1) = \omega_l(n) + \mu e(n) x_l' e(n) \tag{2.48}$$

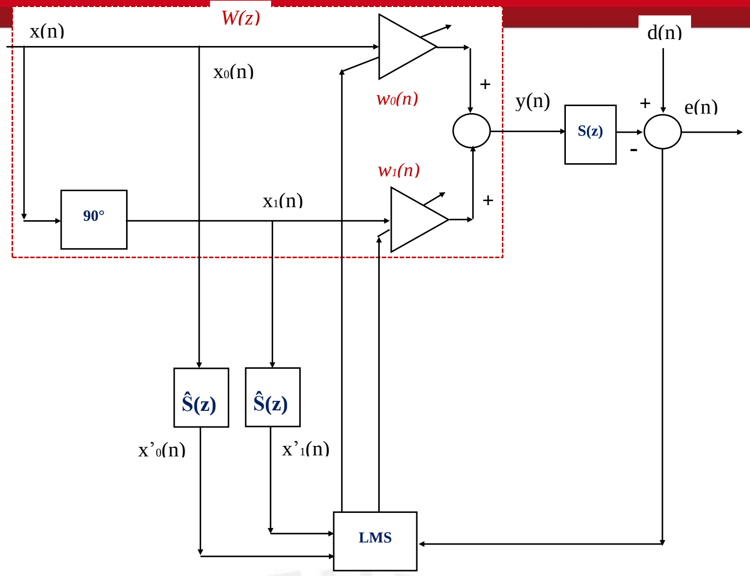

Figure 2.14.: Block diagram of single frequency ANC FXLMS algorithm [KM99].

where $l = 0$ or $l = 1$.

The transfer function of the narrow band ANC system is given by the equation 2.49, as was proposed by authors [KM99], in which A is amplitude of input signal $X(n)$, $z = e^{j\omega_0}$ or $z = e^{-j\omega_0}$, j is complex index, μ is step size, $\beta = \mu A^2 A_s$, A_s is the amplitude of $S(z)$, and ϕ_Δ is the phase difference between $S(z)$ with its estimated.

$$H(z) = \frac{z^2 - 2zcos(\omega_0) + 1}{z^2 - [2cos\omega_0 - \beta cos(\omega_0 - \phi_\Delta)]z + 1 - \beta cos(\phi_\Delta)} \tag{2.49}$$

From equation 2.49 the stability condition is given by:

$$cos(\phi_\Delta) > 0 \; or \; -90 < \phi_\Delta < 90 \tag{2.50}$$

2.2.5. Multiple Frequency ANC

In practical applications, periodic noise usually contains multiple tones at the fundamental frequency and at several harmonic frequencies. This type of noise can be attenuated by a filter with multiple Notches. In general, realization of multiple Notches requires higher orders filters, which can be implemented by direct, parallel, direct/parallel or cascade forms [KM99].

Direct form: A method for eliminating multiple sinusoids or other periodic interference was proposed by Glover as indicate [KM99]. The reference input is a sum of M sinusoids where A_m and W_m are the amplitude and frequency respectively as is shown in the equation 2.51. Analysis from [KM99]

$$x(n) = \sum_{m=1}^{M} A_m cos(\omega_m n) \tag{2.51}$$

“Parallel form: For the case in which the undesired primary noise contains M sinusoids, M two weight adaptive filters can be connected in parallel to attenuate this narrow band components. The canceling signal is a sum of M adaptive filter outputs, where each output $Y_m(n)$ is derived as in the single frequency case; as is shown in the equation 2.52” [KM99]:

$$y(n) = \sum_{m=1}^{M} Y_m(n) \tag{2.52}$$

“Direct parallel form is a configuration of multiple reference signal generator and corresponding adaptive filters has been developed to improve the performance of ANC system for automotive applications. The idea is to separate a collection of many harmonically related sinusoids into mutually exclusive sets that individually have frequencies spaced out as far as possible” [KM99].

“Cascade form, ideally, multiple sinusoids references are more effectively employed in a cascade of M second order single frequency notch filters. The overall response of such an arrangement is shown in the equation 2.53”, as was analyzed by the authors [KM99]:

$$H(z) = \prod_{m=1}^{M} H_m(z) = \prod_{m=1}^{M} \frac{1}{1 + S(z)W_m(z)} \tag{2.53}$$

2.2.6. Active Noise Equalizer

The equalizer retains small amount of information from error signal in order to perform bandwidth control and obtain maximal attenuation of periodic noise. Therefore, Active Noise Equalizer system usually attenuates incoming noise in engines because of its periodic behaviour. A block diagram of the general narrow band active noise equalizer system for controlling periodic noise is shown in figure 2.15, as was proposed by authors the [KM99]:

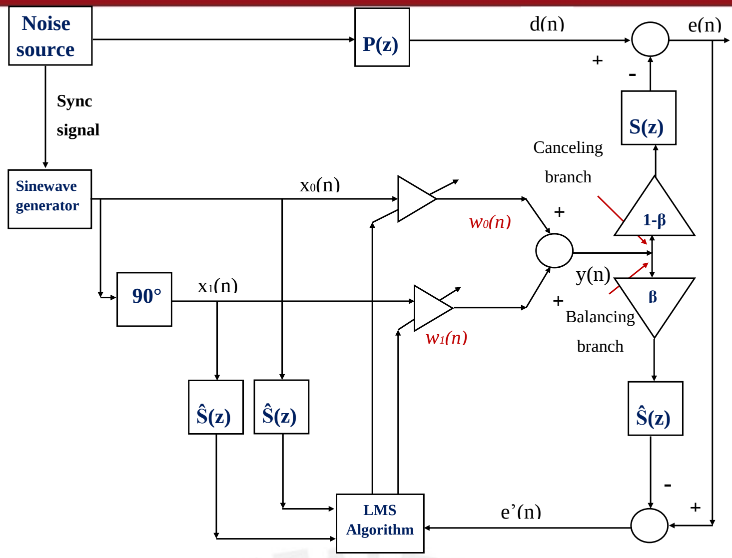

Figure 2.15.: Block diagram single frequency equalizer [KM99].

2.3. Adaptive feedback Active Noise Control Systems

"The basic idea of an adaptive feedback ANC is to estimate the primary noise and use it as a reference signal x(n) for the ANC filter". The error sensor output is processed by an ANC system to generate the secondary signal [KM99]. That is shown by equation 2.54 [KM99].

$$X(z) \equiv \hat{D}(z) = E(z) + \hat{S}(z)Y(z) \tag{2.54}$$

In figure 2.16, the primary noise is expressed in the Z domain as $D(z) = E(z) + S(z)Y(z)$, where $E(z)$ is the signal obtained from the error sensor and $Y(z)$ is the secondary signal generated by the adaptive filter. If the estimated of $S(z)$ is approximatly the value of $S(z)$, it is possible to estimate the primary noise d(n) and use this as an synthesized reference signal $x(n)$. This was explained by the authors [KM99]:

The complete single channel adaptive feedback ANC using the FXLMS algorithm is ilustarted in the figure 2.17, as was proposed by authors [KM99]:

X(n) synthesized is shown in the approximation, as was proposed by authors [KM99]:

$$X(n) \equiv \hat{d}(n) = e(n) + \sum_{m=0}^{M-1} \hat{s}(m)y(n-m) \tag{2.55}$$

If the estimated of S(z) is S(z), the overall transfer function H(z) of feedback ANC from d(n) to e(n) is given by the equation 2.56 [KM99]:

$$H(z) = \frac{E(z)}{D(z)} = (1 - S(z)W(z)) \tag{2.56}$$

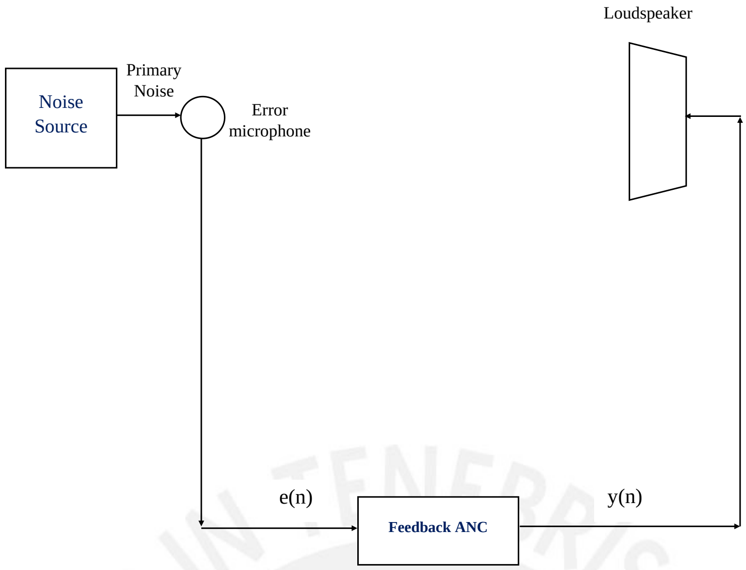

Figure 2.16.: Block diagram of single channel Feedback ANC [KM99].

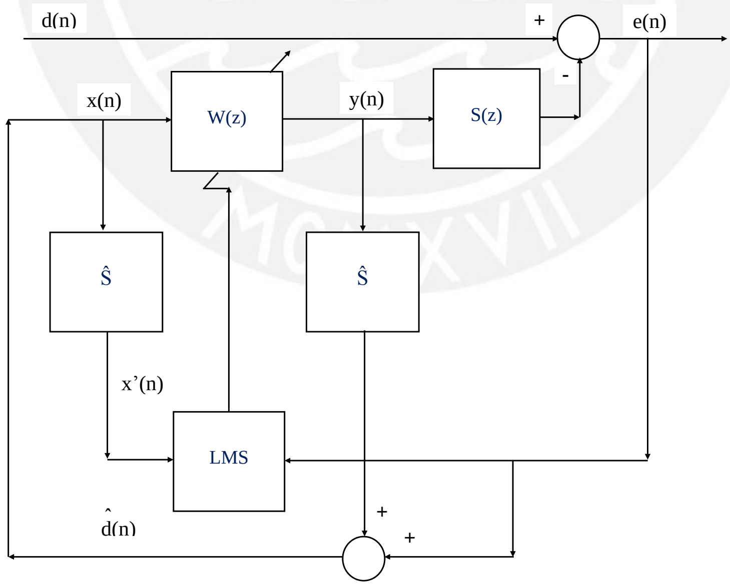

Figure 2.17.: Block diagram single channel Feedback ANC [KM99].

Figure 2.18 shows a block diagram for an adaptive predictor, which generates its own reference signal based on the adaptive filter output, as was analyzed by authors the [KM99]:

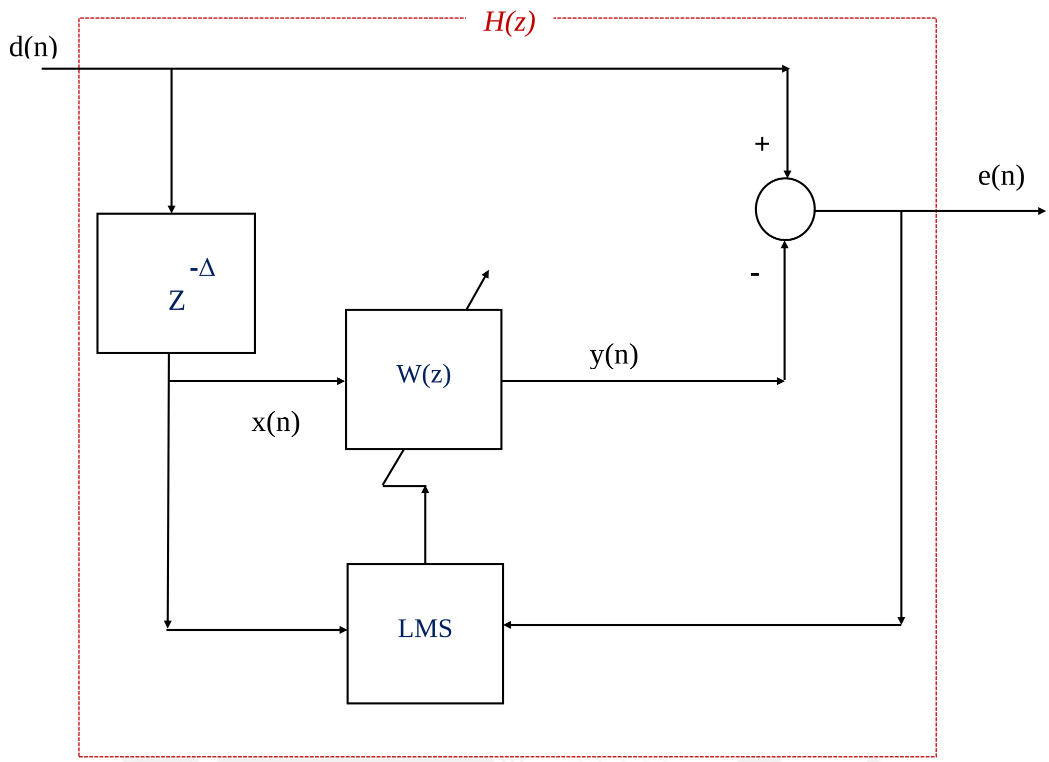

Figure 2.18.: Block diagram Adaptive Predictor [KM99].

2.3.1. Other Feedback ANC Algorithm

"The output whitening feedback ANC method assumes that the primary noise d(n) is formed by passing white noise through a moving average (MA) filter P(z). The secondary source is assumed to be placed close enough to the primary source so that the secondary path transfer function S(z) has minimum phase. Then from linear estimation theory, the optimal controller is expressed in the equation 2.57", as was analyzed by the [KM99]:

$$W^0(z) = \frac{\hat{P}(z) - 1}{\hat{S}(z)} \tag{2.57}$$

"The prediction part is then formulated using the standard Kalman filter setup. The output of the error sensor contains much more information about the future values of the primary noise at the error sensor than the output of the reference sensor" [KM99].

2.4. Hybrid controller

The junction of feedback and feedforward controllers, for ANC, achieves to keep the system controlled in presence of external disturbances. However, the time it takes to generate the estimated signal to attenuate the primary signal (noisy wave) will be longer than the time needed by the feedforward controller, but shorter compared to the time needed by feedback controller.

In hybrid ANC System, the feedforward ANC uses two sensors: a reference sensor and an error sensor. The reference sensor measure the primary noise to be canceled, while the error sensor monitors the performance of the ANC system. The adaptive feedback ANC system uses only an error sensor and cancels only the predictable noise components of the primary noise. A combination of the feedback and feedforward control structures is called a hybrid ANC System, as illustrated in figure 2.19 and as was proposed by the [KM99]:

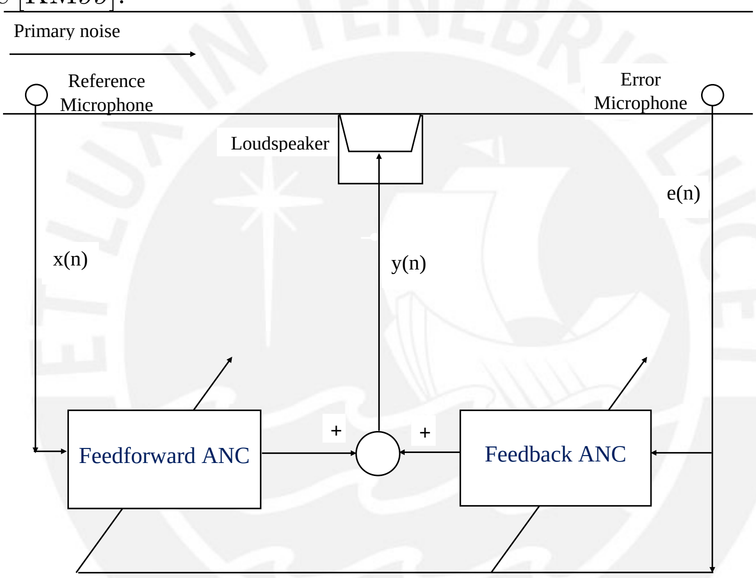

Figure 2.19.: Block diagram Adaptive Predictor [KM99].

Table 2.1 shows some properties of the described algorithms for ANC control systems, which are verified by simulations and experiments.

Algorithm	settling time	disturbance rejection
Feedforward	short	bad
Hybrid	average	average
Feedback	long	good

Table 2.1.: Algorithms comparison

“The Hybrid ANC system using FIR feedforward ANC and the adaptive feedback ANC is illustrated in figure 2.20, where the secondary signal y(n) is generated using the outputs of both the feedforward ANC filter A(z) and the feedback ANC filter C(z). The combined controller W(z) has two reference inputs: x(n) from the reference sensor and the estimated from d(n) (the estimated primary signal). Filtered versions of the reference signals x'(n) and d'(n) are used to adapt the coefficients of the filters A(z) and C(z) respectively", as was proposed by the authors [KM99].

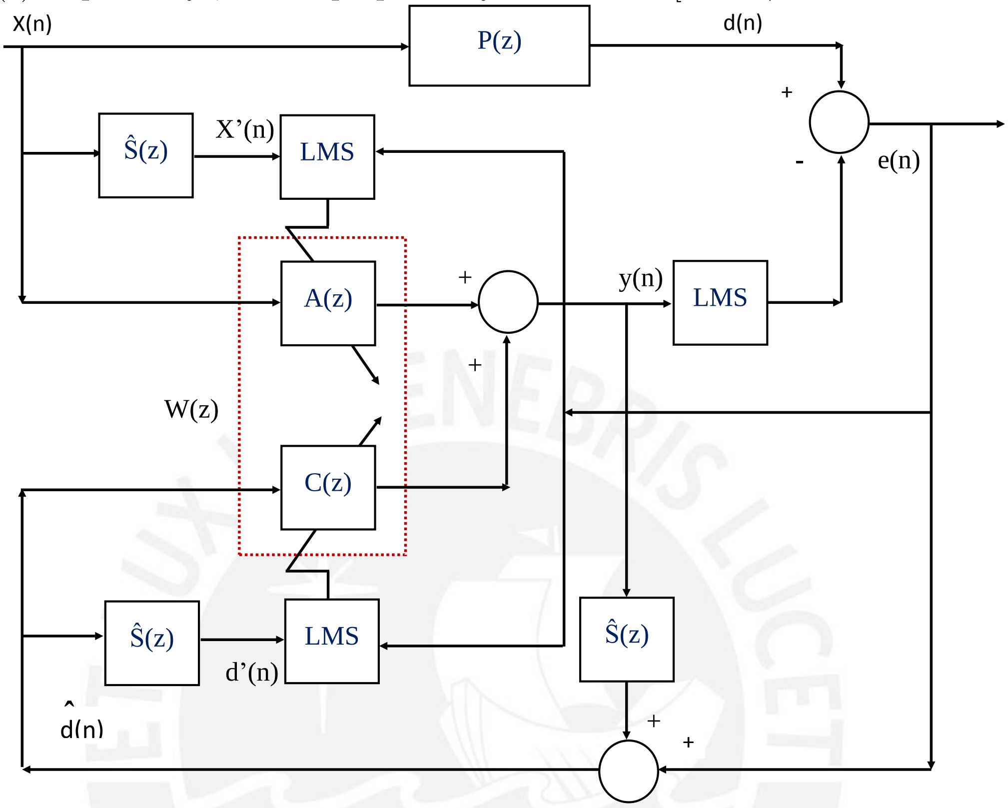

Figure 2.20.: Block diagram for hybrid algorithm.

CHAPTER 3

Simulation study of strategies for Active Noise Control

3.1. Selection, modeling and evaluation of control algorithms for Active Noise Control

In order to design control algorithms for ANC is necessary to identify the system. System identification provides the coefficients for the adaptive control algorithm design, using the FXLMS filter.

The FXLMS algorithm for simulations was designed analyzing the codes of the authors [Che15] and [Oey10].

The primary signal, for every simulation test, is a sine wave with 3 kHz and 88.17 dB (was obtained by equation 3.1), and sampling frequency of 15 kHz. Additive White Gaussian Noise (AWGN) was added to the input signal, with 20 dB as Signal to Noise Ratio (SNR) in order to simulate a real signal. This SNR value was chosen by analyzing the amplitudes of the sinusoidal signal and the actual signal. It is observed that the simulation error during system has a settling time of 0.06s and filtered signal decrease its Sound Pressure Level (SPL) in 12.99 dB.

Sound waves are usually described using SPL value in dB, therefore the electrical signal analyzed in the algorithms should be expressed in SPL terms in order to obtain an uniform units conversions between electrical and acoustic values.

Equation 3.1 converts electrical signals to SPL, which is appropriate to describe sound waves. This equation is being used in simulation and experimental results, to explain some changes produced by the execution of ANC in dB units.

$$microphSPL = msen + 20log10(microphRead/msenTF) + refSPL \tag{3.1}$$

where $microphSPL$ is the microphone measurement, $msen$ means microphone sensitivity which is known for the tests: -46 dB re $1V/Pa$, $microphRead$ is the signal sent to the ADC from microphone lecture, $msenTF$ is the microphone sensitivity transfer factor which is known for the tests: $5.0119E10^{-}3V/Pa$, $refSPL$ is reference SPL which is known for the tests: 94 dB SPL.

Figure 3.1 shows the simulation of identification error, which is obtained by executing the FXLMS alforithm with the input signal data, and the adaptive coefficients are generated.

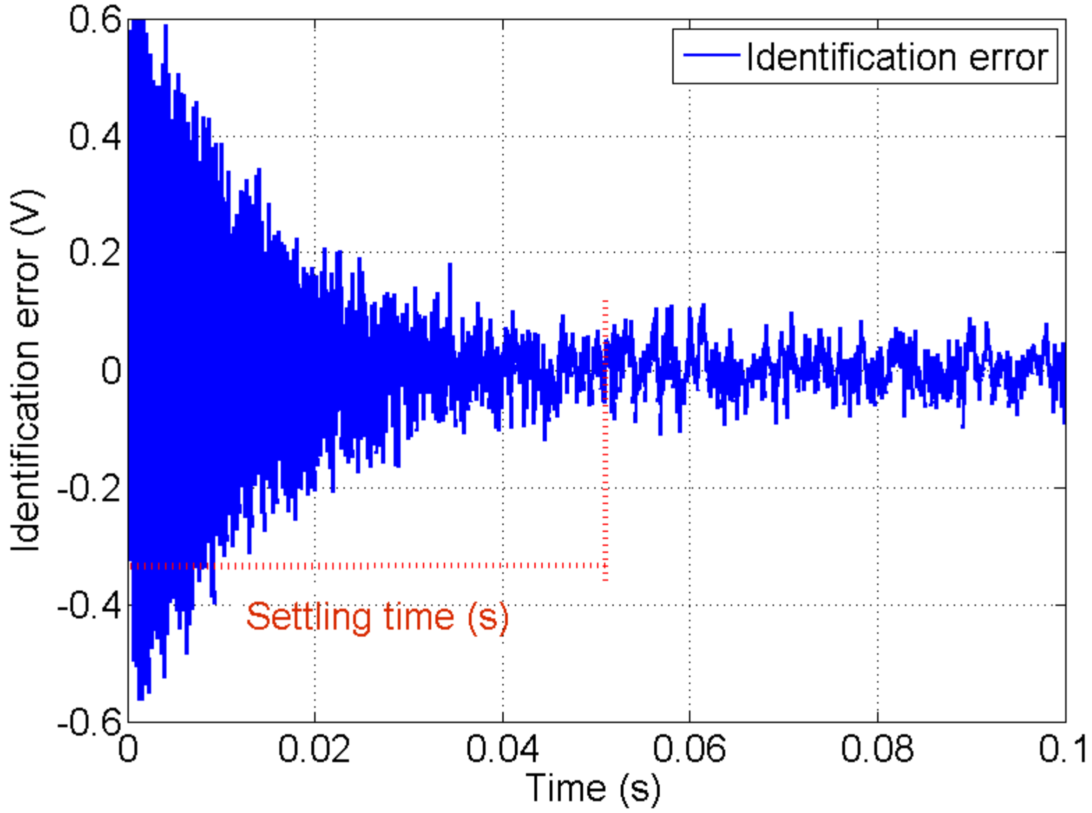

Figure 3.1.: Identification error

Feedforward, feedback, and hybrid (feedforward/feedback) algorithms were designed and simulated in order to analyze their properties in ANC (reference algorithms [Che15] and [Oey10]).

Figure 3.2 shows execution results of ANC feedforward algorithm, the settling time was 0.07 s, which is analyzed from the noise residue curve. The algorithm tries to create a control signal that should be more similar possible to the input signal, as it is shown in figure 3.2, finally the output signal decrease its SPL in 23.22 dB.

Figure 3.3 shows a zoomed view figure 3.2 in time range: 0.070s to 0.077s, in order to show that estimated control signal should be more similar to primary signal; with the aim to attenuate it.

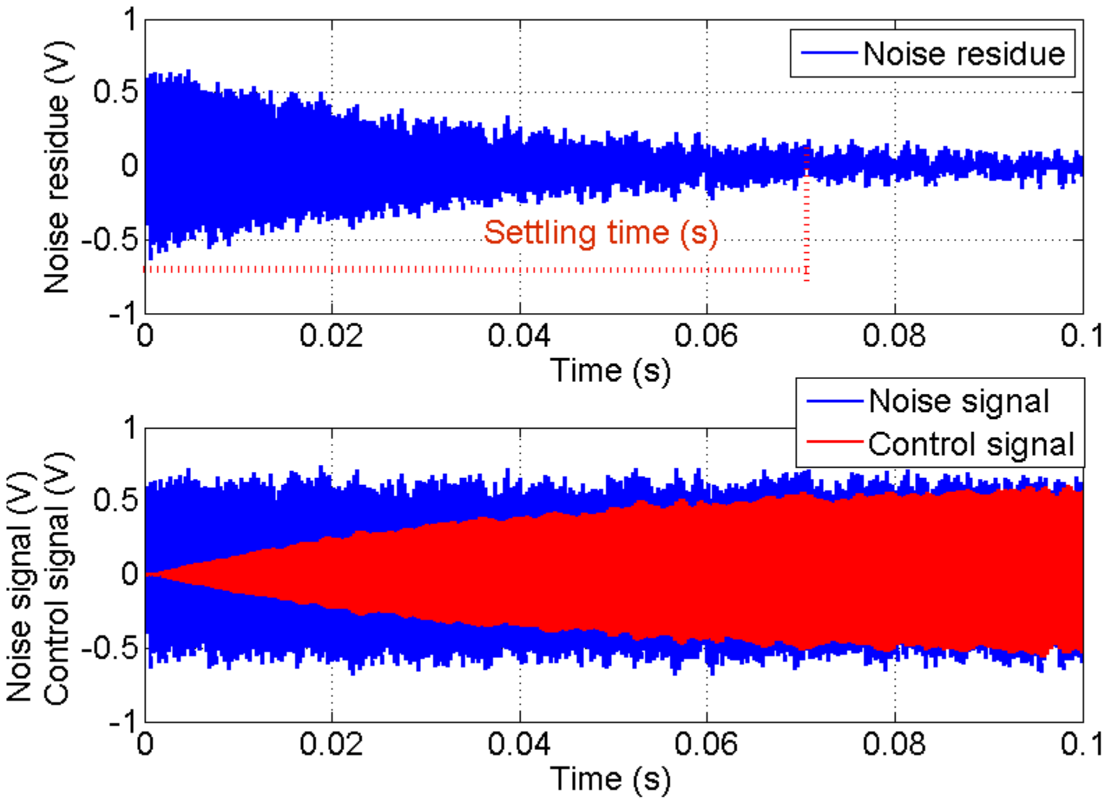

Figure 3.2.: Feedforward ANC algorithm

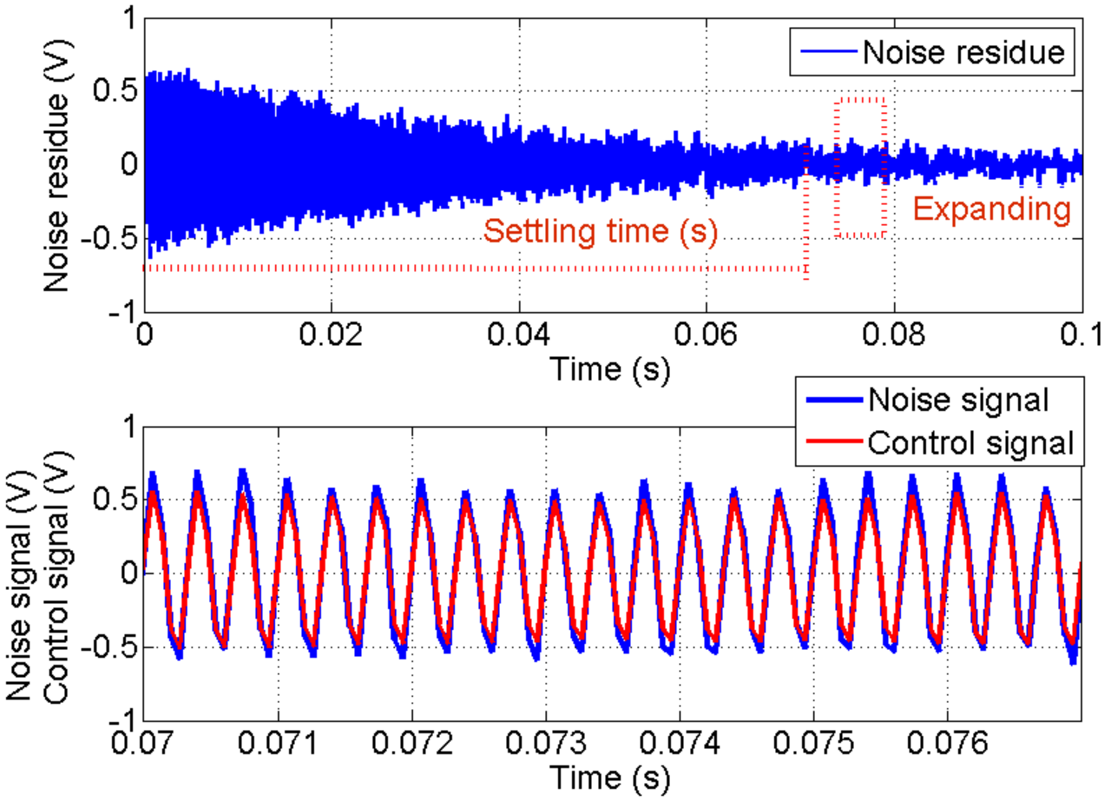

Figure 3.3.: Feedforward ANC algorithm

Execution results, of ANC feedback algorithm, is shown in figure Figure 3.4, the settling time was 0.5 s, which is analyzed from the noise residue curve, the output signal decrease its SPL in 10.39 dB.

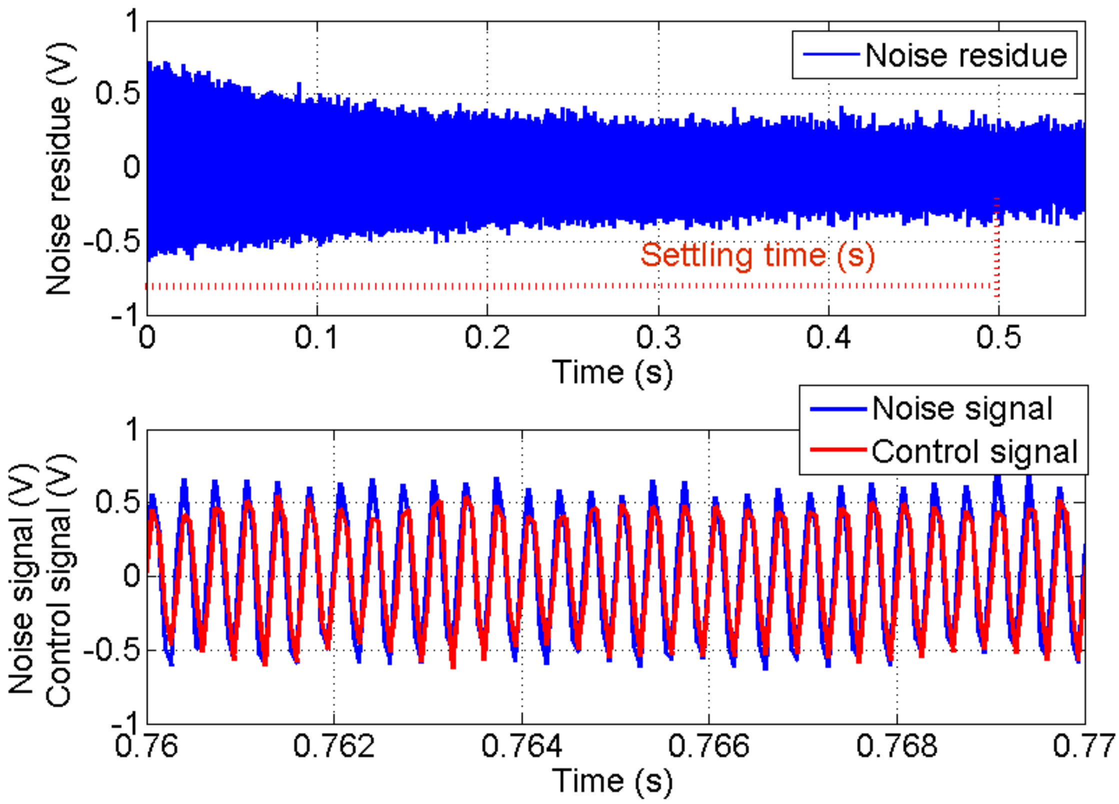

Figure 3.4.: Feedback ANC using algorithm

Figure 3.5 shows execution results of ANC using feedback algorithm, the settling time was 0.4 s, which is analyzed from the noise residue curve, the output signal decrease its SPL in 16.02 dB.

Simulation results related to the noise signal attenuation for feedforward, feedback and hybrid algorithms are summarized in table 3.1. It is shown that feedforward algorithm has shorter settling time in comparison to other algorithms as well as bigger SPL attenuation.. Nevertheless, hybrid algorithm has not short settling time, but hybrid algorithm longer settling time but is more stable to signal disturbances.

Algorithm	Decrement SPL (dB)	Settling time (s)
Feedforward	23.22	0.07
Hybrid	16.02	0.4
Feedback	10.39	0.5

Table 3.1.: Algorithms comparison

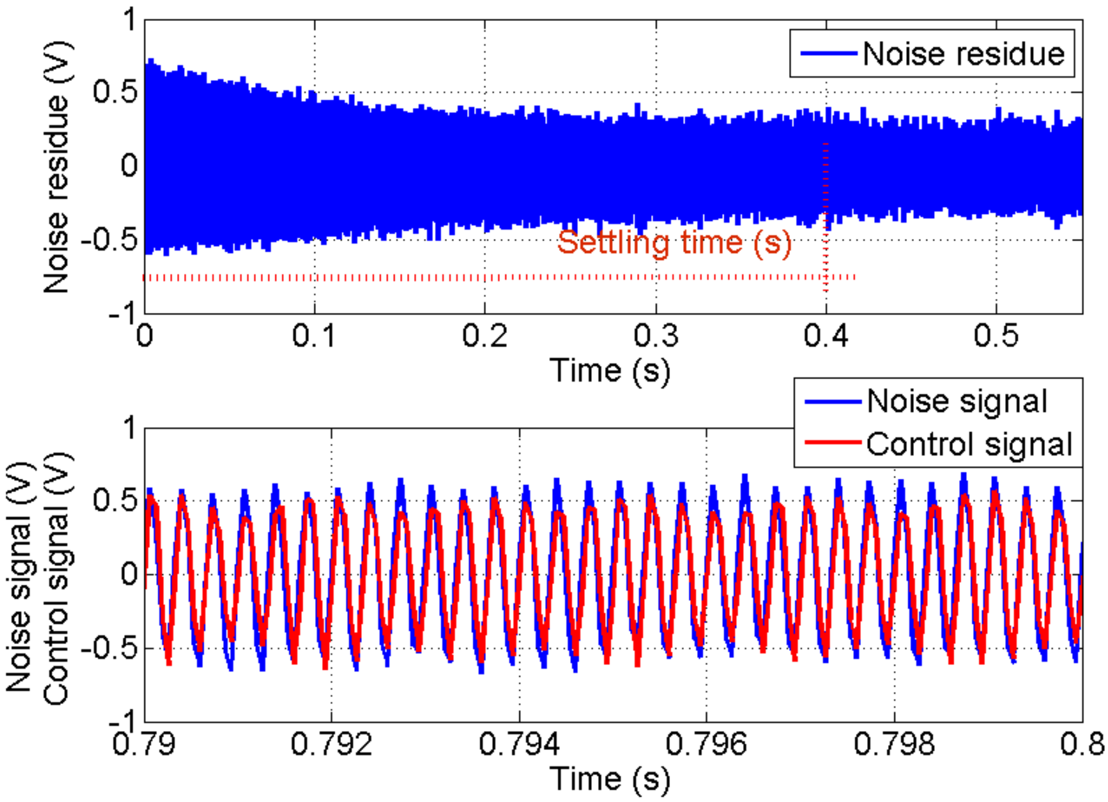

Figure 3.5.: ANC using hybrid algorithm

CHAPTER 4

Experiments for ANC

4.1. Experimental Results

Experimental tests were performed using adequate instrumentation and implementation of an algorithm for feedforward control designed [Che15]. However, before the experimental tests, curves response of the system was obtained by sinus input signals. The input signals allowed recognize the ranges of work where the control algorithm will be operating properly, for certain amplitude and frequency values.

4.1.1. System characterization

The purpose to make initial tests on the system, by sinusoidal input signals for different values of frequency and amplitude, is to recognize the permissible operating ranges for the system. The aim of test is to find the ranges of frequency and amplitude of the primary signal for the system being recognized by the microphones or to find the ranges of frequency and amplitude of the signal emitted by the antinoise loudspeaker.

The responses of the system (in SPL and frequency) sinusoidal input signals are shown in figures 4.1, 4.2 , and 4.3, the amplitudes of these signals are $0.02V$, $0.03V$ and $0.04V$, which are emitted from the anti noise speaker. The amplitudes increase in Voltages for every SPL value in the sound wave, that is generated from the initial electrical signals.

Figures 4.1, 4.2 , and 4.3 show three different moments, the increase in SPL for each signal is due to the power supply increase voltage with amplitudes from $2V$, $3V$ to $4V$.

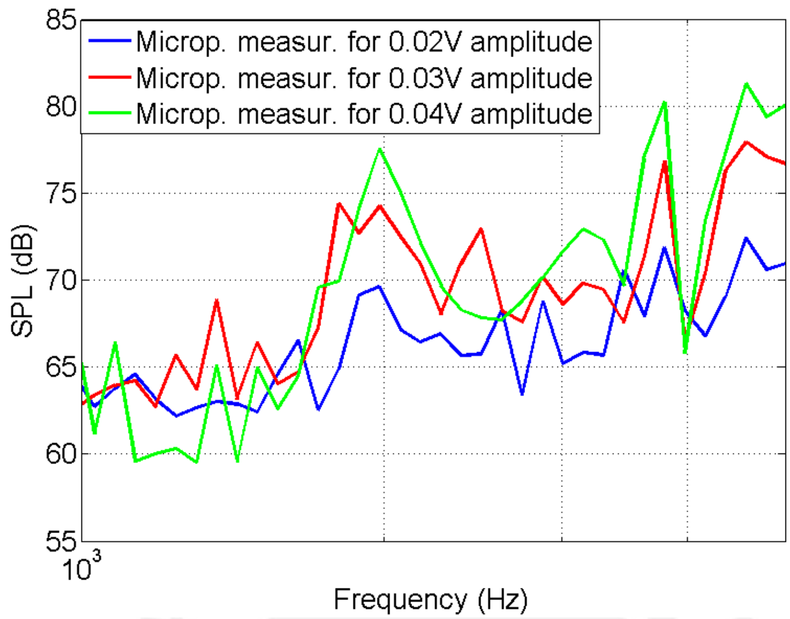

Figure 4.1.: Characteristic curves for 2V of Power Supply.

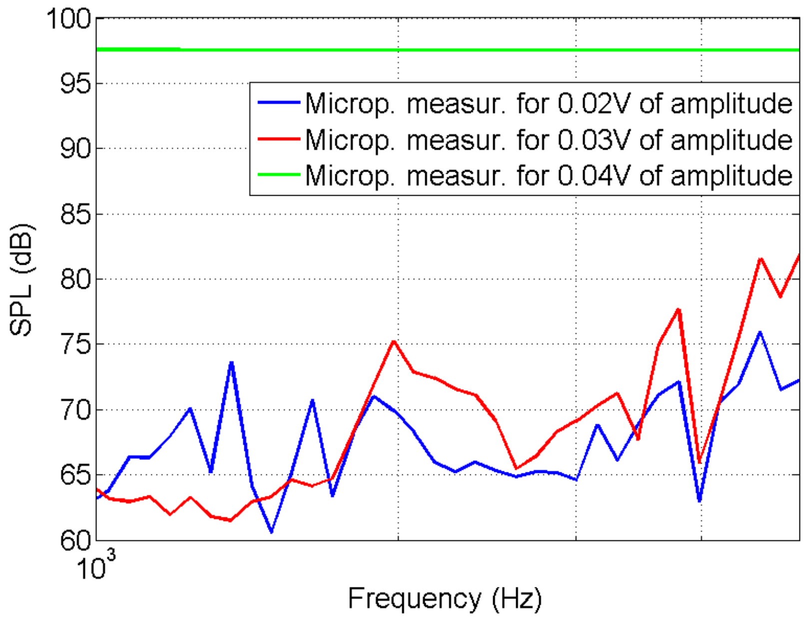

Figure 4.2.: Characteristic curves for 3V of Power Supply.

Figure 4.2 shows the signal amplitude $0.03V$ has a different behavior from the other two, which may not provide useful system information to generate an apropiate control signal. Therefore, the signal amplitude $0.02V$ maintains a similar behavior to changes in supply voltage, it will increase in SPL when increasing the supply voltages. Therefore the value of $0.02V$ amplitude excitation signal at a frequency of 3 kHz is chosen, and voltage providing from power supply should be $2V$.

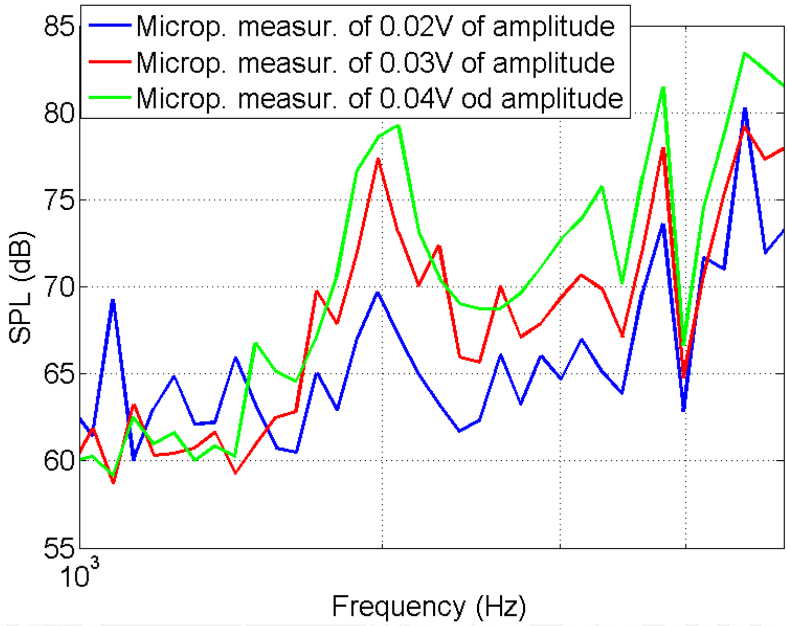

Figure 4.3.: Characteristic curves for 4V of Power Supply.

The similar behavior are shown in Figures 4.4 and 4.5, both coming from primary noise signal speaker.

4.1.2. Identification and ANC tets

Topology for ANC in the ventilation system is shown in figure 4.6. The control algorithm is executed by the ADWIN system. Also the personal computer is being involved in the ANC process for analyzing and visualization of related signals. The primary signal is generated and emitted from the ADWIN system, this signal enters the ventilation duct and is detected by the reference microphone.

The information received from the reference microphone is processed by the ADWIN system; ADWIN executes the control algorithm in order to generate an estimated signal, that is as similar as possible to the measured signal. The estimated signal is emitted by the loud speaker, which seek to mitigate the primary signal, depending on a correct location of its phases.

The difference of both signals (error) is measured by the error microphone, this signal is sent to ADWIN system to optimize the generation of the estimated signal.

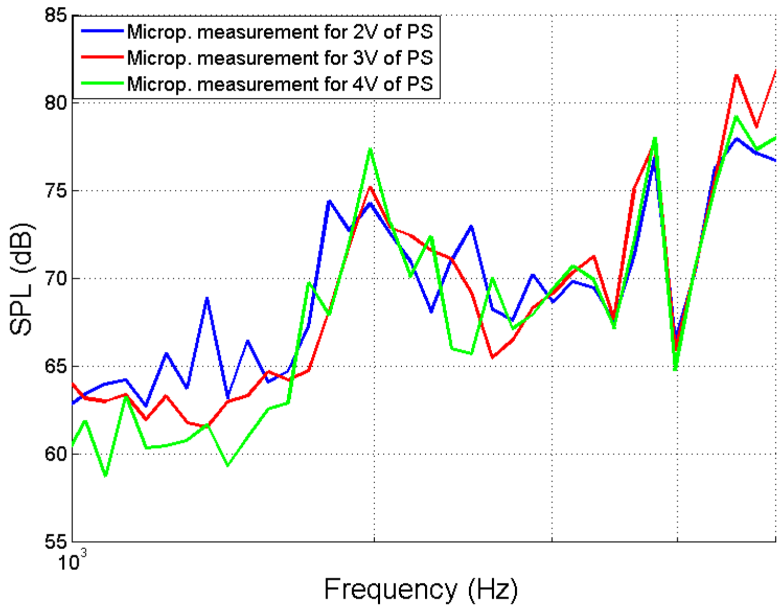

Figure 4.4.: Characteristic curves with 0.02V amplitude and 2V, 3V and 4V of Power Supply

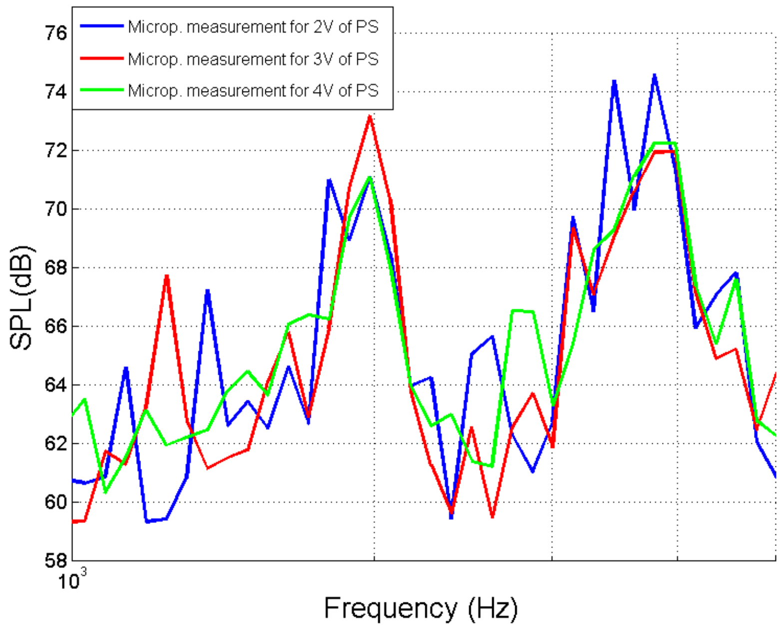

Figure 4.5.: Characteristic curves with 0.02V amplitude and 2V, 3V and 4V of Power Supply.

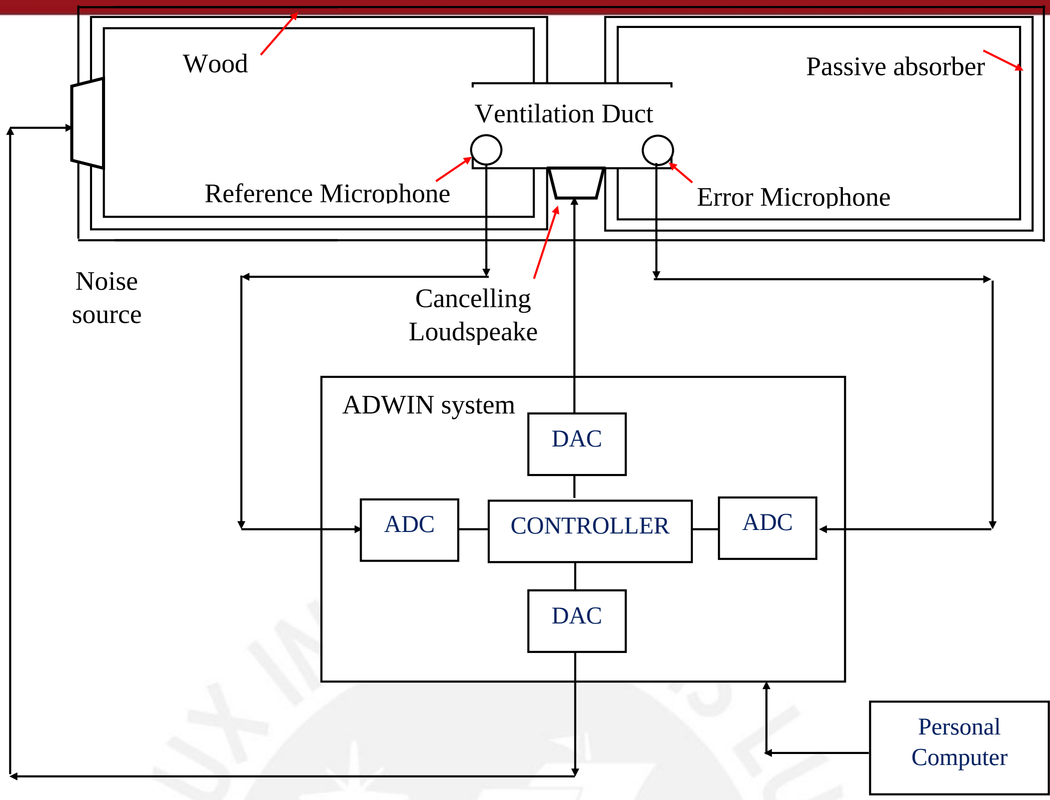

Figure 4.6.: Topology of ANC for the experimental tests.

Filter FXLMS was used to identify the system and to find the adaptive coefficients, that were used to execute the control algorithm.

Figure 4.7 shows the identification error, which was achieved by the difference between primary signal with estimated signal. The tests were disturbed by external mechanical movement, for this reason appears oscillations in the signal filtered by FXLMS. Nevertheless, the system was identified after to be executed the FXLMS filter and identified its adaptive coefficients. The best range of values chosen, in order to get ANC of the system described in figure 4.7, are: Sinusoidal input signal with 0.02V of amplitude at 3kHz, 2V proportioned by power supply to energize the system.

The identification algorithm to be executed by ADWIN was designed by the author [Che15], this code was executed with different frequencies values, in order to analyze the performance of the identification algorithm, but the best response time and noise reduction (in dB) was obtained by the range of values analyzed from the characteristic curves and table 4.1.

It's suggested no to work with 4 V power supply because of the overloading of the measured microphone signal that results in inproper identification and control.

After the system identification a control algorithm was designed. The feedforward algorithm implemented for using with ADWIN was adopted from [Che15].

It is possible to design many control algorithms for different values of amplitude of input signal, other frequencies. Nevertheless, the best performance is achieved with

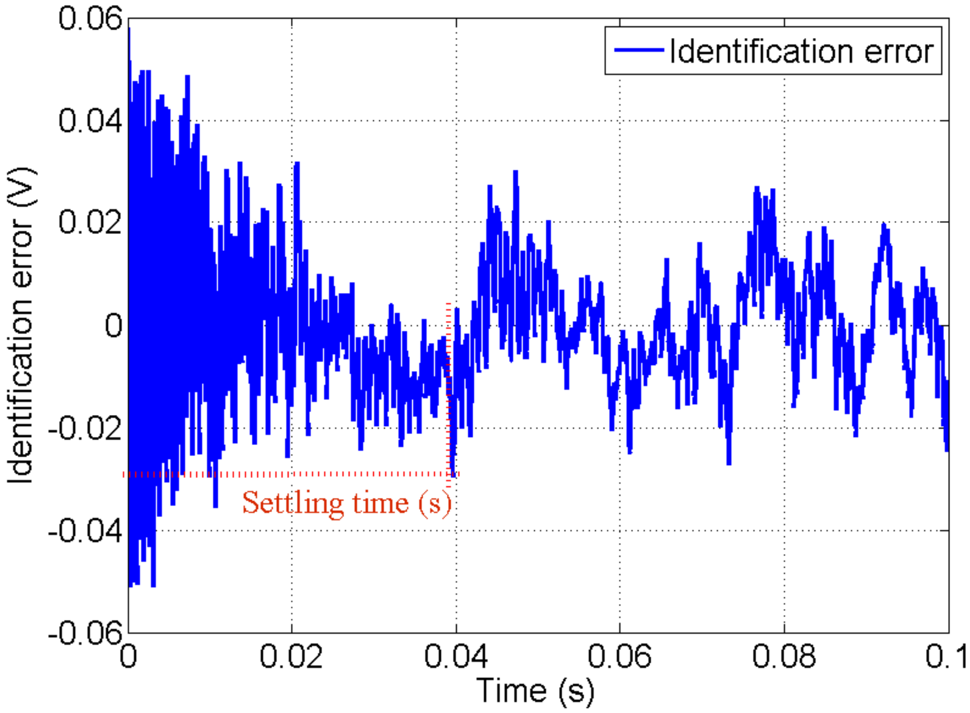

Figure 4.7.: Identification error.

Frequency (kHz)	decrement SPL (dB)	Settling time (s)
2	4.07	0.43
2.5	5.83	0.53
3	5.84	0.040
4	4.95	0.037
5	9.58	0.018

Table 4.1.: Identification results

range values analyzed in the characteristic curves.

Table 4.2 shows this range at 3 kHz (in this frequency value was identified the characteristic curves 4.1 to 4.5), settling time in the system was 0.45 s, when ANC by feedforward algorithm was executed, also the sound signal decrement 8.15 dB as is shown in figures 4.8 and 4.9.

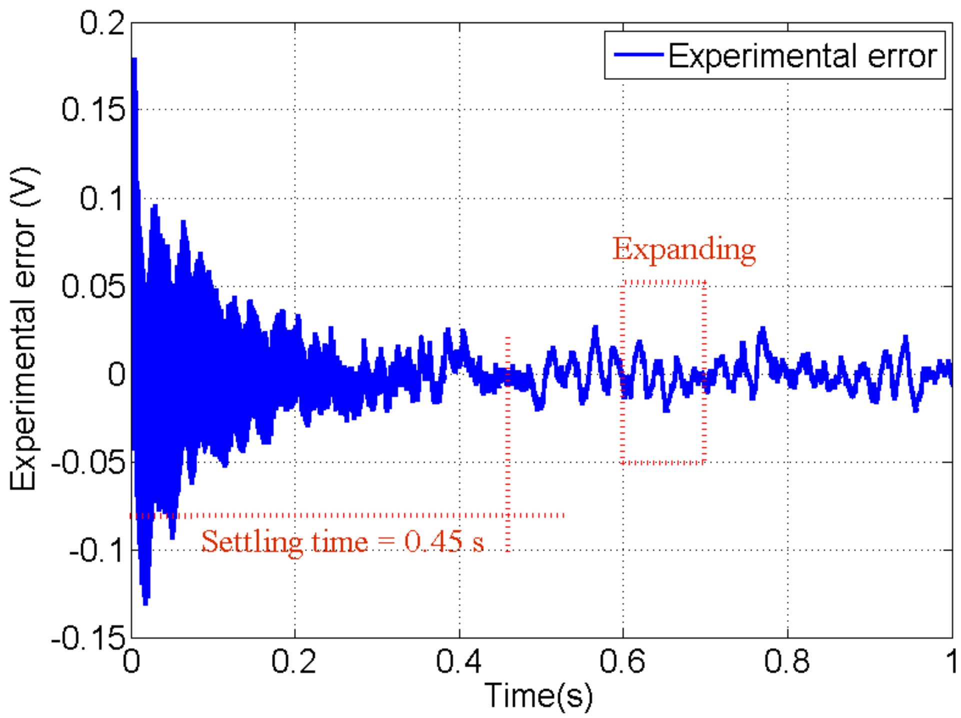

Figure 4.8.: Control error.

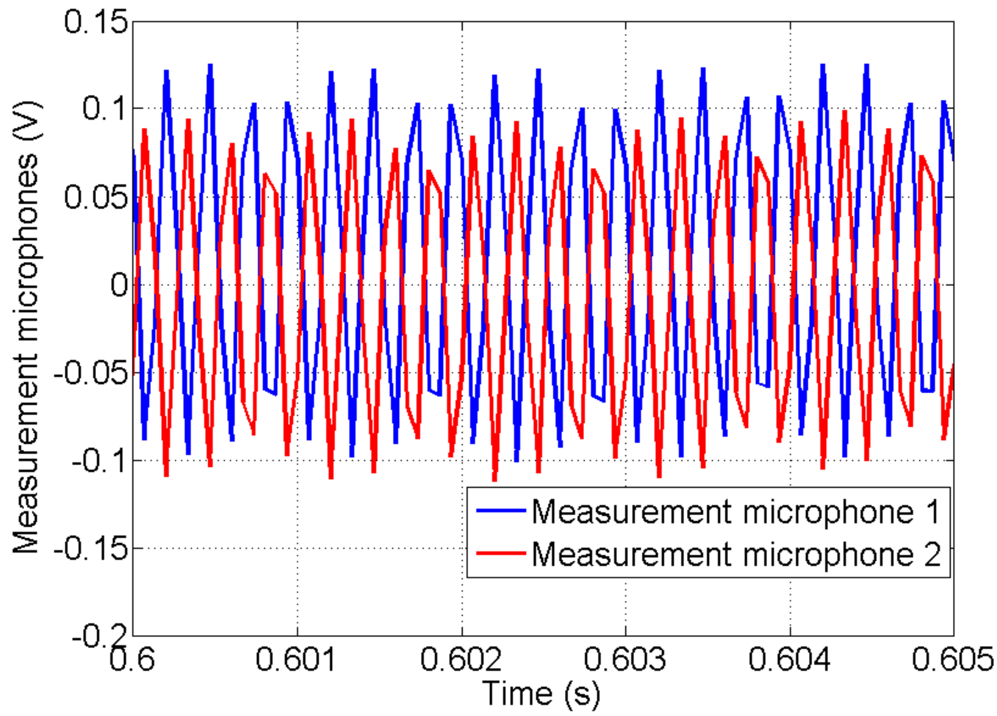

Figure 4.9.: Primary signal and output signal.

Frequency (kHz)	Decrement SPL (dB)	Settilng time (s)
2	1.6	11.3
2.5	7.66	3.67
3	8.15	0.45
4	10.47	0.33
5	2.10	9.33

Table 4.2.: Control algorithm comparison at 2kHz, 2.5kHz, 3kHz, 4kHz,5kHz of primary signal frequency

CHAPTER 5

Summary and Outlook

The ventilation systems produce sound by mechanical consequences of operation [Lar11]. For this reason it is necessary to design the devices that may attenuate and delete undesired sound waves. Furthermore, they can provide satisfactory performance for users. In this thesis it is an overview of some existing as well as design and implementation of new algorithm for ANC applicable for ventilation systems is given.

Simulated and experimental tests of an adaptive controller designed for a prototyped ventilation system are shown. The characteristics of the adaptive controller achieves attenuation of sound signals produced by the ventilation system and external disturbances which affect the system control. The system designed has a quickly response in finding the appropriate weights for the filter representing a system behavior.

A general review of many systems in the actual laboratories and industry, that have a good performance in reducing of the sound disturbances produced by mechanical effects in ventilation system is shown.

It is shown the mathematical equations to describe the process of attenuation and deletion of undesired noise [Lar11]. Furthermore, an analytical description for testing of the designed algorithm as well as a strategy for comparison between simulation and experimental results are proposed.

For testing of the designed algorithms a ventilation system prototype described in [Che15] was used. In order to design the algorithms for ANC some characteristic of this prototype were measured. The algorithms were simulated using Matlab and compared with experimental responses of designed prototype.

5.1. Summary

Algorithms and strategies for ANC were designed, implemented and experimentally tested for the prototype of the duct ventilation system based on the algorithms implemented by [Che15] and [Oey10].

Piezoelectric loudspeakers were used as anti noise sources. The using of the EAP actuator for the ANC was not considered in this work and is to be implemented in the future.

Characteristic curves were found in order to define the best range of work, where the system can measure properly the primary signal and estimate the antinoise signal.

Feedforward, feedback and hybrid algorithms were designed to simulate ANC [Che15] and [Oey10].

Feedforward algorithm was tested to get ANC in the system [Che15].

5.2. Own scientific contributions

Feedback and hybrid ANC algorithms were designed and simulated for the prototype of the duct ventilation system based on the algorithms implemented by [Che15] and [Oey10].

A model of online hybrid ANC algorithm was designed and simulated for the prototype of the duct ventilation system based on the algorithms implemented by [Che15] and [Oey10]. This online model propose to identify the system by FXLMS algorithm as an internal function of the main algorithm. The error signal can be defined as an output condition with dependence of changes in the step size μ, which is explained by C.

5.3. Outlook

Appendix C shows graphs obtained by simulating of the Hybrid algorithm, that executes the FXLMS filter as an internal function to main control algorithm. The objective of this algorithm is to find adaptive parameters in order to identify the system online. This way could offer a good response in presence of disturbances.

This is the reason that it is suggested for further researches: Online FXLMS Identification of the system as a dependence of desired error in the hybrid ANC algorithm.

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APPENDIX A

Flowcharts of the algorithms

A.1. Flowcharts

LMS for every of the controllers are shown in the following flowcharts. Figure A.1 shows the flowchart to evaluate the LMS:

Figure A.2 shows the main for all the controllers

Figure A.3 shows the feedforward controller

Figure A.4 shows the feedback controller

Figure A.5 shows hybrid controller

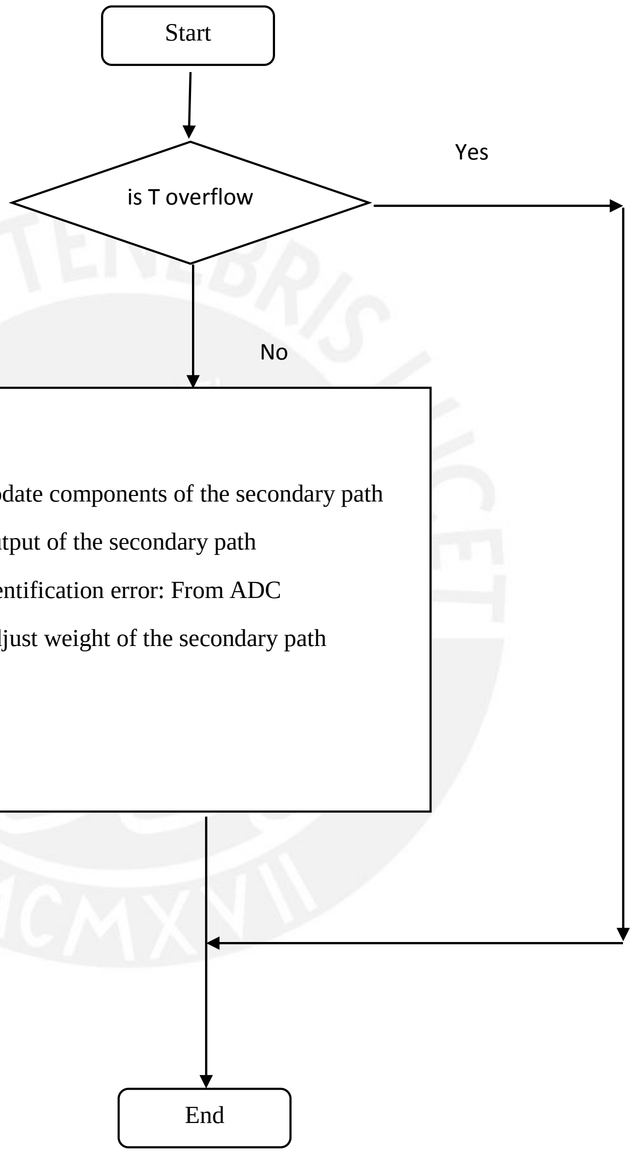

Figure A.1.: flowchart to evaluate the LMS

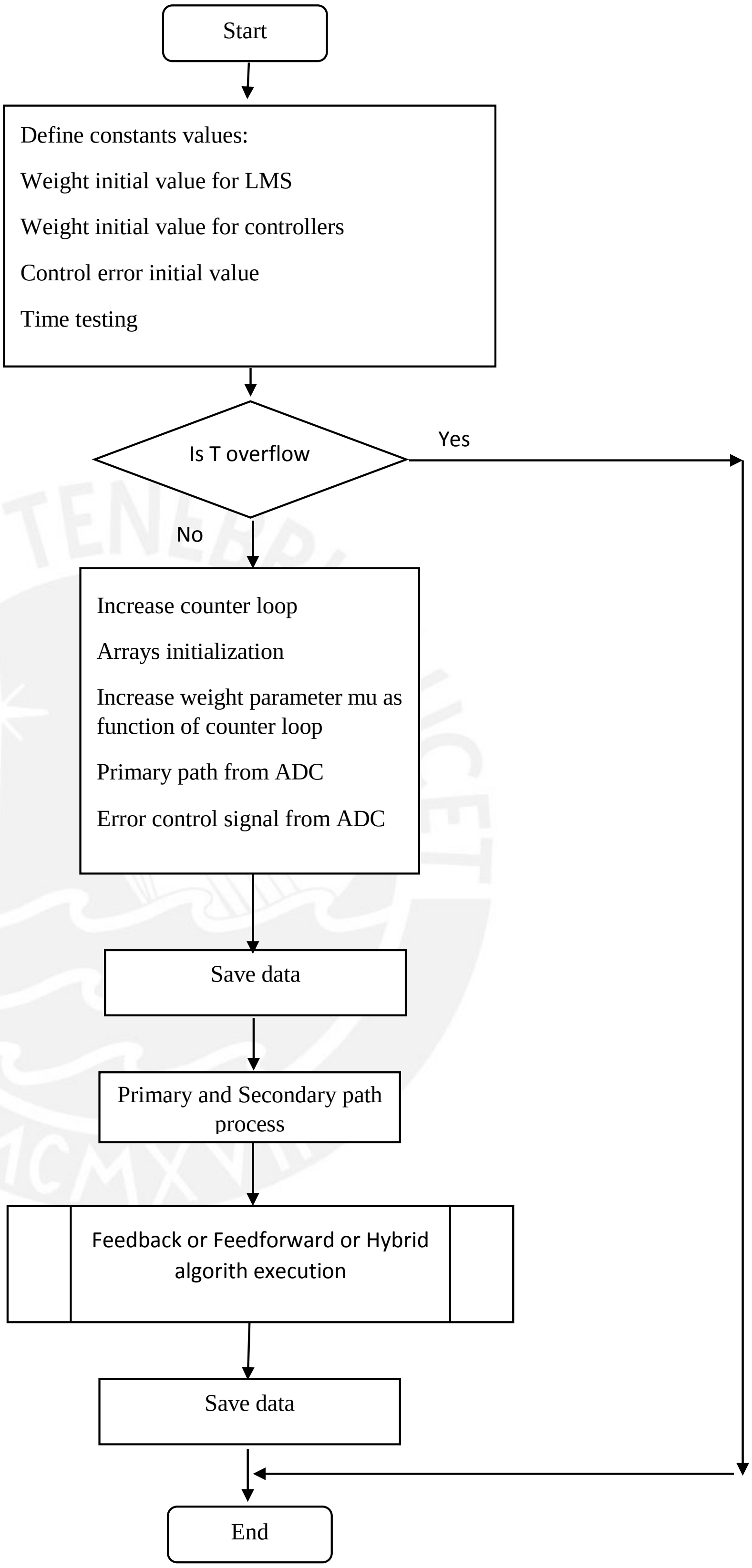

Figure A.2.: the main for all the controllers

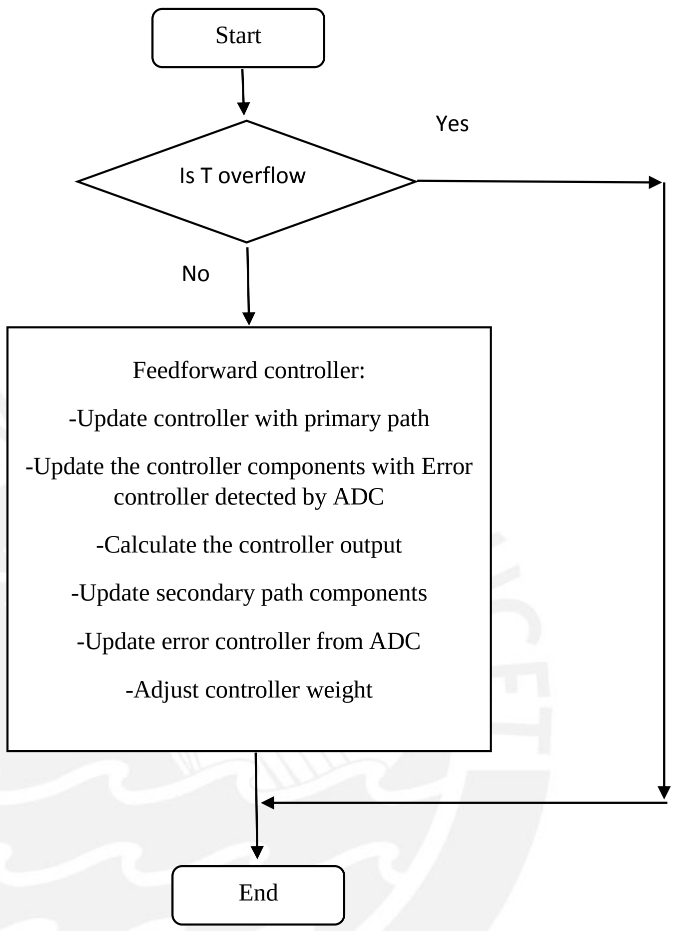

Figure A.3.: feedforward controller

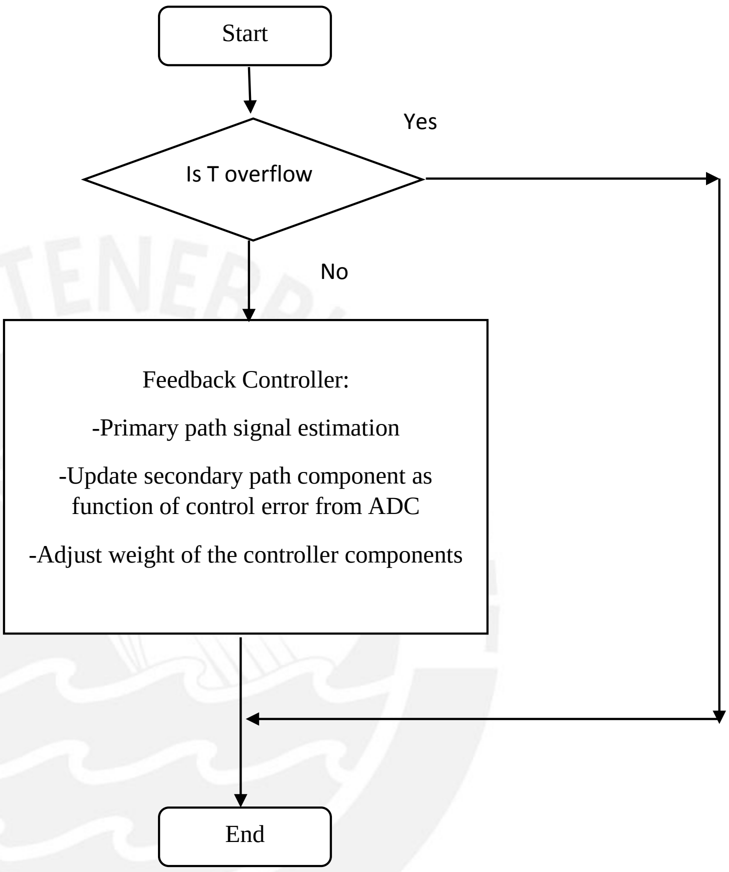

Figure A.4.: feedback controller

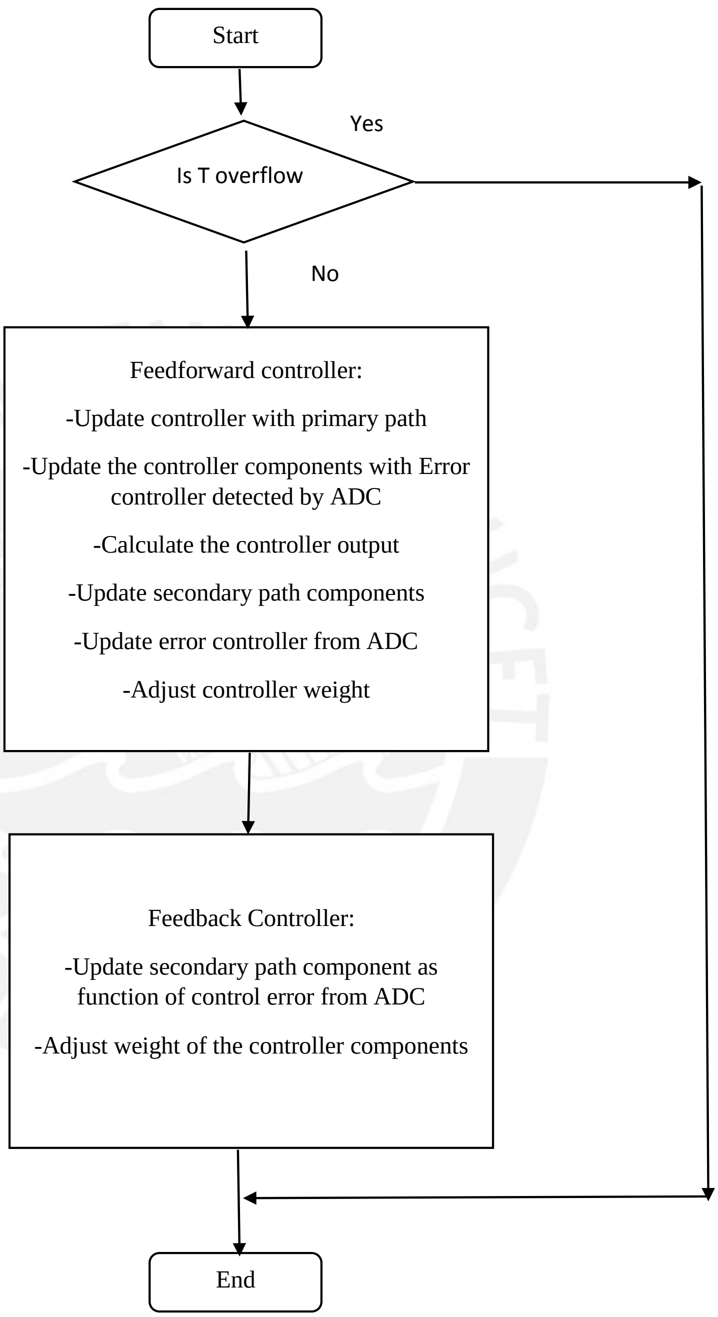

Figure A.5.: hybrid controller

APPENDIX B

Equations and pseudocodes

B.1. Equations and pseudocodes

B.1.1. Some additional equations

Methods to design the algorithms:

As was proposed by [Zan94], the primary signal should be attenuated by a secondary signal which should be slightly the same primary signal but inverse, in order to generate the same pressure of sound.

It is possible to design algorithms with the objective to achieve a better error signal as a dependency of the weights of the system. The following equations are designed to allow better performance of the desired error signal.

In the following equations analyzed by author [Sid13], $q(x,t)$ represents the sound source, $P(x,t)$ and $P'(x,t)$ represent the pressure field, X is the space variable, Γ represents spacial domain, c represents the speed of sound, and t is the time variable.

This equation describes the process to get a pressure field $P(x,t)$ from the sound source $q(x,t)$:

$$\frac{1}{c^2}\frac{\partial^2 P(x,t)}{\partial t^2} - \nabla^2 P(x,t) = q(x,t) \tag{B.1}$$

This equation describes the process to get a pressure field $P'(x,t)$ from the sound source $q(x,t)$:

$$\frac{1}{c^2}\frac{\partial^2 P'(x,t)}{\partial t^2} - \nabla^2 P'(x,t) = q(x,t) + \frac{1}{c^2}\frac{\partial^2 q(x,t)}{\partial t^2} - \nabla^2 q(x,t) \tag{B.2}$$

This equation describes the process to get a pressure field $P'(x,t)$ - $q(x,t)$ from the sound source $q(x,t)$:

$$\frac{1}{c^2}\frac{\partial^2(P'(x,t)-q(x,t))}{\partial t^2} - \nabla^2(P'(x,t)-q(x,t)) = q(x,t) \tag{B.3}$$

Outside the space domain, if it is assumed the sound source $q(x,t)$ is null, then the pressure or acoustic field $P(x,t)$ should be the same as $P'(x,t)$, which is the target of ANC algorithm, because from equation 2.1 to equation 2.54 an estimated signal is generated, like the input signal. On the other hand, in order to implement this algorithm, the author [Sid13] proposes using an inverter to achieve a difference between the input signal and the estimated signal.

B.1.2. Pseudocodes to design algorithms

In order to design FXLMS ANC algorithm for simulations and implementation, it was necessary to find a relation between the control error $e_conty(k)$ with the adaptive coefficients C_y, C_x, C_w and parameter weights μ. The following pseudo code describes the characteristic of the feedforward algorithm designed in this work, which was analyzed by author [Oey10].

LMS pseudocode for feedback, feedforward and hybrid controllers:

$$error = \frac{Shw(k+1) - Shw(k)}{\mu} - Shx(k) \tag{B.4}$$

feedforward control pseudocode:

$$C_x(k+1) = [X(k), C_x(k)] \tag{B.5}$$

$$C_y(k+1) = \sum_{i=1}^{T} C_{x_i}(k)C_w(k) \tag{B.6}$$

$$S_x(k+1) = [C_y(k), S_x(k)] \tag{B.7}$$

$$e_conty(k) = Y_d(k) - \sum_{i=1}^{T} S_{x_i}(k)S_w(k) \tag{B.8}$$

$$Sh_x(k+1) = [X(k), Sh_x(k)] \tag{B.9}$$

$$Xh_x(k+1) = [\sum_{i=1}^{T} Sh_{x_i}(k)Sh_w(k), Xh_x(k)] \tag{B.10}$$

$$Cw(k+1) = Cw(k) + \mu e_{cot}(k) X h_x(k) \tag{B.11}$$

feedback control pseudocode:

$$e_cont(k) = Ydk - \sum_{i=1}^{T} Sx_{x_i}(k) Sw \tag{B.12}$$

$$\beta = \sum_{i=1}^{T} Sh0_{x_i}(k) Sh_w 2(k) + error_{control} \tag{B.13}$$

$$Cx_0(k+1) = [\beta, C_x 0(k)] \tag{B.14}$$

$$Cy_0 = \sum_{i=1}^{T} Cx0_{x_i}(k) Cw_0 \tag{B.15}$$

$$Sx(k+1) = [Cy0, Sx(k)] \tag{B.16}$$

$$Shx0(k+1) = [\beta, Shx0(k)] \tag{B.17}$$

$$Xhx(k+1) = [\sum_{i=1}^{T} Shx0_{x_i}(k) Shw2, Xhx0(k)] \tag{B.18}$$

$$Cw0(k+1) = Cw0(k) + \mu e_{cot}(k) X h_x(k) \tag{B.19}$$

Hybrid control pseudocode: Feedforward component of hybrid pseudocode:

$$C_x(k+1) = [X(k), C_x(k)] \tag{B.20}$$

$$\beta = \sum_{i=1}^{T} Sh_{x_i}(k) Sh_w(k) + error_{control} \tag{B.21}$$

$$Cx_0(k+1) = [\beta, C_x(k)] \tag{B.22}$$

$$Cy_0 = \sum_{i=1}^{T} Cx0_{x_i}(k) Cw_0 \tag{B.23}$$

$$Cy = \sum_{i=1}^{T} Cx_{x_i}(k)Cw \tag{B.24}$$

$$Sx(k+1) = [Cy0 + Cy, Sx(k)] \tag{B.25}$$

$$e_cont(k) = Ydk - \sum_{i=1}^{T} Sx_{x_i}(k)Sw \tag{B.26}$$

$$Sx(k+1) = [X(k), Shx(k)] \tag{B.27}$$

$$Xhx(k+1) = [\sum_{i=1}^{T} Shx_{x_i}(k)Shw, Xhx(k)] \tag{B.28}$$

$$Cw(k+1) = Cw(k) + \mu e_{cot}(k)Xh_x(k) \tag{B.29}$$

Feedback component of hybrid pseudocode:

$$Shx(k+1) = [\beta, Shx0(k)] \tag{B.30}$$

$$Xhx(k+1) = [\sum_{i=1}^{T} Shx0_{x_i}(k)Shw2, Xhx0(k)] \tag{B.31}$$

$$Cw0(k+1) = Cw0(k) + \mu e_{cot}(k)Xh_x0(k) \tag{B.32}$$

APPENDIX C

Online ANC algorithms

C.1. Online ANC algorithms

The control algorithms, designed and executed in this thesis, require the calculation of the adaptive system weights in the identification stage. For this reason, it is necessary a previous Identification by FXLMS filter; Notwithstanding, What happen if some properties (like temperature or another disturbance) in the air will change?.

The sound wave is propagated in the air, also this changes can produce modifications in the adaptive weights, thereby the ANC needs another adaptive coefficients (Repeating of the system identification).

The system identification, as is explained for this thesis, is an external process taking place before the control. However, the flowchart C.1 proposes a sequence to identify the system by FXLMS as a subfunction inside the ANC algorithm.

The algorithm identify the system and achieve the adaptive coefficients continuously, before to control loop. The criterium for optimize this process depends of the time assigned to execute the LMS as a function inside the main of the algorithm.

The objective of this algorithm to calculate the adaptive weights online, periodically repeating the system identification process inside of the main algorithm. The algorithm will be running till the condition, predefined by user, is not satisfied. This condition can be determined by desired error as a function of the changes in step size.

Figure C.1 shows the flowchart of the main loop of the hybrid algorithm. Figure E2 was obtained by simulating the Hybrid algorithm which executes the FXLMS filter internally as an internal function to principal algorithm, in order to obtain adaptive control system parameters and identify the system shown soon as possible before taking control action. This way could offer a good response in presence of disturbances.

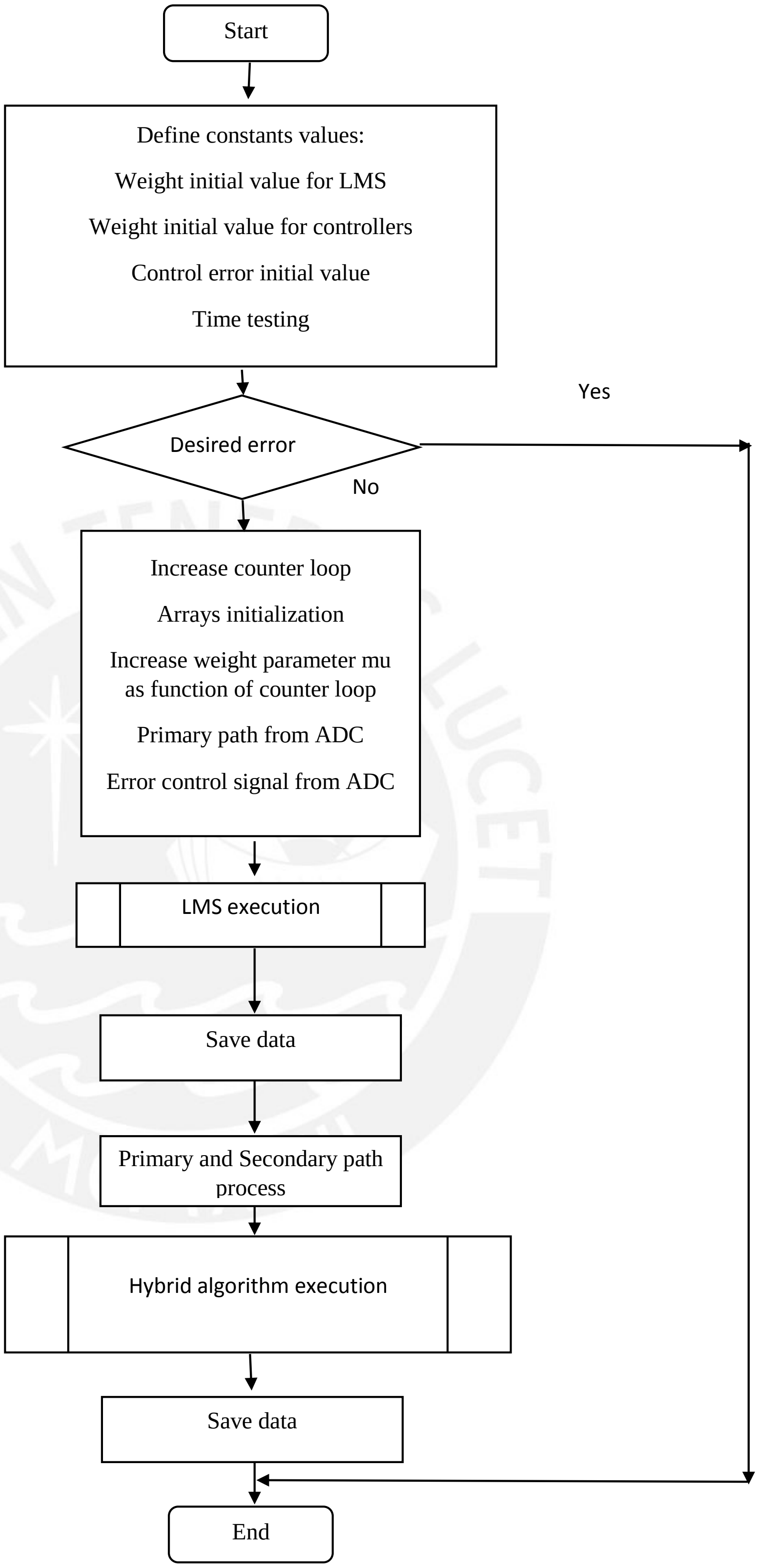

Figure C.1.: ANC applications

This is the reason that it is suggested for further researches: Online FXLMS Identification of the system as a dependence of desired error in the hybrid ANC algorithm.

Figure C.2

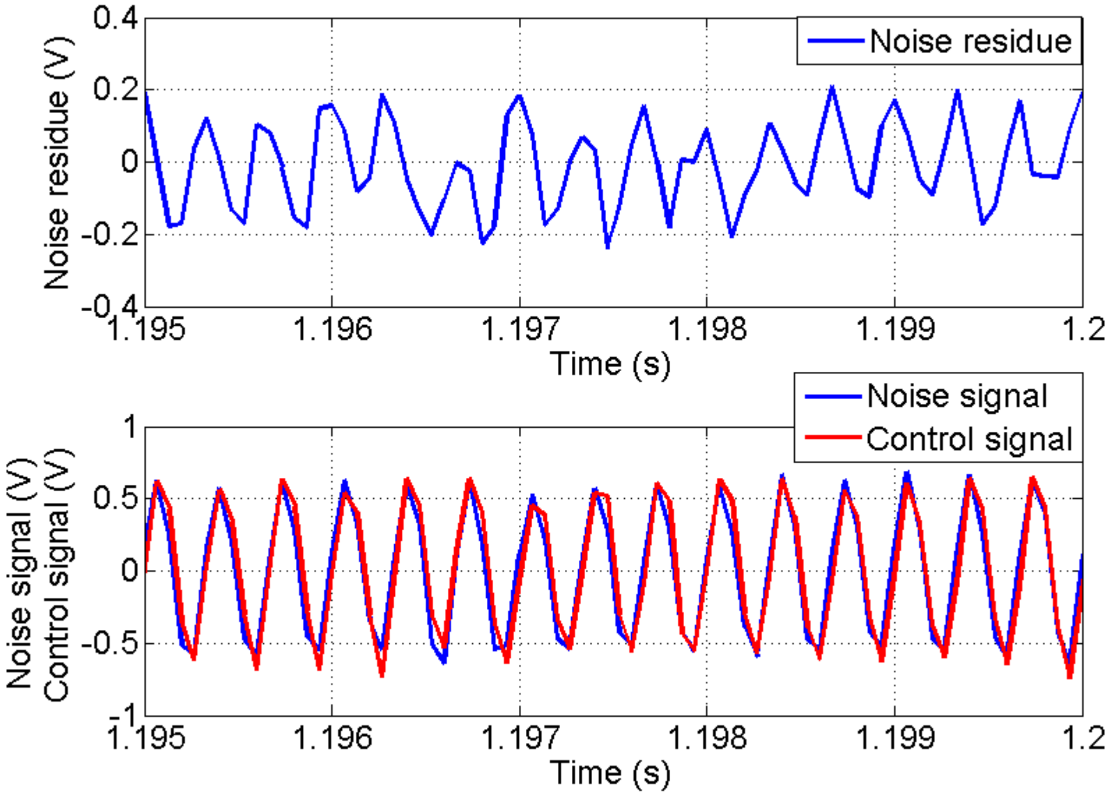

Figure C.2.: ANC applications