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Performance Evaluation of Different Active Noise Control (ANC) Algorithms for Attenuating Noise in a Duct

Muhammad Moazzam

Muhammad Shoaib Rabbani

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Supervisor:
Prof. Dr. Lars Håkansson / Imran Khan

Blekinge Institute of Technology
Department of Applied Signal Processing
School of Engineering
Karlskrona, Sweden.

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Contact Information:

Author:
Muhammad Moazzam
E-mail: moazzam017@gmail.com

Muhammad Shoaib Rabbani
E-mail: shoaibrabbani11@gmail.com

Supervisor/Examiner:
Dr. Lars Håkansson
School of Engineering (ING)
E-mail: lars.hakansson@bth.se
Phone: +46 455 385 712

Supervisor:
Imran Khan
School of Engineering (ING)
E-mail: Imran.khan@bth.se
Phone: +46 455 385 747

School of Engineering
Blekinge Institute of Technology
371 79 Karlskrona
Sweden

Internet : www.bth.se/ing
Phone : +46 455 38 50 00
Fax : +46 455 38 50 57

To My Family

Muhammad Moazzam

To My Parents

Muhammad Shoaib Rabbani

ABSTRACT

Adaptive filter algorithms are extensively use in active control applications and the availability of low cost powerful digital signal processor (DSP) platforms has opened the way for new applications and further research opportunities in e.g. the active control area. The field of active control demands a solid exposure to practical systems and DSP platforms for a comprehensive understanding of the theory involved. Traditional laboratory experiments prove to be insufficient to fulfill these demands and need to be complemented with more flexible and economic remotely controlled laboratories.

The purpose of this thesis project is to implement a number of different adaptive control algorithms in the recently developed remotely controlled Virtual Instrument Systems in Reality (VISIR) ANC/DSP remote laboratory at Blekinge Institute of Technology and to evaluate the performance of these algorithms in the remote laboratory. In this thesis, performance of different filtered-x versions adaptive algorithms (NLMS, LLMS, RLS and FuRLMS) has been evaluated in a remote Laboratory. The adaptive algorithms were implemented remotely on a Texas Instrument DSP TMS320C6713 in an ANC system to attenuate low frequency noise which ranges from 0-200 Hz in a circular ventilation duct using single channel feed forward control.

Results show that the remote lab can handle complex and advanced control algorithms. These algorithms were tested and it was found that remote lab works effectively and the achieved attenuation level for the algorithms used on the duct system is comparable to similar applications.

Keywords: Active Noise Control, Adaptive Algorithms, L-LMS, N-LMS, FuLMS, RLS

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Last but not the least; Words are lacking to express our feelings and thoughts to our dear friends for their love, devotion, care and concern. Any tribute will be less for them.

Muhammad Moazzam
Muhammad Shoaib Rabbani

Contents

List of Figures

List of Tables

List of Abbreviations

ADC	Analog to Digital Converter
ANC	Active Noise Control
BTH	Blekinge Tekniska Hogskola, Sweden
CCS	Code Composer Studio
DAC	Digital to Analog Converter
DSP	Digital Signal Processor
Fu LMS	Filtered-u Least Mean Square
GPIB	General Purpose Interface Bus
HP	Hewlett-Packard
IC	Internal Combustion
IEPE	Integrated Electronics Piezo Electric
LMS	Least Mean Square
LLMS	Leaky Least Mean Square
NI	National Instrument
NLMS	Normalized Least Mean Square
PSD	Power Spectral Density
RDE	Remote development Environment
RLS	Recursive Least Square
SAR	Successive Approximation Register
TI	Texas Instrument

Chapter 1 : Introduction

1.1 Overview

Our world is successively becoming more and more technologically advanced. Although, the technological advancements have brought in a significant comfort to our life's they may on other hand also introduced new challenges that influence our environment negatively. For instance, the acoustic noise in our environment is nowadays common because of products such as compressors, vacuum cleaners, dish washers, ventilation systems, transformers, fans, engines, blowers, motors etc. The associated mechanical vibration is also a problem in industries, all areas of transportation and in household appliances [1]. The awareness of the degrading influence of acoustic noise on humans has increased in recent years as a consequence of the increasing understanding of it's the associated health risks.

Acoustic noise in our surrounding environment can be broadly classified as broad band noise and narrowband noise[2]. Broadband noise has typically stochastic properties. Examples of machinery and other sources which generate broadband noise are heating, ventilation and air condition systems, noise of explosion etc. While narrow-band noise basically has its energy concentrated at discrete frequencies. This type of noise is mostly periodic or near to periodic. Examples of such noise includes fan noise, noise of internal combustion engines in vehicles, vacuum pumps etc. Either of the two, narrowband noise or broadband noise is annoying to humans working in industries or offices as well as to peoples living in homes. Exposure to acoustic noises may be hazardous to humans from both physical and psychological point of view [4].

Several techniques have been developed for the reduction or control the acoustic noise. Out of these, two classes are: the Passive and Active noise control methods. Passive noise control technique consists typically of absorbers and/or reflectors/barriers[3]**.** Acoustic energy is converted to thermal energy by absorbers. On the other side, in order to prevent the noise to propagate from one space to another reflectors /barriers are used which reflects the incident waves. For frequencies over approx. 300Hz passive noise control technique are generally very useful. However, for the attenuation of noise at lower frequencies passive techniques are generally impractical because of their weight and bulk [3].

The concept of active noise control is not a new idea and it can be traced back to 1936 by a Unites States patent by Paul Leug [2]. A renewed interest in ANC during the latest decades may primarily be related to the availability of low cost DSP platforms and the development of control algorithms. Compared to passive noise control, active noise control (ANC) is well suited for the attenuation of low frequency noise. Great interest has been shown in active noise control in order to overcome problems that are not practical to approach with passive noise control. The successful application of active control is determined on the basis of its effectiveness compared with passive control in the low-frequency range. Active technique for noise control may be attractive to achieve large amounts of noise reduction in a small package especially for low frequencies below 300Hz and generally offers a real advantage over passive control in the low-frequency range [2].

The ANC is a wide field consisting of knowledge of different specialized subjects such as statistics, adaptive signal processing, acoustics, signal processing and DSP implementation, etc. The importance of practice and experience in these fields is beyond any doubt and therefore, comprehensive experimentation is required to understand the theoretical concepts that have been developed in the classroom and in particular to be able to implement an ANC system that results in some noise reduction. A typical ANC experiment requires a setup which includes suitable sensors, secondary sources and analysis equipment e.g. microphones, loudspeakers and signal analyzer. A suitable DSP platform is also required to implement the active control algorithms usually implemented via in C/C ++ programming or Assembly language. Thus, a number of important practical learning tools that demands e.g. a substantial amount of practice for an engineering student to acquire an adequate knowledge level in respective discipline. At present a single setup for experimentation on remotely controlled ANC laboratory lab is available. The equipment involved in an active control laboratory is advanced and expensive. To increase the usability of the limited resources, remotely controlled laboratories for active control experiments prove to be a feasible solution.

The emergence of remotely controlled laboratories with real measurement equipment and experiment objects can be seen as major development in education sector. These remote laboratories provide a number of advantages compared to traditional on campus laboratories. Institutions having remotely controlled laboratories can share them with other institutions around the world and thus increasing the mutual collaboration and capacity in research and education.

Remote laboratories can be accessed via the Internet and students can perform their experiments with convenience using real equipment and physical systems.

1.2 Motivation

To keep up an adequate level in the experimental parts of courses such as signal processing, adaptive signal processing, sound and vibration measurements, DSP, etc. remotely controlled laboratories are a suitable complement to on campus laboratories. Blekinge Institute of Technology (BTH) has developed a prototype remotely controlled ANC laboratory, accessible via Internet. The remote laboratory was initially tested using the Filtered-x least mean square algorithm (FxLMS). However a more comprehensive evaluation of the remotely controlled ANC laboratory using more complex and advanced algorithms, e.g. FuRLMS and RLS, is of interest.

This master thesis focuses on the use of the remote laboratories in the field of Active noise control. The suitability of the laboratory resources for more advanced and computation demanding algorithms will be investigated. The expected outcome is two dimensional. First, to learn and to implement active control algorithms on a DSP for the attenuation of noise in ventilation duct. Secondly, providing guidelines for students concerning how to use the ANC remote laboratory implement adaptive algorithms and perform ANC experiments.

1.3 Research Question

This thesis presents the ANC algorithms implemented in a digital signal processor system that is a part of a recently built remotely controlled laboratory for ANC experiments on a ventilation duct. Research question behind this thesis work are as

1. What are the possibilities of different algorithms for controlling the noise?
2. How effective are different ANC algorithms for the attenuation of the noise in a duct?
3. How effective are remote laboratory work concerning implementation of control algorithms?
4. What are the computational demands of the considered algorithms?

1.4 Thesis Outline

This thesis describes the performance of different ANC algorithms for narrowband feed forward ANC of noise in a ventilation duct using the remotely controlled ANC laboratory at BTH. The application for ANC considered in this thesis is a ventilation duct and Chapter 2 describes the propagation of sound in ducts followed by chapter 3 where the basic concept of ANC with different ANC applications and systems are described. Chapter 4 focuses the different adaptive algorithms that are implemented for experiments and for performance evaluation. In chapter 5, experiment hardware and the user interface of the remotely controlled laboratory are described in detail. Chapter 6 presents duct noise attenuation results for different implemented ANC algorithms, while chapter 7 concludes the thesis work.

Chapter 2 : Duct Acoustics

In Heating, Ventilation and Air conditioning (HVAC) systems and exhaust appliances, ducts are used as a transmission path of hot/cold air. Random noises are generated along the duct inner surface, at the inner boundaries of the duct, as well as in bends where the airflow causes boundary layer turbulence. ANC may be utilized to attenuate the low-frequency range of duct noise in HVAC systems. A substantial amount of research has been carried out concerning acoustics in ducts [4]. Extensive work has been carried out on ANC systems for the control of noise in ventilation ducts [4] . In current remote laboratory, the test bed is a HVAC duct. Noise attenuation in such duct via active methods is a classical well known application. To attenuate noise effectively it is important to understand sound wave propagation in ducts. This chapter will describe basic duct acoustics briefly.

2.1 Classification of Sound waves in a Duct

Sound waves inside the ducts can be classified as plane waves and higher order modes. Plane waves remain constant over the cross section of the duct and vary only along the length of the duct below a certain frequency. While for higher modes, the sound pressure also varies along the duct cross section and have a resonant behavior in the radial direction of the duct producing more complicated waveforms [5].

To determine the type of ANC system to be implemented on duct system, and positioning and number of sensors and secondary source, it is important to determine whether the sound waves propagate only as a plane wave in the frequency range where ANC is to be applied in a certain duct system [7]. To produce a sufficient attenuation in the plane wave propagation region, a single channel ANC system may be enough to produce a desired attenuation [8] [9]. Otherwise, in order to attenuate higher order modes, a multichannel ANC system based on several sensors and control sources may be required to control these higher order modes [4][10]. The cut-on frequency of different acoustic modes can be obtained from the modified wave equation and depends on the shape and size of the cross section of the duct [13].

2.2 Wave Equation

Sound waves travel in air both as a function of time and space. The linearized one dimensional wave equation can be derived using the three fundamental fluid equations [8][9]. The one dimensional wave equation given by [11],

$$\frac{\partial^2}{\partial x^2} p(x,t) - \frac{1}{c^2}\frac{\partial^2}{\partial t^2}\, p(x,t) = 0 \qquad (2.1)$$

Here c represents the speed of sound in air and p is the sound pressure. The general solution to this equation is given by two arbitrary functions $p(x,t) = f\,(t - \frac{x}{c})$ and $p(x,t) = g\,(t + \frac{x}{c})$, i.e. two pressure waves traveling in opposite direction one in positive x-direction and the other in negative x-direction. For harmonic pressure waves traveling in opposite direction the functions which satisfy Eq. 2.1 can be written as,

$$p(x,t) = \left\{Ae^{-kx}e^{j\omega t}\right\} \qquad (2.2)$$

And

$$g(x,t) = \{Be^{kx}e^{j\omega t}\} \quad (2.3)$$

The terms $A = |A|e^{j\varphi_A}$ and $B = |B|e^{j\varphi_B}$ are complex numbers and their modulus represents the amplitude of the sound pressure and φ is arbitrary phase angle, ω is the angular frequency and $k = \omega/c$ is the wave number [12]. Thus we can write Eq. 2.2 and Eq. 2.3 as,

$$Re\{Ae^{-kx}e^{j\omega t}\} = |\mathrm{A}|\cos(\omega t - kx + \varphi_A) \quad (2.4)$$

$$Re\{Be^{kx}e^{j\omega t}\} = |\mathrm{B}|\cos(\omega t + kx + \varphi_B) \quad (2.5)$$

For a three dimension Cartesian coordinate system (x, y, z) which is suitable for the analysis of rectangular ducts, the equation of wave motion can be written as,

$$\nabla p(x,y,z,t) - \frac{1}{c^2}\frac{\partial^2}{\partial t^2}\, p(x,y,z,t) = 0 \quad (2.6)$$

Where ∇ is the Laplace or divergence operator according to [8] and is given by,

$$\nabla = \frac{\partial^2}{\partial x^2} + \frac{\partial^2}{\partial y^2} + \frac{\partial^2}{\partial z^2} \quad (2.7)$$

Similarly, the sound pressure in polar coordinates system which is suitable for analyzing circular ducts is given by $p(r, \theta, z, t)$ and the Laplace operator is given by [8] ,

$$\nabla = \frac{\partial^2}{\partial r^2} + \frac{1}{r}\frac{\partial}{\partial r} + \frac{1}{r^2}\frac{\partial}{\partial \theta^2} + \frac{\partial^2}{\partial z^2} \quad (2.8)$$

The solution of the lineraized wave equation i.e., Eq. 2.6 can be used to calculate the acoustics modes for a particular duct system. Also it is important to mention that in practical HVAC systems the air inside has a mean air flow and its influence must be taken into consideration. When the mean air flow influence in the duct is taken into account, the general wave equation is modified. The modified wave equation known as the converted three dimensional wave equation or modified three dimensional wave equation is given by [8],

$$\nabla p(x,y,z,t) - \frac{1}{c^2}\left(\frac{\partial}{\partial t} + U\frac{\partial}{\partial z}\right)^2 p(x,y,z,t) = 0 \quad (2.9)$$

Comparing equations Eq. 2.6 and Eq. 2.9, it can be seen that the only difference is the mean air flow speed U in the duct for the propagation of air along the length of the duct, in this case along the $z - axis$ direction.

2.3 Cut-on Frequencies

In HVAC systems, the ducts may be circular or rectangular. For a particular duct system the cut-on frequencies for higher order modes may be calculated based on the solution of the modified wave equations Eq. 2.10 and Eq. 2.21. For the design of a ANC system for control of duct noise it is important to have knowledge concerning the plane wave region and higher order mode region of the duct for that reason a brief discussion for both cases is presented in the following section.

2.3.1 Circular Ducts

A circular duct assumed to be infinitely long in the z-direction and having stiff walls is shown in Figure 2.1:

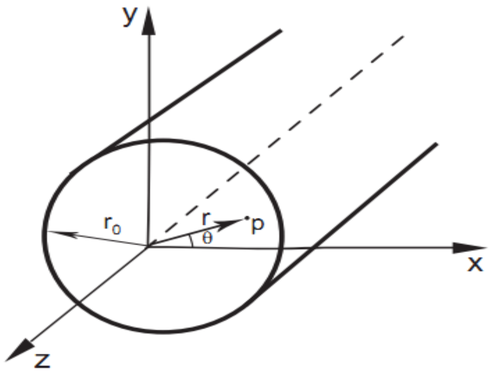


Figure 2.1: The Cross Section of a circular HVAC duct represented in a cylindric coordinate System.

Where z is the longitudinal dimension and shows the direction of sound propagation, θ is the angle, r_0 is the radius of the duct and p represents the sound pressure at a position described by the radial distance r from the center of the duct. This circular duct is assumed to be long in z-direction and to have stiff walls the assumed wall boundary conditions the particle velocity perpendicular to the duct wall is zero [8][9]. The solution to the modified wave equation, given by Eq. 2.7, can be expressed in terms of cylindric coordinates as

$$p(r,\theta,z,t) = \sum_{m=0}^{\infty}\sum_{n=0}^{\infty} J_m\,(k_{r,m,n}\,r)\,e^{jm\theta}e^{j2\pi ft} * \{\,p_{1,m,n}\,e^{-jk^{+}_{z,m,n}z} + p_{2,m,n}\,e^{jk^{-}_{z,m,n}z}\,\} \tag{2.10}$$

Where J_m represents the Bessel function of the first kind of order m in Eq. 2.10, which is a part of the general solution to the specific second-order linear differential equation termed as Bessel's equation [8]. Further m and n represents the mode numbers where $(m,n) = (0,0)$ corresponds to plane wave propagation and P_1 and P_2 are amplitude of the sound waves in the positive and negative z-direction i.e., along the length of the duct. The nodal lines for sound pressure for the circular duct are shown below in Figure 2.2;

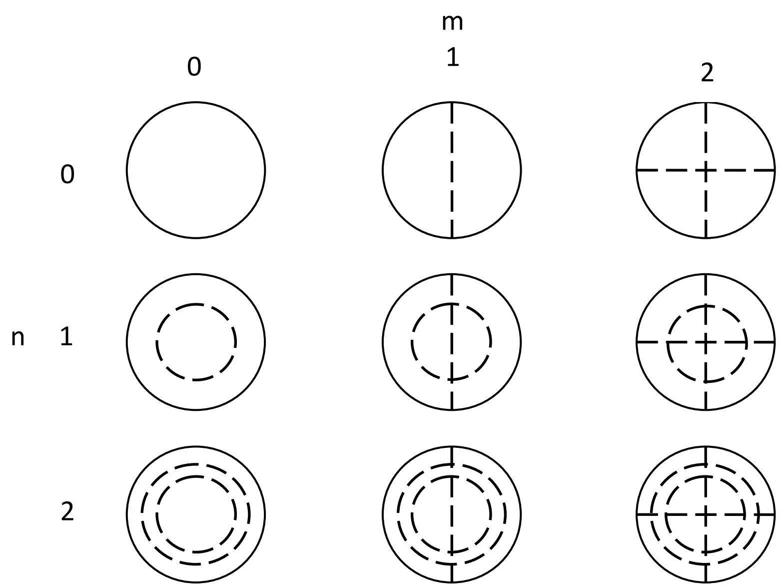


Figure 2.2: Nodal lines for the sound pressure in a circular duct

The transmission wave numbers for longitudinal propagation for the (m, n) mode in Eq. 2.10 becomes [8],

$$k_{z,m,n}^{\pm} = \frac{\mp Mk + \sqrt{(k^2) - (1 - M^2)k_{r,m,n}^2)}}{1 - M^2} \tag{2.11}$$

Where, $k = 2\pi f/c$ is a wave number and M is known as Mach number and is given as $M = U/c.U$ represents the mean air flow speed in duct and c represents the speed of sound.

For modes to be unattenuated, Eq. 2.11 must be real, while the modes will be attenuated if Eq. 2.11 is complex. To achieve these conditions the terms inside the square root must be greater or less than zero i.e., for the modes to propagate unattenuated,

$$k^2 - (1 - M^2)k_{r,m,n}^2 > 0 \tag{2.12}$$

While for attenuated modes,

$$k^2 - (1 - M^2)\, k_{r,m,n}^2 < 0 \tag{2.13}$$

Alternatively the above conditions can be used to get the cut-on frequencies for a particular mode i.e., and cut on if,

$$k^2 - (1 - M^2)\, k_{r,m,n}^2 = 0 \tag{2.14}$$

Or

$$k = k_{r,m,n}\sqrt{(1 - M^2)} \tag{2.15}$$

The cut on frequency for the higher order modes $(m, n) = (1,0)\ or\ (0,1)$ can be calculated from Eq. 2.14 which on the other hand is used to calculate the frequency under which only plane wave propagates. The cut-on wave numbers for the first two higher order modes $(m,n) = (1,0)$ and $(m,n) = (0,1)$ for a duct with radius "r_0", is given by [8][9]

$$k_{r,1,0} = \frac{1.84}{r_0} \tag{2.16}$$

And

$$k_{r,0,1} = \frac{3.83}{r_0} \tag{2.17}$$

Substituting the Eq.2.16 and Eq. 2.17 into Eq.2.15, Cut-on frequencies $f_{cut-on,m,n}$ for the modes $(m,n) = (1,0)$ and $(m,n) = (0,1)$ are obtained.

The diameter of the circular duct is $D = 2r_0$, taking wave number k as $k = 2\pi f/c$ and Mach number M as $M = U/c$ (U represents the mean air flow speed in duct), for the modes $(m,n) = (1,0)$, the cut on frequency can be found by putting values of Eq. 2.16 into Eq. 2.15 [8][7];

$$f_{cut-on,1,0} = \frac{1.84c}{\pi D}\sqrt{(1 - M^2)} \tag{2.18}$$

Similarly the cut-on frequency for the mode $(m,n) = (0,1)$ becomes,

$$f_{cut-on,0,1} = \frac{3.83C}{\pi D}\sqrt{(1 - M^2)} \tag{2.19}$$

Since $f_{cut-on,1,0} < f_{cut-on,0,1}$ only plane sound waves will propagate if,

$$f < \frac{1.84C}{\pi D}\sqrt{(1 - M^2)} \tag{2.20}$$

A single-channel ANC system may be used for this frequency range [7][13] . The 315*mm* diameter of the duct ensures plane wave propagation below 630*Hz*. For frequencies higher than 630*Hz*, multichannel ANC system or a single channel together with a reasonable size of passive silencer can be used to achieve a sufficient amount of attenuation. Thus, it may not be required to control higher order modes with an ANC system if it is combined with a reasonable size passive silencer [9].

2.3.2 Rectangular Ducts

It is also worthwhile to investigate the plane wave propagation limit for the rectangular HVAC duct. Figure 2.3 shows a rectangular duct in which z is the direction of propagation of sound, l is the height of the duct and d is the width of the duct. The duct is assumed to have stiff walls and it is infinitely long in the z-direction.

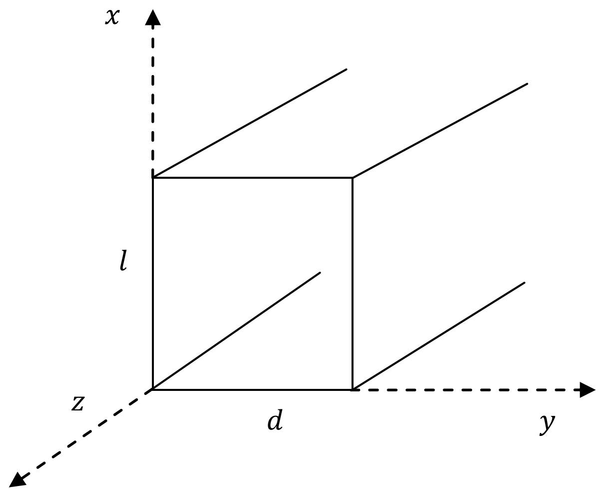


Figure 2.3: A duct with rectangular cross-section shown in a Cartesian coordinate system.

To get the values for the cut-on frequencies of higher order modes, the wave equation for the Cartesian coordinate system is used. The solution for the modified equation is [8],

$$p(x,y,z,t\,) \;=\; \sum_{m=0}^{\infty}\sum_{n=0}^{\infty} cos\left(\frac{m\pi x}{l}\right) cos\left(\frac{n\pi y}{d}\right)\{\,p_{1,m,n}\; e^{jk^{+}_{z,m,n}z} + \; p_{2,m,n}\; e^{jk^{-}_{z,m,n}z}\}e^{j2\pi f t} \qquad (2.21)$$

Where f is the frequency in Hertz (Hz) in Eq 2.21. P_1 and P_2 are amplitude of sound waves. The mode numbers and the number of nodes for sound pressure are represented by m and n in respective direction and are shown in Figure 2.4. $(m,n) = (0,0)$ corresponds to plane wave propagation. $k^{+}_{z,m,n}z$ and $k^{-}_{z,m,n}z$ are the transmission wave numbers for longitudinal propagation in positive and negative direction along the z-axis.

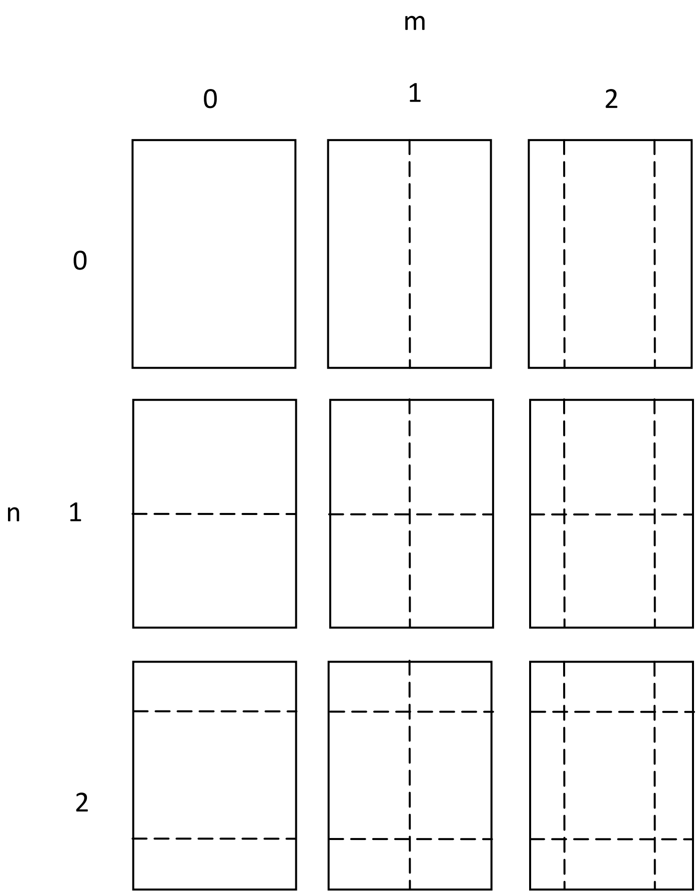


Figure 2.4: Nodal lines for the sound pressure in a rectangular duct having stiff walls.

The transmission wave numbers for longitudinal propagation for the (m, n) mode is given as [8],

$$k_{z,m,n}^{\pm} = \frac{\mp Mk + \sqrt{(k^2) - (1 - M^2)\{\left(\frac{m\pi}{l}\right)^2 + \left(\frac{n\pi}{d}\right)^2\}}}{1 - M^2} \tag{2.22}$$

Here

$$k = 2\pi f/c \tag{2.23}$$

is a wave number and M is a Mach number, is given by

$$M = U/c \tag{2.24}$$

Where, U is the mean air flow speed in the duct. Similar to the circular ducts, higher order modes in the duct are attenuated if

$$k^2 - (1 - M^2)\{\left(\frac{m\pi}{l}\right)^2 + \left(\frac{n\pi}{d}\right)^2\} < 0 \tag{2.25}$$

And cut on if

$$k^2 - (1 - M^2)\left\{\left(\frac{m\pi}{l}\right)^2 + \left(\frac{n\pi}{d}\right)^2\right\} = 0 \qquad (2.26)$$

It can be written as,

$$K = \left\{\left(\frac{m\pi}{l}\right)^2 + \left(\frac{n\pi}{d}\right)^2\right\}\sqrt{(1 - M^2)} \qquad (2.27)$$

Eq. 2.27 can be utilized to calculate cut on frequency for the first higher modes and thus can be used to calculate the frequency under which plane wave propagates. The cut-on frequency for the mode $(m, n) = (0,1)$ can be calculated as,

$$f_{cut-on,0,1} = \frac{C}{2h}\sqrt{(1 - M^2)} \qquad (2.28)$$

The cut-on frequency for the mode $(m, n) = (1,0)$ is given by

$$f_{cut-on,1,0} = \frac{C}{2b}\sqrt{(1 - M^2)} \qquad (2.29)$$

Plane wave propagates only if,

$$f < min\,(f_{cut-on,0,1}, f_{cut-on,1,0}) \qquad (2.30)$$

Chapter 3 : Active Noise Control Systems

Humans are generally exposed to acoustic noise in industry, in homes, in means of public transportation, in hospitals, in aircrafts, etc. and are caused by large number of sources. Common noise sources are e.g. engines, blowers, transformers, fans, compressors, MRI cameras and ventilation systems. The expansions of high density housing have increased the exposure of the population to noise from different sources as well. Vibration caused by machinery is also a type of noise creating problems in transportation and in manufacturing [4]. A common source of acoustic noise is the low frequency noise produced in heating ventilation and air-conditioning (HVAC) ducts by turbulences of the air flow and due to fan interaction with the air it move. Such noise is not only annoying to humans, but may also cause irritation and reduced performance of humans.

Recent awareness against noise pollution demands more effective control of noise sources. Traditionally, techniques applied to acoustic noise control e.g. in ventilation ducts employ passive methods i.e, silencers, barriers and enclosure to attenuate the undesired noise. Passive silencers use the concept of energy loss and the attenuation depends upon the thickness, length and sound absorbing quality of the silencer material used. Passive silencers are generally very effective over a broad frequency range. The effectiveness of passive control products for ventilation noise is generally insignificant at low frequencies as such methods tend to be too bulky and costly for low-frequency noise attenuation [4]. Another noise control technique that is well suited for low frequency noise attenuation is "Active Noise Control" where additional secondary noise sources are used to cancel the unwanted noise originating from the actual noise source.

Active Noise Control (ANC) rely on the principle of "destructive interference"[12]. The basic principle of ANC is "to generate a secondary sound or control sound (anti noise) that is equal in amplitude but 180 degree out of phase with the original, "primary" sound". The superposition of these two sounds i.e., primary and secondary sound will result in a residual sound whose amplitude is reduced as compared to primary sound. The attenuation level depends on the generated secondary sounds phase and amplitude accuracy as compared to the primary noise [2, 4, 11]. Active noise control is gaining popularity because it may, depending on application, offer low-frequency noise attenuation to low cost, volume and weight [1].

A simple example of the superposition principle is shown in Figure 3.1. The unwanted noise (primary noise), the canceling noise (anti noise) and the residual noise after superposition is shown.

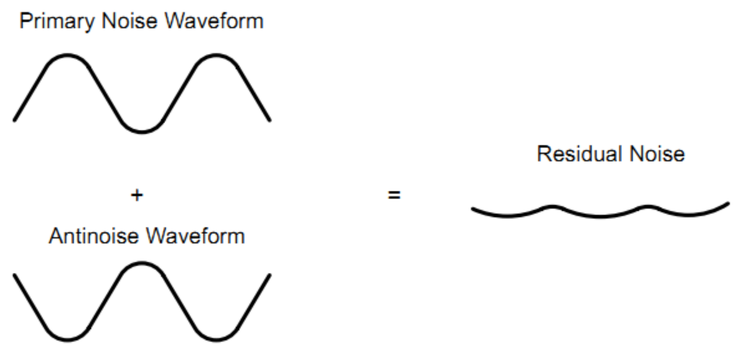


Figure 3.1: Physical Concept of Active Noise Cancellation (ANC)

The Active Noise Control (ANC) concept is not new. The design of acoustic ANC utilizing a microphone and an electronically driven loudspeaker to generate a canceling sound was first proposed in 1936 by Paul Lueg [1][14] in a US patent. Until 1970, most of the work on ANC was theoretical. The introduction of Adaptive algorithms by B. Widrow [15] and the advancement in digital signal processing technologies hardware encouraged extensive research on ANC. 1981 was the beginning for the practical implementation of ANC systems with adaptive filters, the filtered-X LMS (FXLMS) algorithm was presented by B, Widrow [15].

In 1980's, with tremendous development in technology especially in computers and microprocessors, which started getting cheaper and more powerful, research on the use of ANC in different applications got momentum. For digitized signals, to be processed numerically in real-time, specific Digital Signal Processors (DSPs) were designed. Such hardware's facilitated low-cost implementation of powerful adaptive algorithms[2] and encouraged the extensive development and application of ANC systems based on digital adaptive signal processing technology. The advancement in DSP hardware allow more sophisticated and computational demanding algorithms to be implemented in real time to enhance system performance [1]. One of the first area where ANC applied concerned "Low frequency noise produced by HVAC systems" [7] [17][18]. Such applications and a wide range of solutions are described in large number of book chapters, research articles and conference proceedings [7] [17] [18][19]. An overview can be found in [1] [4].

Although Figure 3.1 presents the ANC concepts in a very simple manner but in practice the environment and acoustic noise source/sources characteristics are generally non stationary, noise transfer paths and noise source/sources are non-stationary, therefore an ANC system must be adaptive in order to maintain noise control under such conditions [1][2]. A basic ANC system is shown in Figure 3.2**.** Basically, Active noise control systems consist of four main components:

- Sensors: Microphones and/or accelerometers, etc. are used to feed the controller with suitable reference and error signals.
- Actuators: Loud speakers and/or electro dynamic shakers, etc
- The Plant: The physical system under control, e.g. in plane wave noise control of a HVAC duct, the plant comprise of signal conditioning filter, an amplifier, a loudspeaker, an acoustic path between loudspeaker and a error microphone, etc.
- Controller: DSP where the adaptive control algorithms are implemented and control the actuators via the signals received by the sensors.

A fundamental features of an ANC system is reliability, precise control and temporal stability [4]. In order to obtain significant noise attenuation, the phase and amplitude of generated (secondary) noise should match the primary (source) noise but with a 180 degrees phase difference. It is therefore advantageous for the controller to be digital in nature [4].

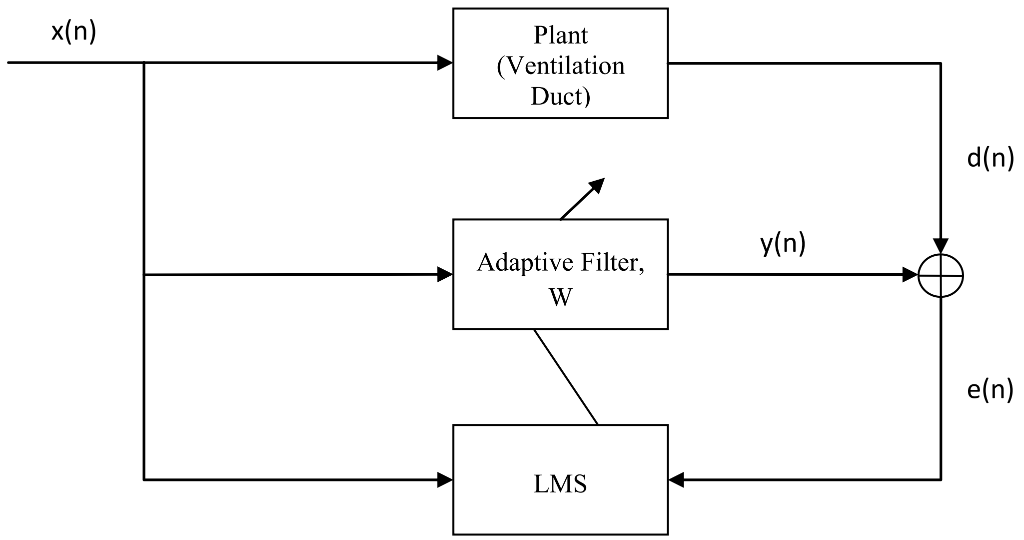


Figure 3.2: Block diagram of a basic ANC system.

Figure 3.2 shows block diagram of a basic ANC system. Where, $x(n)$ is the reference signal and $e(n)$ is the error signal. The sensors used to provide reference and error signals are usually microphones or accelerometers. The controller is usually adaptive and is generally based on a version of the Filtered-X Least Mean Square (FXLMS) algorithm implemented on a DSP module which controls the actuator based on the reference and error signals.

3.1 ANC Applications

At low frequencies, Active Noise Control technique is an efficient way to provide substantial attenuation as compared to Passive Noise Control[2]. A large number of applications rely on ANC and some of the applications where ANC is used for attenuation of unwanted noise [2][20] are as follows,

Appliances: large number of electronics appliances such as vacuum cleaners, washing machines, and air conditioning ducts use ANC technique to attenuate low frequency noise.

Industrial: In industry there is a large amount of noise from fans, transformers, compressors, chimneys etc. Noise control technique is used extensively, either Active methods or Passive methods or combinations thereof.

Automotives/Transportation: Noise reduction issue in automotive and in transportation is taken seriously now a days. Lot of work is carried out to provide noise attenuate in e.g. cars, trucks, ground moving machines, airplanes and helicopters[2] [4].

3.2 ANC Systems

Active Noise Control is based on either "*feed forward*" control or "*feedback*" control where the active noise controller attempts to cancel the noise without the benefits of a separate reference input signal [4][22][23]. Feed forward ANC is generally more robust as compared to feedback ANC, particularly when the feed forward system has a reference input signal independent of the secondary anti-noise [4].

Noise may be classified into broadband and narrowband noise [2] [4]. Hence, ANC may also be classified as broadband or narrowband. Broadband noise generally has stochastic properties and its energy is distributed over a frequency range, etc.

In contrast to broadband noise is narrowband noise whose energy is found at discrete frequencies or approx. discrete frequencies. Hence, periodic or nearly periodic noise [2]. In this thesis, feedforward approach will be applied.

3.2.1 Broadband Feed forward ANC Systems

A significant amount of broadband noise is produced in Heating, Ventilation and Air conditioning (HVAC) systems. A very simple single channel feed forward ANC control system for a narrow and long duct is illustrated in Figure 3.3 . It has a single reference sensor, a single secondary source and a single error sensor. A reference microphone is placed near the noise source to sense the noise produced by it and generates the time continuous version of the reference signal $x(n)$ before it reaches the secondary source loudspeaker. The ANC system processes the reference signal in order to produce a control signal to drive the secondary source loudspeaker. An error microphone is placed downstream of the loudspeaker. This microphone senses the duct noise and produces the time continuous version of the error signal $e(n)$ which is feed to the ANC controller. The main goal of the ANC controller is to minimize the noise in the duct by generating so called anti-sound $y(n)$ via loudspeaker which is 180 degree out of phase but equal in magnitude with the primary sound when it reaches the error microphone in the duct. The ANC controller adjusts itself automatically based on the information provided to it by the reference and error signals to minimize the acoustic noise sensed by the error microphone in the minimum mean-square sense.

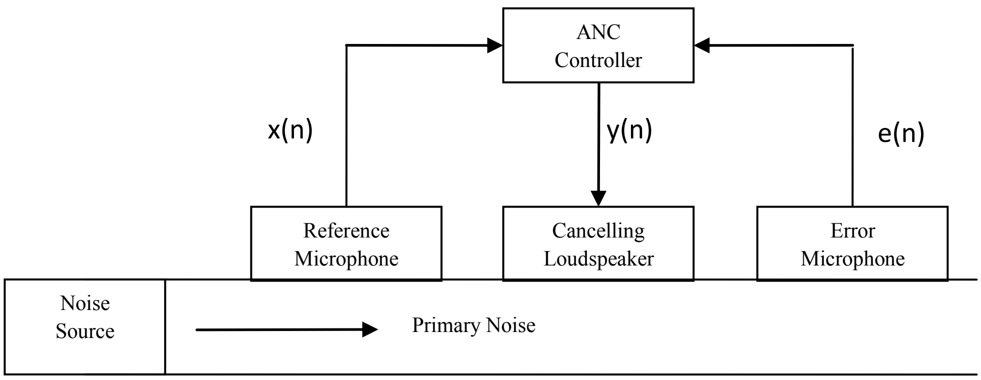


Figure 3.3: Single Channel Broadband Feedforward ANC system for Duct noise control.

3.2.2 Narrowband Feed forward ANC Systems

Noises generally generated by engines, compressors, propellers, motors and fans are of the periodic type. By using appropriate sensors, we can directly observe the mechanical motion of these sources which produce such noise. The sensors used to observe the motion gives us the information in the form of electrical signals which contains all the harmonics of primary generated noise and fundamental frequency. Block diagram is shown in Figure 3.4, which explains narrowband ANC for attenuating noise in a duct.

In narrowband ANC, the reference microphone may favorably be replaced by a non-acoustic sensor which eliminates eventual problems of acoustic feedback. An error microphone is still required to measure the noise to be controlled to adjust the coefficients of the adaptive filter to minimize the mean-square error [2].

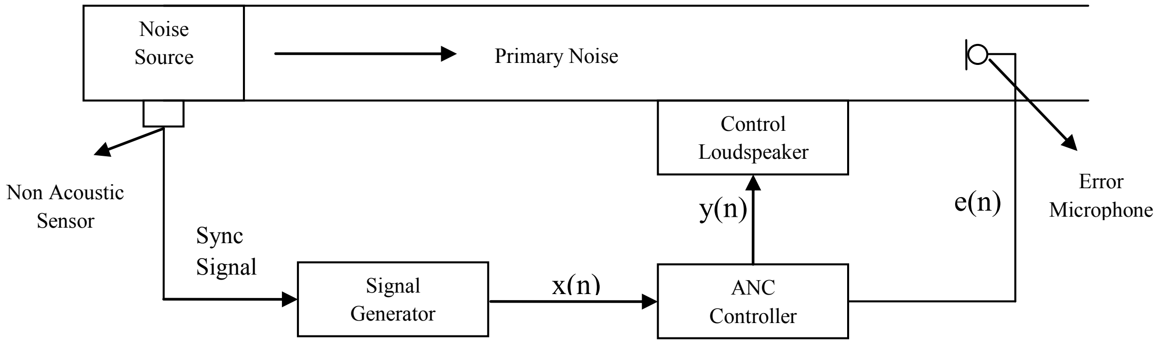


Figure 3.4: Narrowband Feedforward ANC system for Duct noise control.

3.2.3 Feed Back ANC Systems

A Feedback ANC configuration is illustrated in Figure 3.5. In cases where a suitable reference signal is not available, such configuration may be utilized.

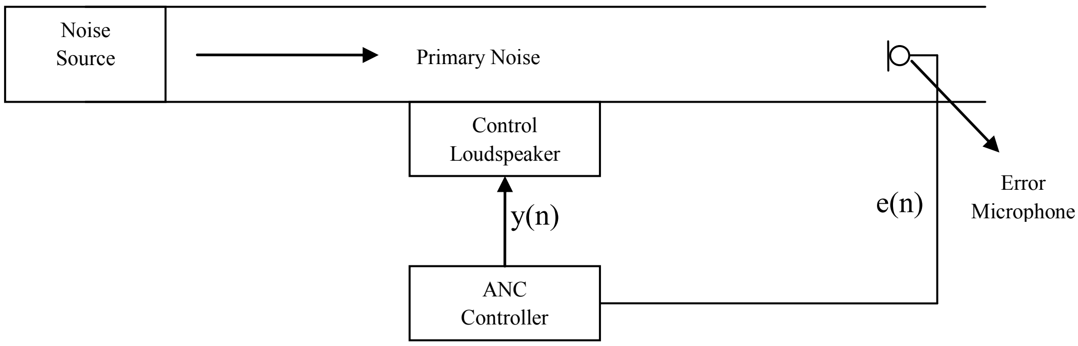


Figure 3.5: Feedback ANC system

3.2.4 Multi-Channel ANC Systems

In large scale applications, for instance large ducts or enclosures, passenger compartments of aircrafts and automobiles etc, it is desirable to attenuate or cancel complex acoustic sound field noise in cavities with substantial volume [25][26]. In such cases single channel systems are ineffective and multi-channel ANC is generally required [4].

A multi channel ANC system generally requires a number of reference sensors and a number of error sensors and actuators. In Figure 3.6 , a block diagram for multi channel ANC system is shown [2].

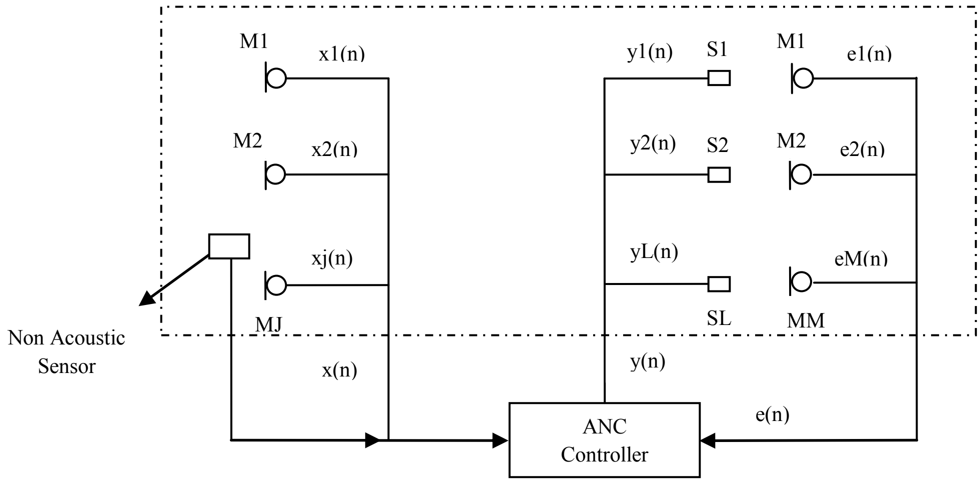


Figure 3.6: Block diagram of a Multi Channel ANC system

In Figure 3.6, $x(n)$ is the vector reference signal obtained from the $x_1(n)$, $x_2(n)$ ...$x_j(n)$ obtained from multiple reference sensors i.e microphones $M1, M2, \ldots Mj$. $y(n)$ is the vector of control signal obtained from ANC controller and is fed into multiple number of control speaker's denoted by $S1, S2, \ldots SL$ through control signal $y_1(n)$, $y_2(n)$ ...$y_L(n)$ respectively. $e(n)$ is the vector of residual noise of $e_1(n)$, $e_2(n)$ ...$e_M(n)$ sensed by the error microphones $M1, M2, \ldots MM$ and is given back to ANC controller for further processing. If multi-channel ANC is considered for a particular application; a careful investigation of the acoustic characteristics of it is required. Both, the number of reference sensors, error sensors and actuators, and their properties as well as their position and size have to be considered. To motivate an implementation of an ANC system certain attenuation performance requirements generally have to be fulfilled as well as possible constraints concerning the weight and positions of the reference and error sensors, and actuators. The price for an adequate ANC system for a particular application may also be a limiting factor. For multiple reference and multiple output broadband feedforward ANC, signal processing structures have been introduced in[27].

Chapter 4 : Adaptive Control Algorithms

ANC system consists of adaptive filter which has two vital parts. These two parts include adjustable digital filter and a control algorithm. The digital filter performs the required signal processing while the control algorithm adjust the digital filter's coefficients. Adaptive filter algorithms play an important role in ANC systems as they generally are used in such system as adaptive controller. Considering the importance of the control algorithms, this chapter will highlight the advanced and more complex control algorithms that have been deployed as adaptive controllers in ANC systems.

4.1 Adaptive Algorithms

The design of digital filters with fixed coefficients requires well defined specifications [30]. However, if adaptive filters are considered they adjust their filter coefficients automatically to minimize a specified objective function [30]. A basic adaptive filter structure is shown in Figure 4.1.

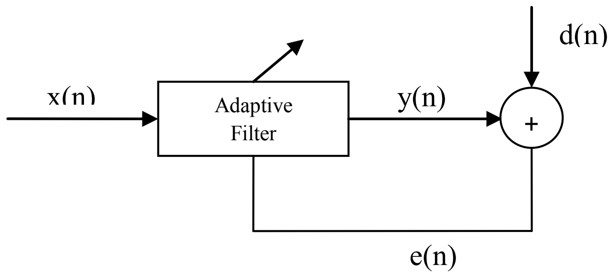


Figure 4.1: Figure showing Basic Adaptive Filter Structure

In Figure 4.1, The adaptive filter's output "$y(n)$" is compared with a desired signal "$d(n)$" to yield an error signal "$e(n)$" which is fed back to the adaptive filter. Here, n is the iteration number. An adaptive algorithm updates the adaptive FIR filter coefficients during each iteration generally by extracting information from error signal and reference signal. The error signal $e(n)$ is determined as,

$$e(n) = d(n) - y(n) \tag{4.1}$$

Basically, the error signal $e(n)$ in combination with the reference signal is generally used by the adaptive algorithm by its coefficient adjustment algorithm for the adjustment of the filter coefficients with purpose to minimize e.g. the mean-square error or the least-squares error [30]. Thus, the error signal shows how well the adaptive filter output signal matches the desired signal [30].

Several crucial aspects arise when selecting adaptive filters i.e, the average the filter coefficients converge to a biased solution, computational complexity, divergence problems, etc. The choice of the adaptive filter algorithm will have an impact on such issues [30]. There is a great variety of adaptive filter applications which have their own challenges. The choice of adaptive algorithms for a particular application should be made with care because an adaptive filter solution to one application may not be suitable for other application. In the selection of adaptive algorithm factors like computational cost, performance and robustness should also be considered for any adaptive application[31].

There exist a large number of different adaptive algorithms; such as LMS (Least Mean Square) algorithm, NLMS (Normalized Least Mean Square Algorithm), LLMS (Leaky Least Mean Square Algorithm), RLS (Recursive Least Square) algorithm. In this thesis, only a few of their Filtered-X versions have been considered for attenuation of duct noise.

4.1.1 Least Mean Square (LMS) Algorithm

The LMS algorithm is a gradient based algorithm which adjusts its filter coefficients in the negative direction of the instantaneous gradient of the squared error signal with respect to the coefficient vector [32]. The method of steepest decent can be utilized theoretically to adapt a filter to minimize mean-square value of the error signal, however, this is a theoretic method and may not be used in reality. However, in the LMS algorithm, the instantaneous square- forms a direct estimate of the mean-square value of the error signal and thus the gradient estimate used to update the filter coefficients. The update equation for LMS algorithm can be given as [33] [34],

$$y(n) = \boldsymbol{w}^T(n)\boldsymbol{x}(n) \tag{4.2}$$

$$e(n) = d(n) - y(n) \tag{4.3}$$

$$\boldsymbol{w}(n+1) = \boldsymbol{w}^T(n) + \mu\, e(n)\boldsymbol{x}(n) \tag{4.4}$$

Where, $y(n)$ is the output signal of adaptive FIR filter, $\boldsymbol{w}(n)$ is the coefficient vector of the adaptive FIR filter and at time n it can be defined as

$$\boldsymbol{w}(n) = [w_1(n)\ \ w_2(n)\ \ \ldots\ w_L(n)]^T$$

$\boldsymbol{x}(n)$ is the reference signal vector and can be defined as,

$$\boldsymbol{x}(n) = [x(n), x(n-1), \ldots, x(n-L+1)]^T$$

$e(n)$ is the error signal and μ is the step size and for the LMS algorithm to convergence in the mean-square it is recommended that the step size may selected as [33] [34],

$$0 < \mu < \frac{1}{L.E\,[x^2(n)]} \tag{4.5}$$

Where, L is the filter length and $E\,[x^2(n)]$ is the signal power of the input signal $x(n)$.

4.1.2 Filtered-x Least Mean Square (F-x LMS) Algorithm

The utilization of LMS algorithm assumes that the error signal is given by the difference between the desired signal $d(n)$ and the filtered output signal $y(n)$ [33][34]. In ANC applications, the output signal of the adaptive filter does not form an estimate of the so called desired signal. In such applications the estimate of the desired signal is produced by a dynamic system whose input is the adaptive filters output signal. This dynamic system is known as forward path or called control path. This path constitutes D/A converter, low pass filter, amplifier, anti-noise loudspeaker, acoustic path between anti-noise loudspeaker and error microphone, error microphone, Low pass filter, Amplifier and A/D convertor [4]. Thus, a standard adaptive filter algorithm whose adaptive filter produce an estimate of the desired signal as output signal is not defined for ANC applications. In such case a so called Filtered-X version of the standard adaptive filter algorithm has to be used to adjust the coefficients to minimize e.g. the mean-square error or the least-squares error.

The Filtered-x LMS algorithm is one of the common adaptive filter algorithms that are defined for ANC applications. The filtered-x LMS algorithm can be used in control applications where forward path is present. This algorithm generally uses an FIR-filter estimate of the forward path to filter the reference sensor signal and this filtered reference signal is used to form the gradient estimate. The filtered x-LMS algorithm compensate for the forward path by filtering the input signal $\boldsymbol{x}(n)$ by an estimate of the forward path which produces a filtered reference signal $\boldsymbol{x}_{\boldsymbol{C}'}(n)$. This filtered reference signal then becomes input for the weight adjustment algorithm LMS. $d(n)$ is the desired signal and it is propagated through the primary physical path P. By filtering $\boldsymbol{x}(n)$ with the adaptive FIR- filter W, the output from the adaptive filter is obtained. This output is denoted as $y(n)$. The output $y(n)$ is the input to the canceling loudspeaker and error signal $e(n)$ and is then achievd in error microphone by acoustic interference of $y_C(n)$ which is output $y(n)$ filtered by the forward path, with the desired signal $d(n)$. The block diagram in Figure 4.2 illustrates the filtered-x LMS algorithm. The error signal can be written as,

$$e(n) = d(n) + y_C(n) \qquad (4.6)$$

The filtered-x LMS algorithm can be written using vector notation as [4] [7],

$$y(n) = \boldsymbol{w}^T(n)\,\boldsymbol{x}(n) \qquad (4.7)$$

$$e(n) = d(n) + y_C(n) \qquad (4.8)$$

$$\boldsymbol{w}(n+1) = \boldsymbol{w}(n) - 2\mu\, e(n)\boldsymbol{x}_{\boldsymbol{C}'}(n) \qquad (4.9)$$

In order to converge in the mean square of the filtered-x LMS algorithm, the step size should be selected according to [12],

$$0 < \mu < \frac{1}{E[x^2{}_{C'}(n)](L_W+\Delta)} \qquad (4.10)$$

Here, L_W is the length of the adaptive filter and Δ is the number of samples corresponding to over all delay in the forward path.

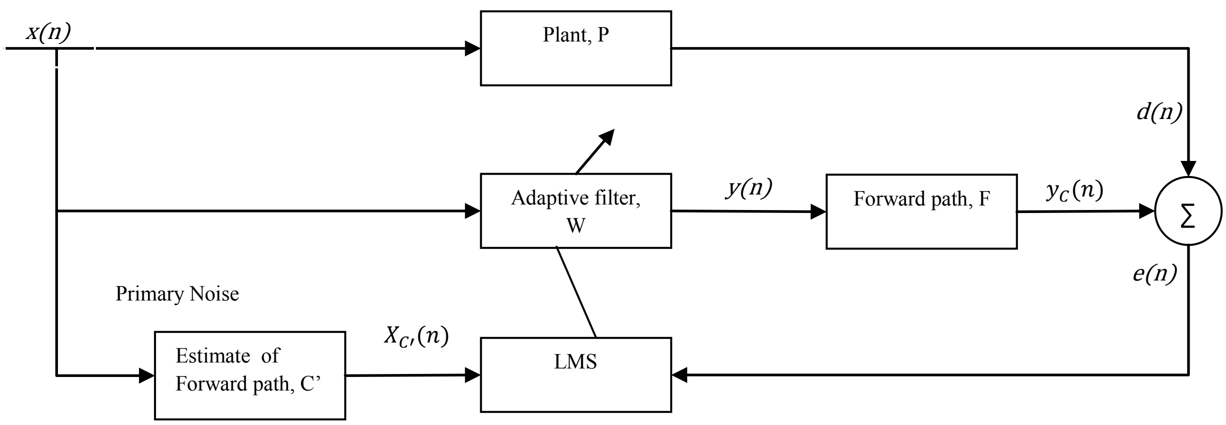


Figure 4.2: Block diagram of ANC system using the filtered-x LMS algorithm

4.1.3 Normalized Least Mean Square (NLMS) Algorithm

By normalization the step size with respect to the energy in the reference signal vector in the LMS algorithm the algorithm known as NLMS algorithm is obtained. It is one of the important techniques to maintain the effective speed of convergence while maintaining the desired steady state response [20][35]. Thus, NLMS algorithm solves the problem of instability of LMS algorithm due to variation in the power of the reference signal. The co-effects adjustment of Normalized LMS algorithm is given by,

$$\boldsymbol{w}(n+1) = \boldsymbol{w}(n) + \beta \frac{\boldsymbol{x}(n)}{\varepsilon + \|\boldsymbol{x}(n)\|^2} e(n) \tag{4.11}$$

Here 'β' is the new step size and " $\|\boldsymbol{x}(n)\|$" is the L_2 norm which reduces the sensitivity of LMS by affecting the step size in a negative gradient direction and "ε" is a small positive real value which avoids division by zero in case $\boldsymbol{x}(n)$ becomes zero. The NLMS algorithm converges when 'β' obeys the inequalities [34][35],

$$0 < \beta < 2 \tag{4.12}$$

A block diagram illustrating the working of filtered-x NLMS algorithm for an ANC system is shown in Figure 4.3.

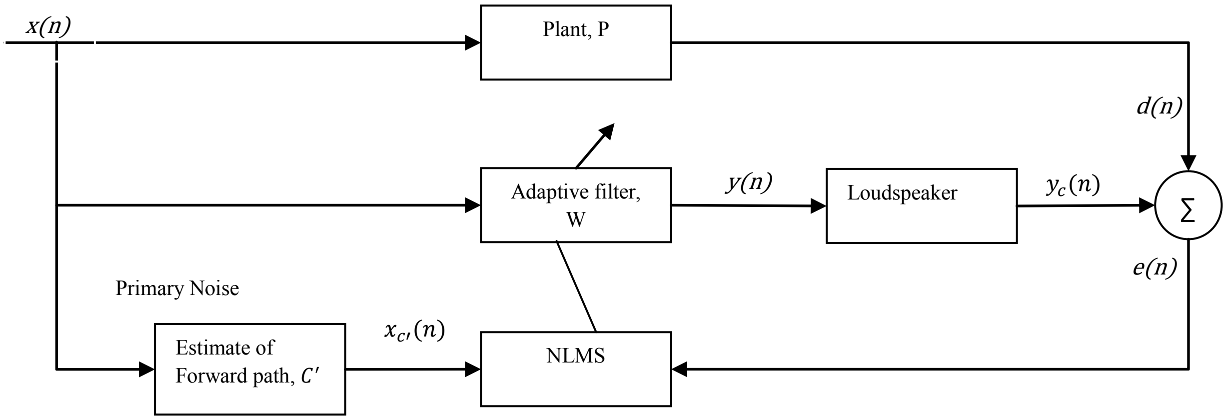


Figure 4.3: Block diagram of ANC system using the filtered-x NLMS algorithm

In Figure 4.3, the input signal $\mathbf{x}$(n) is filtered by an estimate of the forward path, which produces a filtered reference signal $\mathbf{x}_{c'}$(n). This filtered reference signal then becomes input for the weight adjustment algorithm NLMS. d(n)is the desired signal and it is propagated through the primary physical path P. By filtering x(n) with the adaptive FIR- filter W, the output from the adaptive filter is obtained. This output is denoted as y(n). The output y(n) is the input to the canceling loudspeaker and error signal e(n) and is then achievd in error microphone by acoustic interference of y_c(n) which is loudspeaker output y(n) filtered by the forward path, with the desired signal d(n). The algorithm update equation will be as follows,

$$\boldsymbol{w}(n+1) = \boldsymbol{w}(n) + \beta \frac{x_{C'}(n)}{\varepsilon + \|x_{C'}(n)\|^2} e(n) \tag{4.13}$$

4.1.4 Leaky Least Mean Square (LLMS) Algorithm

In cases where the reference signal to the LMS algorithm is poorly conditioned the problem of bias accumulation in the coefficients of the adaptive filter is likely to occur [34]. Thus, the LMS algorithm is likely to exhibit divergence problem.

The Leaky LMS algorithm is effective for addressing the problem of poorly conditioned reference signal and thus to overcome associated divergence problems of LMS algorithm. The problem of bias accumulation in the coefficients of the adaptive filter is generally solved by using a "leakage" technique. Basically, an adaptive algorithm is modified to behave as if white noise is added to the reference signal without actually adding any white noise to the reference signal. The co-efficient adjustment of FXLMS algorithm is given by,

$$\boldsymbol{w}(n+1) = v\boldsymbol{w}(n) + \mu\, e(n)\boldsymbol{x}(n) \tag{4.14}$$

Where "v" is the leakage factor and $v = 1 - \mu\gamma$ and its value lies between,

$$0 < v \leq 1 \tag{4.15}$$

Here, γ is weighting factor. Therefore, the value of $1 - v$ should be kept smaller than the value of μ [7].

A block diagram in Figure 4.4 shows the implementation of filtered-x LLMS algorithm. The input signal $\boldsymbol{x}(n)$ is filtered by an estimate of the forward path, which produces a filtered reference signal $\boldsymbol{x}_{C'}(n)$. This filtered reference signal then becomes input for the weight adjustment algorithm LLMS. $d(n)$ is the desired signal and it is propagated through the primary physical path P. By filtering $\boldsymbol{x}(n)$ with the adaptive FIR- filter W, the output from the adaptive filter is obtained. This output is denoted as $y(n)$. The output $y(n)$ is the input to the canceling loudspeaker and error signal $e(n)$ and is then achievd in error microphone by acoustic interference of $y_c(n)$ which is loudspeaker output $y(n)$ filtered by the forward path, with the desired signal $d(n)$. The weight update equation for the filtered-x LLMS algorithm is as follow,

$$\boldsymbol{w}(n+1) = v\boldsymbol{w}(n) + \mu\, e(n)\boldsymbol{x}_{C'}(n) \tag{4.16}$$

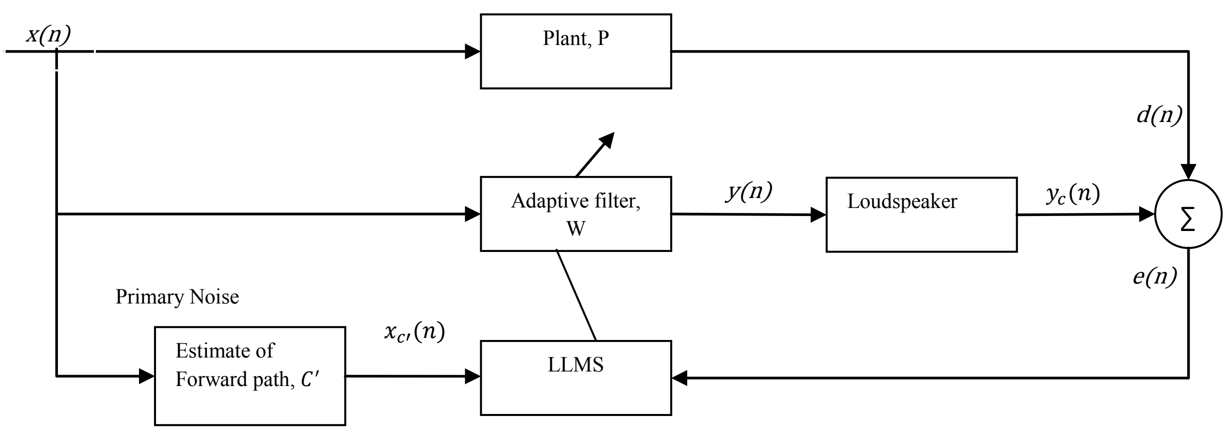


Figure 4.4: Block diagram of ANC system using the filtered-x LLMS algorithm

4.1.5 The Exponentially Weighted Recursive Least Square (RLS) Algorithm

RLS algorithm is a recursive adaptive filter algorithm that uses a reference signal $\boldsymbol{x}(n)$ and a desired signal $d(n)$ to calculate the least-squares solution for the adaptive filters coefficients $\boldsymbol{w}(n)$ in each iteration n. The RLS algorithm provides a computationally efficient method for calculating the least-squares solution for the adaptive filters coefficients $\boldsymbol{w}(n)$ in each iteration n. The RLS algorithm has a smaller steady state error and faster convergence but the computational complexity is higher as compared to LMS [4]. The RLS algorithm can be written as

$$\boldsymbol{k}(n) = \frac{\lambda^{-1}\,\boldsymbol{Q}(n-1)\boldsymbol{x'}_C(n)}{x'^{\mathrm{T}}{}_C(n)\lambda^{-1}\,\boldsymbol{Q}(n-1)\boldsymbol{x'}_C(n)+1} \tag{4.17}$$

$$y(n) = \boldsymbol{w}^T(n)\boldsymbol{x}(n) \tag{4.18}$$

$$e(n) = d(n) - y_C(n) \tag{4.19}$$

$$\boldsymbol{w}(n+1) = \boldsymbol{w}(n) - \boldsymbol{k}(n)e(n) \tag{4.20}$$

$$\boldsymbol{Q}(n) = \lambda^{-1}\boldsymbol{Q}(n-1) - \lambda^{-1}\boldsymbol{k}(n)\boldsymbol{x'}^{\mathrm{T}}{}_{\boldsymbol{C}}(n)\boldsymbol{Q}(n-1) \tag{4.21}$$

Where $\boldsymbol{k}(n)$ is the gain factor , $\boldsymbol{Q}(n)$ is the inverse of the input signal autocorrelation matrix evaluated recursively, $\boldsymbol{x'}_C(n)$ is the vector of the filtered input signal, filtered by C' and $\boldsymbol{x'}^{\mathrm{T}}{}_{\boldsymbol{C}}(n)$ is the transpose of $\boldsymbol{x'}_C(n)$ and $y_C(n)$ is the output signal of the forward path. $\boldsymbol{Q}(0) = I\delta^{-1}$ Where delta is small positive number and $\boldsymbol{w}(0) = 0$.

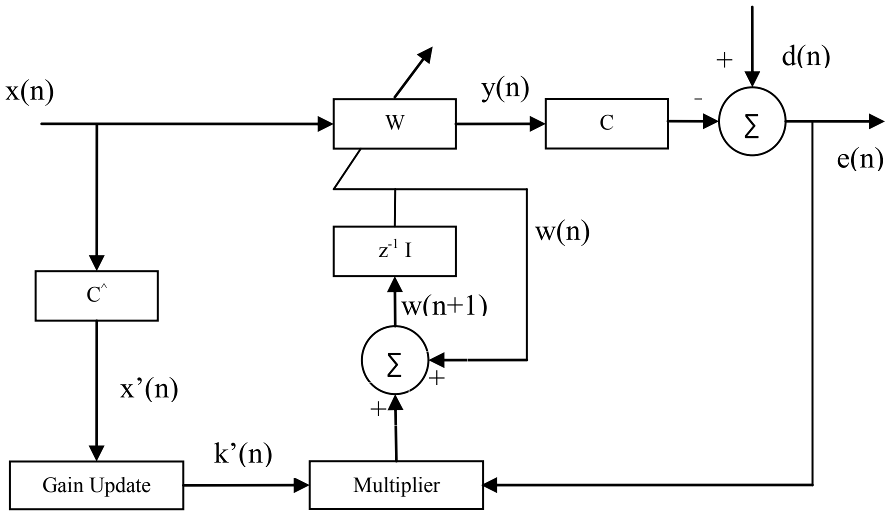


Figure 4.5: Block diagram of FXRLS Algorithm

4.1.6 Filtered–u-Recursive Least Mean Square (F-u-RLMS) Algorithm

Infinite impulse response (IIR) filters provide an advantage in case of acoustic feedback present during the implementation of active noise control in a duct. The filtered U recursive LMS algorithm by Feintuch is one of the several algorithms used for ANC using IIR filters [2]. The poles introduced by the acoustic feedback are eliminated by the poles of adaptive IIR filter [2]. IIR filters require reduced number of arithmetic operations as their poles provide well matched characteristics with a structural order possessing lower order [36]. The poles of the IIR filter provide same performance as FIR filter but with a much lower order because the presence of feedback. This feedback leads to infinite impulse response with only a finite number of coefficients. That is why IIR filters require less computation for each sample than FIR [36]. The convergence rate is slower as compare to adaptive FIR filters and its poles may introduce instability problems. Moreover, it might converge to local minimum and the error signal is not guaranteed to be reduced at every iteration [34].

A block diagram of adaptive IIR based FuRLMS algorithm implementation is shown in Figure 4.6. The output of the controller for the FURLMS is given by

$$y(n) = \sum_{i=0}^{N-1} \boldsymbol{a_i}(n)\, x(n-i) + \sum_{j=1}^{M} \boldsymbol{b_j}(n)\, y(n-j) \qquad (4.21)$$

Where $\boldsymbol{a_i}(n)$ and $\boldsymbol{b_j}(n)$ are the weight vectors of filters A and B respectively and N and M are the order of filters A and B respectively. The weight update equations for filtered- U Recursive LMS algorithm can be given as [2] [4],

$$\boldsymbol{a}(n+1) = \boldsymbol{a}(n) - \mu e(n)\boldsymbol{x_C}(n) \qquad (4.22)$$

And,

$$\boldsymbol{b}(n+1) = \boldsymbol{b}(n) - \mu e(n)\boldsymbol{y_C}(n-1) \qquad (4.23)$$

The residual error signal is given as [20],

$$e(n) = d(n) - c(n) * y(n) \qquad (4.24)$$

Where $c(n)$ is the impulse response for the filter C. The IIR filter uses the same residual error in the adaptation process for direct and feedback filters. When the residual error is minimum, both filters stop adapting. At this moment, the filter A models the plant P and filter B models the feedback path completely. C represents the forward path as well as feedback path. After both A and B have converged, the measured residual error $e(n)$ is minimal in the minimum mean square sense [20]

Although, the Filtered-U Recursive LMS Algorithm eliminates the acoustic feedback by introducing poles yet it has some disadvantages too. They have slow convergence rate compared to FIR filters. Selection of small step size leads to slow convergence which is undesirable in some applications[19]

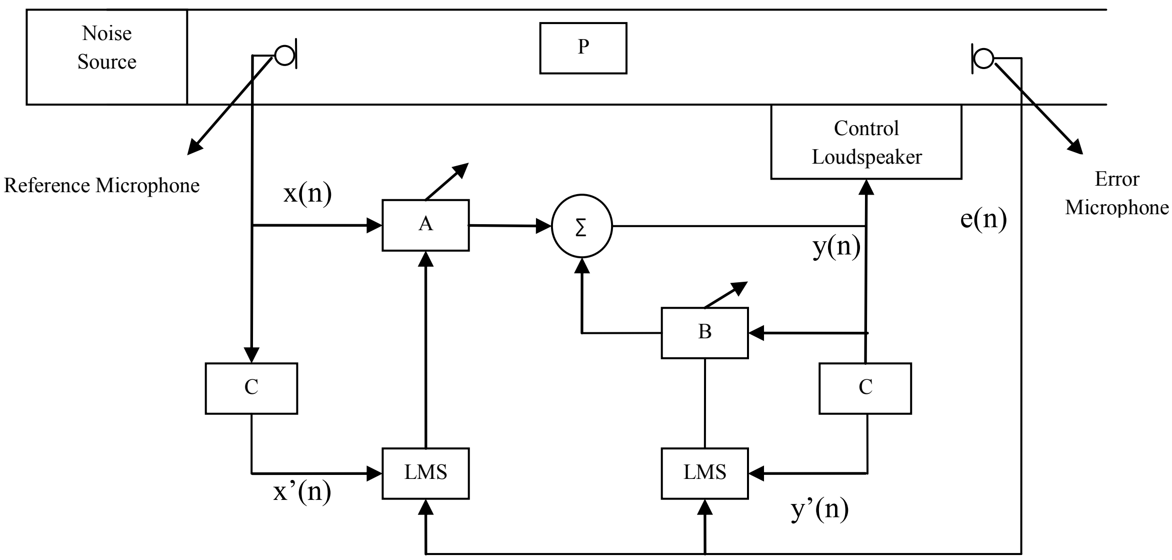


Figure 4.6: Block diagram of Filtered-u-Recursive LMS Algorithm

Chapter 5 : Remote ANC Laboratory

Experimental knowledge is essential in order to get better insight of the theoretical knowledge. Without laboratory experience, theoretical knowledge gained from books is abstract. The main part of thesis work is to implement adaptive control algorithms on real hardware in a remote laboratory and to evaluate their performance in the remote laboratory. To acquire adequate understanding of Active Noise Control comprehensive experimentation is required.

ANC experimentation either performed in a hands-on laboratory or in a remotely controlled laboratory has proved to be challenging for the students. In this chapter the remotely controlled ANC laboratory is described.

5.1 Remote laboratory and Its Benefits for ANC

Educational laboratories can be traditional hands-on laboratories or remotely controlled laboratories, with real instruments and equipment.

In traditional ANC laboratories for duct noise control, physical presence in the laboratory is necessary in order to access the hardware and to carry out ANC related experiments. Experimental wiring, setting of the hardware etc. it all needs to be done manually. Each time a new system has to be configured the wiring has to be changed manually.

An ANC remote laboratory is an attractive alternative to traditional ANC laboratory. In remotely controlled laboratories, real hardware can be accessed and controlled over the Internet remotely. There is no need to be in the laboratory for performing the experiments. Remote laboratories are gaining popularity all over the world. One such remote laboratory is recently built by Blekinge Institute of Technology (BTH), Sweden. Real time ANC experiments can be performed now remotely via remote laboratory of BTH. This remote lab was built for carrying out different types of experiments which are very useful for the students and can be performed remotely. Experiments related to digital signal processing, acoustics and ANC can be performed via the prototype laboratory [37].

A remotely controlled ANC laboratory has a number of benefits over traditional laboratory. Remote laboratory can be accessed 24/7 from anywhere in the world over Internet. It enables distance learning education. Courses generally given on campus can now be offered as distance courses. Moreover, students and researchers can collaborate with students and researchers at other universities and may work together on ANC.

This remote lab hardware can be used easily from remote end for carrying out ANC experiments. The reason to describe remote laboratory is to facilitate the readers with understanding of the system on which ANC experiment will be performed. A photo of the hardware in the remote ANC laboratory is shown in Figure 5.1. A brief introduction of the hardware and user interface of the remotely controlled ANC lab is described in this section.

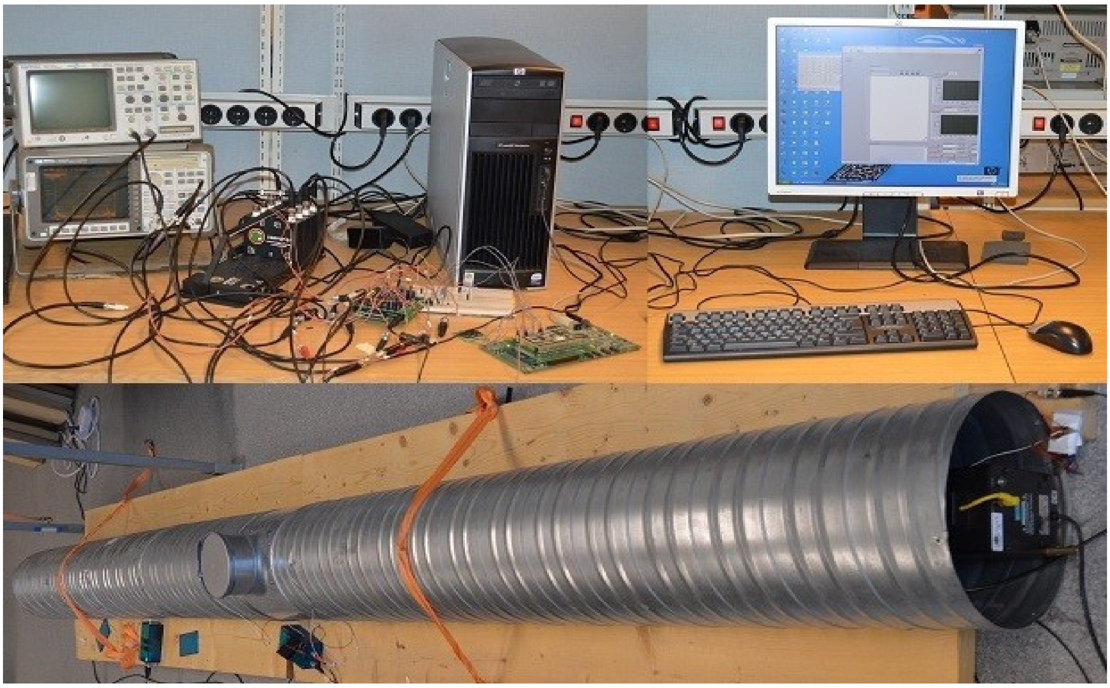


Figure 5.1: A photo of the ANC remote Lab at BTH

5.2 Remote laboratory Hardware

A brief description of the hardware used in remote laboratory for carrying out ANC experiments is given below.

5.2.1 Ventilation Duct

The duct used for remote laboratory is 4m in length with inner diameter of 315mm.

5.2.2 Microphones

In the ventilation duct of remote ANC lab, five microphones are attached at different positions. Four of them are can be used as reference sensors and one is used as error sensor. The reference microphones are used to sense the noise coming from the noise source in the duct and the error microphone is used to sense the residual noise downstream of the reference microphones in the duct. All the microphones have flat response in frequency range 20Hz-16000Hz [37].

5.2.3 Speakers

The primary and secondary noises are generated by two Fostex 6301B3 loudspeakers. The primary speaker is placed at one end of the duct and the secondary speaker is placed at the other end of the duct.

5.2.4 Signal Analyzer

To generate random noise to the primary noise speaker and analyze microphone signals during ANC experiments, a dynamic signal analyzer (Hewlett-Packard (HP) (HP35670A)) is used. The signal analyzer is connected to the server via a General Purpose Interface Bus (GPIB). The signal analyzer can be used to make estimates of measured signals in ANC experiments such as; coherence, cross-correlation, power spectral density (PSD), frequency response function of duct.

5.2.5 Digital Signal Processor (DSP)

A 32 bit floating point DSP, TI TMS320C6713 DSK is used for implementation of the adaptive controller and other signal processing tasks.

TI TMS320C6713 DSK is 32-bit floating point DSP. A 16-bit resolution is used as ADC in daughter card. The reason to use daughter card is DSP has less number of inputs and outputs and this card supports four analog inputs and four synchronized analog outputs. Anti aliasing units and programmable gain is available in daughter card[38]. A brief feature of the DSP is described in Table 5.1 below and DSP with daughter card used is shown in Figure 5.2 below respectively.

TI based DSP Specifications	
Processor	TMS320C6713 DSP
Main processing unit	TMS320C6713 DSK
Daughter Card	S-Module 16
ADC and DAC resolution	16 bit
Type of ADC	Successive Approximation Register (SAR)
Reconstruction filter	2nd order Butterworth filter (output)
Anti-aliasing filter	4th order Butterworth filter (input)
No. of I/P	4
No. of O/P	4

Table 5.1: Shows specifications of TI based DSP TMS320C6713

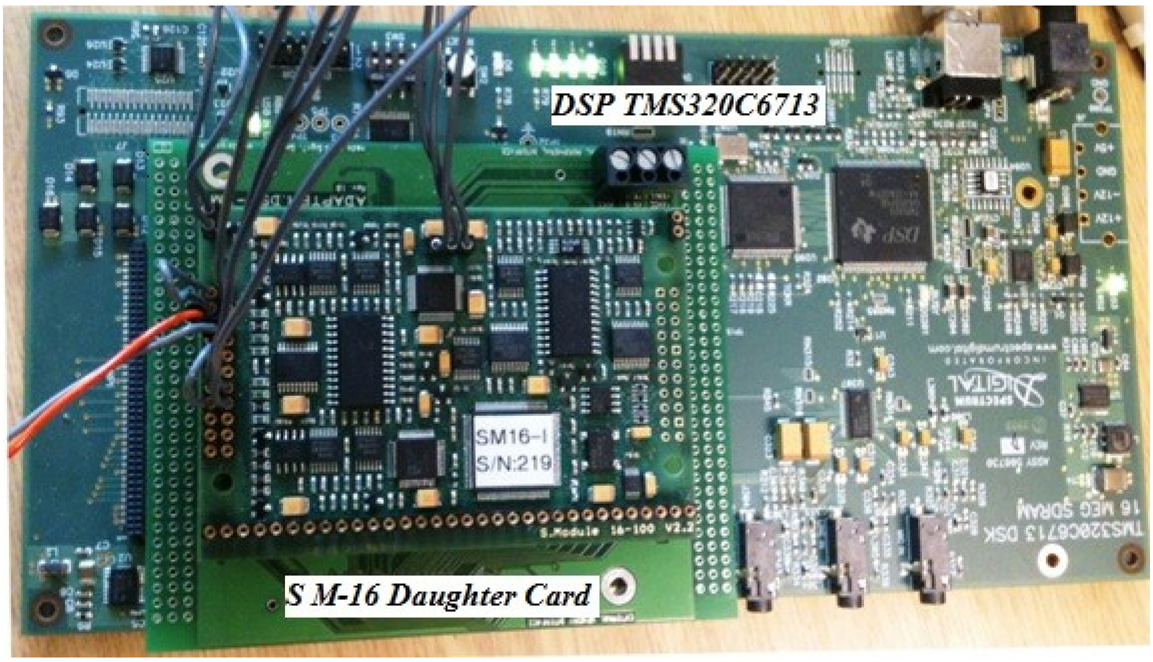


Figure 5.2: TMS320C6713 DSK with mounted S-Module 16 daughter card

5.2.6 Switching Matrix

Performing ANC experiments requires different experimental configuration of the equipment such as selecting a proper reference microphone, turning ON/OFF the primary speaker etc. In normal laboratories, cables are used for establishing the connections between speakers, microphones and DSP which has to be manually changed to establish different experimental setups. To perform the same tasks in ANC experiments remotely, a switching matrix is used which is developed under VISIR project[39]. The switching matrix is comprised of switching relays which can be controlled through USB interface. Connection between different hardware's for carrying out experiments on lab can be easily established through switching matrix without any problem and difficulty and again and again assembling the wires to these hardware's.

Switching matrix with 14 switching relays is there in remote lab which is connected with computer and can be controlled remotely without any problem. Switching matrix used in remote lab of BTH is shown in Figure 5.3 below.

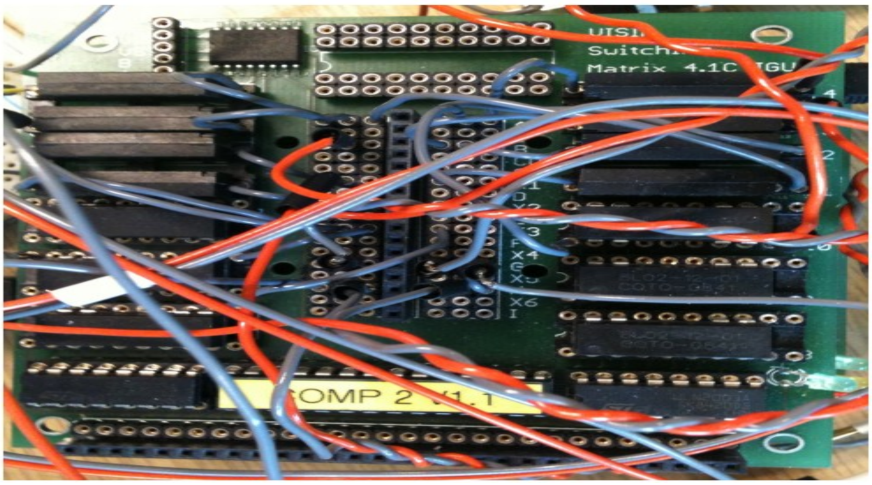


Figure 5.3: Switching Matrix used in Remote lab of BTH

5.2.7 Signal Conditioning Module

In general ANC system, signals (like microphone signals) needs to be conditioned before they are feed to the DSP via data acquisition card. Amplifiers are needed to utilize the dynamic range of the ADC and filters are used to attenuate the energy in signals above the Nyquist frequency. The filter amplifier module USBPGF-S1/L by Alligator technologies are used for this purpose [40].

The USBPGF-S1 / L are software programmable. The module is equipped with programmable amplifiers and followed by normalized Bessel filters. Table 5 below describes some specification for the signal conditioning module for remote lab.

Signal Conditioning Module	
No of I/p	2
No of O/p Channels	2
Sampling Frequency of Data acquisition card	1000 Hz Fixed
Input Coupling	AC/ DC
Input type	Single ended / Differential
Gain	1 to +1000

Table 5.2: Specification of Signal conditioning Module for the Remote ANC lab at BTH

5.3 Remote Laboratory User Interface Description

In this section a brief description of the server and the user interface which helps to access the remote laboratory and perform the experiments is presented.

5.4 Measurement and Equipment Server

Remote ANC laboratory was built on the client-server architecture. The laboratory measurement and equipment server also hosts the web server. The server provides two different web services. One for measurement and configuration of ANC system and the other is to access the Remote Development Environment (RDE) [41]. Following sections describes about these servers in details.

5.4.1 Measurement and Configuration Module

A client page was developed to access the remote ANC lab. This client page has a user interface which helps to establish hardware setup for experiments and to measure and analyze the noise. The user interface has 4 different functionalities listed as below and shown in Figure 5.4.

- Hardware Connection Setting
- Controlling Signal Conditioning Module
- Accessing Signal Analyzer
- Launching remote debug environment

5.4.1.1 Hardware Connection Setting

Web interface has a schematic diagram which shows how the system is wired for carrying out experiments. This helps to establish connections between the hardware which on the back end gets changed through switching matrix. Different ANC system configuration can be built easily by connecting or disconnecting the components i.e. amplifiers and microphones.

Only one reference microphone out of four can be used at a time. In order to filter the signal analyzer source output signal before giving to DSP or control speaker an optional band pass filter is available which can be connected.

For system identification experiments, ON-OFF button facility is available for primary speaker on web interface. Toggle buttons are available in the ANC system configuration table of the Measurement and Configuration client interface which helps to control the primary speaker and band pass filter.

5.4.1.2 Controlling Signal Conditioning Module

Low pass, Band pass filters with amplifiers are available in signal conditioning module. Parameters which can be set through the measurement and configuration client for signal conditioning are as follows and can also be seen in Figure 5.4.

- Amplifier Gain
- AC or DC coupling
- Low pass and Band pass filter Cut-off frequency

5.4.1.3 Accessing Signal Analyzer

The signal analyzer can be accessed through an Adobe Flash front end module. Most of the basic functionalities required for an ANC experiment can be implemented.

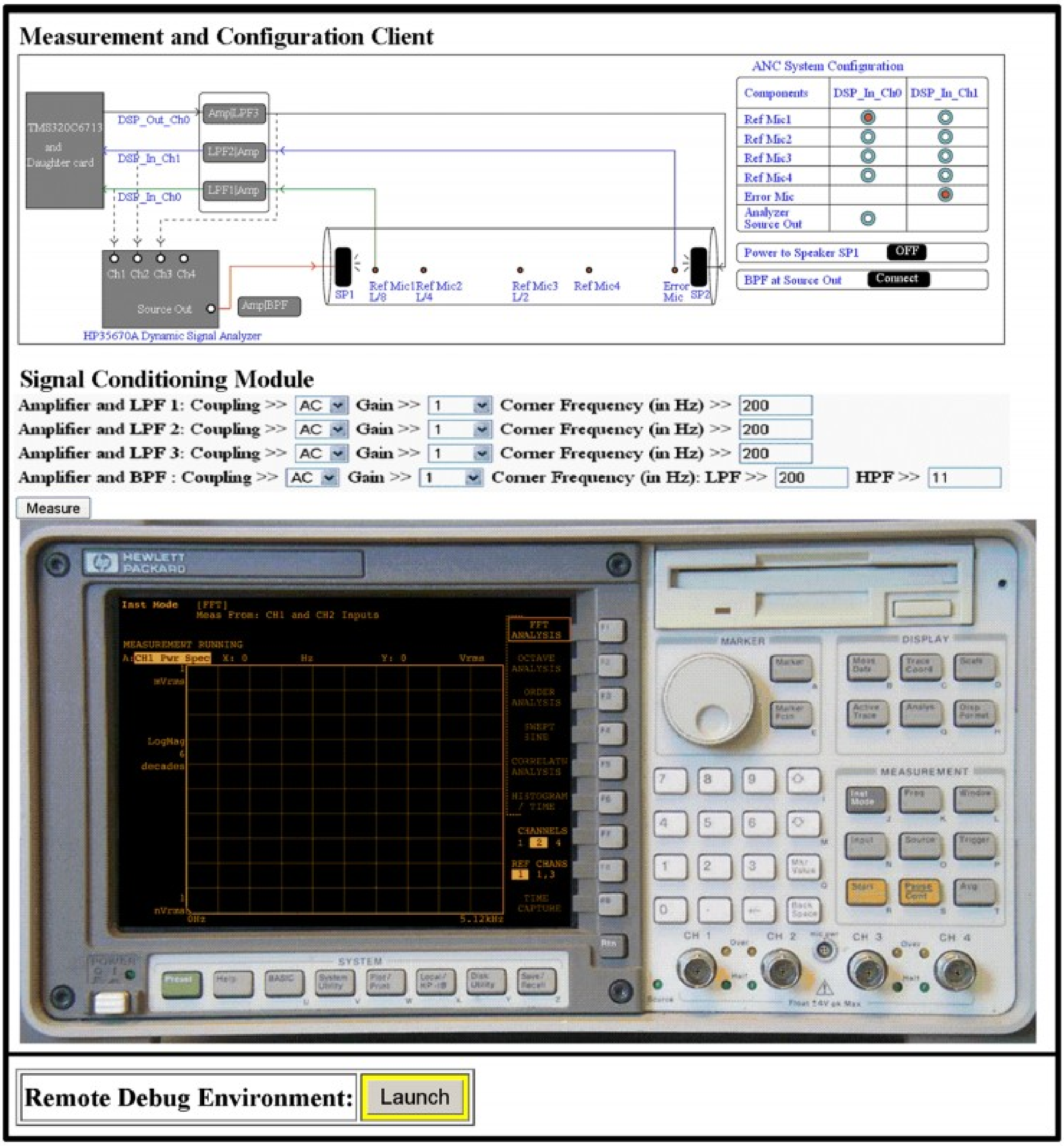


Figure 5.4: User Interface for Accessing Remote Laboratory

5.4.1.4 Launching Remote debug Environment

In order to program DSP, remote debug environment is available on Measurement and Configuration client which can be launched by pressing the “Launch” button when required hardware settings are done.

5.4.2 Remote Development Environment

In order to program the hardware i.e, DSP, an Integrated Development Environment (IDE) is available. An IDE generally helps the user to program and download an executable code to a target device [42]. It provides a connection between hardware i.e, DSP and computer through which user can program, test and debug different algorithms.

DSP used in remote prototype laboratory is TMS320C6713 which can be programmed by Code Composer Studio. Code Composer Studio is however installed on the server computer. LabView Runtime Engine is required to access the front panel which can be installed for the user's browser on their computer. A screen shot of web based remote development environment is shown in Figure 5.5. This front panel has all the necessary features which are required to program the DSP i.e. Project, load, run, Bug/Debug, Halt, Plotting etc.

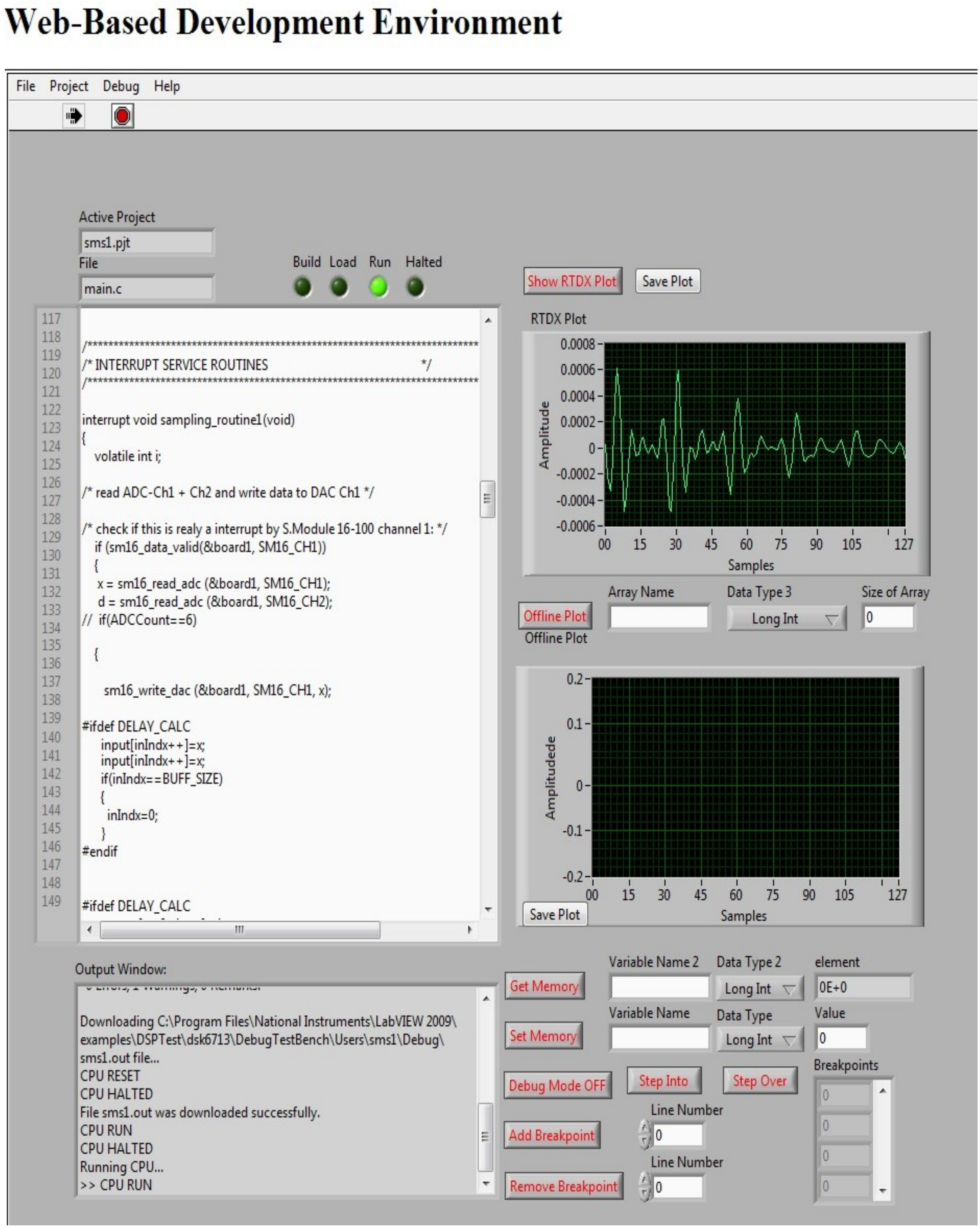


Figure 5.5: Web Based Development Environment for User at Remote end

Chapter 6 : Experiment and Results

The main focus of the thesis was to perform the ANC experiments remotely by implementing different Adaptive algorithm and to check their performance. In this chapter ANC experiments and their results are described carried out on remote laboratory. First the system identification i.e., the forward paths and the feedback paths are estimated then ANC implementation on remote lab is described followed by the results of algorithms for noise attenuation and performance comparison.

6.1 ANC Experiment

ANC experiments were carried out in the remotely controlled ANC lab with the circular ventilation duct. Microphones are available at 5 different locations inside the duct, one microphone act as an error microphone. One of the four other microphones may be used as reference microphone. Two speakers, one acting as source speaker also known as primary loudspeaker and other as control loudspeaker, are available inside the duct at respective duct ends. A broadband noise (random signal) with selectable bandwidth may be generated by the signal analyzer, for instance in range from 0-200 Hz. To carry out ANC experiment, a good estimate of forward path is necessary as described in the section below. In order to perform ANC experiments remotely, it is required both a remote development environment and a measurement and configuration client.

6.1.1Forward Path Estimation

The basic concept of forward path has been discussed in chapter 3. This path has to be estimated and subsequently used by the controller algorithm because otherwise the adaptive control algorithm will not adjust its filter coefficients towards their "optimal" solution.

Estimation of forward path is very important for ANC experiments where adaptive algorithms are to be implemented. DAC, Low pass filter, Amplifier, Anti-noise speaker, Acoustic path between secondary speaker, error microphone, Low pass filter, Amplifier, and ADC constitute the forward path in a duct. This forward path is also named as "*secondary path*" or "*control path*".

In order to estimate the forward path, a random signal (identification signal) $x(n)$ with the bandwidth from 0-200Hz was used to excite the forward path via the control speaker. The input signal, after low pass filtering and amplification, is fed to the control speaker. The signal $x(n)$ is also feed via the low pass filter and ADC is to an adaptive FIR filter producing the output $y(n)$. The signal sensed by the error microphone is named desired signal $d(n)$ which is fed into DSP after amplification and low pass filtering. The difference between the desired signal $d(n)$ and the adaptive filter output signal $y(n)$ is the error signal $e(n)$. The LMS algorithms may be implemented to estimate the forward path. The goal of adaptive algorithm is to minimize the mean square error of the error signal. A block diagram of forward path estimation is shown in Figure 6.1.

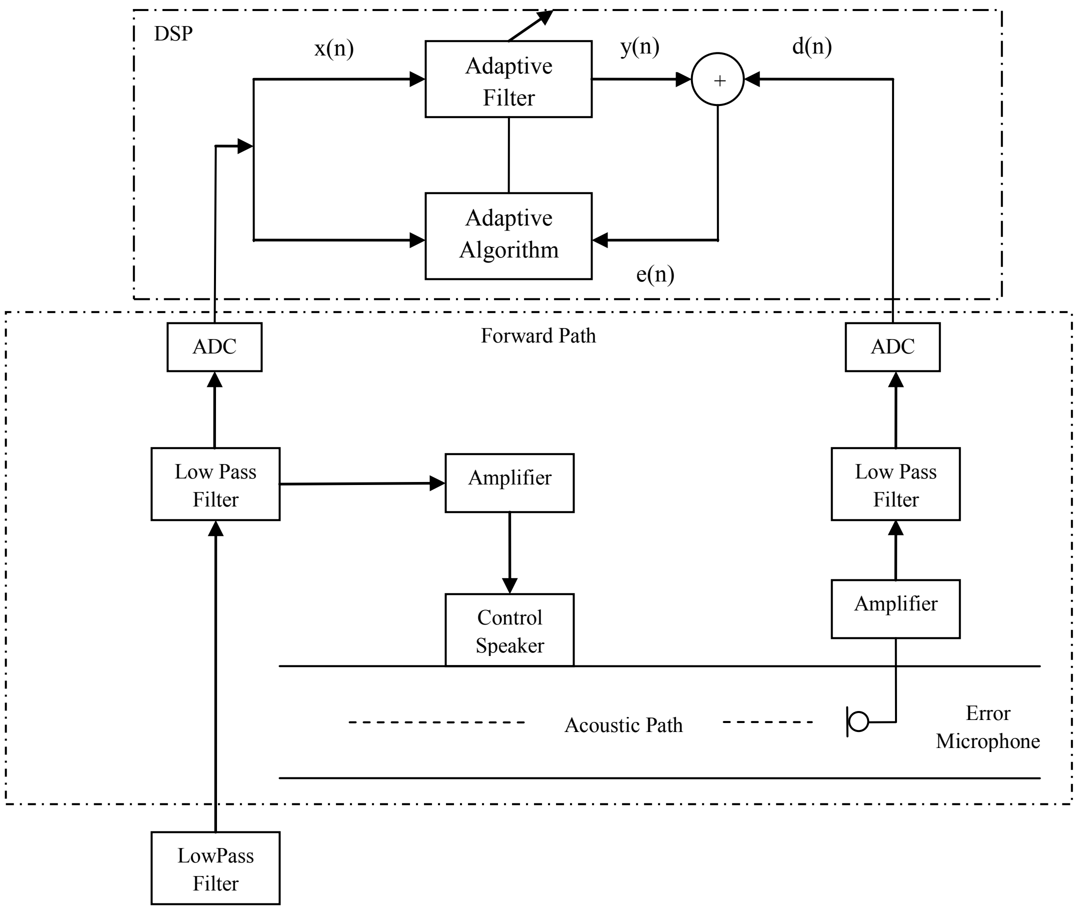


Figure 6.1: Forward path estimation Block diagram.

Remote Lab Setup for Forward Path Estimation

In order to estimate the forward path remotely using the remote ANC laboratory, *Measurement and Configuration client* is used which have been described in section 5.4.1. Forward path hardware setting on Measurement and Configuration client is shown in Figure 6.2.

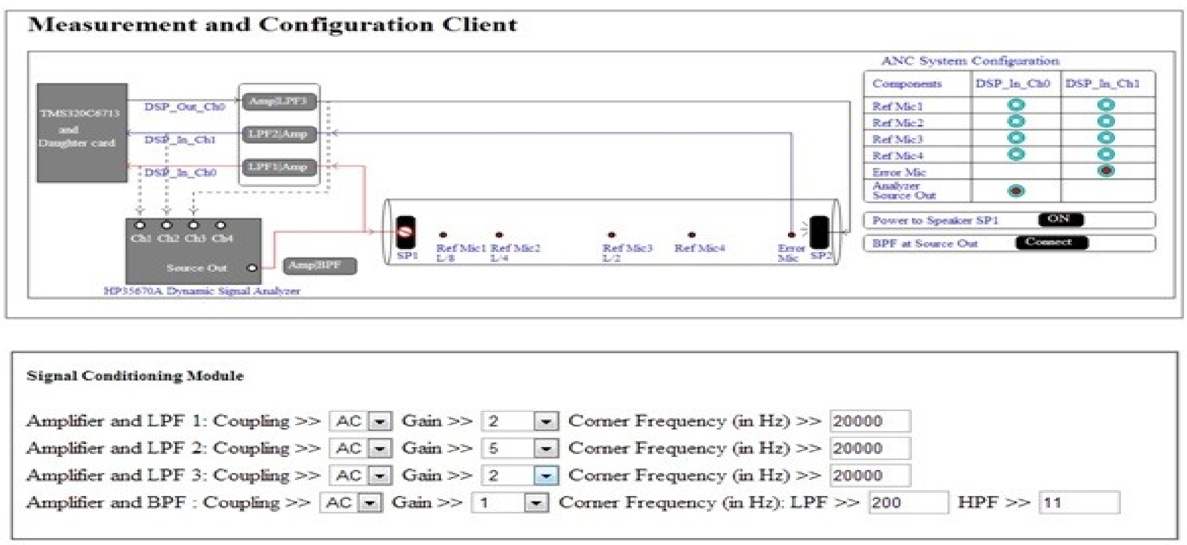
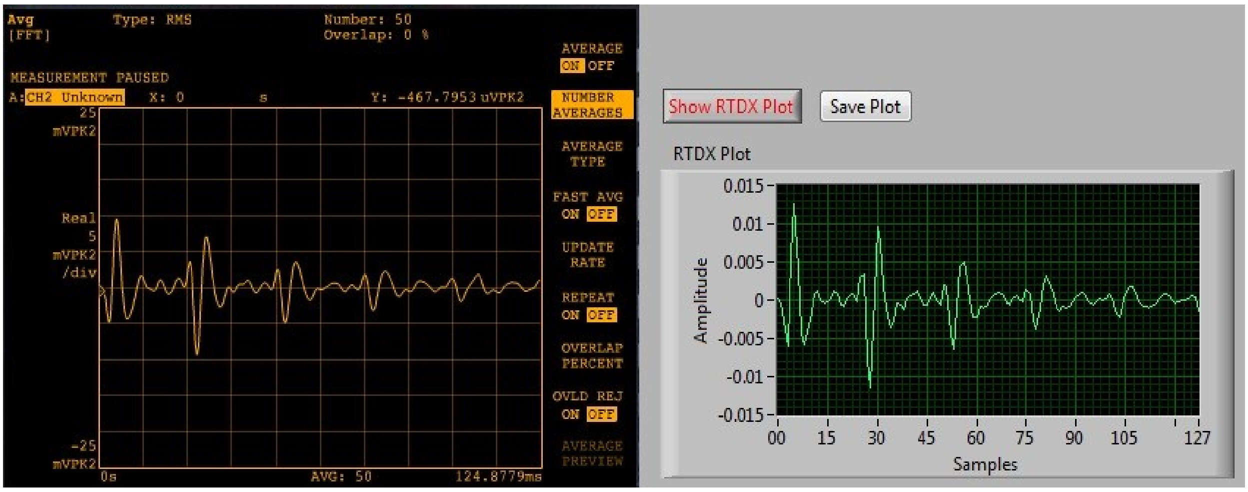


Figure 6.2: User Interface illustrating Forward path estimation setup and Signal conditioning module settings.

In forward path estimation identification signal is produced by secondary speaker $SP2$, and the primary speaker $SP1$ is muted. The power to the speaker $SP1$ can be turned off by a button available on right at the bottom of the client interface. Signal Analyzer source out is connected to amplifier and filter module 1. This Signal Analyzer source out signal is used as identification signal which is feed to $SP2$. After amplifying and filtering the signal is feed to *DSP-In-Ch0* and from *DSP-Out-Ch0* it is feed to $SP2$. The acoustic noise in the duct is sensed by the error microphone which is connected to the amplifier and filter module 2 which feed to *DSP-In-Ch1*. Hence a complete forward path setup is established and is illustrated in the above Figure 6.2.

Signal conditioning module which is also shown in Figure 6.2, is used for amplifying and low pass filtering the primary (input) signal error signal as well as control signal**.** Gain setting is also there on signal conditioning module. All these settings are required in order to use full dynamic range of the ADC:s. When all these settings have been carried out, the remote development environment can be launched to program the DSP.

The FIR filter estimate of the forward path can be compared with a corresponding cross correlation function estimated between the input signal to the forward path and the error microphone signal with the aid of the signal analyzer . A comparison of a Forward path FIR filter estimate and a corresponding cross correlation function estimated by the signal analyzer is shown in Figure 6.3. Forward path estimation can be carried out by executing the system identification algorithm (the LMS algorithm) on DSP. During the identification of the forward path with the LMS algorithm the filter coefficients may be observed via RDE and compared with corresponding cross correlation function. After cross checking, the estimated coefficients can be saved and can subsequently be used in the ANC experiment.



Signal Analyzer cross correlation

Filter coefficients in RDE

Figure 6.3: Forward Path estimates based on Signal Analyzer and the LMS algorithm

6.1.2 Feedback Path Estimation

Single channel feed forward ANC suffers from another problem. The primary microphone senses the anti-noise which is produced by the secondary or anti-noise loudspeaker in duct. This phenomenon is so-called acoustic feedback. This feedback path includes the DAC before the secondary speaker, low pass filter, amplifier, anti-noise speaker, acoustic path, reference microphone, low pass filter, amplifier, ADC. The Feedback path can also be estimated in a way similar to forward path. The desired signal is in this case sensed by the reference microphone instead of the error microphone as in the case of forward path estimation. A block diagram for feedback path estimation is shown Figure 6.4.

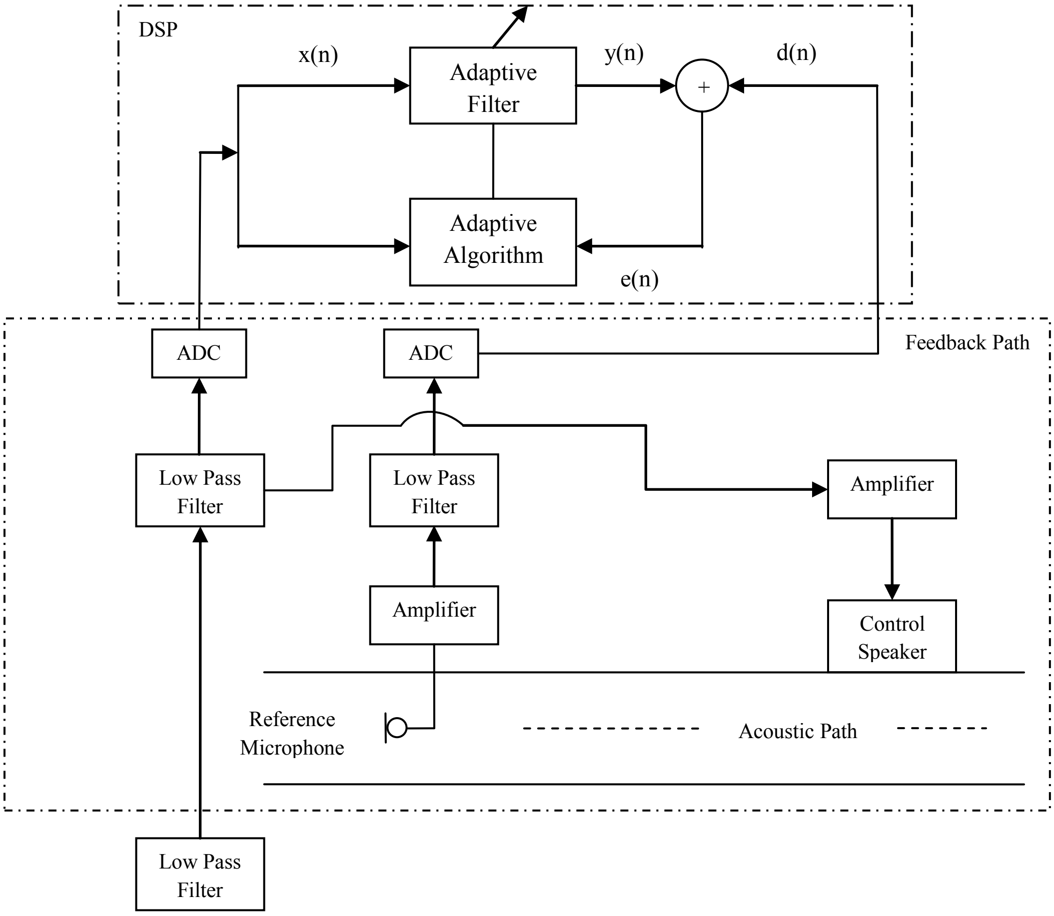


Figure 6.4 : Feedback path estimation Block diagram.

Remote Setup for Feedback Path Estimation

In establishing setup for feedback path estimation we can see that every setting for the identification is similar to that of the forward path estimation except that reference microphone Ref Mic 2 is used instead to provide error signal to DSP input channel 1 DSP-In-Ch1. A typical setting for feedback path estimation at client end is shown in Figure 6.5 below.

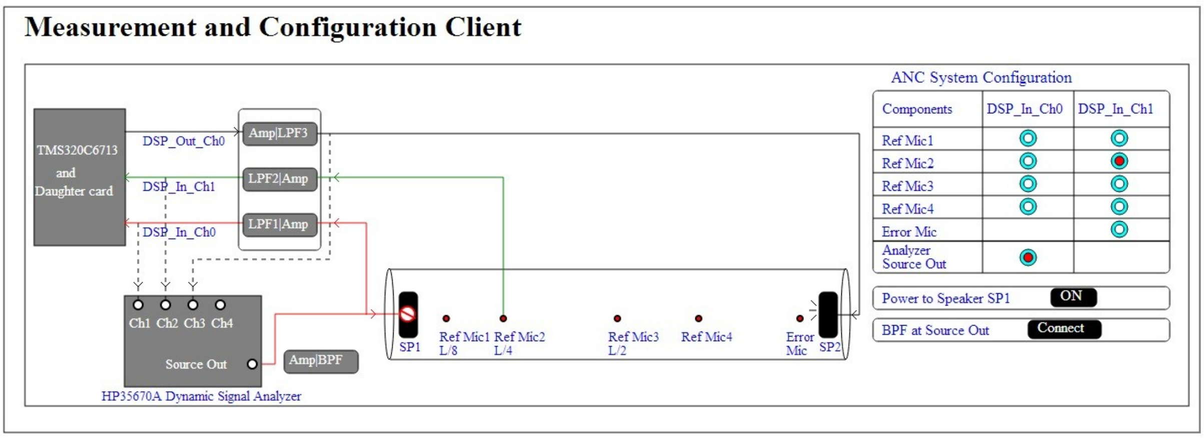
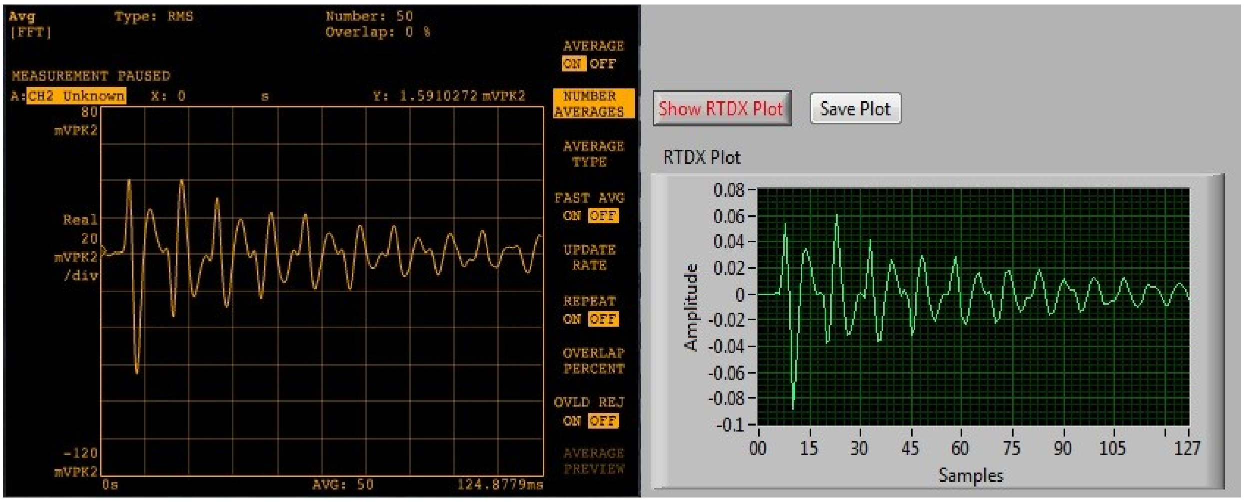


Figure 6.5: User Interface illustrating Feedback path estimation setup.

Once the setup for feedback path estimation is completed, the LMS algorithm is started on the DSP to produce a FIR filter estimate of the feedback path. After the algorithm has converged its filter coefficients can be stored and subsequently used for the ANC experiments similar to the case of forward path estimation. Also in this case; the estimated FIR filter coefficients can be compared using RDE and signal analyzer estimated cross correlation function. Estimation of feedback filter coefficient is shown in Figure 6.6 for feedback path.



Signal Analyzer cross correlation　　　　　Filter coefficients in RDE

Figure 6.6: Feedback path estimate.

6.1.3 ANC Implementation

Single channel feedforward ANC system is already discussed in detail in section 2.4.1. In this section the implementation of ANC using the remote lab is discussed. Single channel feedforward ANC system block diagram is shown in Figure 6.7.

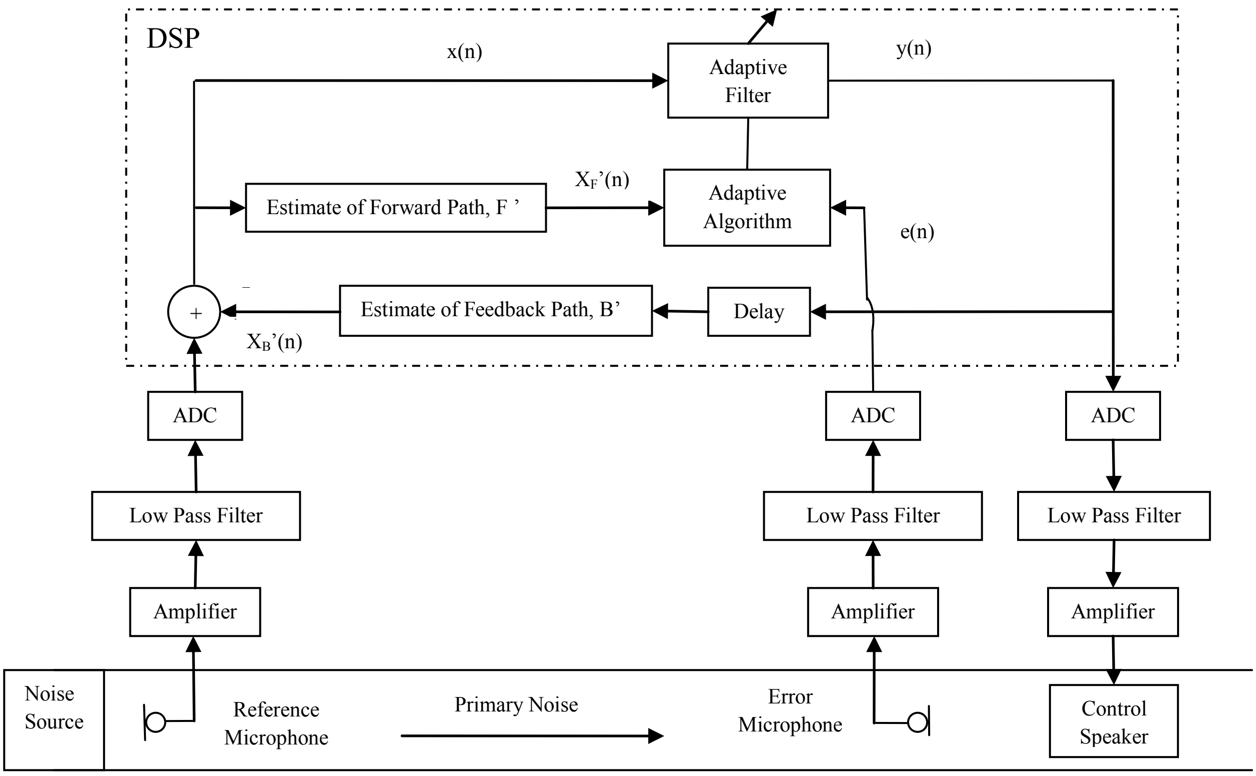


Figure 6.7: Block diagram of Single channel feedforward ANC system.

Remote Setup for Single Channel feedforward ANC

In order to perform ANC experiments, it is important that the user have necessary knowledge on the acoustic properties the duct and to investigate the applicability of ANC. To get an idea about the acoustic properties of duct, estimates of e.g. coherence and frequency response functions may be utilized. These two functions may be estimated between reference microphone signals and error microphone signal. Frequencies where the coherence assumes values close to 1 may indicate that ANC is feasible at these frequencies with acceptable performance. Estimates of a Coherence function and a frequency response function for the duct are shown in Figure 6.8 and Figure 6.9 respectively.

From the coherence plot in Figure 6.8 it can be seen that the coherence below 60 Hz is lower than 0.9. This indicates that output, the error signal, in this case cannot be completely linearly explained from the input i.e. the primary speaker signal and measurement are noisy at this frequency. The lower frequency limit of the speaker is at approx. 60 Hz. Thus, ANC should be applied above 60Hz and below 200Hz in this case for a single channel ANC system. Similarly, the frequency response function (FRF) is also used to estimate the resonance frequencies of the duct. From Figure 6.9, the FRF of the duct, it can be observed duct resonance frequencies exists at the frequencies 80, 119 and 160Hz in the frequency range 0-200Hz.

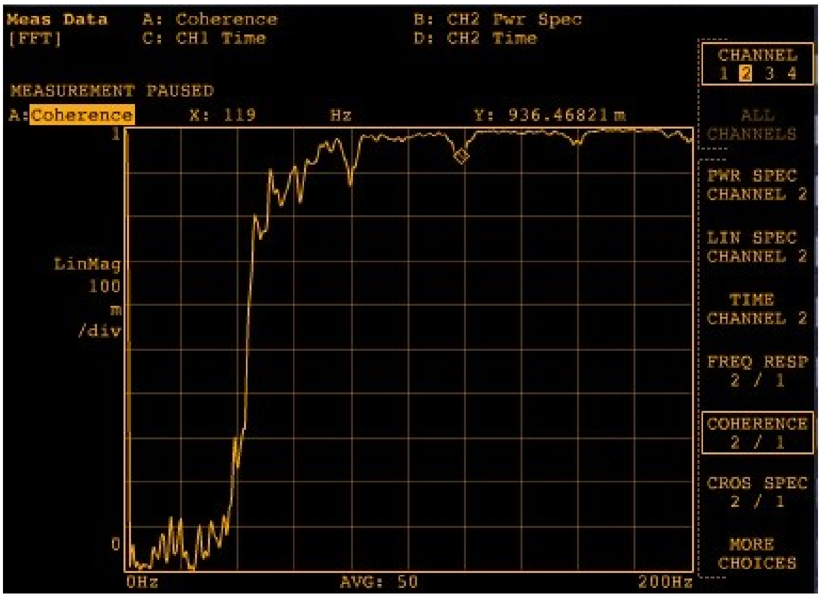


Figure 6.8: Estimate of the coherence function between error microphone and primary speaker Signals for the duct

The hardware settings for ANC experiment in the measurement and configuration client is shown in **Measurement and Configuration Client**

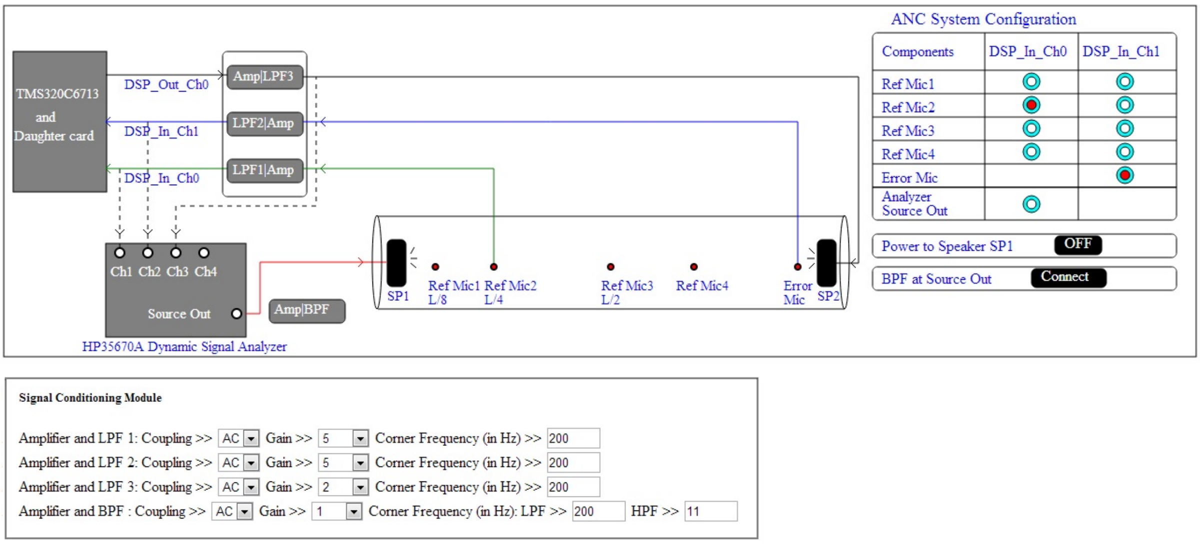


Figure **6.10**. After analyzing the coherence and frequency response functions of the duct, a random noise of suitable frequency range between 0-200Hz was used to excite the duct as primary noise signal. This random noise signal is fed to the primary speaker *SP1* after passing through amplifier and band pass filter. Reference microphone *Ref Mic. 1* receives the signal from the SP1 and after passing from amplifier and low pass filter is fed to DSP at *DSP-In-Ch0*. Control speaker *SP2* receives the signal from DSP from *DSP-Out-Ch0* after passing from amplifier and low pass filter. Error microphone then receives the signal from SP2 and is connected to *DSP-In-Ch1*. Power to speaker *SP1* can be turned on from the toggle button.

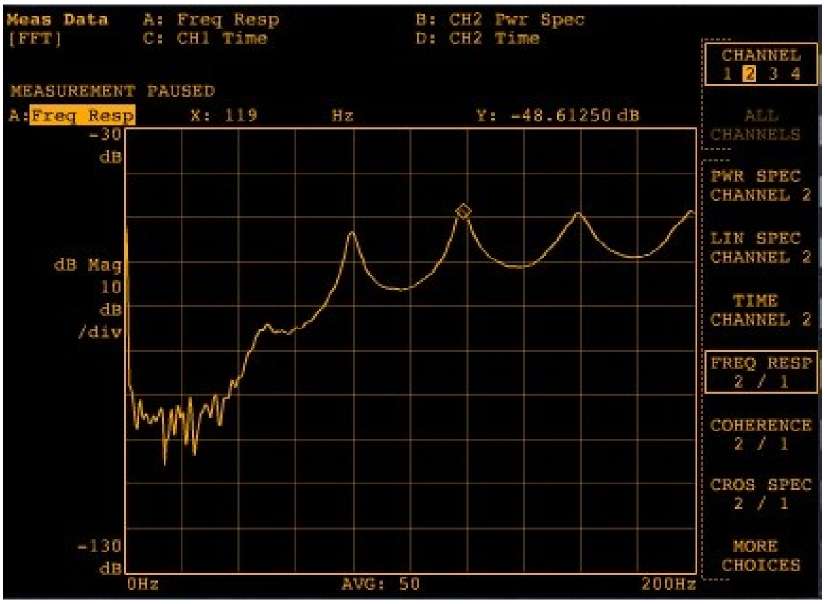


Figure 6.9: Duct frequency response function estimate between error microphone and primary speaker signals.

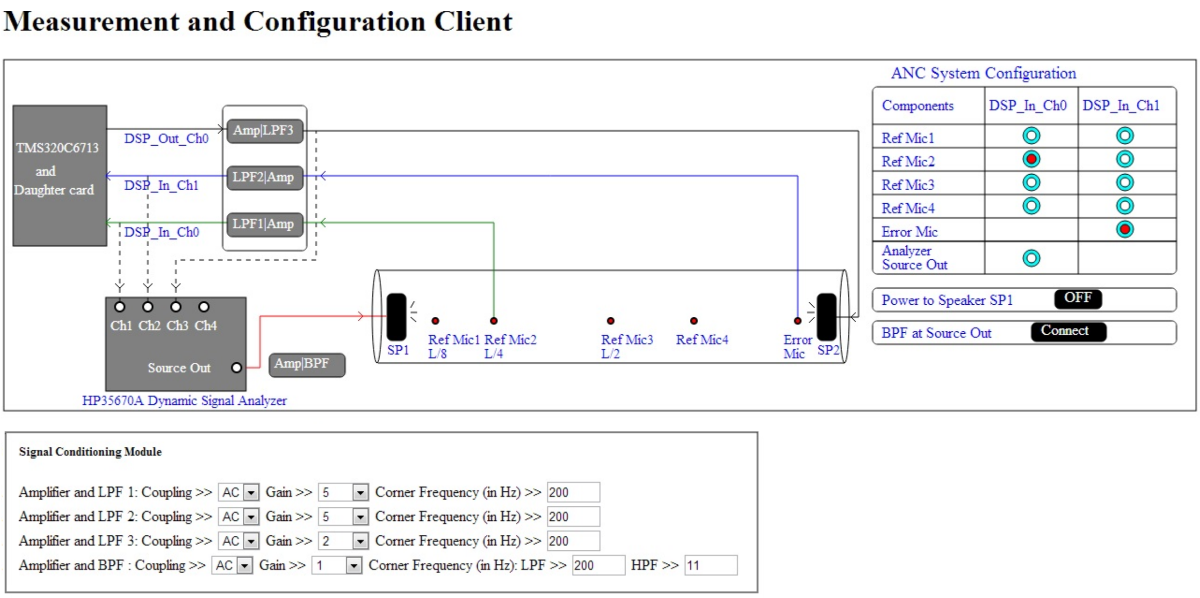


Figure 6.10: Part of the Measurement and configuration client showing the configuration of the equipment for conducting an ANC experiment.

Signal level on signal analyzer is set at 2.5V in order to utilize the full dynamic range of ADC. After completing all hardware setting, gain and frequency value, remote development environment is launched to program the DSP for carrying out ANC experiment. The same procedure was adapted for all other algorithms. The necessary configuration was done on the measurement and configuration client.

The algorithm written in C-programming language can be downloaded to the DSP via the web based development environment. After completing the necessary hardware settings remote debug environment can be launched and code can be written. In order to download and run the algorithm code in DSP three necessary steps are done i.e, build, load and run. Build option built the code and tells that weather there is any error in code or not. After the errors are cleared, code is ready to load on

DSP and can be done by the load button. After the code has been loaded to DSP run button can be pressed to start the working of the algorithm. After the successful completion of experiment, Halt button can be pressed to stop the process. Working of all these four buttons can be monitored on the LED's available on remote development environment.

Adaptive algorithms which are implemented for ANC are Leaky-LMS, Normalized- LMS, filtered U recursive LMS and RLS. These algorithms are already described in detail in chapter 3. Three peaks originate in the duct at frequencies 80, 119 and 160 Hz. The peak at 119Hz is taken into consideration for noise reduction and the highest attenuation is achieved at this frequency from all algorithms. Power spectral density of error microphone with ANC OFF and ANC ON for these algorithms is shown below in figures.

6.2 Results

In order to check the results of different adaptive algorithms on remote laboratory for attenuation of noise, experiments were conducted and power spectral density (PSD) was estimated for the error microphone signal with and without ANC. All the experiments are conducted in real time environment which means that excessive computations have to be carried out during implementation of each algorithm. The DSP of the remote lab is very helpful in this process.

6.2.1 Filtered-X Leaky-LMS (L-LMS) Algorithm Performance:

Power spectral density of the error microphone signal without Fx leaky LMS control is shown as ANC OFF and PSD with Fx leaky LMS control is shown as ANC ON in Figure 6.11. After setting the input gain, leaky factor and appropriate filter length which was taken as 256, The FxLLMS algorithm was run on the system. From the plot it can be seen that maximum noise attenuation of approx. 24.5dB was achieved at the frequency 119 Hz. Significant differences are also visible at other frequencies.

ANC-OFF **ANC-ON**

Figure 6.11: PSD of Error Microphone signal with and without FxLeaky-LMS (L-LMS) control.

6.2.2 Normalized-LMS (N-LMS) Algorithm Performance:

Power spectral density of the error microphone signal without Normalized LMS is shown as ANC OFF and PSD with N-LMS algorithm is shown as ANC ON in Figure 6.12. After setting the input gain, leaky factor and appropriate filter length which was taken as 256, N-LMS algorithm was run on the system. From the plot, it can be seen that maximum noise attenuation of approx 23.5dB was achieved at the frequency 119 Hz. Significant differences are also visible at other frequencies.

ANC-OFF **ANC-ON**

Figure 6.12: PSD of Error Microphone signal with and without Normalized-LMS Algorithm.

6.2.3 Filtered-U-Recursive LMS Algorithm Performance:

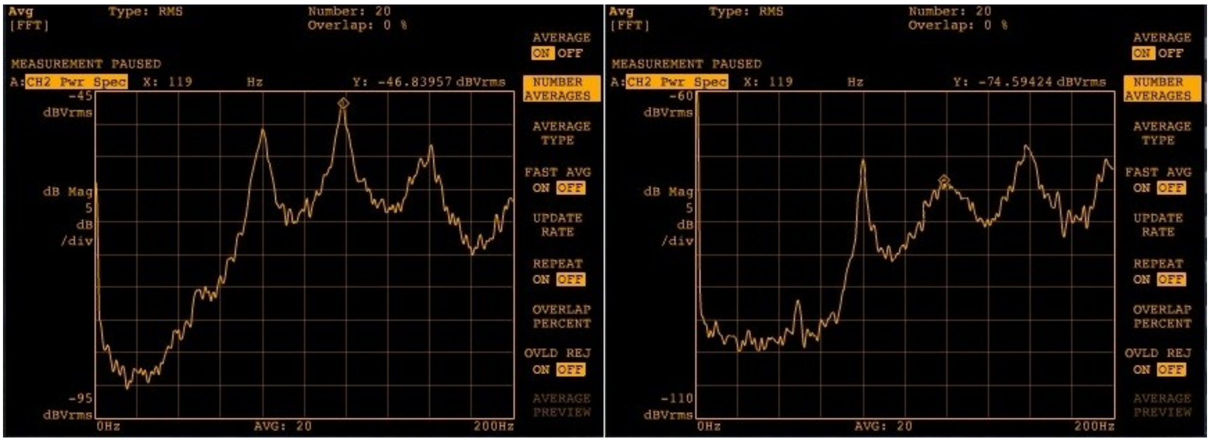
Power spectral density of the error microphone signal without Filtered U recursive LMS is shown as ANC OFF and PSD with FuRLMS algorithm is shown as ANC ON Figure 6.13. After setting the input gain, leaky factor and appropriate filter length which was taken as 128, FuRLMS algorithm was run on the system. From the plot it can be seen that maximum noise attenuation of approx 20.0dB at the frequency 119Hz. significant differences are also visible at other frequencies.

ANC-OFF **ANC-ON**

Figure 6.13: PSD of Error Microphone signal with and without filtered-u-recursive LMS (FuRLMS) control.

6.2.4 Recursive Least Square (RLS) Algorithm Performance:

Power spectral density of the error microphone signal without RLS control is shown as ANC OFF and PSD with RLS control is shown as ANC ON in **Figure 6.14** below. After setting the input gain, leaky factor and appropriate filter length which was taken as 256, The RLS algorithm was run on the system. From the plot, it can be seen that maximum noise attenuation of approx 27.7dB was achieved at the frequency 119 Hz. Significant differences are also visible at other frequencies.



ANC-OFF **ANC-ON**

Figure 6.14: PSD of Error Microphone signal with and without Recursive Least Square (RLS) control.

6.3 Performance Comparison

The performance of the noise control for the different algorithms was evaluated on the basis of noise reduction, filter order and computational resources. Filter order plays an important role in ANC. The length of the filter depends e.g, on the acoustics of the duct. The length of the controller filter should at least be selected to take into account the physical distances, the sound propagation time between the reference microphone and the control loudspeaker to enable noise attenuation [43]. To accomplish this, adaptive filter is utilized to estimate the path from the reference microphone to the secondary loudspeaker. According to real time signal processing, the processing time T_p should be less than the sampling period.[20][2]

$$T_p < T = \frac{1}{f_s} \qquad (6.1)$$

Where f_s is the sampling frequency in Hz and should be held high to satisfy Nyquist criteria and it is given as,

$$f_s \geq 2.56\, f_{max} \qquad (6.2)$$

Where f_{max} is the highest frequency of interest. The sampling resolution distance can be expressed as,

$$\Delta_{s=} {}^{c_0}/_{f_s} \qquad (6.3)$$

Where c_0 is the speed of sound in air and given as 343 m/s at 75°F. The path between the reference microphone and the control loudspeaker can be modeled by FIR filter of at least the order L as,

$$D = L\Delta_s \qquad (6.4)$$

Where D is the time it takes for sound to propagate from the reference microphone to the control loudspeaker and putting the value of Δ_s in the above equation, D becomes,

$$D = L\ {}^{c_0}/_{f_s} \qquad (6.5)$$

The length of the adaptive filter was represented as "W" for all algorithms implemented in this thesis. The forward path used in all algorithms utilizes the same filter length, a common symbol "C" to represent forward path for all algorithms. The filter "A" and the filter "B" have been used only in FuRLMS algorithm and both of them work as adaptive filters. Since FuRLMS algorithm deals with forward path and feedback path, the filter "A" models the path between the reference microphone and the control loudspeaker, plant "P" and "B" models the complete feedback path.

The Computational requirement is another important factor for evaluating the performance of control algorithms. Computational resources requirement must be considered while selecting the algorithm. Computational resources involve arithmetic operation including multiplication, addition and memory usage. These resources tell us that how many multiplication and addition are required for an algorithm and how much memory does it consume when implemented on DSP. Computational resources for the whole algorithm are described in the **Error! Reference source not found.** below. Where N is the ength of the adaptive filter and M is the length of the estimated impulse response of the secondary path.

In short, performances of algorithms are evaluated on the basis of following factors;

- Noise reduction
- Filter order
- Computation Complexity/resources Requirements, which includes
  - Multiplication
  - Addition
  - Memory Usage

Comparisons of all algorithms on above mentioned criteria are presented in **Error! Reference source ot found.** below. The Noise attenuation comparison is also shown in 3-D plot in Figure 6.15

Algorithms	**Noise Reduction in dB's**	**Filter Order**			**Computation Complexity/ Resources**		
		C	**B**, A	**W**	**Multiplication**	**Addition**	**Memory Usage**
L-LMS	**24.5**	**128**	**--**	**256**	**3N+1**	**2N+2**	**2N**
N-LMS	**23.5**	**128**	**--**	**256**	**2N+4**	**2N+3**	**2N**
F-u-RLMS	**20.0**	**128**	**128, 128**	**128**	**2N+3**	**2N+5**	**2N+M**
RLS	**27.7**	**128**	**--**	**256**	**$2N^2$+4N**	**$1.5N^2$+2.5N**	N^2+2N

Error! Reference source not found.: Performance comparisons of Algorithms in terms of omputational requirements

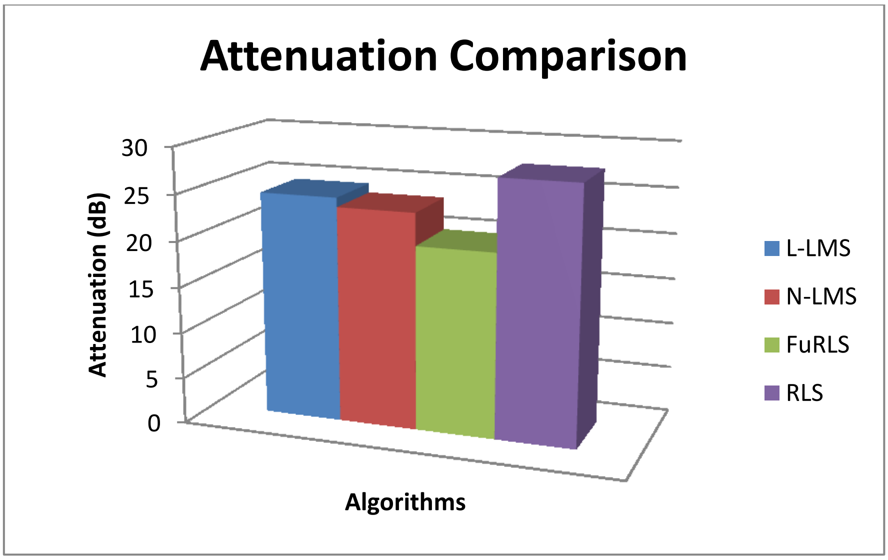


Figure 6.15**:** 3-D plot of the noise attenuation produced by the implemented algorithms

Chapter 7 : Conclusion and Future Work

7.1 Conclusion

In this thesis, performance of remote lab has been analyzed by executing control algorithms FxN-LMS, FxL-LMS, FuRLMS and FxRLS. The results of these algorithms have been observed to reduce the low frequency noise in a ventilation duct. It was observed that each of these algorithms can be utilized in a ANC system to provide substantial noise attenuation. After comparing the noise attenuation produced by the ANC system for respective algorithms, it was found out that RLS algorithm provided the highest noise attenuation as compared to the other algorithms but at the expense of a high computational COST. fXLLMS, fXNLMS and FuRLMS results approx. in a noise attenuation of 24.5dB, 23.5dB and 20.0dB respectively at a substantially lower computational cost while the FxRLS algorithm results in a noise attenuation of approx. 27.7dB.

ANC laboratory was tested using the aforementioned algorithms to find out whether it can provide an attenuation level that is comparable to similar applications. The GUI in the remote lab works in an effective way to carry out the experiment by controlling speakers and transducers according to the demand of experiment. Besides GUI, DSP working in the ANC has proved to be a remarkable platform to perform the complex experiments in a simple and efficient way. The remote laboratory can be used to perform experiments throughout the world via internet connection, thus, facilitating the students, teachers and researchers to access the lab around the globe.

7.2 Future work

Being a prototype, the laboratory's performance is satisfactory. Working on this remote laboratory helps to perform all the required steps for performing the ANC experiment. The remote lab can prove to be efficacious to perform experiments relating to courses like Digital Signal Processing, Optimal Signal Processing, Sound and Vibration Analysis, Adaptive Signal Processing etc. Students can perform time domain analysis and frequency domain analysis of a signal using this lab. Moreover, it can be used to perform experiments pertaining to sound measurement in the course of Physics being taught at high schools. It was very delightful to work without any time limitation as compared traditional laboratories where working session is time limited. The hardware deployed in the remote lab provides the facility to use it anytime and anywhere when it is needed. Working on remote lab provides the same results as achieved in a traditional laboratory and students are introduced to the real world technical challenges that are not possible with simulation tools.

Work can be done in the area of audio and video processing in remote lab to make the student's perception of being physically present more real. The hardware being used in the lab should be given a proper covering pertaining to its delicacy. The other issue is the hanging problem of the lab that can be focused on. Sometimes it gets hanged when a student is performing experiment. Due to this hanging problem the student has to wait for the lab to regain working state to perform the experiment. The filters being used in the lab need to be calibrated because sometimes they show incorrect readings when signal is being analyzed. The other thing that can be improved is to give more control of the DSP resources to the students, thus, facilitating them to perform experiments with more ease. For this purpose, the Web-Based development Environment client of the laboratory can also be also be improved. The remote laboratory can also be extended to handle multi-channel ANC system. Moreover, a sound module can be integrated to the current hardware that will help the students to listen to the duct noise when ANC is ON and when it is OFF.

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