NOISE CANCELLATION IN HEADPHONES FOR AUDIOMETRY

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Master of Technology

by

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Abstract

Audiometric tests involve presentation of sound stimuli, often using headphones, in order to quantify hearing thresholds and speech discrimination. The objective of this project is to develop a noise canceling headphone, so that headphone based audiometry can be carried out without the need for an acoustically isolated cabin. In this application, test stimuli are presented for short intervals and there is a need for high attenuation of ambient noise inside the headphone earcup. It is proposed to use feedforward adaptive noise cancellation using LMS algorithm, for reducing the background noise in the earcup without affecting the stimulus. It uses a reference microphone outside the earcup, and an error microphone inside the earcup, and the output of the adaptive filter is added as anti-noise to the stimulus input to the headphone transducer. The inter-stimulus duration is used for tuning the adaptive filter, and the filter coefficients are kept frozen during the stimulus presentation. A numerical simulation of the technique was carried out using Matlab for various noises. Real-time implementation is carried out on ADSP BF533 based DSP board with multiple input and output channels. Noise reduction of about 25 dB is achieved for tone swept over 200 Hz to1.5 kHz. Owing to limitations in sampling rate, and processing speed of the DSP board, broadband noise cancellation was not effective.

Acknowledgements

I would like to express my deep sense of gratitude to Prof. P. C. Pandey for his invaluable help and guidance during the course of the project. I am indebted to him for constantly encouraging me by providing his critics on my work. I am thankful to Mr. Bhaskar for his valuable suggestions during this project.

N. Narahari

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Contents

List of abbreviations

NCH	Noise canceling headphone
ANC	Active noise cancellation
PNA	Passive noise attenuation
LMS	Least mean square
FIR	Finite impulse response
IIR	Infinite impulse response
ADC	Analog-to-digital converter
DAC	Digital-to-analog converter
ECM	Electret condenser microphone
CPU	Central processing unit
DMA	Direct memory access
EMIF	External memory interface
PC	Personal computer
HL	Hearing level
I/O	Input-output
SPI	Serial peripheral interface
SPORT	Serial port
TDM	Time division multiplexing
I^2S	Inter-IC sound

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Chapter 1

INTRODUCTION

1.1 Project overview

Earcups of conventional headphones block background noise to a certain degree by passive noise attenuation (PNA). In noise canceling headphone (NCH), the background noise is further reduced by active noise cancellation (ANC), by “actively" introducing a secondary, sound called anti-noise, which reduces the unwanted noise by destructive interference. NCHs are particularly useful for workers, working near heavy machinery and engines. The noise is selectively eliminated thus enabling the reception of desired sounds, such as speech and warning signals. Cabin noise in small aircrafts is a combination of noise from a variety of sources, the major ones being the engine, wind, and propeller. Aircraft pilots routinely wear noise attenuating headsets, which usually employ PNA in the form of an annular cushion carried on the rim of each earcup. NCHs have proved to be effective in attenuating noise to a further level, enabling the pilots to hear warnings and instructions effectively. In military applications, the soldiers in noisy environments use NCHs for hearing instructions through mobile sets. NCHs are also being used for music listening along with audio gadgets. NCH for audiometry is one of the noise canceling applications, used to conduct audiometric tests such as to find hearing thresholds and speech discrimination along with an audiometer. To conduct audiometric tests, an acoustically isolated room is needed which is expensive and it is difficult to set-up and maintain such rooms in hospitals especially where space and cost is critical**.** Conventional NCHs reduce the noise effectively but the test signal may get affected.

1.2 Project objective

The objective of this project is to study and develop a NCH for audiometry which will be interfaced with an audiometer. This NCH can be used along with audiometer to conduct audiometric tests without having an acoustically isolated environment. The basic requirement will be that noise reduction should match that of an acoustically isolated room over the audiometric frequency range of 125 Hz to 8 kHz and the test signals should not be affected. Assuming that, most patients undergoing hearing tests, may have hearing thresholds of 40 dB or higher, the ambient noise leakage into the headphone must be lower than 30 dB. Assuming the ambient noise present in the clinic to not exceed 80 dB, and a passive attenuation of at least 10 dB, the active noise cancellation should provide an attenuation of 40 dB or better. Actually as the ambient noise is broadband or speech shaped and generally not concentrated at single frequency, attenuation less than 40 dB also be acceptable.

1.3 Dissertation outline

Next chapter gives a review of noise canceling techniques, which includes passive noise attenuation and active noise cancellation. In active noise cancellation, analog feedback, adaptive feedback, and adaptive feedfoward methods are discussed. Brief information on audiometry and noise canceling headphones for audimetry using the ANC techniques are also presented.

In Chapter 3, a proposed method to implement NCH is discussed. Results are included for various noises, simulated using Matlab. The performance of the algorithm for different noises is discussed. Characterization of a headphone, simulated results for tone noise, and frequency response of microphones are also included.

Real time implementation of active noise cancellation using a digital signal processor (DSP) is presented in Chapter 4. A choice of processors, architecture overview and simulated results are also discussed in this chapter. In Chapter 5, a summary of the work carried out, and suggestions for the future work of the project are presented.

Chapter 2

NOISE CANCELING TECHNIQUES

Noise cancellation techniques are broadly classified into passive noise attenuation (PNA) and active noise cancellation (ANC). PNA depends on acoustic absorbers such as earcups of headphones. ANC works on the principle of destructive interference between unwanted noise and secondary out of phase noise known as anti-noise. NCHs rely on the passive acoustic isolation of headphones as well as active noise reduction to provide broadband noise reduction as shown in Figure 2.1. At high frequencies PNA is significant and ANC is predominant at low frequencies [1].

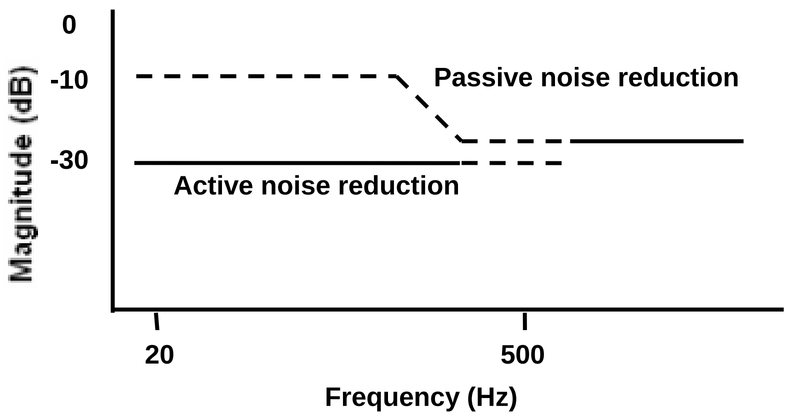


Figure 2.1: Operational range of active and passive noise reduction, from [1].

A noise reducing headphone should have good attenuation at lower as well as higher frequencies. Noise cancellation for entire broadband is generally not practical using PNA because it is difficult to stop low frequency sounds transmitting from outside to inside the earcup unless the intervening barrier is very heavy. Broadband noise cancellation is achieved using a combination of PNA and ANC. Where passive attenuation begins to weaken at low frequencies, ANC takes over.

2.1 Passive noise attenuation

Passive noise absorbers block some degree of noise at frequencies only above 500 Hz as shown in Figure 2.1. This is because at low frequencies the acoustic wavelengths become large compared to thickness of a typical earcup of a headphone. It is understood from the literature that a well-designed closed-ear headphones can passively block high frequency noise down to about 500 Hz by nearly 30 dB [1]. The degree of noise reduction depends on the quality of the seal on the ear cushion, the construction of the earcups and the sound-absorbent materials used in the earcups

Passive attenuation in an earcup is a function of the front and rear cavity volumes and the driver compliance below the free air resonance. Increased driver compliance can compensate the reduction of passive attenuation if the front cavity is kept small for maximum efficiency [2].

An optimized headphone construction, described by Sapiejewski [3] is shown in Figure 2.2. An ideal cavity is characterized by rigid walls and constant pressure amplitude for wavelengths much larger than the distance across the cavity. The ear cushions should be compliant enough to affect a seal that prevents leaks and has sufficient high density and flow resistance to create such an "ideal" cavity. The diaphragm should be small - less than 1/3 wavelength of the highest audio frequency to be reproduced. There is a trade off between a small front cavity which will minimize the sound pressure required to cancel low frequencies and a large front cavity which offers superior passive attenuation.

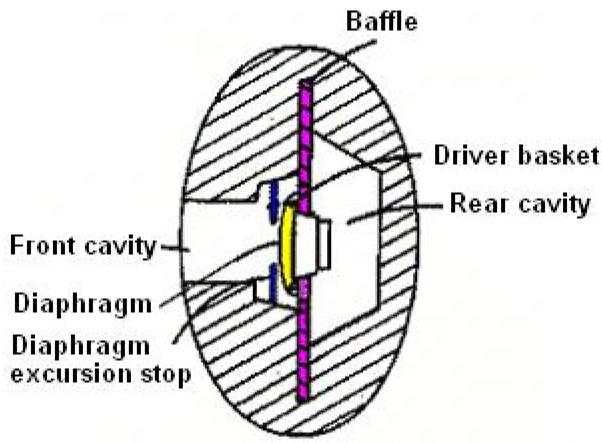


Figure 2.2: High compliance drive for headphone ANC, as described in [3].

The disadvantage with high compliance driver is its sensitivity to overpressure. The pressure waves due to the movement can pull the voice coil outside the gap or distort the diaphragm when the user moves or takes off the headphones. A tiny barrier or mesh in front of the diaphragm can prevent its distortion. This diaphragm can also be minimized by indenting it so that it quickly recovers when distorted [3].

2.2 Active noise cancellation

ANC is based on either feedforward control or feedback control. In feedforward control, a reference input coherent with the noise is sensed before it propagates past the secondary source. In feedback control, the active noise controller attempts to cancel the noise without the benefit of an “upstream” reference input [4]. Structures for feedforward ANC are classified into (1) broadband adaptive feedforward control with a control field reference sensor, (2) narrowband adaptive feedforward control with a reference sensor that is not influenced by control field.

2.2.1 Analog feedback ANC

Feedback systems use only an error sensor and create a quite region using feedback control of a secondary source located in the environs of the error sensor. Feedback ANC is required for applications in which it is not possible to sense a reference signal coherent with the disturbance. The block diagram of analog feedback ANC described in [5] is shown in Figure 2.3. In this method, W represents the analog controller and H is the transfer function of the headphone. The external noise is $d(t)$, error signal is $e(t)$, and audio signal is $s(t)$. The output is given as

$$Y = D\,(1+WH)^{-1} + S\,W\,H\,(1+WH)^{-1} \tag{2.1}$$

In the absence of audio signal the error can be expressed as

$$E = D\,(1+WH)^{-1} \tag{2.2}$$

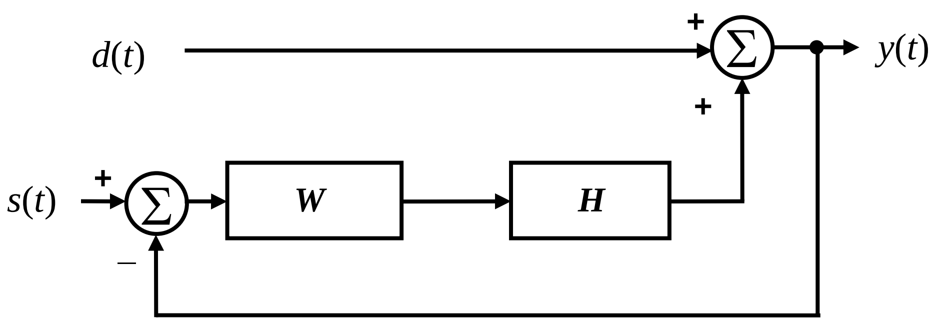


Figure 2.3: Block diagram of analog feedback ANC system, from [5].

We see that the error is eliminated and the signal remain undistorted if WH >>1. At lower frequencies the gain of the controller is high to reduce the noise effectively. The phase shift in the transfer function increases with frequency. As it approaches 180°, the desired negative feedback becomes positive feedback, and the control system becomes unstable. Instead of reducing, the controller tends to amplify the noise. At higher frequencies the gain is very low to maintain stability of the feedback system; hence noise reduction is not significant. There is a trade-off between the performance and stability. It is observed from the literature that analog feedback ANC is effectively used to control the noise at lower frequencies. An adaptive feedback system can be used to reduce the noise over wide frequency range [5].

An NCH developed by Bose Corporation was proposed by Bose, Waylad, and Mass [6]. This headphone was built based on a feedback control as shown in Figure 2.4. In this approach a microphone is kept in the earcup adjacent to the diaphragm, provides a feedback signal. The microphone output signal is pre-amplified and combined with the audio signal to be reproduced, and then sent to the speaker through compensation circuit and power amplifier. The signal combiner provides the combined signal to the compensator which limits the level of high level signals. This in turn provides the compressed signal to compensator, ensures the open loop gain meets Nyquist stability criteria, so that the system will not oscillate in closed loop. Power amplifier energizes the headphone driver to produce an acoustic signal in cavity that gets combined with the outside background noise that enters the cavity through the earcup of the headphone. Microphone pre-amplifier amplifies the transduced signal and delivers it to signal combiner.

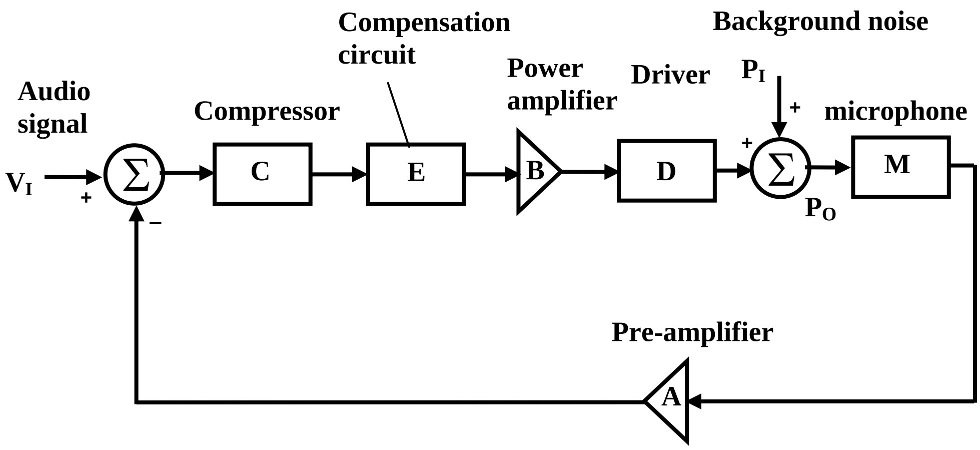


Figure 2.4: Block diagram of feedback control ANC, from [6].

The amount of noise reduction where P_I corresponds to acoustical noise input is

$$N_R = 1 + CBDEMA \tag{2.3}$$

Where C,B,D,E,M, and A are the gain of the respective blocks in Figure 2.4.The passive and active noise characteristics of the Bose noise canceling headphone are discussed in the next chapter.

2.2.2 Adaptive feedback ANC

In feedback systems unlike feedforward method, primary noise is not available during ANC, since noise is cancelled by the anti-noise. The reference signal is regenerated from error and adaptive filter output. The adaptive filter is a digital filter in which the coefficients of the filter are updated using least mean square (LMS) algorithm to minimize the error as shown in Figure 2.5. The output of the error microphone is subtracted from the signal input and then given as input to adaptive filter. The error microphone output is also given to the LMS algorithm.

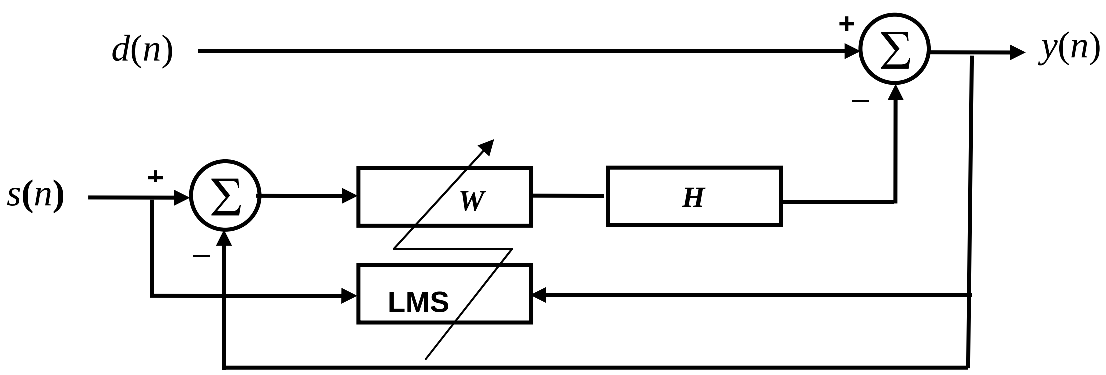


Figure.2.5: Block diagram of an adaptive feedback ANC system, modified, from [4].

Use of adaptive feedback and analog feedback method for a noise cancellation headset system has been outlined by McIntosh [7] in a US patent. This is a two channel system, consisting DSP, A/D, and D/A converters. Scheme for a single channel is shown in Figure 2.6. The D/A output which consists of the noise cancellation signal and the audio signal is given to a speaker mounted within the earcup. A microphone within the earcup senses the error signal which is given to an analog filter to generate an analog noise cancellation signal. The analog error signal is given to an A/D converter. A high pass and low pass filters are provided in the audio signal for each channel. The low frequency component of the audio signal is given to DSP via A/D where it is subtracted from the microphone output otherwise acoustic signal cancels the desired audio signal. The high frequency signal is directly given to summing amplifier. The DSP takes the digital error signal and generates a digital tonal noise cancellation signal using an adaptive digital feedback filter. A D/A then converts the digital tonal noise cancellation signal to an analog tonal noise cancellation signal so that it can be combined with the analog broadband noise cancellation signal. The resultant composite cancellation signal is provided to the speakers in the earcups to cancel noise within the earcups. The broadband analog cancellation is effective to reduce overall noise within the earcup.

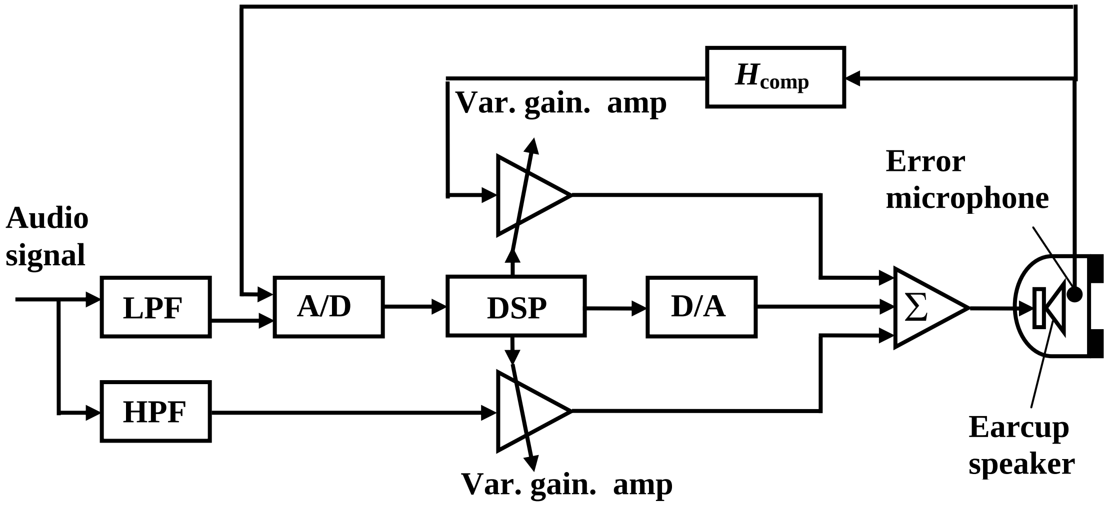


Figure.2.6: Schematic diagram of noise cancellation circuit for a single channel headset system, as proposed in [7].

The DSP not only provides active control of the analog cancellation loop gain to maximize the effectiveness of the broadband analog cancellation but also uses the adaptive feedback filter/algorithm to substantially reduce at least the loudest tonal noises penetrating through the earcup. The adaptive digital feedback filter may be a regenerative feedback filtered-X least-means-square (FXLMS) algorithm [4]. This system can be viewed as an “adaptive feedforward system that, in effect, synthesizes or regenerates its own reference signal based only on the adaptive filter output and the error signal” [7].

2.2.3 Adaptive feedforward ANC

Feedforward ANC is a method of canceling noise by using a reference input coherent with the noise. It is generally more robust than feedback ANC, particularly when the feedforward system has a reference input isolated from the secondary anti-noise source [4]. The block diagram of feedforward ANC without an audio signal input is as shown in Figure 2.7. In this method, ANC systems have a reference sensor, and an error sensor. The undesired noise, $d_1(n)$, from a primary noise source is measured by reference input microphone. This primary noise is transmitted through headphone earcup with transfer function H_1 and presented as inside noise, $d_2(n)$. An adaptive filter, filters the primary noise and gives output $d_4(n)$, used to drive a secondary sound source such as speaker, to cancel the noise inside the headphone. The input signal from the reference sensor must be well correlated with the noise from primary source. “In systems that control broadband noise, the reference signal provides advance information about the primary noise before it reaches the canceling speaker, which is necessary requirement for a causal controller”, [4]. The residual noise, $x_1(n)$, is detected by an error microphone and is used to update the coefficients of the adaptive filter to minimize the error. Feedforward doesn’t mean that it excludes feedback. It actually uses the feedback to update the coefficients of the adaptive filter.

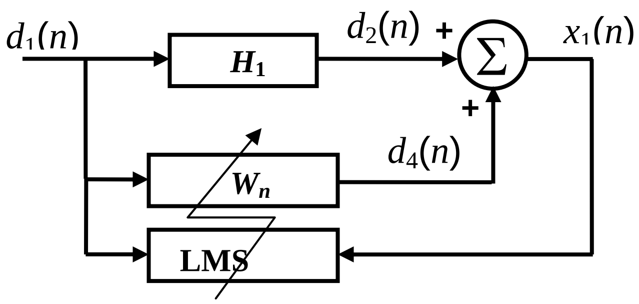


Figure 2.7: Block diagram of a feedforward ANC system without audio signal input, from [4].

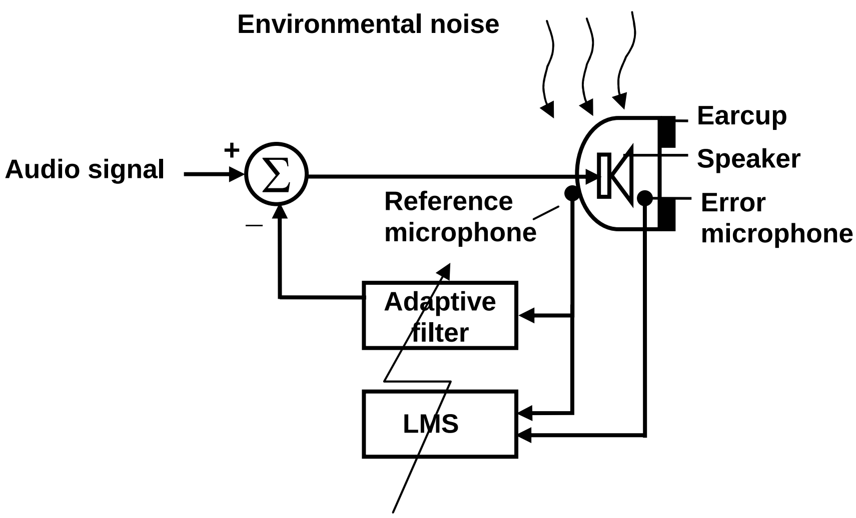


Figure 2.8: Adaptive noise canceling headphone, as described in [1].

A NCH using feedforward adaptive noise reduction, as shown in Figure 2.8, was proposed by Chu Moy [1]. In this approach, the noise signal is sensed by a reference microphone at the top of the headphone. The reference signal is passed through an adaptive filter which models the headphone system and produces anti-noise. This is added to the desired audio signal and then sent to the headphone transducer. An error microphone inside the earcup measures the resulting sound and generates an error signal to minimize the error for more accurate anti-noise [1].

2.3 Audiometry: An application of NCH

Audiometry is a technique to identify and quantitatively determine the degree of hearing loss of a person by measuring his or her hearing sensitivity, so that suitable medical treatment or one of the appropriate hearing aids and assistive devices can be prescribed. In audiological investigations, the hearing sensitivity is tested for pure tones, speech or other sound stimuli. The electronic instrument used for measuring the hearing threshold level is called an audiometer. In pure tone audiometry, test tones of different frequencies and levels are presented and hearing thresholds are determined on the basis of patient’s response. The instrument consists of an audio oscillator basically for generating pure tone sounds of various frequencies, usually of 125, 250, 500, 1000, 1500, 2000, 3000, 4000, 6000, 8000 Hz. The presentation level can be varied over 0-100 in a 110 HL, in steps of 5 dB [8].

To diagnose the hearing defects, a test stimulus (tone of specific frequency or speech, etc) is presented to the human ear as shown in Figure 2.9, usually for a short duration with long pauses. The level is varied and threshold is determined as the lowest level reported as audible by the patient. Audiometric tests are usually conducted in an acoustically isolated room, where the level of noise in the test environment is lower than the test stimulus, so that it may not cause a threshold shift. In normal rooms, when a test tone is sent to the headphone transducer, gets combined with background noise transmitted through the earcup and the measurement of hearing thresholds gets affected. Conventional headphones reduce the noise up to some extent but the reduction of noise is not uniform over the audible frequency range. Hence these headphones cannot be used without acoustically isolated rooms. NCH reduces the noise to a further level using noise canceling techniques such as PNA and ANC. The level of attenuation achieved using an NCH may often be adequate to use it along with an audiometer to conduct audiometric tests even in normal rooms. However, most of the existing NCHs affect the signal to some extent.

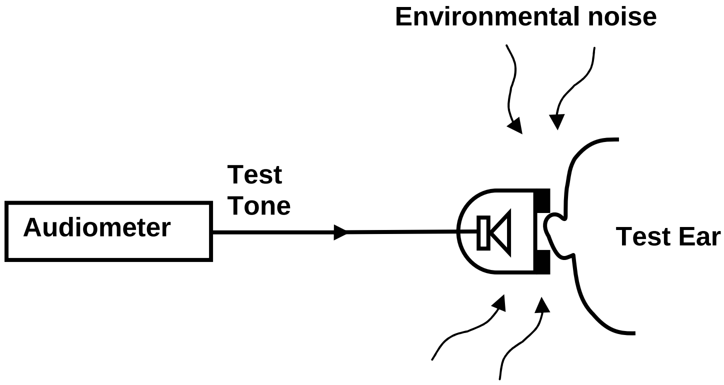


Figure 2.9: Tone audiometry test.

2.4 Audiometric NCH techniques

Audiometric testing requires very low ambient noise levels in order to determine a subject's threshold level. This has been achieved in two ways in the past, without degrading the test stimuli: (a) by installing soundproof chambers, or (b) by adding more passive attenuation materials to the existing headphones. The former is difficult to set-up and maintain, and the later is generally not preferred because of improper fitting of such headphones to the wearer's head. Recently, insert earphones were introduced as an alternative to acoustically isolated cabins. Although they are capable of providing accurate test signals, they are not effective against low frequency noise. The technology of ANC was incorporated into audiometry using one of the analog feedback, adaptive feedforward, and adaptive feedback methods to reduce ambient noise.

Vaudrey and Saunders have outlined an NCH for audiometry in a US patent [9], using feedback and/or feedforward ANC techniques without distorting test stimulus, as shown in Figure 2.10. In this method, a microphone kept inside the earcup, senses the combined test signal and transmitted noise as error signal, which is amplified and given to speaker along with the test signal through an amplifier. Since the error microphone picks up both test signal and residual noise, the input test signal needs compensation, which is carried out through a filter P that minimizes the control loop's effects on the test stimulus. The error signal $e(t)$, can be represented as,

$$E = SPG/(1+GH) + D/(1+GH) \tag{2.4}$$

In order to reproduce the test stimulus without any distortion, we need to have

$$P = (1 + GH)/G \tag{2.5}$$

In Figure 2.10, H^1 represents controller that can be implemented using analog electronics or digital software. Feedback method using digital control is shown in Figure 2.11, where H^1, P can be implemented using FIR or IIR filters. Digital design offers superior features over analog design despite the additional hardware requirement.

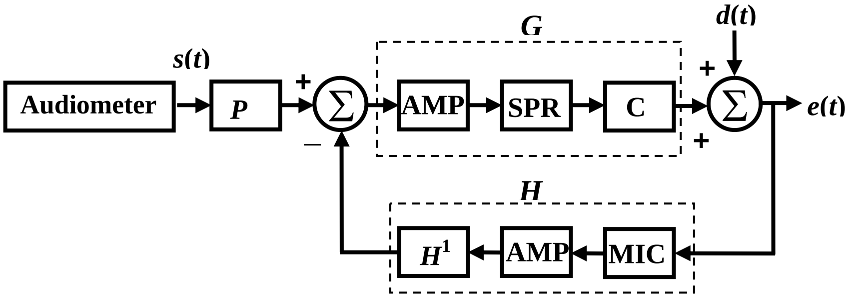


Figure 2.10: Audiometric NCH using analog feedback method, from [9].

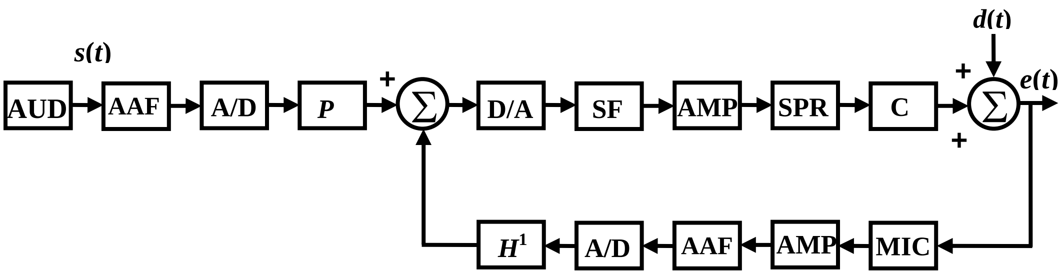


Figure 2.11: Audiometric NCH using digital feedback method, from [9].

ANC using feedforward method, by the same authors [9] is shown in Figure 2.12. Because of the complexity of the algorithms, the filters can be implemented using digital software. The implementation is carried out using adaptive algorithms such as LMS and FXLMS discussed in the next section. The noise outside and inside the earcup are obtained using a reference microphone and error microphone respectively. The two signal are given to inputs of adaptive controller consists of an FIR filter and an algorithm to update the filter coefficients. Antinoise generated by the adaptive filter is added to the test stimuli from the audiometer and sent to the earcup speaker. Inside the earcup, due to acoustical superposition of transmitted noise and antinoise, the test signal is reproduced. In Figure 2.12, G^1 represents the secondary path compensating filter. This method is effective for reducing tonal noises as the correlation between reference and error signal is highest for sinusoidal waveforms [9].

The feedback method has limited levels of performance over a pre-specified bandwidth due to stability constraints and the performance of feedforward method is limited by the correlation between reference and error signals. Hence it is concluded by the author that the feedback control tends to perform better for broadband noise fields while feedforward control performs better for tonal noise fields. A combination of feedback and feedforward control can effectively reduce ambient noise. The combination of feedback and feedforward approach is shown in Figure 2.13. The feedback control shown is analog implementation; the controller can also be entirely digital. As mentioned earlier, the feedback control affects the test stimulus and the compensation has to be made through a filter *P*. Since the feedforward control does not affect the test signal, the error signal is same as shown in Equation 2.4.

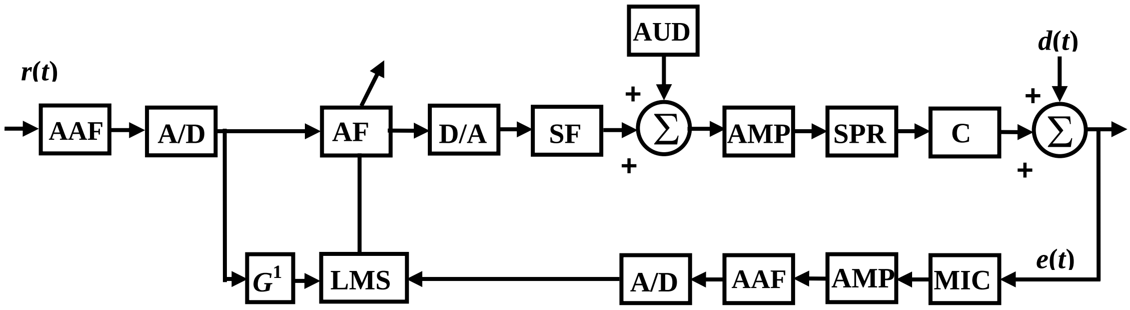


Figure 2.12: Audiometric NCH using feedforward method, from [9].

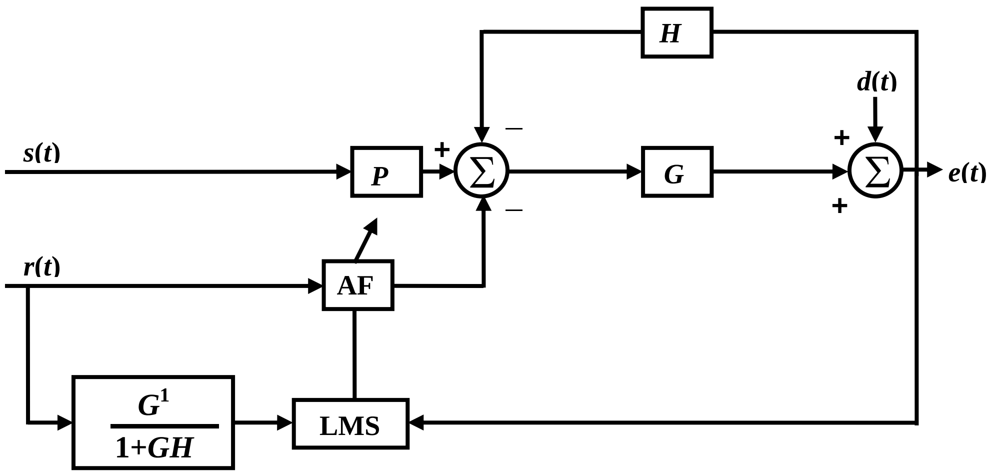


Figure 2.13: Audiometric NCH using combined feedback and feedforward method, from [9].

The methods discussed earlier are based on single actuator, to deliver both noise control and test signals. The authors in the patent [9] have proposed that the single actuator can be replaced by two actuators as shown in Figure 2.14. In this method, the insert earphone delivers test stimulus and a separate actuator delivers the active noise control signal. The control signal is generated using feedback ANC method. The signal at the eardrum can be represented by,

$$E_2 = SP(1+GH)(1+GH+G_1)^{-1} + D(1+GH+G_1)^{-1} \tag{2.6}$$

Where G_1 is the gain of the feedback loop from test signal to residual noise. If G_1 is very small the gain of the filter P becomes unity and the test signal is not affected by the feedback loop, which a major advantage in using two actuators. The feedback effect has to be compensated if the gain G_1 is not small. Since placing an error microphone inside the earcup and noise cancellation, may change the characteristics of the test stimulus; hence the audiometer needs to be calibrated. The calibration is different for each of the noise canceling methods discussed earlier. In two actuators method as the correlation between the test stimulus and noise canceling signal is very low, the test signal may not get affected.

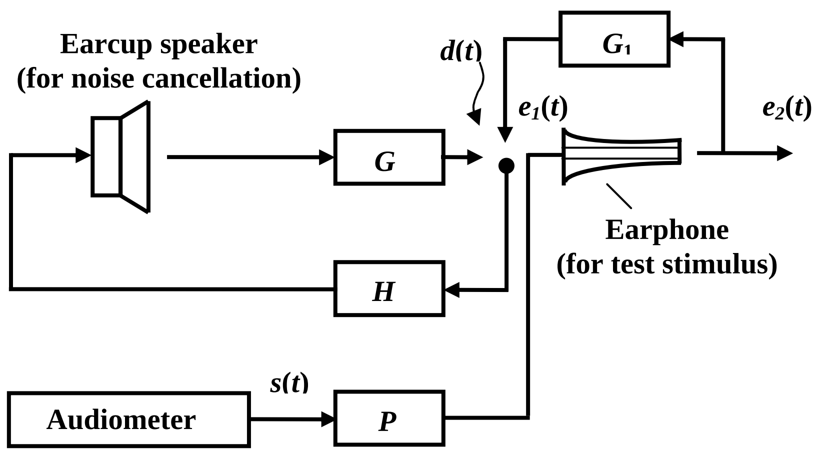


Figure 2.14: Audiometric NCH using feedback ANC with separate actuators for test stimulus, from [9].

2.5 Adaptive algorithms

The implementation of ANC is done using adaptive algorithms such as least mean square (LMS) algorithm to update the coefficients of the filter and filtered-X least mean square (FXLMS) algorithm to model the path transfer function exists between noise canceling speaker and error microphone.

2.5.1 LMS algorithm

LMS algorithm is the widely used adaptive algorithm. It is used for the descending on the performance surface. The weights of adaptive filter are updated using this algorithm. This algorithm is important because of its simplicity and ease of computation. If the adaptive system is an adaptive linear combiner, and if the input vector $d_1(n)$ and the desired response $d_2(n)$ are available at each iteration, the LMS algorithm is generally the best choice for many different applications of adaptive signal processing [4]. The LMS algorithm is used to create a filter with filter coefficients $w_n(k)$ as shown in Figure 2.7.

The output is given as,

$$d_4(n) = \sum_{k=0}^{M-1} d_1(n-k) w_n(k) \tag{2.7}$$

where M is the length of the FIR filter.

We can obtain the filter coefficients updated at time index n as a vector $\boldsymbol{w}_n$. The error $x_1(n)$ is defined as

$$x_1(n) = d_2(n) + d_4(n) \tag{2.8}$$

The filter coefficients can be updated according to

$$w_{n+1} = w_n(k) - \mu x_1(n) d_1(n-k) \tag{2.9}$$

where μ is an update coefficient that determines the speed of the adaptation.

The main drawback of the LMS algorithm is the speed of convergence, becomes very slow if there is a change in the feedback path. The existence of secondary path between noise canceling speaker and error microphone affects the performance of ANC filter and the convergence of the LMS algorithm [4].

2.5.2 Filtered-X least mean square (FXLMS) algorithm

The use of adaptive filter is complicated by the fact that electrical reference signal must be obtained from the acoustic pressure using a microphone. Also, an electrical error signal must be obtained from the residual acoustic noise using an error microphone. Finally, the canceling sound must be produced from the electrical output signal using a loud speaker. Therefore, a number of other transfer functions must be included. The summing junction represents acoustic superposition in the space from the canceling speaker to the error microphone, where the primary noise is combined with the output of the adaptive filter. Therefore it is necessary to compensate for the secondary path transfer function H_3 from d_4 to x_1, which includes the D/A converter, reconstruction filter, power amplifier, error microphone etc. To compensate the effects of secondary path transfer function H_3, the conventional LMS algorithm needs to be modified. To ensure convergence of the algorithm, the input to the error correlator is filtered by a secondary path estimate H_4 [4].

The compensation can be carried by placing, an inverse filter in series with the loudspeaker or an identical filter in the reference signal path to the weight update of LMS algorithm. The placement of the secondary path transfer function, following the digital filter W, controlled by the LMS algorithm is as shown in Figure 2.15.The output $d_4(n)$ is computed as,

$$d_4(n) = \sum_{k=0}^{M-1} d_1(n-k)w_n(k) \tag{2.10}$$

The filter coefficients can be updated according to

$$w_{n+1} = w_n(k) - \mu x_1(n)d_3(n-k) \tag{2.11}$$

where
$$d_3(n) = \sum_{k=0}^{M-1} d_1(n-k)h_{4n}(k) \tag{2.12}$$

$$x_1(n) = d_2(n) + x_4(n) \tag{2.13}$$

$$x_4(n) = \sum_{k=0}^{M-1} d_4(n-k)h_{3n}(k) \tag{2.14}$$

where h_{3n} is the coefficient vector of the secondary path estimate, H_3.

The input vector $d_1(n)$ is filtered by H_3 before updating the weight vector. However in practical applications, H_3 is unknown and must be estimated by the filter H_4.

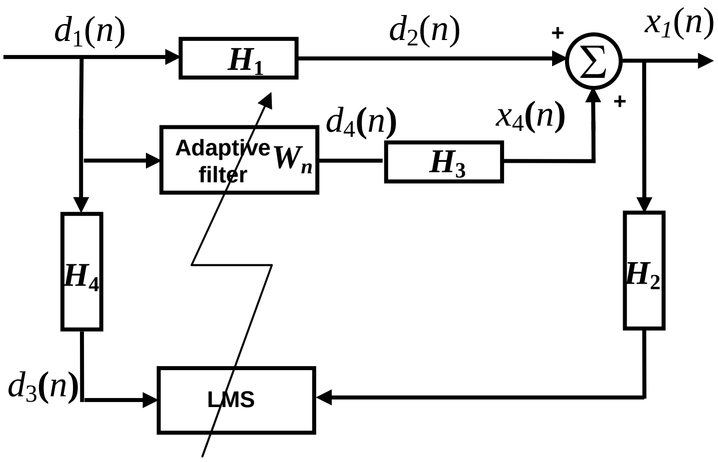


Figure 2.15: Block diagram of the FXLMS algorithm for ANC, from [4].

Chapter 3

THE PROPOSED NCH METHOD AND NUMERICAL SIMULATION

In audiometric application, it will be desirable to have headphones which will cancel the noise without affecting the test stimulus. A method has been proposed for NCH for audiometry based on adaptive feedforward technique, described in the next Section. LMS algorithm is selected to implement feedforward ANC and its implementation is simulated using Matlab. Simulation of the algorithm for various noises and their results are discussed in the subsequent sections.

3.1 Proposed NCH method for audiometry

The analog feedback method discussed in the previous chapter, can effectively reduce the noise over low frequency band but becomes unstable for high frequency noise cancellation. Hence it cannot be used for broadband noise cancellation, and hence it is not suitable for audiometry. The adaptive feedback and feedforward methods are based on continuous adaptation of filter coefficients. The error microphone picks up the error as well as audio signal; hence the generated anti-noise may affects the test signal, which is the primary requirement in audiometry. Hence these methods may distort the test signal.

Audiometry involves presentation of stimuli for brief duration, with significant interval between the successive presentations. Making use of this feature, it is proposed to modify the adaptive feedforwad method for audiometry. This method also uses two microphones, one as reference and the other as error sensor as shown in Figure 3.1. It is proposed to use LMS based feedforward method with filter adaptation suspended during the stimuli presentation of the signal. The output of the reference microphone is given as input to LMS as well as adaptive filter. The noise inside the earcup is picked up by the error microphone and given to LMS as error input. The LMS algorithm adjusts the coefficients of the adaptive filter in such a way as to reduce the error to minimum. The coefficients of the adaptive filter are adjusted using LMS algorithm only when there is no stimulus presentation. These are treated as adapted and frozen during the stimulus presentation and LMS is not used. The anti-noise, produced by the adapted filter is added to the stimulus and then sent to the headphone transducer. Since the stimulus does not contribute to the adaptation, it should not get affected by adaptation.

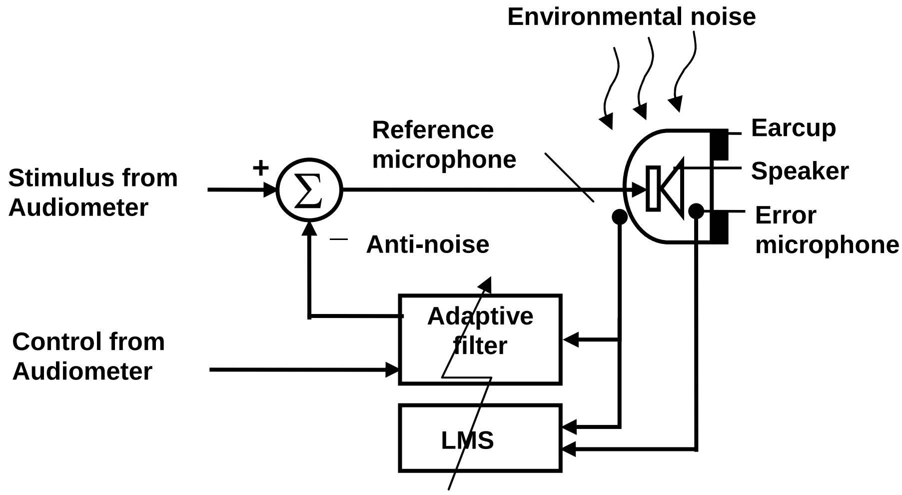


Figure 3.1: Block diagram of proposed NCH technique.

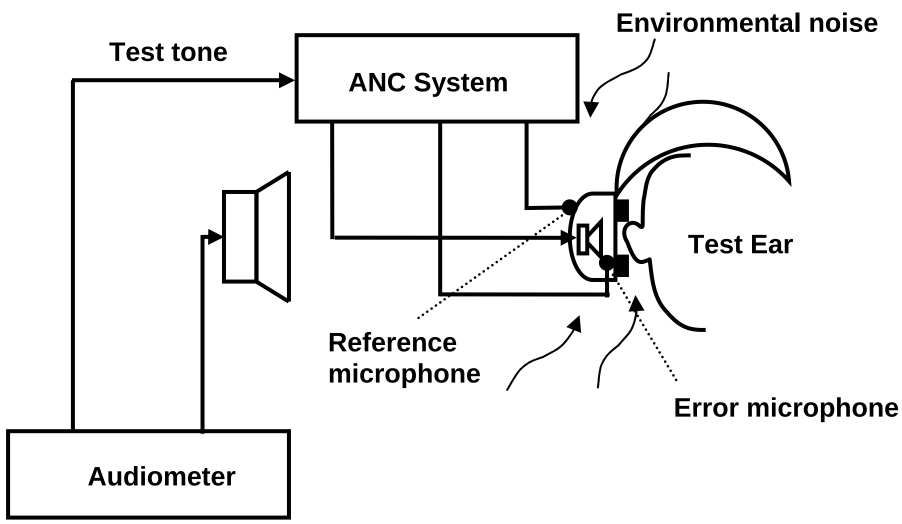


Figure 3.2: Proposed NCH set-up for audiometry

In the proposed method, filter coefficients are updated in the absence of test signal. Hence the adaptive filter can update its coefficients using LMS algorithm in the pause interval available between the tests. For ensuring effective noise cancellation over the entire audio range, the system may employ an external broadband noise source (generally available in the audiometer as masker) as shown in

Figure 3.2. Further, the error sensing microphone may not necessarily be used as part of audiometric set-up. It may be used in a calibration set-up and filter coefficients may be frozen then onwards.

3.2 Simulation and results

LMS algorithm for feedforward method is simulated using Matlab. Results are plotted and analyzed in the subsequent sections for various noises. Characterization of headphone is also done by actually recording noise outside and inside the headphone through a microphone and PC sound card.

3.2.1 Simulation of LMS algorithm using Matlab

A simulation of feedforward NCH was carried out using Matlab, for various noises such as explosion, truck, train, plane, and music. A signal flow representation of the noise cancellation system during adaptation is shown in Figure 3.3(a). The incident noise d_1 is picked up by the reference microphone, represented by the transfer function H_2. The reference microphone output d_3 is input to the adaptive filter, which generates the anti-noise d_4. Reference noise, d_3 also provides input to the LMS algorithm. Noise, d_1 is also transmitted through transfer function H_1 that models the headphone system. Residual noise inside the headphone x_1 is picked up by the error microphone with transfer function H_4, and given as error input to the LMS. The LMS algorithm updates the coefficients $w_n(k)$ of the adaptive filter so as to minimize the error. Anti-noise d_4 produced by the adaptive filter, is added to the test signal or audio signal and then sent to the headphone transducer, which is represented by the transfer function H_3. In our proposed method audio signal is zero during adaptation process.

A signal flow representation of the noise cancellation system after adaptation and while signal is present, is shown in Figure 3.3(b). As described earlier once the coefficients are updated, they are frozen. Anti-noise, produced by the adapted filter is added to the desired audio signal, $s(n)$ and then sent to the headphone transducer. Inside the headphone, noise is cancelled by the anti-noise and original signal is preserved. Simulation was carried out taking the impulse response of the reference microphone h_2, headphone speaker h_3, and error sensing microphone h_4 as unit impulse. Simulation equations are,

$$d_2(n) = d_1(n) * h_1(n) \tag{3.1}$$

$$d_3(n) = d_1(n) \tag{3.2}$$

$$d_4(n) = d_3(n) * w(n) \tag{3.3}$$

$$x_2(n) = x_1(n) \tag{3.4}$$

$$x_3(n) = s(n) - d_4(n) \tag{3.5}$$

$$x_4(n) = x_3(n) \tag{3.6}$$

$$x_1(n) = x_4(n) + d_2(n) \tag{3.7}$$

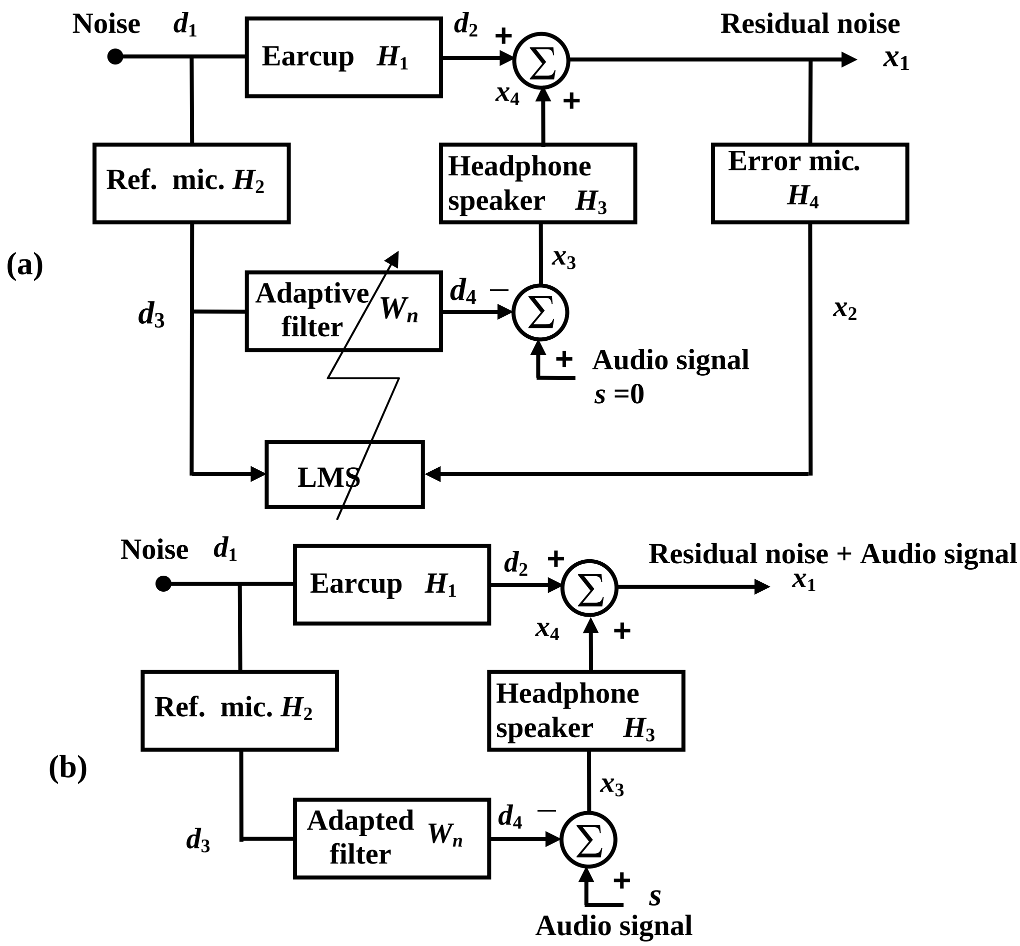


Figure 3.3: Signal flow diagram of NCH (a) without audio signal; (b) with audio signal.

3.3.2 Results for various noises

The sound waveform of 3 s duration was digitized at sampling rate of 24 ksa/s with 16- bit resolution and stored as "`.wav`" files. The order of the adaptive FIR filter is 48. The transfer function of the passive transmission of the noise through the earcup, H_1 is modeled as a 4^{th} order Butterworth lowpass filter with cutoff frequency 500 Hz. Simulation is carried out in two steps. First filter coefficients are updated without giving audio signal input. In the second step anti-noise is produced by adapted filter, added to audio signal. The noise used for adaptation and noise during noise cancellation were the same. The results for some combinations of background noises and signal are shown in Figures 3.4, 3.5, and 3.6. In addition to the waveforms, these figures also show the power spectrum (dB) of the noise inside the earcup, with and without ANC. In Figure 3.4, a white Gaussian noise is the background noise, and the audio signal is a tone of 1 kHz. In Figure 3.5, a white Gaussian noise is the background noise, and the audio signal is speech. In Figure 3.6, a tone of 500 Hz is the background noise, and the audio signal is a tone of 1 kHz. In the waveforms of these figures, x_{1a} is audio signal mixed with primarily transmitted noise, i.e. $x_{1a} = s + d_2$, and x_{1b} is the audio signal mixed with residual noise, i.e. $x_{1b} = s + d_2 - h_3 * d_4$. It is observed from the results that noise reduction is 35 dB for the 500 Hz tone and 25 dB for the broadband noise. Noise reduction depends on a number of factors such as filter order, step size, adaptation time, and intensity of the noise. Simulations were done for a combination of these parameters to select an optimal set for maximum noise reduction. Noise reduction as a function of step size and filter order is discussed in the subsequent sections.

Simulation is also carried out by taking white Gaussian noise as adaptive noise for updating the coefficients of the FIR filter and actual noise as music, tone of 500 Hz, and train noise, and with audio signal being speech, and 1 kHz tone. The power spectra of the waveforms for these simulations are shown in Figures 3.7, 3.8, and 3.9.

In another simulation, the average noise reduction is observed for a white Gaussian noise by varying the filter order of IIR filter, represents the passive transmission path transfer function, and adaptive FIR filter order. The IIR filter is an M^{th} order Butterwoth lowpass filter with cutoff frequency 500 Hz. The average noise reduction as a function of order of FIR and IIR filters is shown in Table 3.1.

Table 3.1: Noise reduction as a function of FIR and IIR filter orders

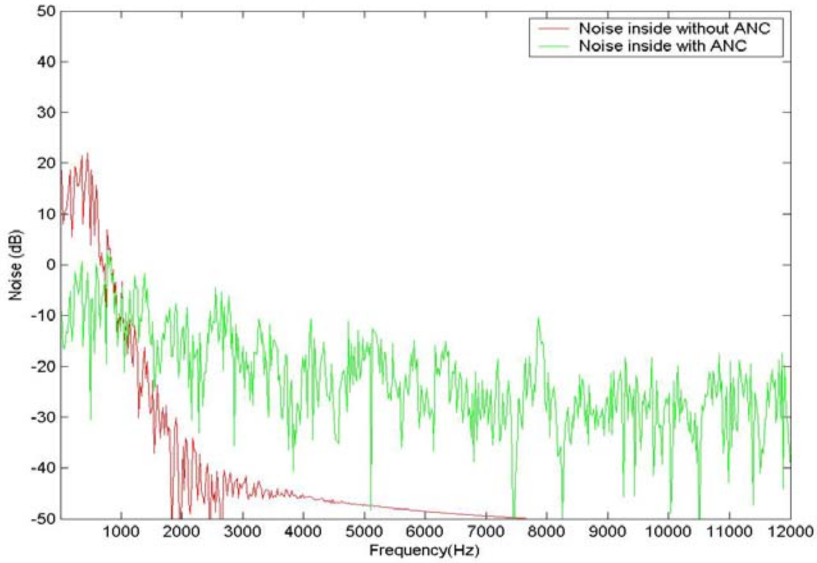
Adaptive FIR Filter order (N)	Avg. noise reduction (dB) for M=4	Avg. noise reduction (dB) for M=8	Avg. noise reduction (dB) for M=12
16	-5	-4	-2
32	-4	-4	-2
48	11	-10	3
64	14	0	-3
80	19	-1	-2
96	40	11	5
112	33	8	6
128	43	10	6
144	51	23	13
160	54	19	12
176	69	20	10
192	68	23	12
208	74	27	15
224	87	29	17
240	86	37	21
256	92	36	24
272	101	37	23
288	100	41	24
304	71	45	28
320	46	39	26
336	20	14	19
352	-7	-10	-7
368	-32	-36	-31
384	-56	-61	-59
400	-85	-91	-86

From the simulations discussed earlier, the following observations are made.

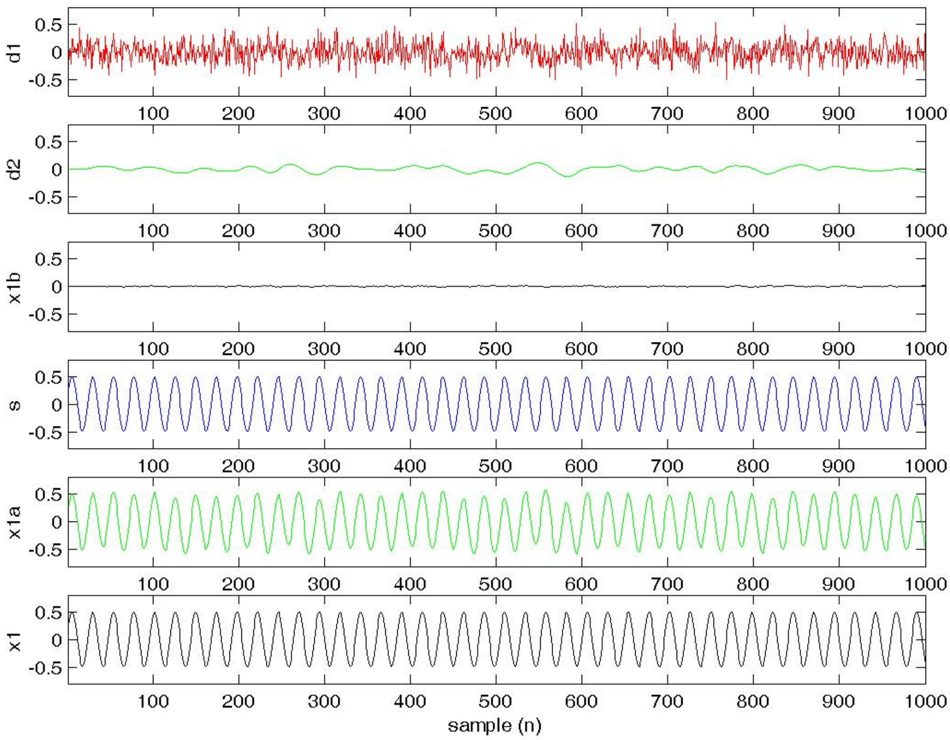
(i) For filter order of 48, if the filter is adapted for a tone we got a very high attenuation (35 dB), but the attenuation decreases if the actual noise is a tone of another frequency or broadband noise.

(ii) If the filter is adapted for broadband noise, it gives an attenuation of about 20 dB for different types of noises and 25 dB for tone as actual interference.

(iii) As the FIR filter order increases the noise reduction increases up to a certain filter order, and after that it decreases and for higher order IIR filter, the noise reduction is less for a certain FIR filter order.

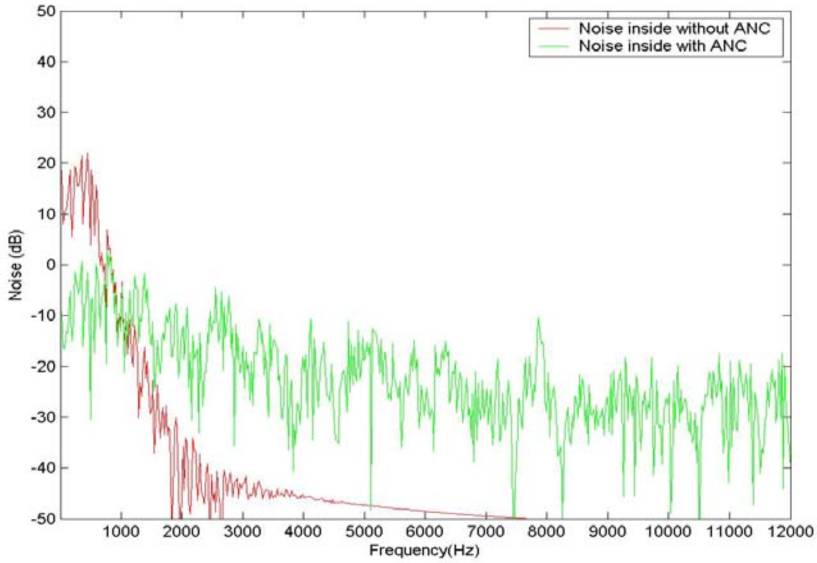


(a)

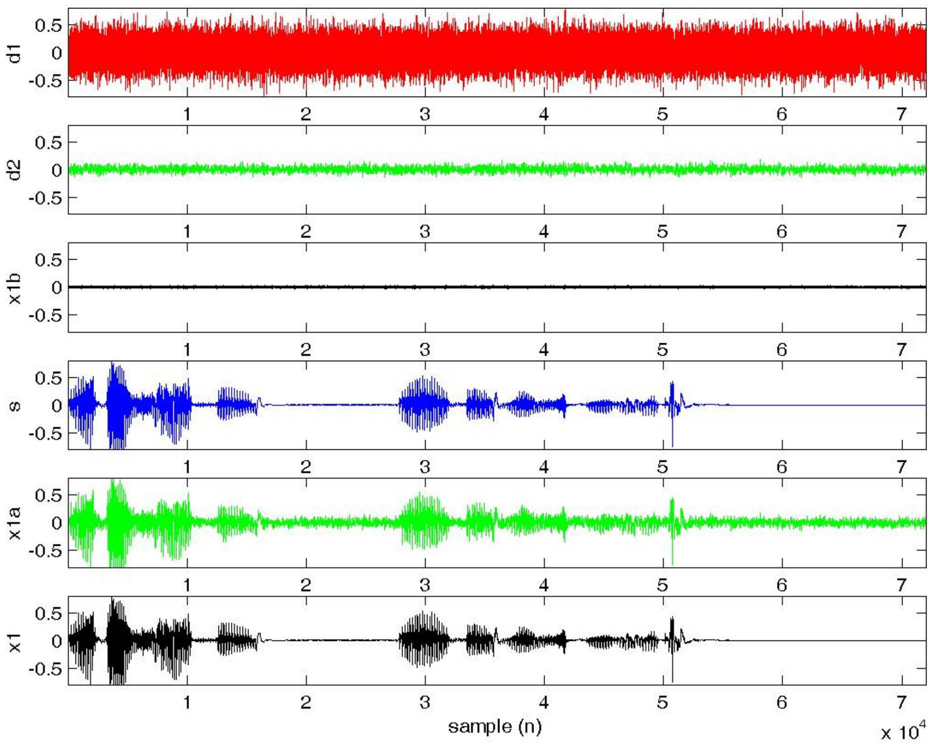


(b)

Figure 3.4: (a) Power spectra of transmitted noise with and without ANC; (b) Plots of the signal at various points of NCH when test signal is a tone of 1 kHz and noise is a white Gaussian noise.

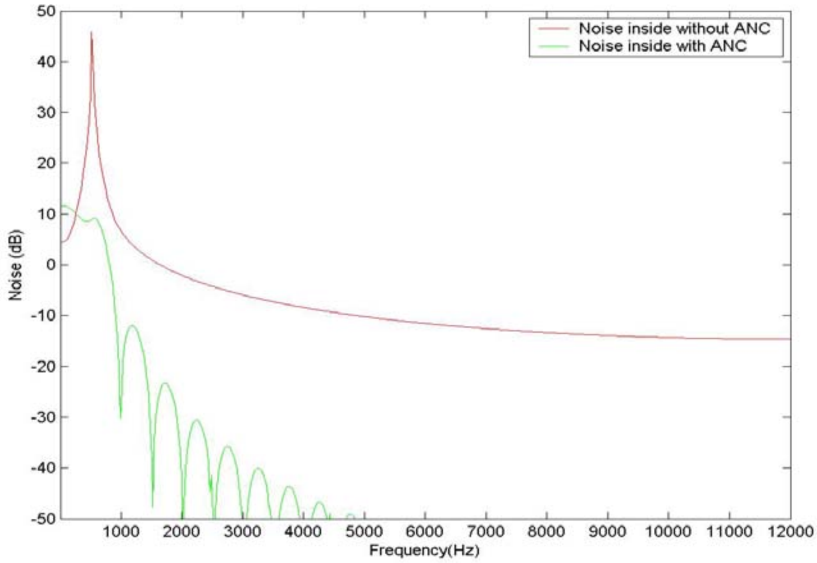


(a)

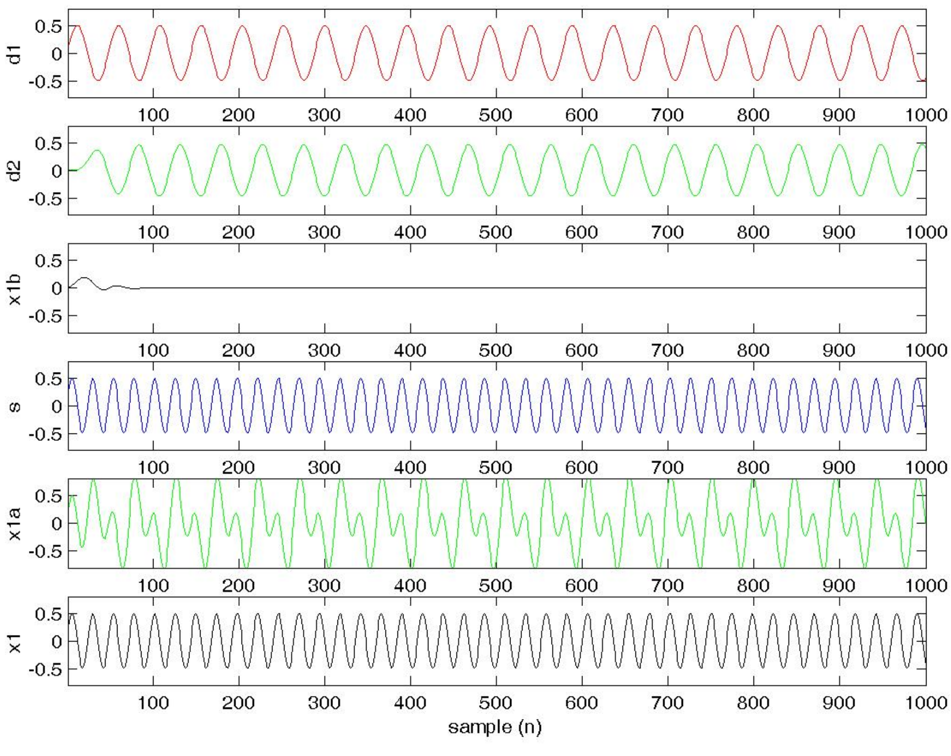


(b)

Figure 3.5: (a) Power spectra of transmitted noise with and without ANC; (b) Plots of the signal at various points of NCH when test signal is a speech and noise is a white Gaussian noise.

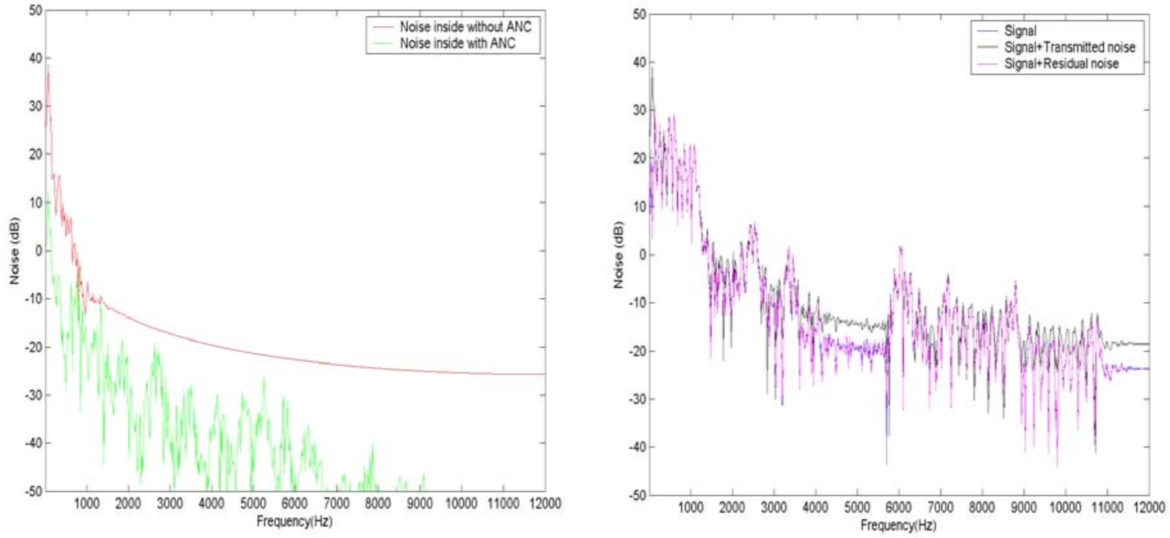


(a)



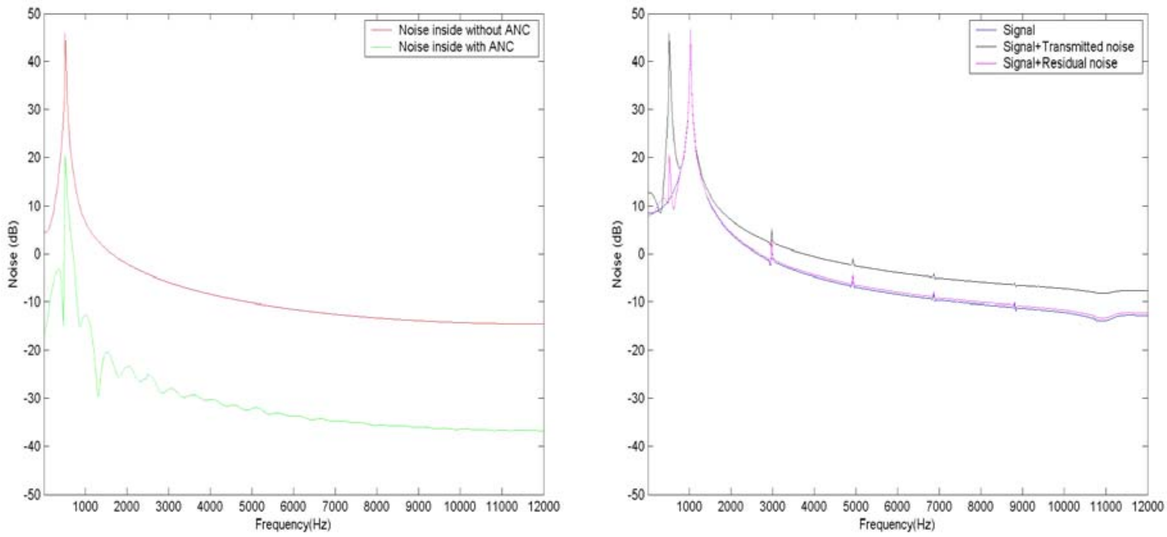
(b)

Figure 3.6: (a) Power spectra of transmitted noise with and without ANC; (b) Plots of the signal at various points of NCH when test signal is a tone of 1000 Hz and noise is a tone of 500 Hz.



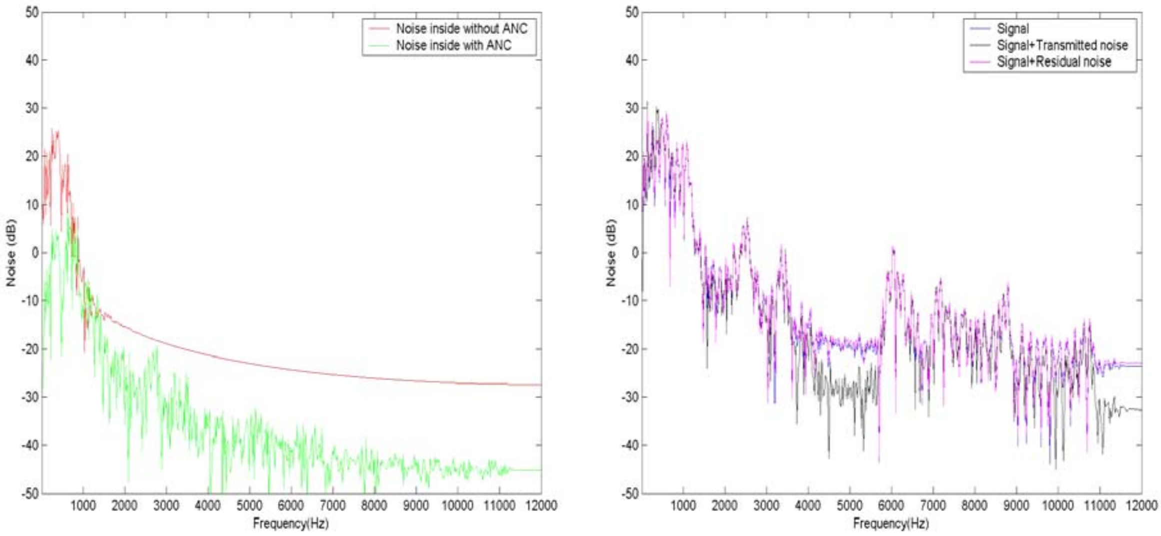
(a) **(b)**

Figure 3.7: Power spectra of (a) noise; (b) noise and signal; with music as the background noise and speech as audio signal.



(a) **(b)**

Figure 3.8: Power spectra of (a) noise; (b) noise and signal; with 500 Hz tone as the background noise and 1 kHz tone as audio signal.

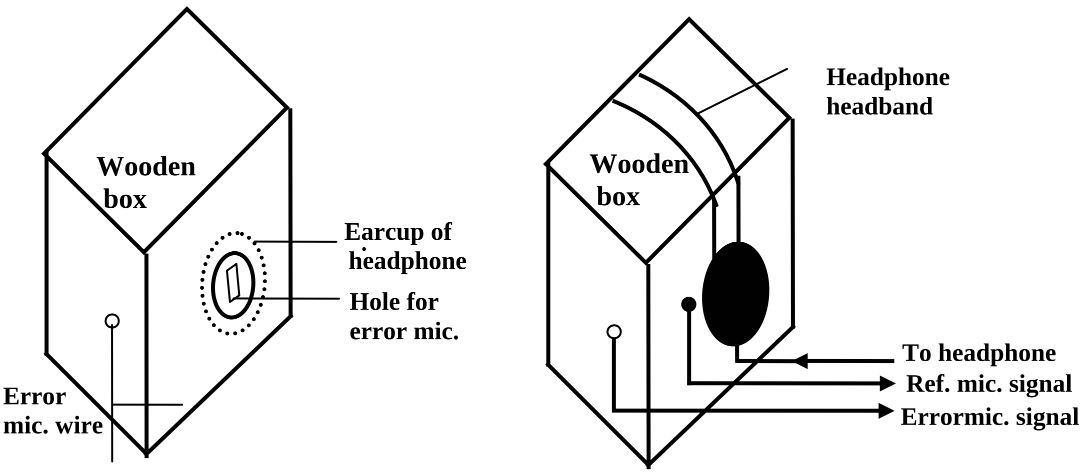


(a) (b)

Figure 3.9: Power spectra of (a) noise; (b) noise and signal; with train noise as the background noise and speech as audio signal.

3.2.3 Passive noise reduction of a headphone

Passive noise reduction characteristics are observed for three different headphones. (i) Bose, (ii) HB-575, and (iii) Nova, using an experimental setup as shown in Figure 3.10. A wooden box, on which, a hole is made for headphone earcup. Inside the hole, a microphone is kept to pickup inside noise. Another microphone is kept on the top of the earcup to pick up the outside noise.



(a) (b)

Figure 3.10: Head simulator wooden block for mounting the headphone and error microphone; (a) Wooden box; (b)Wooden box with headphone and microphones.

A White Gaussian noise was applied through a signal generator and speaker, and noise inside and outside is recorded through a PC sound card. The sound files are stored in ".wav" format. Power spectra of the noise inside and outside are shown in Figure 3.11 for the three different headphones. It is observed from the plots that attenuation of the noise by PNA below 500 Hz is less and it is significant above 1000 Hz. The level of attenuation is also different for each of the headphones tested. Bose headphone has higher attenuation compared to the headphones from Nova and HB.

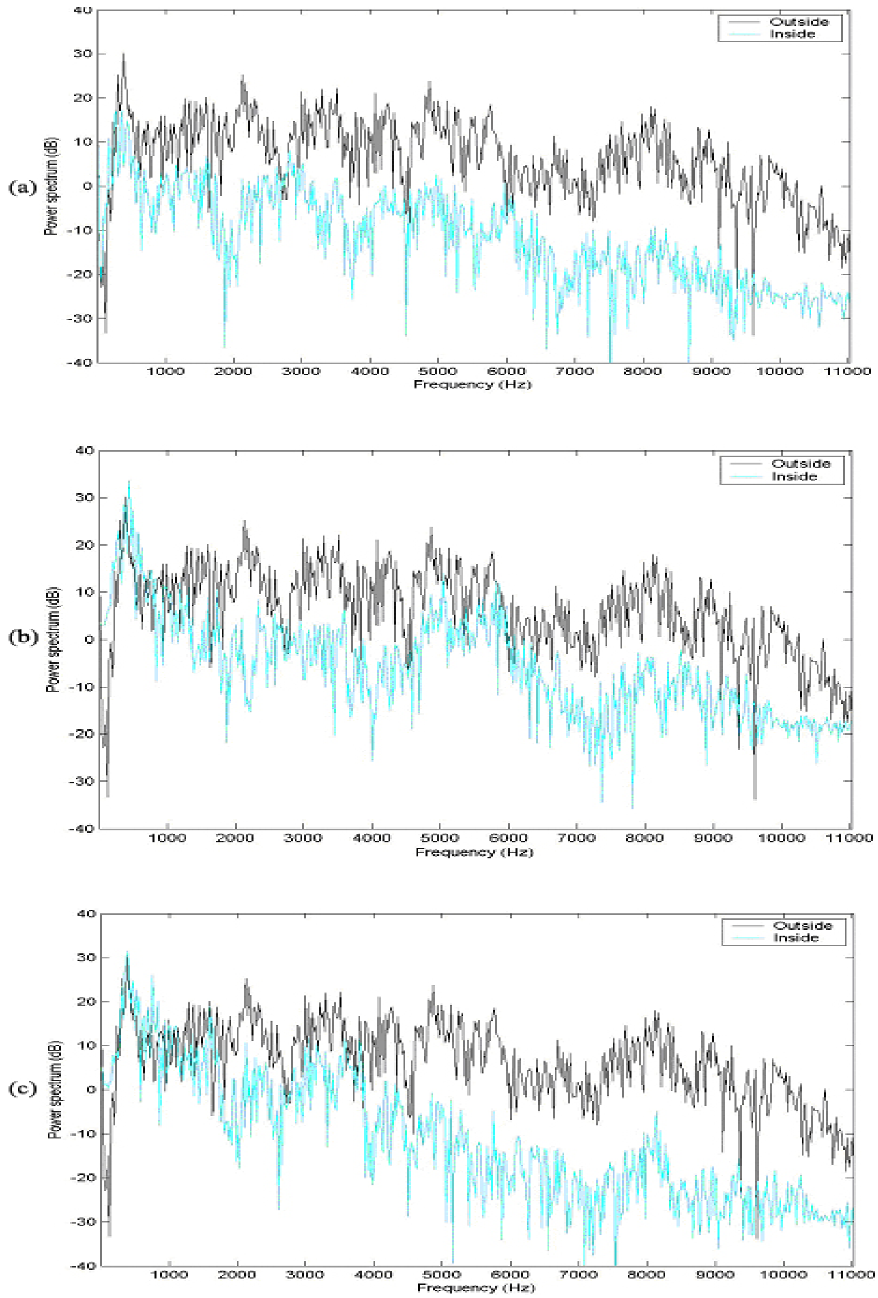


Figure 3.11: Passive noise reduction characteristics of headphone (a) Bose;(b) Nova;(c) HB-575.

Noise canceling headphone from Bose Corporation was tested by applying tone of 200 Hz - 1.5 kHz as well as broadband noise as background noise. The passive and active noise cancellation for a tone input is as shown in Figure 3.12. A maximum noise reduction of 17.5 dB at tone of 200 Hz is observed, and at other frequencies, noise is getting amplified instead of reduction. For broadband noise, attenuation is as shown in Figure 3.13. The noise reduction for tone frequencies below 500 is 10 -15 dB.

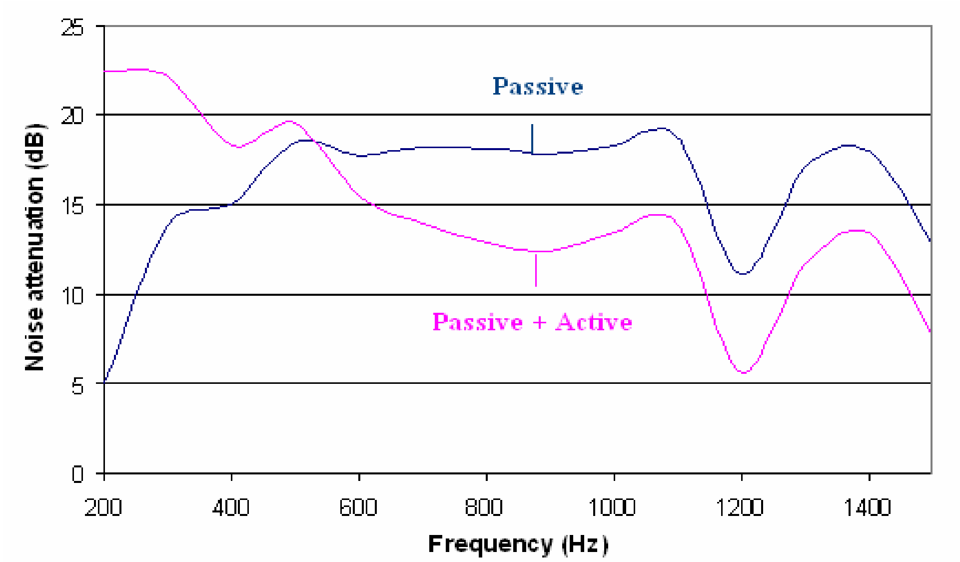


Figure 3.12: Passive and active noise attenuation of Bose headphone, noise taken as swept tone over 200 Hz-1.5 kHz.

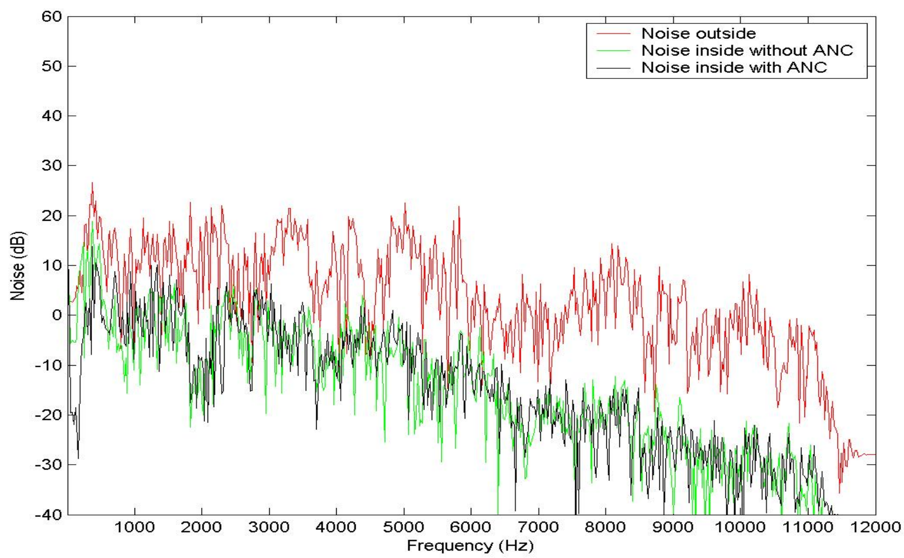


Figure 3.13: Passive and active noise attenuation of Bose headphone, noise taken as a white Gaussian noise.

3.2.4 Headphone characterization

In the earlier simulation, we have taken a 4^{th} order Butterworth IIR filter to model the headphone system. For realistic simulation, we need to characterize the passive transmission of the headphone. Actual model of the headphone system is synthesized using the characteristics of the headphone and LMS algorithm. A white Gaussian noise is applied through a signal generator and a speaker which has an inbuilt amplifier. Two microphones kept outside and inside capture noises outside and inside. The microphone outputs are amplified through a pre-amplifier and recorded using a PC sound card. An adaptive FIR filter is used to model the headphone whose coefficients are updated using LMS algorithm for which the recorded noise inside and noise outside are the inputs. The frequency responses of the FIR filter are as shown Figure 3.14.

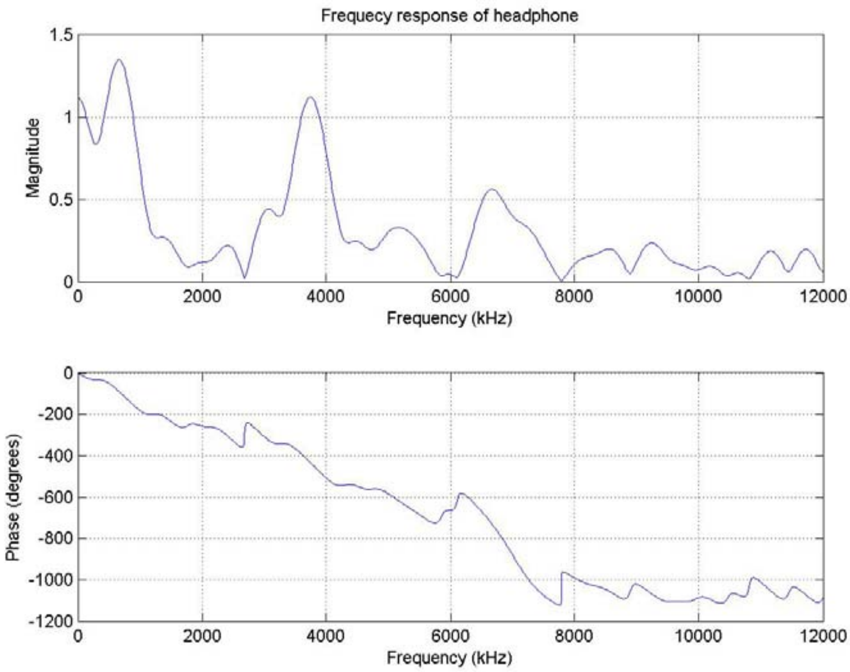


Figure 3.14: Frequency response of the headphone.

The modeled adaptive filter is taken to as adapted FIR filter in noise cancellation. Actual noise is taken as swept tone over 200 Hz to 1.5 kHz, is applied and recorded in the similar manner discussed earlier. Noise reduction as a function of filter order is as shown in Figure 3.15. It is observed from the results that maximum attenuation for a particular frequency occurred at different filter order and vice-versa.

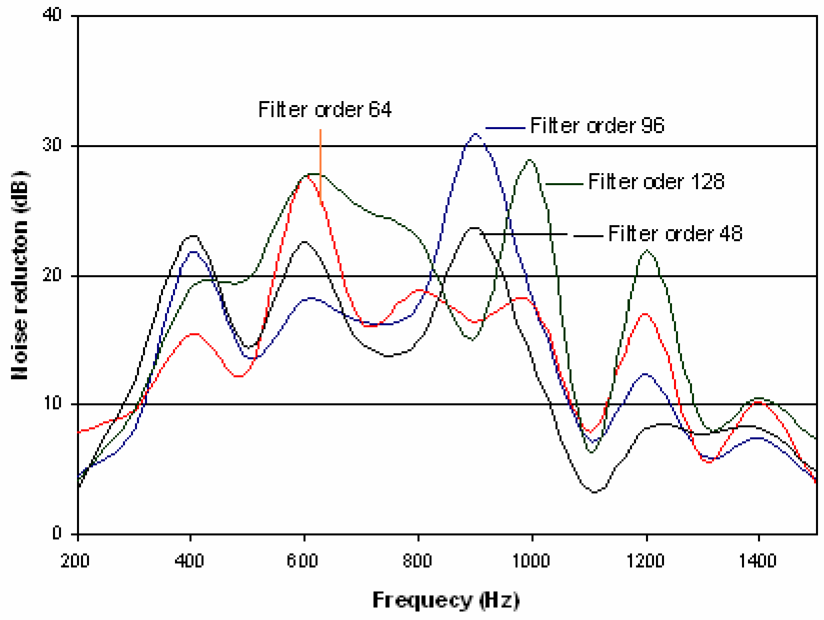
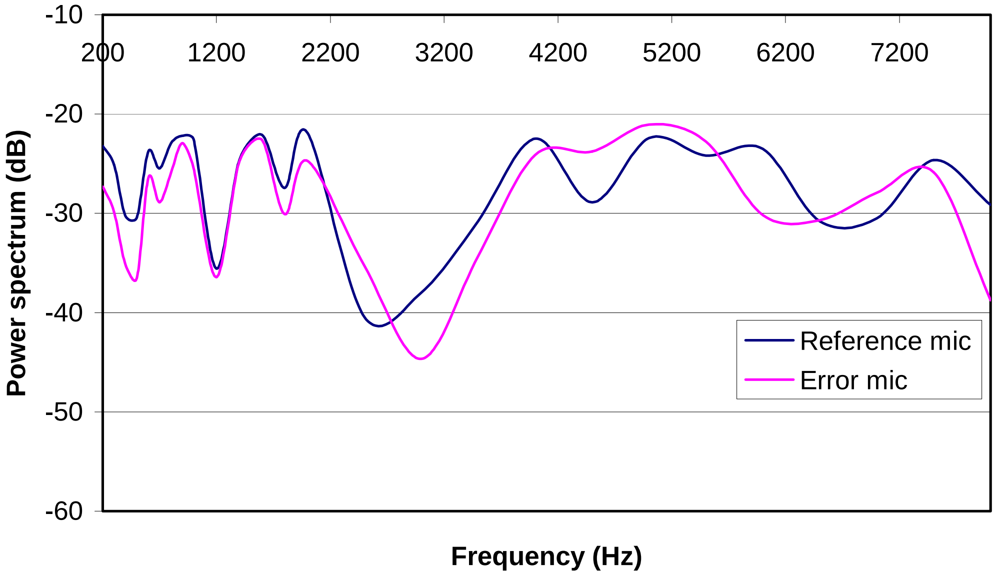


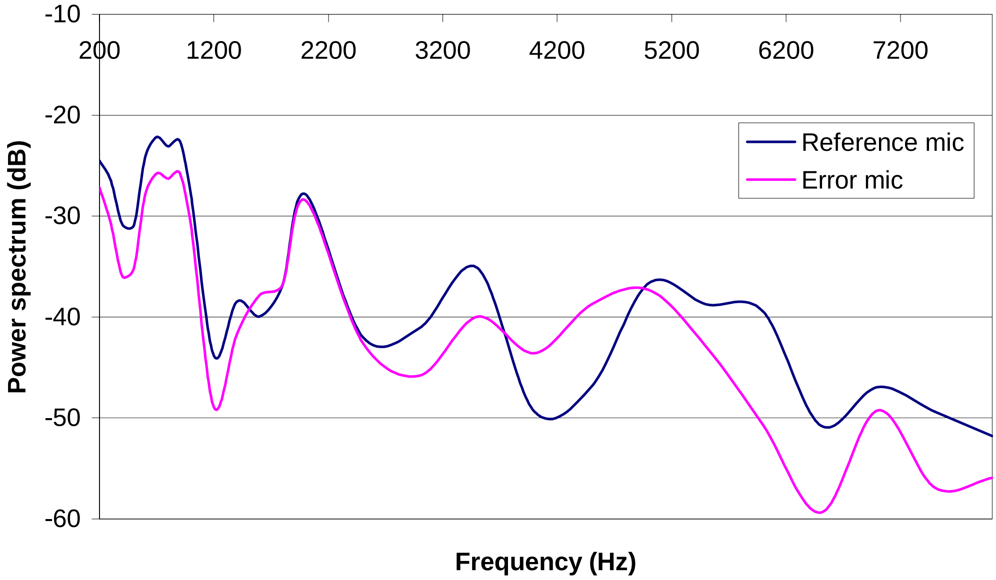
Figure 3.15: Noise reduction as a function of filter order.

3.2.5 Frequency response of microphones

In ANC application, two microphones are used, one used as reference microphone outside the earcup and the other one as error microphone inside the earcup. To understand the behavior of noise cancellation, the frequency response of the two microphones is experimentally observed. The microphones used are electret microphones from the manufacturer "AIWA". The experimental set-up is same as shown in the Figure 3.10. Tones swept over 200 Hz to 8 kHz are applied through a signal generator and a speaker with in-built amplifier. The acoustic intensity of the tones is kept constant at 70 dB, measured using a sound level meter (B&K 2238). The output of each of the microphones is amplified through an amplifier with gain 3 and average magnitude spectrum is observed using a signal analyzer. The frequency response of the microphones, when kept outside the earcup and inside the earcup is shown in the Figure 3.16. It is observed from the results that the two microphones do not form a matched pair, and hence will contribute to an increase in the required filter order for the adaptive filter.



(a)



(b)

Figure 3.16: Frequency response of reference and error microphones, kept (a) outside the earcup, (b) inside the earcup.

Chapter 4

REAL -TIME IMPLEMENTATION USING DSP

This chapter discusses real-time implementation and analog interfacing with DSP board. Next section gives an outline of experimental setup and components used in it. Selection of a DSP board, implementation using the board and results are discussed in the subsequent sections.

4.1 Setup for ANC system

The block diagram of the DSP based ANC system hardware is shown in Figure 4.1. The hardware acquires the two input signals, one from the reference microphone outside the earcup and other from the error microphone inside the earcup. The signal output from the two microphones are amplified and then given to analog inputs of the DSP board. The DSP board consists, on board anti-aliasing filter to reduce the effect of aliasing for ADC inputs. The two inputs are simultaneously sampled and digitized by the ADCs. Digital output of the DSP is converted back to analog signal using a DAC, which is then passed through a smoothening filter. The output signal is added to the stimulus from the audiometer in an output amplifier for driving the earcup speaker. Essential parts of the hardware are discussed in the subsequent sections.

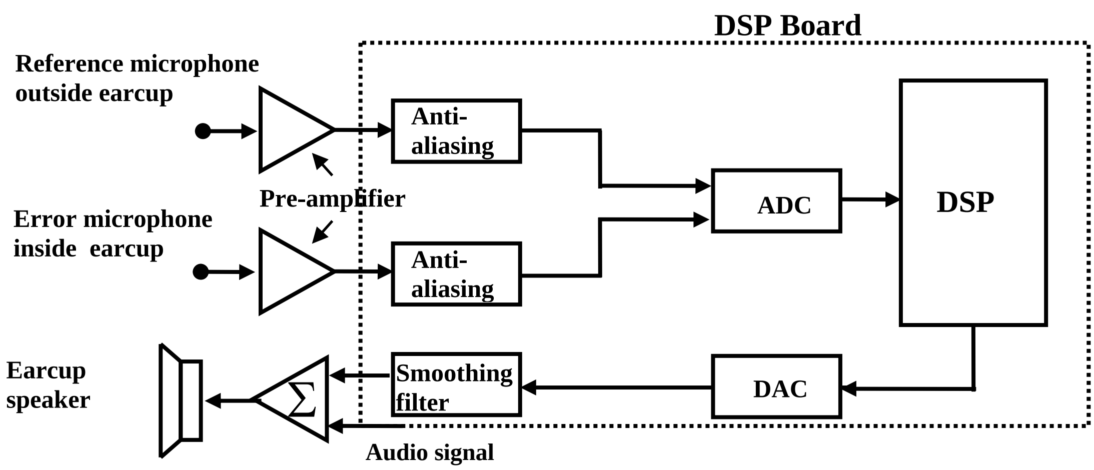


Figure 4.1: DSP based ANC system hardware for single channel NCH.

4.1.1 Microphone and pre-amplifier

The microphones used for NCH are electret condenser microphones (ECMs). An electret microphone consists of a pre-charged, non-conductive membrane between two plates that form a capacitor. One of the plates is fixed and the other plate moves with sound pressure. Movement of the plate results in a capacitance change, which in turn results in a change in output voltage due to the non-conductive, pre-charged membrane. An electrical representation of such an acoustic sensor consists of a signal voltage source in series with a source capacitor. The most common method of interfacing this sensor has been a high impedance buffer such as a single junction field effect transistor (JFET) with its gate connected to the sensor plate as shown in Figure 4.2. The high input impedance of a JFET gate provides a good interface to the capacitive sensor, and the relatively low resistance connected to the drain provides sufficient output drive towards a signal-conditioning block such as an ADC [10].

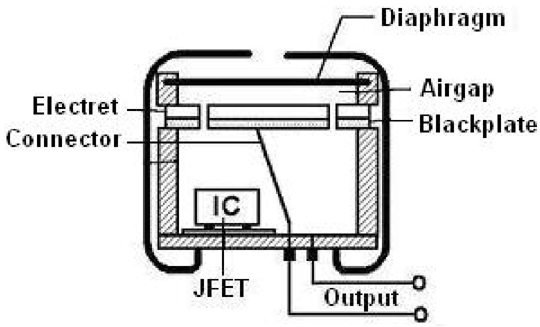


Figure 4.2: Cross section of an ECM with JFET buffer, from [10].

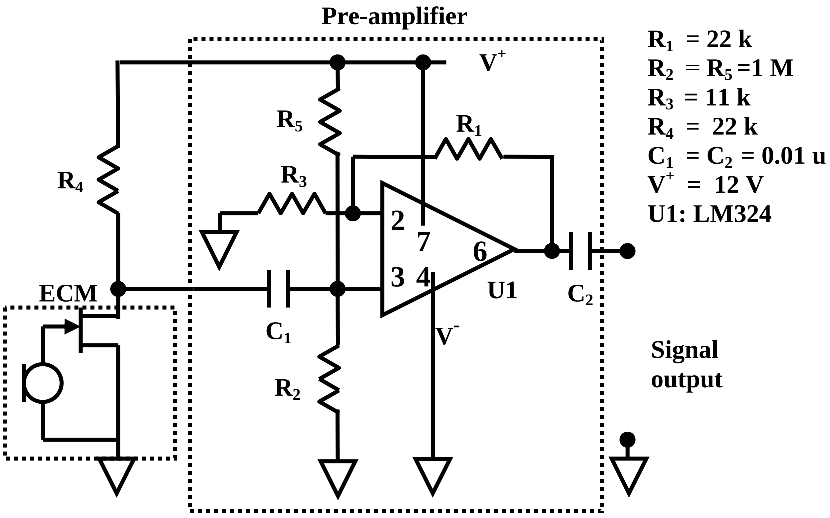


Figure 4.3: Circuit diagram of ECM with pre-amplifier

The method of biasing and interfacing the ECM is called "phantom biasing" [10] through a load resistor and a series capacitor. The main benefits of this type of ECM are its small size, low cost and relatively low noise. Having a low sensitivity of the ECM results in small output voltages of the sensor (on the order of 0.1-10mV average, 100mV peak) [10]. Also, to supply the signal to an analog to digital (ADC) converter, significant pre-amplification is necessary as shown in Figure 4.3. Gain of the amplifier can be changed by varying resistors (R_1or R_3) value.

4.1.2 Signal acquisition unit

Signal acquisition unit of the DSP board contains on chip ADC, DAC, MUX and lowpass filters for signal processing. The input signals are amplified with a gain of 3 and then filtered by anti-aliasing filters before being applied to their respective ADC input ports. The ADC converts the signal into discrete output digital words, corresponding to the analog signal value at the sampling time. These digital words, representing sampled values of the analog input signal, are sent to the host through a serial port interface for their respective channels. Most of the DSP boards contain ADCs which are over sampling sigma-delta modulators. These ADCs provide high resolution and low noise performance using over sampling techniques and the noise shaping advantages of sigma-delta modulators [11].

The DAC receives the data words from the host through the serial port interface for each channel. The data is converted to analog voltages by their respective sigma-delta DACs comprised of a digital interpolation filter and a digital modulator. The outputs of the DACs are each then passed to internal low pass filters to complete the signal reconstruction resulting in an analog signal. These analog signals are then sent to an output driver, driving a load such as headphone transducer.

4.2 DSP board for ANC

NCH is a real-time embedded system application, in which the adaptive filter, secondary path modeling and ANC will be implemented using a DSP board. The suitable DSP is selected on the basis of power of speed, flexibility, cost and architecture optimized for adaptive signal processing. The DSP board should contain at least two signal inputs (for each channel) and one output. There are several processors and boards available from different manufacturers such as Texas Instruments, Analog Devices, and Freescale Semiconductors.

Texas Instrument provides DSK (DSP starter kit) which includes TMS320C6211 processor. The DSK also contain code composer studio, C/C++ compiler/debugger. The board includes analog interface with only one input signal and one output signal, which is a major drawback in using it for NCH. Analog Devices, ADSP-BF533 Blackfin processor is one of the suitable processors. Analog Devices offers this processor in a kit called EZ-Kit Lite. The package contains an evaluation copy of Visual DSP++, which includes C/C++ compiler/debugger. Information on ADSP processors is given in the processor's manual [11], [12]. ADSP-BF533 processor based DSP board was chosen to implement NCH and its features are discussed in the next section.

4.3 Overview of ADSP BF 533 EZ Kit Lite

The ADSP-BF533 EZ-KIT Lite provides a cost-effective method for initial evaluation of the ADSP-BF533 Blackfin processor for a wide range of applications including audio and video processing [13]. The EZ-KIT Lite includes an ADSP-BF533 desktop evaluation board and fundamental debugging software to facilitate architecture evaluations via a USB-based PC-hosted tool set. Real-time debugging is made possible via the background telemetry channel (BTC) feature. Through BTC, data can be streamed both to and from the processor over the JTAG connection between host and embedded processor without the overhead involved with halting the target application, getting the desired information, and then restarting the target application. The ADSP-BF533 EZ-KIT Lite provides an evaluation suite of the VisualDSP++ integrated development and debugging environment with C/C++ compiler, advanced plotting tools, statistical profiling, VisualDSP++ component software engineering (VCSE), and the VisualDSP++ kernel (VDK). Other features of VisualDSP++ include: assembler, linker, libraries, and splitter [11], [12].

The ADSP-BF531, ADSP-BF532, and ADSP-BF533 processors are enhanced members of the Blackfin processor family that offers significantly higher performance and lower power than previous Blackfin processors while retaining their ease-of-use and code compatibility benefits. The three new processors are completely pin compatible, differing only in their performance and on-chip memory. The Blackfin processor core architecture combines a dual MAC signal processing engine, an orthogonal RISC-like microprocessor instruction set, flexible single instruction multiple data (SIMD) capabilities, and multimedia features into a single instruction set architecture. The processor system peripherals include parallel peripheral interface, serial ports (SPORTs), serial peripheral interface (SPI), general-purpose timers, universal asynchronous receiver transmitter (UART), real-time clock (RTC), watchdog timer, and general-purpose I/O (programmable flags).These peripherals are connected to the core via several high bandwidth buses, as shown in Figure 4.4.

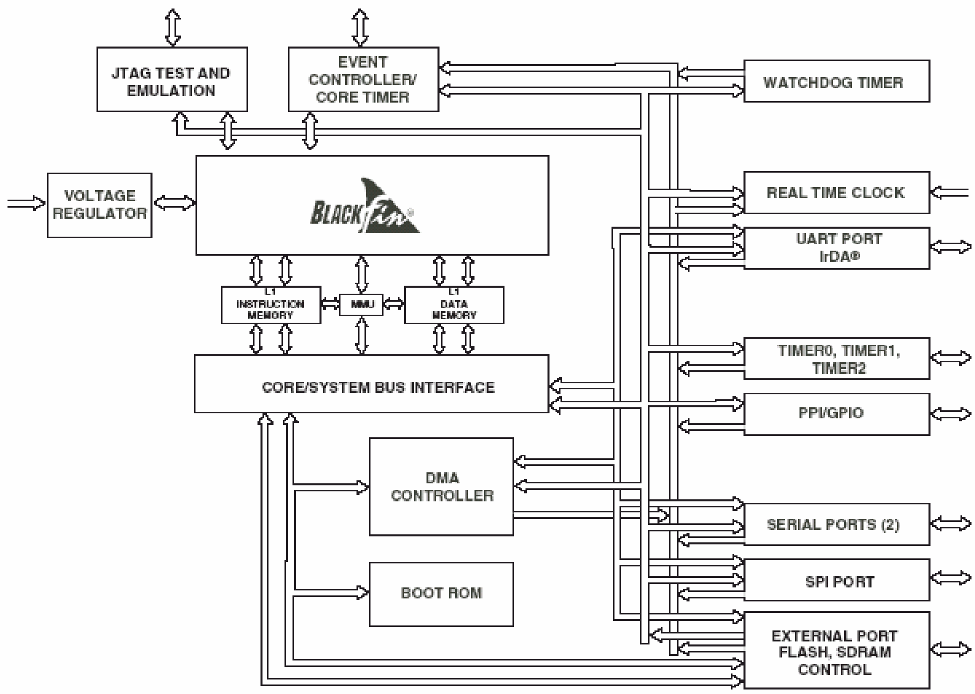


Figure 4.4: Block diagram of ADSP BF 533 processor, from [11].

The block diagram of AD1836A codec is shown in Figure 4.5. This codec is a high performance, single-chip codec that provides three stereo DACs and two stereo ADCs using multibit Σ-Δ architecture patented by Analog Devices [11]. A serial peripheral interface (SPI) port is included, allowing a microcontroller to adjust volume and many other parameters. The AD1836A operates from a 5 V supply, with provision for a separate output supply to interface with low voltage external circuitry. There are four ADC channels in the AD1836A configured as two independent stereo pairs. One stereo pair is the primary ADC and has fully differential inputs. The second pair can be programmed to operate in one of three possible input modes (programmed via SPI ADC control register 3). The ADC section may also operate at a sample rate of 96 kHz with only the two primary channels active. The ADCs include an on-board digital decimation filter with 120 dB stop-band attenuation and linear phase response, operating at an over-sampling ratio of 128 (for 4-channel 48 kHz operation) or 64 (for 2-channel 96 kHz operation). The primary ADC pair should be driven from a differential signal source for best performance. The secondary input pair can operate in one of three modes: Direct differential inputs, PGA mode with differential inputs. In this mode, the PGA amplifier can be programmed using the SPI port to give an input gain of 0 dB to 12 dB in steps of 3 dB. Single-ended MUX/PGA mode, in this mode, two single-ended stereo inputs are provided that can be selected using the SPI port and. input gain can be programmed same as PGA mode [13], [14].

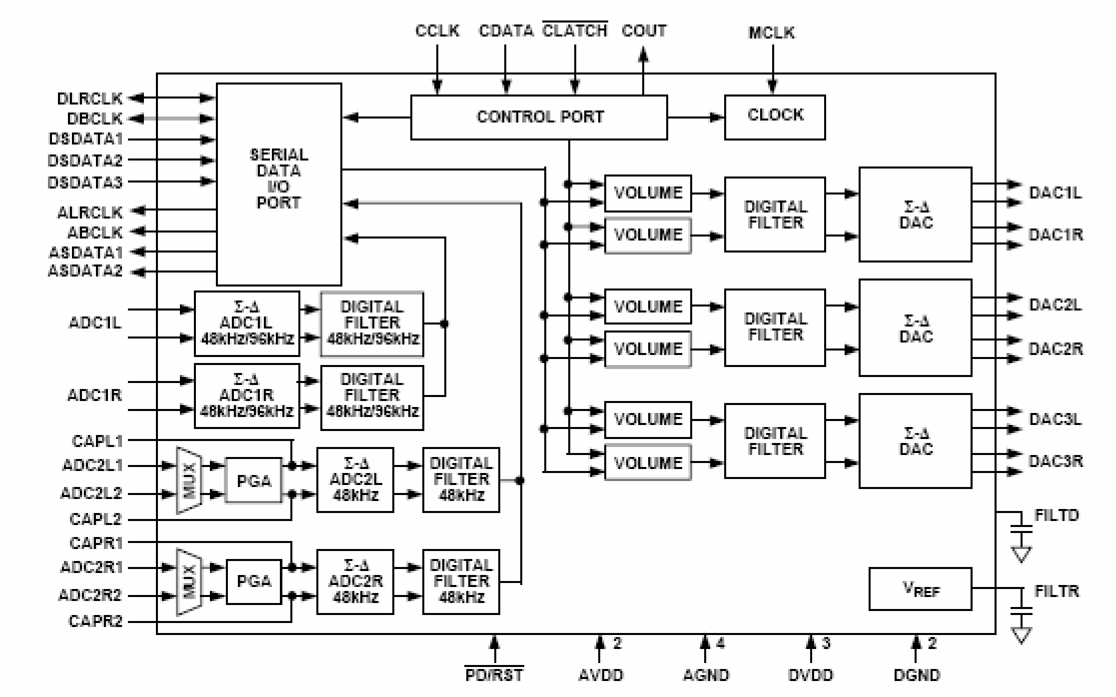


Figure 4.5: Block diagram of AD 1836A, from [11].

The AD1836A has six DAC channels arranged as three independent stereo pairs, with six fully differential analog outputs for improved noise and distortion performance. Each channel has its own independently programmable attenuator, adjustable in 1024 linear steps. ADCs and DACs of AD 1836A can be programmed to set the resolution of 24, 20, or 16 bits. The codec can be programmed using SPI to use it either I2S or TDM mode. In I2S mode two stereo inputs and two stereo outputs can be used to process the signals at sampling frequency of 96 kHz or 48 kHz. In TDM mode, simultaneous processing of two stereo inputs and three stereo outputs can be done at sampling frequency of 48 kHz. ADC and DAC control registers can be programmed to set the sampling rate, resolution, gain of PGA and attenuation.

4.4 Implementation using ADSP BF 533 EZ Kit Lite

Real-time implementation of ANC is carried out as shown in Figure 3.2, in Chapter 3. ANC system hardware includes microphones, pre-amplifier, headphone, DSP board, and noise generator (usually available as masker inside the audiometer). The implementation of ANC comprises of two stages: adaptation process and ANC mode. In the adaptation process, coefficients of the adaptive filter are tuned. A broadband noise is generated through a speaker and masker in the audiometer. A reference microphone kept on top of the earcup picks up the noise presented outside the headphone. Noise presented inside the earcup is picked up by an error microphone kept in the vicinity of noise canceling speaker. Signal outputs from the microphones are amplified through pre-amplifiers and then given to input ports of DSP board. During the adaptation no test signal is given to headphone transducer. LMS algorithm updates the coefficients of the adaptive filter. When updating process is finished, coefficients of the filter are frozen and LMS algorithm and error microphone are disabled. Since the test signal is given for a short duration with long pauses, the filter coefficients can be updated during pause intervals.

In ANC mode, the noise outside the headphone is given as the only input to the DSP board. The input signal is sampled and given as input to the digital filter. Filter output is calculated by point by point multiplication and accumulation of the adaptive filter coefficients and input data. Anti-noise, output of adaptive filter, generated by the ANC system is mixed with the test signal and then given to transducer of the headphone. The anti-noise reduces noise present inside the earcup by acoustical superposition.

Software routine of real-time ANC includes A/D and D/A conversion of input and output signals, LMS algorithm, and FIR filtering. A/D and D/A conversion and transfer of signal data can be done using ADC, DAC, SPI, serial port and direct memory access (DMA). First SPI is configured to initialize the codec to set the parameters like number of input and output channels, sampling rate, resolution of ADC and DAC, PGA gain, and attenuation factor for output. After initialization of the codec, SPI is disabled. Transfer of data between memory, ADC, and DAC is carried out via SPORT and DMA as shown in Figure 4.6. The processor uses DMA to transfer data within memory spaces or between a memory space and a peripheral. The processor can specify data transfer operations and return to normal processing while the fully integrated DMA controller carries out the data transfers independent of processor activity. DMA interrupts the processor after completion of transfer of data (at every sampling interval) from serial port to memory, meanwhile the processor is busy in processing the signal such as filtering, LMS algorithm. In the interrupt service routine the acquired data from the ADC are processed and output sent back to DAC. The example program given along with the Visual DSP++ software from Analog Devies, known as "talkthrough", is modified for using input and output data transfer [12]. Initially the frame synchronization of the data sample is not proper; hence the data got swapped between left and right channels of the two ADC buffers. Numbering the frames in a sequential order from right channel to left has solved the problem.

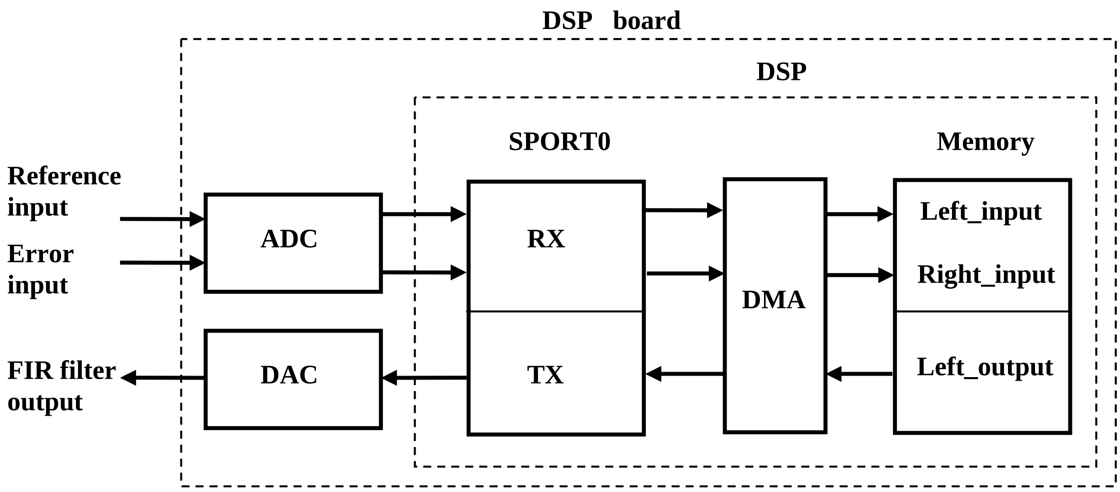


Figure 4.6: Functional block diagram of I/O signals with SPORT, DMA, and memory.

The LMS algorithm, used to update the coefficients of the adaptive filter, is implemented using software routine written in C language. Since the platform chosen is 16-bit fixed point DSP processor based board, for efficient computation the algorithm was implemented in fixed point arithmetic. For 16-bit word length, the fixed point number has the range (-2^{15} ,$2^{15}-1$). All the equations were implemented with appropriate scaling of intermediate results, to keep them within the 16-bit range. During multiply and multiple accumulations of data, there is still a risk of overflow and results in severe distortion of results. Hence a saturation logic is implemented and hence overflow generally does not occur during computation. Only during the use of the result in the next computation, it must be within 16-bit. The processor has a 40-bit accumulator, whenever the computation is likely to exceed result exceeds the 16-bit range while it is to be used as 16-bit input in the next computation saturation logic needs to be provided. The data from the ADC is represented internally in 2's complement format. The I/O data, 24-bit (as the resolution of the ADC and DAC is 24-bit) is represented using an integer (4 bytes) and the coefficients of the adaptive filter are by short integer (2 bytes). Initially the input data is scaled down to 16-bit and stored in the memory. Scaling factors (α, β, γ, and δ) are adjusted in such a way as to make the result not to exceed the range of the corresponding output while maintaining maximum precision (for D_4, range is (-2^{23} ,2^{23}-1) and for W_k, it is (-2^{15} ,2^{15}-1)). The scaling operations are performed using right shifting, where number of right shifts can be carried out in a single instruction. The algorithm implementation is optimized in software routine using macros instead of constants, and global variables instead of local variables used in different functions. For DSP implementation, we are using the ADC inputs scaled down as 16-bit integers. All the filter coefficients are also scaled as 16-bit integers. It is to be noted that this may result in excessive underflow. A better implementation would involve applying 24-bit saturation logic before scaling.

Computations in Matlab simulation which are also involved in the DSP implementation are as the following.

Select step size, μ in the range 0-0.25 (to be selected experimentally).

The reference input d_1 and the error input x_1 are in the range $\{-1,1\}$.

Compute adaptive filter output d_4 as

$$d_4(n) = \sum_{k=0}^{M-1} d_1(n-k)w_n(k)$$

where M = order of the adaptive filter, and $w_n(k)$ are the filter coefficients. The filter coefficients found to be in the range $\{-1,1\}$. These are updated using

$$w_{n+1}(k) = w_n(k) - \mu x_1(n)d_1(n-k)$$

The computation in DSP implementation can be summarized as the following

Take $\beta=2^{15}$ as a scaling factor and scale the step size to set

$$M_u = \beta\mu$$

The reference and error inputs $D_1(n)$ and $X_1(n)$ are 16-bit integers after scaling the 24-bit ADC inputs.

The adaptive filter output is computed as

$$D_4(n) = [\sum_{k=0}^{M-1} D_1^1(n-k)W_n(k)]/\alpha$$

where $\alpha = 2^8$, to make the ouput compatible with 24-bit DAC.

Since the input $X_1(n)$ is the same for all values of *k*, for updating filter coefficients, it is normalized once for updating at each sample.

$$X_{1\mu}(n) = (M_u X_1(n))/\gamma$$

we can select $\gamma = 1/\beta$, actual value used is 2^{-16}

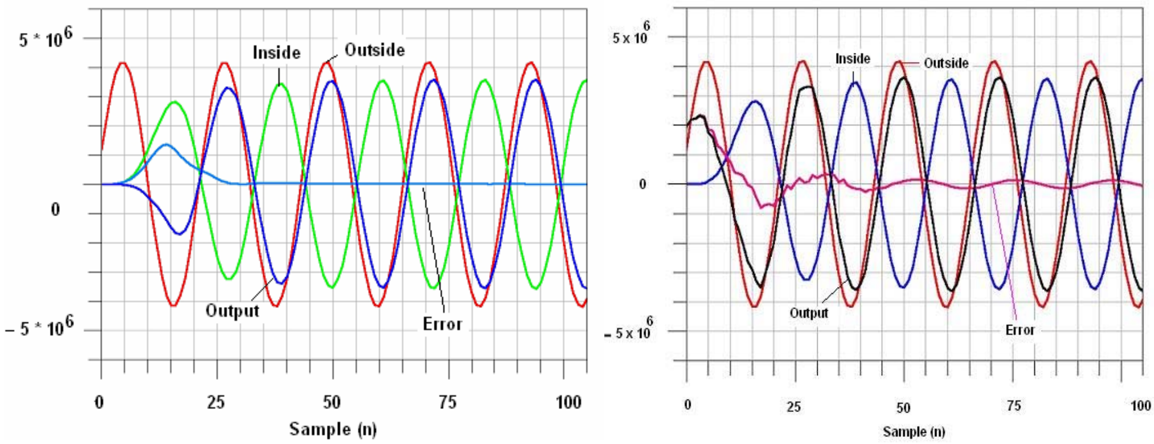
The filter coefficients $W_n(k)$ are updated using the equation

$$W_{n+1}(k) = W_n(k) - [X_{1\mu}(n)D_1(n-k)]\delta$$

where γ is also taken as 2^{-16}.

4.5 Simulation results

LMS algorithm is simulated using Analog Device's Visual DSP++ simulator [12]. Visual DSP++ simulator has a provision to calculate number of instruction cycles taken by the algorithm using linear profiling. Total number of instruction cycles taken by the LMS algorithm is observed to be 5921; hence execution time is 6.99 μs (as the instruction cycle time of ADSP BF 533 processor is 1.32 ns) for a filter order of 48. Simulations were carried out by taking a tone of 1000 Hz as actual noise, and adaptive noise as (i) tone of 1000 Hz, (ii) white Gaussian noise. Headphone is modeled by a 4^{th} order Butterworth lowpass IIR filter with cutoff frequency of 500 Hz. Results were plotted for noise with and without ANC as shown in Figure 4.7. It is observed from the results that even though the filter is tuned for broadband noise, other kinds of noise also get attenuated, provided that the filter order is high enough. If the adaptive noise is a tone, adaptation time and filter order are low to eliminate that particular noise and reduction in noise is 25 dB. If the adaptive noise is a broadband noise, adaptation time and filter order are high and a typical FIR filter order is 48, and the noise reduction is 10 dB. Since the number of samples taken for simulation is 110, the noise reduction is less. The time taken by the simulator to execute the algorithm is long; hence fewer samples have taken for simulation.



(a) **(b)**

Figure 4.7: Plots of noise with and without ANC, (a) Adaptive and actual noise is a tone of 1 kHz; (b) adaptive noise is a white Gaussian noise and actual noise is a tone of 1 kHz.

4.6 Real-time implementation results

The experimental set-up to implement LMS algorithm is as shown in Figure 4.8. Two inputs from the reference microphone and the error microphone are given as analog inputs to the DSP board through a pre-amplifier having a gain of 3. Anti-noise is computed in the DSP and sent to the driving speaker through DAC and a summer. Inside the earcup, noise gets reduced due to acoustical superposition of noise and. Experiments were done with the different combinations of noise, filter order, step size, adaptation time, and intensity of the noise. Variation of noise reduction as a function of step size is as shown in Figure 4.9, for a filter order of 48 and noise as tone of 500 Hz. It can be observed from the plot that for different intensities of the noise, for maximum noise reduction, step size is different.

For sampling rate of 48 ksa/s, the maximum FIR filter order permitted by the code execution speed is 48. To increase the filter order for maximum noise reduction, the sampling rate is reduced to 24 ksa/s in the software routine. A maximum filter order of 96 could be used in the ANC with this sampling rate. The variation of noise reduction as a function of filter order is as shown in Figure 4.10, for 500 Hz tone as background noise. Maximum noise reduction achieved at each of the filter order is at different step sizes for a particular noise.

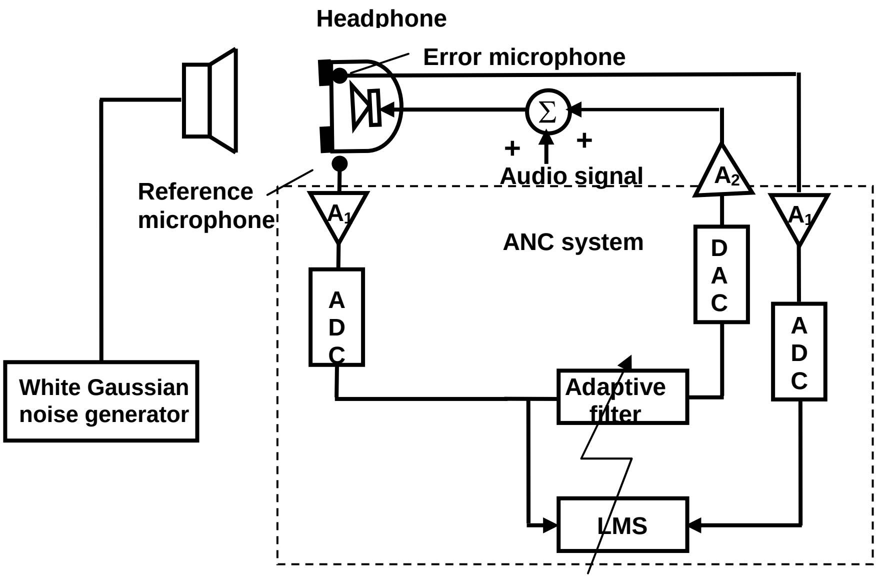


Figure 4.8: Experimental set-up for ANC using LMS algorithm.

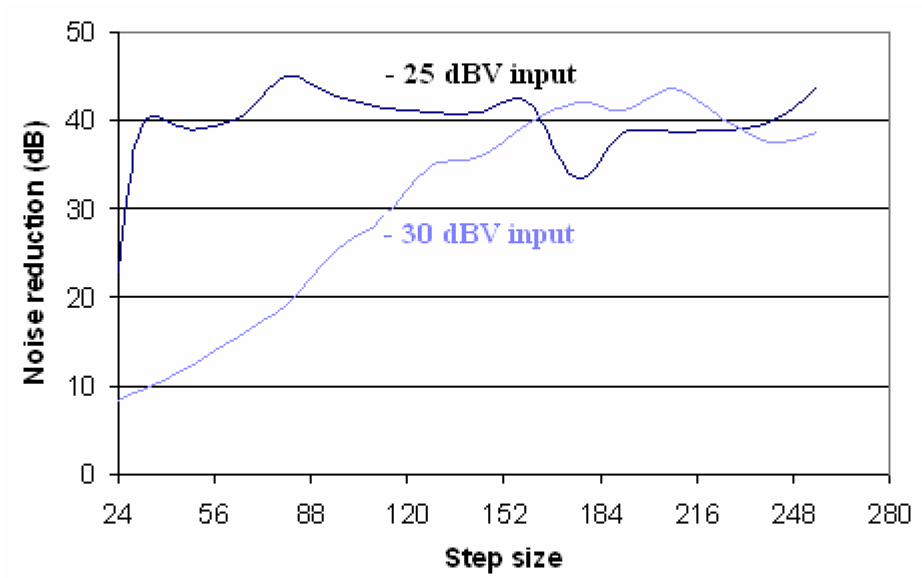


Figure 4.9: Noise reduction as a function of intensity of noise and step size.

Noise reduction as a function of frequency of noise is as shown in Table 4.1. Adaptive noise is same as actual noise and the filter order is 48. The noise reduction is maximum (45 dB) at a frequency of 500, and 600 Hz and is minimum (0 dB) at 200 and 400Hz. For adaptive noise as a broadband noise and actual noise as a tone over 200 Hz-1.5 kHz, noise is reduced up to 10 dB at a couple of frequencies depends on the filter order and step size, and at other frequencies, noise gets amplified. Variation in noise reduction depends on the secondary path transfer function.

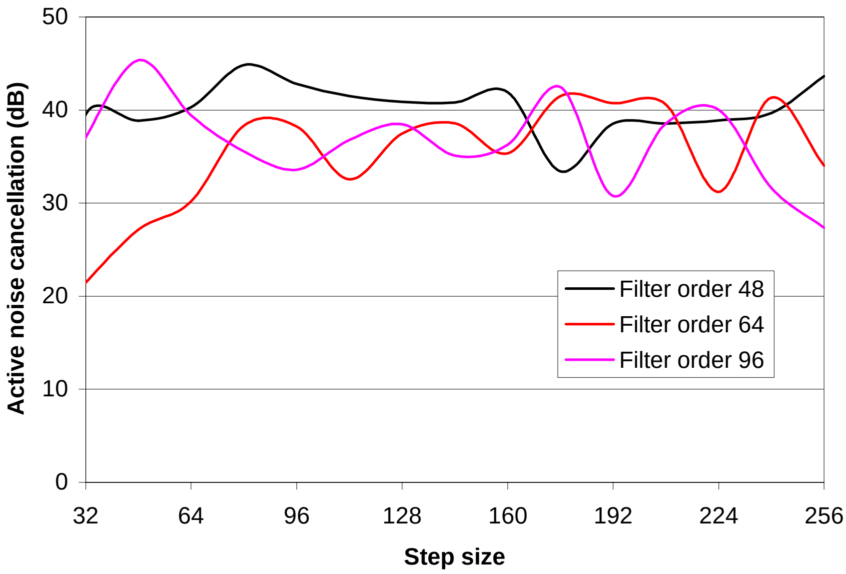


Figure 4.10: Noise reduction as a function of filter order and step size.

Table 4.1: Noise reduction as a function of frequency of noise.

Frequency (Hz)	Passive noise attenuation (dB)	Passive + active noise cancellation (dB)
200	5.73	0.02
300	3.12	23.41
400	0.93	0.04
500	1.72	45.44
600	1.57	45.85
700	3.88	20.28
800	7.89	27.58
900	8.2	11.55
1000	11.8	40.74
1100	15.29	28.3
1200	20.09	25.88
1300	20.2	22.24
1400	18.26	43.79
1500	18.56	27.46

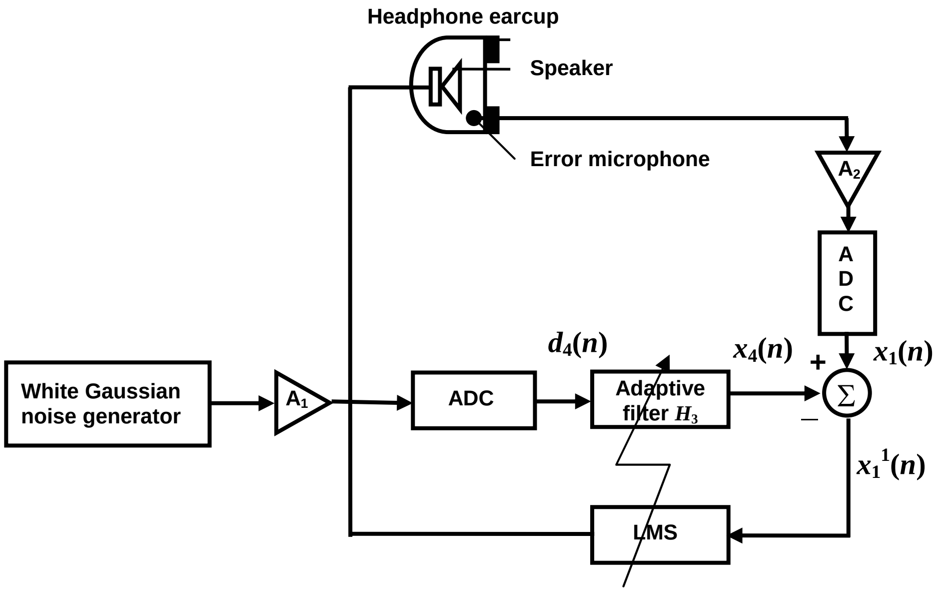


Figure 4.11: Experimental set-up for the offline secondary path modeling

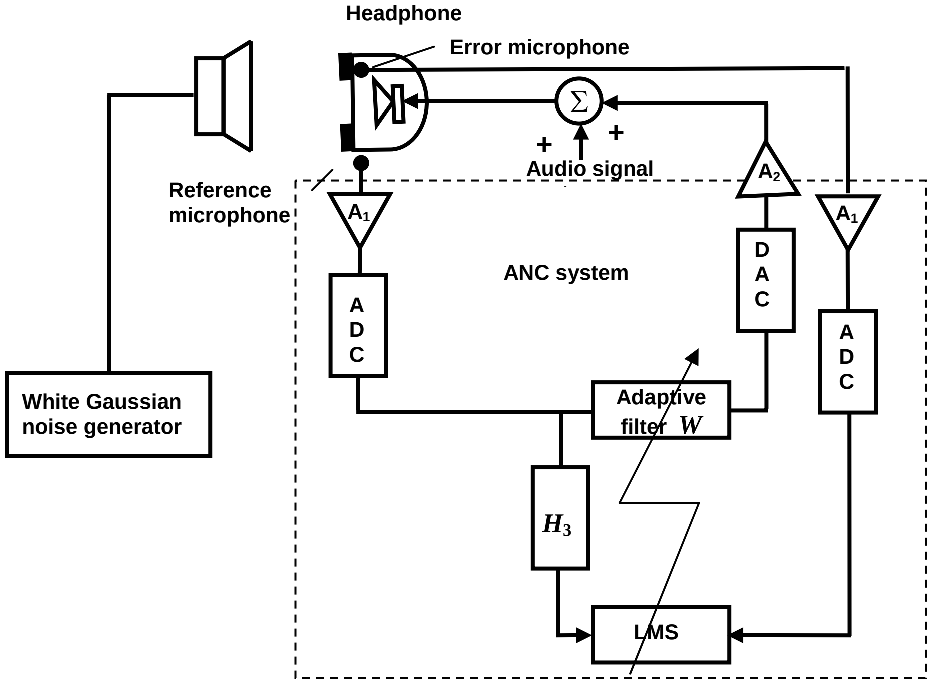


Figure 4.12: Active noise control using FXLMS algorithm

The secondary path effects can be compensated by FXLMS algorithm discussed in Chapter 2. The experimental set-up for modeling secondary path is as shown in Figure 4.11. A white Gaussian noise from a signal generator is given as input to the canceling speaker as well as analog input to the one of the ADCs as reference input. Error microphone output is given to the other ADC through a pre-amplifier as error input [15].

Response of the adaptive model is computed as,

$$x_4(n) = \sum_{k=0}^{M-1} d_4(n) h_{3n}(n) \tag{4.1}$$

Filter coefficients of the adaptive filter are updated according to

$$h_{3n+1}(k) = h_{3n}(k) - \mu x_1(n) x_1^1(n-k) \tag{4.2}$$

where,

$$x_1^1(n) = x_1(n) - x_4(n) \tag{4.3}$$

The coefficients of the adaptive filter, H_3 are saved and used in the noise canceling mode as shown in Figure 4.12. Anti-noise is computed using the adaptive filter W as discussed in Section 2.5.2.

The results lend to the following observations.

1. The average noise reduction, with tones swept over a frequency band of 200 Hz - 1 kHz is 25 dB using LMS algorithm, when adaptive noise is same as actual noise.
2. The noise reduction is not significant if the filter is tuned for a broadband noise. Noise reduction occurred at a couple of frequencies and at other frequencies noise gets amplified.
3. The performance of the FXLMS algorithm for broadband noise is similar to LMS algorithm, for broadband noise cancellation.
4. The reason for less noise reduction could be

   (i) The error path group delay (delay in anti-aliasing filter, smoothening filter, FIR filter, and delay in the electronic circuit) was about 1.5 ms and that of the primary path is about 100 μs. Hence noise canceling system cannot be causal, which is a primary requirement for broadband noise cancellation [16], [17].

   (ii) In order to achieve an error path group delay shorter than acoustic path, it is necessary to reduce the group delay of the electronic components [18].

   (iii) In the experiments, the sampling rate was 48 ksa/s and 24 ksa/s. The group delay of the anti-aliasing filter, smoothening filter was very high (1.45 ms). If much higher sampling rate is used, the delay in the filter will be smaller.

   (iv) With higher sampling rate, the computation time available for the processor will be reduced, but this imposes harder real-time constraints in the real-time processing.

   (v) The algorithm was written in a fixed point format which may impose rounding errors in the tuning of filter coefficients.

(vi) If a floating point processor with high processing speed is used, the noise reduction may be increased.

(vii) Implementation of the algorithm in assembly language may improve the computation time, and the filter order can be higher, which may result in better noise attenuation.

Chapter 5

SUMMARY AND CONCLUSION

5.1 Summary

A noise canceling technique based on feedforward adaptive filtering has been proposed, for noise cancellation in audiometric headphone. The noise reduction for the environmental in the earcup of the headphone should be around 40 dB over the frequency range of 125 Hz to 8 kHz, to use it along with audiometer without the need for an acoustically isolated cabin. In audiometry, the stimulus is applied for short duration with long inter-stimulus pauses. In the proposed method, the adaptive filter is tuned during the inter-stimulus interval and after that the coefficients are treated as adapted and frozen. Since the anti-noise, the output of the adaptive filter, is based on the reference noise input, test stimulus will not be affected.

A white Gaussian noise was taken as the reference environmental noise during the adaptation phase. Simulations were carried out with the same noise, as well as tone, music, and train noise as background interference during stimulus presentation. Passive transmission through the earcup was modeled as 4^{th} order Butterworth filter. Noise reduction was observed for all types of background interferences, and depended on filter order, and step size, and best attenuation values ranged over 20-30 dB for adaptive filter order of 48. Attenuation characteristics of three headphones were experimentally measured and for one of the headphones, transfer function was derived using the LMS algorithm.

Real time implementation included study of debugging interface and actual implementation of ANC. ADSP BF533 (16-bit, fixed point processor from Analog Devices) based DSP board Blackfin 533 EZ Kit Lite was used to implement ANC system. LMS and FXLMS algorithms were implemented. Results were plotted and analyzed for various noises.

5.2 Conclusion

Simulated results supported the proposed method, i.e. if the filter is tuned for a broadband noise, it is sufficient enough to cancel the noise provided the filter order is high enough. The noise reduction depends on filter order, step size, adaptation time, and type of noise. In real-time implementation, the average noise reduction with tones swept over a frequency band of 200 Hz-1.5 kHz as noise is about 25 dB.

With real-time implementation, noise reduction is not significant for a broadband noise. The noise reduction took place only over narrow bands up to a maximum of 10 dB, and at other frequencies the noise got amplified. The filter order may not be sufficient enough for broadband noise reduction. Implementation of the algorithm in assembly language may reduce the code size and increase execution speed; and hence may permit us to use higher filter order. Because of large group delays in the input anti-aliasing filter, and the feedforward adaptive filter, the delay in the noise cancellation path is larger than the acoustic delay in the earcup, making the noise cancellation ineffective for non-periodic interferences. DSP processor with high sampling rate and high processing speed can reduce the electrical group delay. Further, implementation of the algorithm on a floating point processor may improve the ANC.

5.3 Suggestions for future work

The algorithm implementation should be revised to reduce overflow which contribute to reduce the ineffectiveness. Implementation with higher sampling rate needs to be implemented for effective noise cancellation. Once adequate noise reduction is achieved, the NCH can be interfaced to an audiometer for further testing. Placing of the error microphone inside the earcup will change the frequency response of the headphone, and hence audiometer may be calibrated accordingly.

APPENDIX

Noise canceling headphones developed by various manufacturers is listed in the following table. Generally the kind of method used for ANC and performance of NCH is not available.

Sr. No	Manufacturer	Model	Cost (US $)	Specifications and features
1	Aiwa	HP-CN5	50	
2	Brookstone	Foldable Noise Reducing Headphone	99	
3	Bose Corp.	QuietComfort	299	
4	Coby	CV190	15	
5	Creative Labs	HN-505	50	15 dB at 300 Hz
6	Cyber Acoustics	ACM-80050	50	
7	JVC	HANC100	46	
8	Koss	QZ5	200	Average noise attenuation is 10dB in the range 30-1000 Hz and the maximum attenuation is 15 dB from 40 to 200 Hz.
9	MAX Corp.	MAX190400	45	
10	Noisebuster	NB-FX	69	ANC is in the range of 20-1500Hz and maximum attenuation is 15 dB between 150-300Hz.
11	Panasonic	RP-HC70	28	
12	Philips	HN050	60	
13	Plane Quiet	Latitude	30	ANC about 12 dB
14	Plane Quiet	NC-6	80	ANC about 17 dB
15	Sennheiser	PXC 150	74	
16	Sennheiser	PXC250	269	The noise attenuation is around 25 dB up to 1200 Hz.
17	Sennheiser	PXC 300	180	
18	Solitude	Noise canceling Headset	200	Noise reduction is about 18 dB.
19	Sony	MDR-NC6	38	
20	Sony	MDR-NC20	68	NCH features the reduction of outside noise by more than 10dB.

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