Master Thesis
Electrical Engineering
May 2014

Subband Adaptive Filtering Technique employing APA For Stereo Echo Cancellation over Audio Signals

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This thesis is submitted to the School of Computing at Blekinge Institute of Technology in partial fulfilment of the requirements for the degree of Master of Science in Electrical Engineering. The thesis is equivalent to 20 weeks of full time studies.

This Master Thesis is typeset using $\LaTeX$

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Abstract

The world today is relying on hands free devices and teleconferencing equipments but the major problem with the use of these systems is creation of acoustic echo at the time of conversation. The quality of information to be transmitted and received is degraded due to the echo. Hence, echo cancellation has come up as the most interesting as well as challenging part in any communication system. Echo is the degraded or delayed version of an original signal which comes back to the source after successive reflections from the surroundings. The removal of echo without degrading the quality of an original signal is a challenge of research in present time.

Echo cancellation in case of voice communication is removal of echo, improving the quality of an original voice signals. In our thesis we have focused on the acoustic echo cancellation in closed room with the help of adaptive filters. Stereo echo cancellation is a key technique for solving the echo problem in modern teleconferencing systems. There is a fundamental problem in stereo echo cancellation, the misalignment between the converged filter weights and the real room impulse responses. Most existing works explain the misalignment problem by a conceptual stochastic equation, and no studies have investigated the closed form of the final weights in a stereo echo canceller.

In our thesis we have proposed a subband adaptive modeling approach to develop an effective system of stereo echo cancellation with the help of better adaptive filtering techniques. We have adopted the subband adaptive modeling because the fullband adaptive modeling technique may not be able to characterize the electro-acoustic performance of a Multichannel Compression Hearing Aid (MCHA). We also employ the design of DFT filter banks for the analysis and synthesis of the signal and in the place of conventional NLMS or RLS algorithm; we have rather employed the APA (Affine Projection Algorithm) for adaptive modeling in each subband. The performance of the subband adaptive modeling is simulated under various conditions in MATLAB and the results are analyzed. The performance of the subband adaptive modeling system is better and effective as compared to the fullband adaptive modeling system. The results for various input and noise signals are plotted and shown in result section of this report.

Keywords: Stereo Echo Cancellation, NLMS, RLS, MCHA, APA, Subband Adaptive Modeling.

Acknowledgements

We would like to express our special appreciation and thanks to our advisor Mr. Magnus Berggren you have been a tremendous mentor for us. We thank you from the bottom of our heart for encouraging our research and helping us at every point towards the completion of the thesis. We would also take this opportunity to thank Dr. Sven Johansson who has been a constant source of information and motivation.

We express our gratitude towards department of Applied Signal Processing, BTH, Karlskrona for providing the opportunity to present our thesis.

A special thanks to our family for being there with us in the time of need. Words cannot express how grateful we to our mother, father for all the sacrifices they have made on our behalf. These are your blessings and prayers that have sustained us this far.

Also this thesis would not had been complete without the blessings and bestowed strength, ability from the almighty god.

Finally we would like to thank one and all who have been a part of our journey towards the completion of the thesis.

Bhanudurga Sri Satya Simha Edamakanti
Vishnu Chaitanya Gundu

Dedicated to our families and friends

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Acronyms

SEC Stereophonic Echo Cancellation

FIR Finite Impulse Response

IIR Infinite Impulse Response

AEC Acoustic Echo Cancellation

RIR Room Impulse Response

APA Affine Projection Algorithm

PSTN Public Switched Telephone Network

FE Far End

NE Near End

MMSE Minimum Mean Square Estimation

RLS Recursive Least Squares

ERLE Echo Return Loss Enhancement

MIS Misalignment

LMS Least Mean Squares

NLMS Normalized Least Mean Squares

Introduction

Chapter 1

Introduction

1.1 Stereo Echo Cancellation

Stereo or stereo-phonic sound is a method in which the sound is reproduced creating an illusion of directionality and audible perspective. It is a sound reproduction system that uses two or more separate channels to give more natural distribution of sound. It is usually done by using two or more separate microphones to feed two or more loudspeakers through separate channels in order to give a spatial effect to the sound. It is usually applied to a system having 2-channel, 2-speakers as shown in figure 1.1.

Stereo sound systems can be divided into two forms: The first is ”true” or ”natural” stereo in which a live sound is captured, with any natural reverberation or ambience present by an array of microphones. The signal is then reproduced over multiple loudspeakers to recreate as closely as possible, the live sound. Secondly ”artificial” or ”pan-pot” stereo, in which a single-channel (mono) sound is reproduced over multiple loudspeakers. By varying the relative amplitude of the signal sent to each speaker an artificial direction (relative to the listener) can be suggested. The control which is used to vary this relative amplitude of the signal is known as a ”pan-pot” (panoramic potentiometer). By combining multiple ”pan-potted” mono signals together a complete, yet entirely artificial sound field can be created.

The echo cancellation and acoustic echo cancellation can be implemented by an adaptive filter in which the filters have different adaptive algorithms [1]. Using adaptive algorithms the echo cancellation or suppression in the transmission room and receiving room using some filtered weights can be implemented as shown in figure 1.1.

The echoes are different types in which those can be explained further in next chapter briefly. These echoes are cancelled or suppressed by using adaptive algorithms like NLMS, RLS and APA.

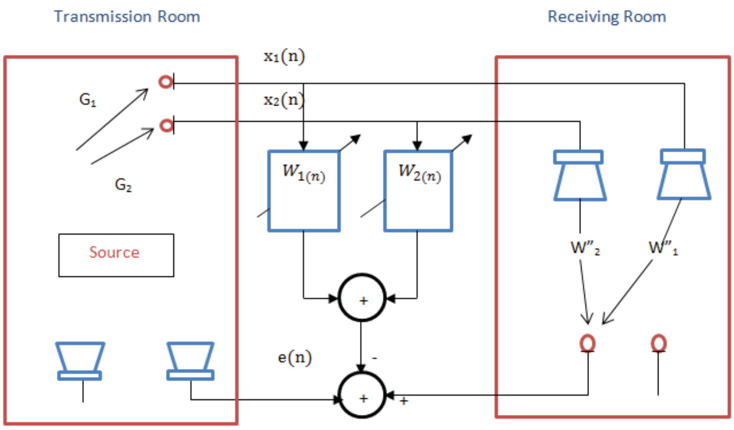

Figure 1.1: Stereo Phonic echo cancellation

1.2 Literature Review and Background Work

Echo cancellation has always been an active area of research. Echo cancellation schemes have been developed in recent times but echo suppressors were developed long before in 1963 P. T. Brady et al [2] studied the echo suppressor design in telephone communication which described an echo suppressor consisting of a single voice operated switch which was located at each end of the 4- wire toll connection. However, speech chopping would take place if effective echo suppression was required in the suggested design.

In 1996 Takatoshi Okuno et al [3] gave the concept of acoustic echo cancellation by short time Fourier transform and cross spectrum processing. The time frame averaged cross spectrum was taken between the input signal and the error signals which also consisted of the echo signals through the echo path. The transfer functions for the acoustic echo paths were estimated from the ratio of averaged cross spectrum and power spectrum of the input signal and it was confirmed by computer simulation that the echo signals can be suppressed as averaging process goes on.

The application of adaptive filters in echo cancellation became an area of research for many researchers and scientists. J. Benesty et al in 1995 [4] explained the difference between the mono and the 2-channel system and also the behavior of 2 channel adaptive algorithm compared to same algorithm in the mono channel case. Also Christina Breining et al in 1999 [5] implied the high order adaptive filtering for the echo cancellation process and also suggested methods to reduce the computational complexity. Shoji Makino in 2001 [6] presented an overview on the stereophonic acoustic echo cancellation and also studied recent solutions in the same field. He also presented an overview on various adaptive algorithms such as Normalized Least Mean Square (NLMS) and Recursive Least Square (RLS) algorithm for echo cancellation and also studied the subband modeling for echo cancellation.

At the place of a single band for adaptive filtering, concept of multi-band in 1992 Andre Gillore et al [7]studied the adaptive filtering in subband with critical sampling. Phillip. L et al in 1995 [8] gave the concept of increased bandwidth analysis filters in oversampled subband acoustic echo cancellers. B. Farhang et al 1997 [9] studied the adaptive filtering in sub-bands considering the design issue and experimental results for the acoustic echo cancellation.

Stereo echo cancellation is a wide area and offers huge scope of research. Many researchers have dedicated their research and time to the field of echo cancellation [10][11]. The problem of echo in any system degrades the quality of the system and also the information to be transmitted is also degraded. Adaptive filters have been used in past for acoustic and stereo echo cancellation but most of them were single band adaptive filtering schemes. The algorithms used for echo cancellation were mostly NLMS (Normalized Least Mean Square Algorithm) and RLS (Recursive Least Square Algorithm) [12]. Thus we have focused our research on subband adaptive filtering which has not been explored much and also we intend to use a new and better algorithm i.e., Affine Projection Algorithm (APA) [13][14] which has less computational complexity than conventional algorithms and is also faster than most other algorithms. Here we find our research questions as what are the effective methods to improve the stereo echo cancellation over audio signals through various adaptive filtering models? Can an Affine Projection Algorithm (APA) be effectively used for the adaptive modeling of subbands? How does a subband adaptive filtering can overcome previous adaptive modeling technique? The stereo echo cancellation depends upon speaker position and transmission room so because of the non-stationary nature of the speaker, the stereo echo cancellation is a key technique adapted for solving such type of problems in modern teleconferencing system. There is a fundamental problem in stereo echo cancellation, the misalignment between converged filter weights and real room impulse responses. Most existing works explain the misalignment problem by a conceptual stochastic equation, and no studies have investigated the closed form of final weights in a stereo echo canceller. Due to the presence of independent compression channels, the conventional fullband adaptive model might not adequately characterize the performance of a Multichannel Compression Hearing Aid (MCHA). Here, we propose a subband adaptive modeling approach to characterize the electro-acoustic performance of a MCHA. In the proposed model we have used uniform oversampled DFT filters for analysis and synthesis and Affine Projection Algorithm (APA) for adaptive modeling in each side band.

1.3 Research Motivation and Thesis Outline

In our day to day life we are hearing so many sounds like music, voices over telephone lines and mobile devices. The common problem in every hearing electronic device is echo. These echo cancellation was studied and improved with different methods by many researchers. In our research we proposed a method to cancel the echo which can be implemented in many research works. The objective of this research is to propose a method which is more effective to implement in echo cancellation using an adaptive filter.

This research can give a glance about how to remove echoes in a sound reproduction system and helps to improve the sound quality using different adaptive filters. The first chapter introduces about Stereo Echo Cancellation and its types. In second chapter a short review and background work. In third chapter it describes about echo and types of echoes. In fourth chapter it describes about MCHA. In fifth chapter we described about impulse response, Room Impulse Response and Room Image Model. In sixth chapter the adaptive filters and various types of filters which are already existed are described thoroughly. In seventh chapter we have found various adaptive filters its convergence rate and errors and explained briefly in this chapter. In eighth chapter the APA and advantages of APA is described in depth. Coming to ninth chapter we have described about ERLE and misalignment. After this part the subband adaptive filtering technique is described in tenth chapter. Following results and conclusion about future work with references are mentioned in this paper.

Echo and Types of Echo

Chapter 2

Echo and Types of Echo

2.1 What is Echo?

Echo is the delayed and distorted version of the original speech which is reflected back to the source. Echo can be acoustic as well as electrical. Electrical echo are mainly due to impedance mismatch at different points across the transmission channel. Electrical echo are very common in Public Switched Telephone Network (PSTN), mobile etc. Acoustic echo may be due to single reflection or due to multiple reflections. The echoes which suffer a number of reflections are the most delayed and are most responsible in degrading the quality of sound at receiving point [10] and thus these echoes must be avoided or cancelled for smooth communication and at this time application of Acoustic Echo Cancellers (AEC) comes in. AEC are required to remove the effect of echo and make the communication smooth and keeping up the quality of communication.

Echo is classified as [15]:

1. Hybrid echo
2. Acoustic echo

2.2 Hybrid Echo

This type of echo is mainly seen in Public Switched Telephone Network (PSTN).These hybrid echoes arise when signals reflect at the point of impedance mismatch in a circuit. This mismatch causes the transmitter signal to appear on receiver signal. This is generally found where the telephone local loops are 2-wire circuits and transmission line is a 4-wire circuit can see in figure 2.1. Each Hybrid produces echoes in both directions, though the far-end echo is usually a greater problem for voice band. For a local call in the telephone network, there is no significant echo because there is no noticeable impedance mismatch on the connecting 2-wire local lines and also because the distance is comparatively small which result is the low delay echo which act as a slight amplification.

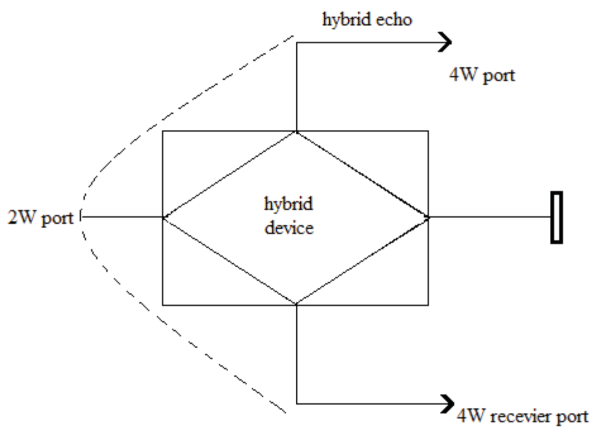

Figure 2.1: Hybrid Echo

In long distance telephone network, at each end of the network, a 2-wire subscriber line must be connected to the 4-wire line at the telephone exchange. The device which serves the purpose of making the connection is known as the Hybrid. Thus the hybrid is usually a 3-port bridge circuit. In such case the hybrid is perfectly balanced, there is no echo but each of the hybrid is designed to serve a number of subscriber lines and which are not equal in length and do not have the same impedance characteristics. Therefore, achieving the perfect balance is not possible and some of the energy becomes convoluted into itself and produces the echo.

Echo suppressors have been developed to suppress the line echoes in the telecommunication networks. Echo cancellers de-couple the 4-wire transmit port when the signal detectors determine that in the network there is a far-end signal at the 4-wire receive port without any near end signal at the two wire receive port. In public switched telephone network (PSTN) any imbalance between the 4/2 wire bridge circuit causes some of the signal energy of the four-wire circuit to be bounced back towards the transmitter and thus causing an echo. If the range of the echo is more than few milliseconds then the echo becomes noticeable and then it disturbs the communication by degrading the quality of communication.

2.3 Acoustic Echo

Acoustic echo takes place when an audio signal is reflected in a surrounding such as rooms, halls which results in original signal and also the reflected signals. Acoustic echo plays a vital role in degrading the quality of communication [16]. Acoustic echo cancellation employs the use of adaptive filtering techniques. The echo is reflected from different surfaces like walls, roofs etc., and after reflection they travel through different paths. These echoes can result in combination of direct acoustic coupling from different surfaces and may be picked up by the microphone. The worst case of acoustic echo occurs when it causes a howling due to the reason that a significant amount of sound energy which is transmitted by the loudspeaker is again received back at the microphone and becomes a part of the feedback network. Howling is mainly a phenomenon which occurs in auditoriums. Acoustic echo results in great discomfort to the users as their own voice is heard by them during communication. The acoustic echo can be said to be a time delayed image of an original signal.

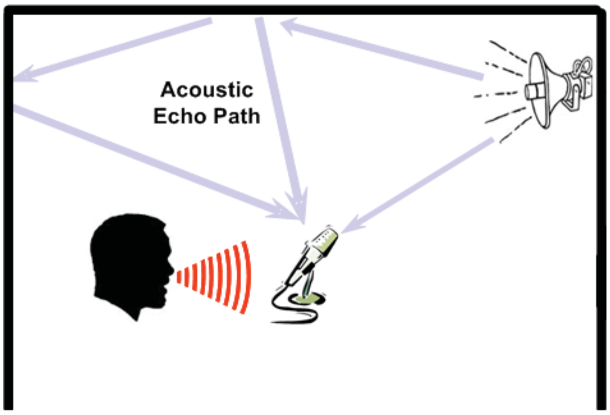

Figure 2.2: Acoustic echo paths

The acoustic signal from the speaker may travel to the microphone directly or it may reach back after single or multiple reflections and the path which the signal takes after reflection is known as acoustic echo path which is shown in figure 2.2.

2.4 Why the Phenomena of Echo Takes Place?

Echo is nothing but the signals which are received after successive reflections from the surroundings. It is the distorted and the delayed version of the original signal. Now the question arises how the echo is produced? When an acoustic signal is produced by a microphone the signal sometimes gets distracted and suffers reflections from the surroundings such as walls, roofs, etc. Now these reflected signals reach the destination after a delay of sometime before the original signal and this phenomenon is known as echo.

The phenomena of echo can be easily described with the help of following figure 2.3.

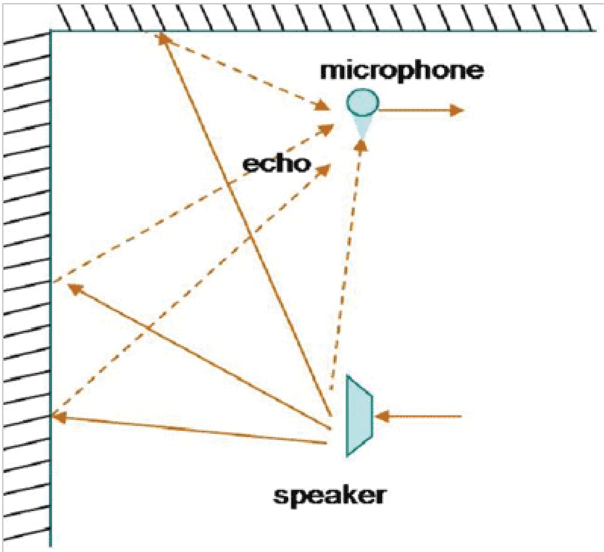

Figure 2.3: Phenomena of echo

The signals from the speaker sometimes reach the microphone directly, sometimes from a single surface and sometimes from multiple surfaces thus produce echo. This echo is the delayed and distorted version of the original signal. The quality of the original signal is degraded because of the echoes in the signal.

Multichannel Compression Hearing Aid (MCHA)

Chapter 3

Multichannel Compression Hearing Aid (MCHA)

Multichannel signal processing offers several advantages over single channel signal processing. Multichannel compression system can accommodate variations in the hearing threshold and also the dynamic range by providing different amounts of gain across channels. Multichannel also has an advantage of wide dynamic range compression which improves the speech intelligibility over the single channel signal compression and also provides greater audibility at low input levels. Multichannel compression systems can also be designed for less vulnerable to the background noise as compared to single-channel compression [17]. Multichannel compression technique also has the advantage of effective noise reduction and availability of acoustic feedback technology. The gain and output of every channel is controlled by its own set of controllers. Thus the multichannel compression offers a wide range of controllability.

Despite the above mentioned advantages, the increasing number of independent channels which does not always have the better listener performance over single channel [18]. In a multichannel compression hearing aid there are design complications for which atleast eight channels are required to obtain the full advantage of multichannel compressing. As the number of channels goes on increasing, the processing complexity also increases which further increases the amount of time required to process the signals. This time delay in the processing of a signal is known as group delay and this produces a delay in an amplified signal respective to the direct sound path. Also increasing negative effects of multichannel compression have been found out recently, but these negative effects are only visible when the compression ratios are high (i.e.,> 3), but for low values of the compression ratio the positive effect outnumber the negative effect.

Multichannel compression hearing aid also improves the audibility of low level signals and ensures the comfort for high level signals. The numbers of compression channels also play a vital role in restoration of audibility; compression has been regarded as a necessary technology in hearing aids as it improves the chances of restoring audibility and comfortable levels of listening with reduced dynamic range.

Room Impulse Response

Chapter 4

Room Impulse Response

4.1 Impulse Response

Impulse response is the output of a dynamic system when a brief input (impulse) is applied to it. It refers to the reaction of a dynamic system is the response for some external change. Impulse response is a way of describing the reaction of a system as a function of time or a function of any other variable. In acoustic or audio application, the impulse response allows to capture the acoustic characteristics of a particular location such as room, conference hall. It refers to the recording of a reverberation that is caused by an acoustic space when an ideal impulse is played.

Impulse responses are classified as Room Impulse Response (RIR), Room Image Model (RIM).

4.2 Room Impulse Response (RIR)

A room impulse response is the graphical image of a loudness of sound along with its reflections with respect to time. Room impulse response is used to simulate the echo mode for speech signals. The room impulse response consists of original sound signal, its reflection and reverberation. Thus to understand room impulse response the phenomena of reverberation is to be considered [18].

4.3 Room Image Model (RIM)

The figure 4.1 shows the image model of a room and different signals and their paths in the room [19]. The sound source S is located near the wall. Let us assume that the direct signal and the reflected signal meet at point D. The length of direct path can be calculated from known locations of source and destination. An image S' of source S is assumed at equal distance from the wall which is equal to the distance of original source from the wall. From the properties of triangles, the triangle SRS' is isosceles and the path length $RD + SR$ will be same as S'D. Thus to calculate the distance reflected path we construct an image of the source and calculate the distance between the image and destination. If the computing is done with one image that means there is one reflected path. The image model with one reflection path is shown in figure 4.1. Where S is the source, D is the destination and S' is an image of the source at other side of the wall.

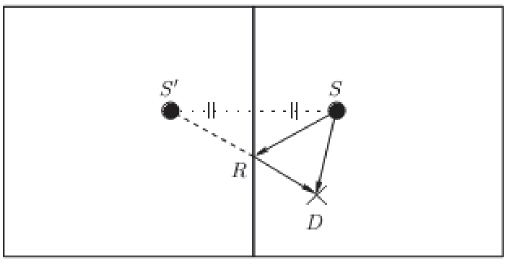

Figure 4.1: Single reflection path

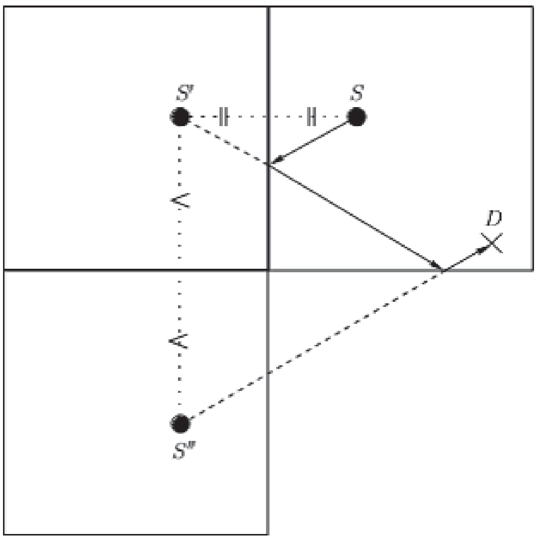

Figure 4.2: Double reflection path

The figure 4.2 shows the distance which has been computed using two images [19]. The length of the path with two reflections can be calculated with distance S"D. The numbers of images involved in calculations are equal to the number of reflections in the path and also the strength of reflections is nothing but the path length and the number of images used.

4.4 Reverberation

The reverberation takes place due to the reflection of sound [20]. The sound from the source travel in the form of wavefronts and these wavefronts suffer reflections from the surroundings like wall in a room and they finally superimpose on the microphone. The phenomenon of reverberation is shown in the figure 4.4. The figure 4.3 shows the direct path and a single path of reflection. The direct path signal reaches the microphone earlier than the signal which suffers the reflection and thus actual signals intelligibility decreases.

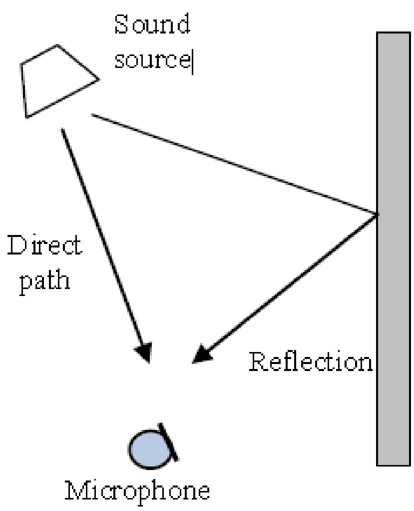

Figure 4.3: Direct path and single reflection path of echo

Besides single reflection, sound may suffer multiple reflections from different walls of a room and these reverberations are more random due to single reverberation sound or the early reverberation [17] and are referred to as "late reverberation" [20]. The set of well-defined and directional. Reflections which are found after a short period of actual sound are directly related to the shape and size of the room and are known as "early reverberations". Figure 4.4 shows the phenomena of multiple reflections.

The figure shows multiple reflections a, b, c, d, e apart from direct path A to B. Also an example of room impulse response for early and late reverberation is shown in figure 4.5.

The first 2-4 reflections are termed as early reflections as they occur comparatively before than the other reflections which are known as the late reflections and are more in their numbers.

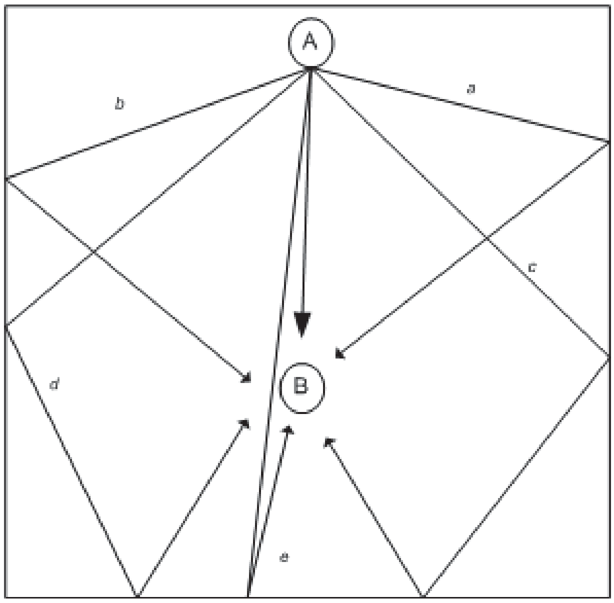

Figure 4.4: Phenomena of Multiple Reflections

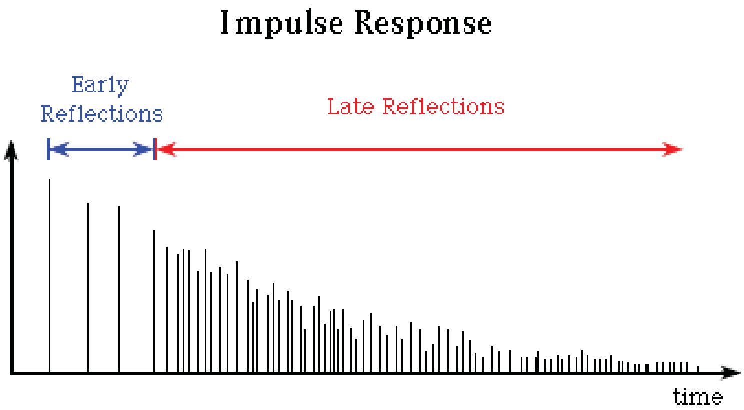

Figure 4.5: Room impulse response

Adaptive Filters

Chapter 5

Adaptive Filters

5.1 Filters

Filters are used as a method of signal processing with an objective to free the signal from an unwanted components or feature. Filtering is used for complete or partial suppression of some of the qualities or features of the signal. The suppression may be of background noise or interfering signals and also it can be the removal of unwanted frequencies. The filters may be linear or non-linear, analog or digital, discrete time or continuous time, Infinite Impulse Response (IIR) or Finite Impulse Response (FIR).

5.2 Adaptive Filters

An adaptive filter is the filter in which the transfer function adjusts itself according to optimization algorithm which is driven by an error signal. The adaptive filters make use of feedback in the form of an error signal to modify its transfer function to be in accordance with changing parameters. An adaptive filter is used where the time-invariant filter cannot satisfy the condition or fixed specifications are unknown. An adaptive filter is a non-linear filter because its characteristics are fully dependent on the input signal [21]. Also the adaptive filters are time varying as their parameters are constantly changing in order to meet the performance criteria. Block diagram of an adaptive filter is shown in the figure 5.1 below with input signal as *x(n)*, *y(n)* is the output of an adaptive filter, *e(n)* is the estimation of an error signal, *d(n)* is the desired signal of finite impulse response filter. The aim of an adaptive filter is to determine the difference between desired output and an adaptive filter output. The error signal is again fed back to an adaptive filter and its coefficients are changed algorithmically in order to minimize this difference.

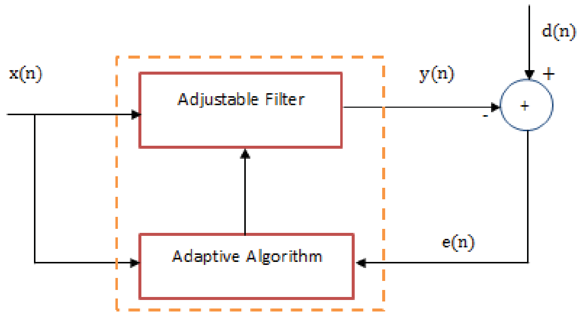

Figure 5.1: Block diagram of an adaptive filter

5.3 Use of Adaptive Filters for Noise Cancellation in Speech Signals

Use of adaptive filters for noise cancellation is shown in figure 5.2 [22]. When the audio signals are to be transmitted in a noisy atmosphere, adaptive filters are used. Two inputs are required for noise cancellation with the help of adaptive filters. One of the input contains the noise contaminated signal and the other signal contains the reference signal which is a noise related to the input signal in some way. The system is so designed that in filters the reference noise signals to make it similar to the signal provided at main input and now this filtered version is subtracted from the main input. This process removes the noise and leaves behind the exact speech signal [23]. However in practical implementations of the system, the noise is not fully removed but it is reduced considerably.

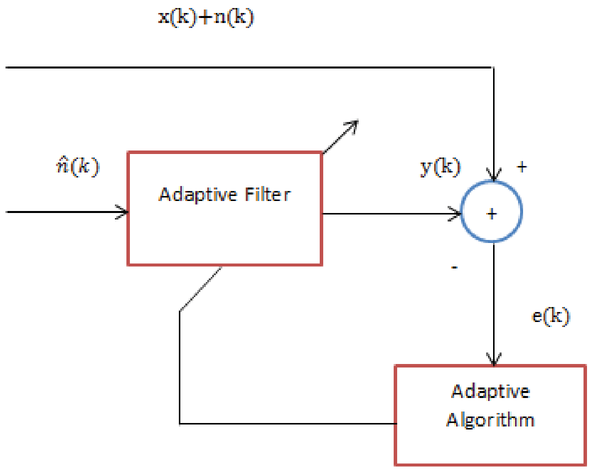

Figure 5.2: Adaptive filter used for noise cancellation

5.4 Use of Adaptive Filters for Interference Cancellation

Adaptive filters can be used as a method to cancel unknown interference in the primary signal. The primary signal is used as the desired response for the adaptive filters and a reference signal is used as the input to the adaptive filter. A signal under consideration *s(k)* is mixed with a noise signal *n(k)*. The noisy signal *n(k) + s(k)* is then used as the reference signal for the adaptive filter. The input of the adaptive filter $\hat{n}(k)$ which is correlated to *n(k)*. The filter adjusts its coefficients in such a way that the output *y(k)* of the filter approximates x(k) forcing the error signal e(k) to be like signal s(k).The figure 5.3 shows the adaptive filters application for interference cancellation [22].

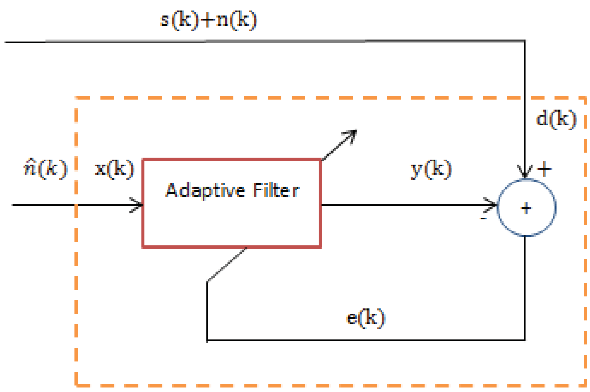

Figure 5.3: Adaptive filter used for interference cancellation

5.5 Use of Adaptive Filters for System Identification

System identification is defined as the ability of an adaptive system to predict a FIR filter that will best reproduce the response of another system. This method is efficient only when the frequency response of the new system matches with the FIR filter. It may never be capable of giving a null output but it may reduce it by converging to an optimum weight vector. The frequency response of the FIR filter is best approximation of the unknown system and never be exactly equal to the system. The system identification using adaptive filter is shown in figure 5.4.

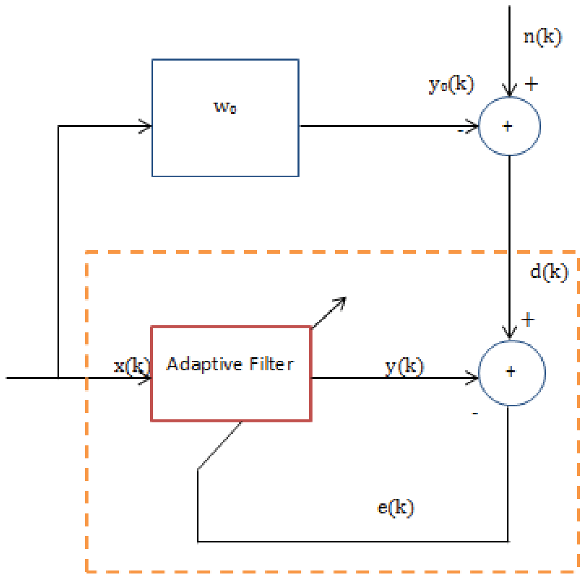

Figure 5.4: Adaptive filter used for system identification

5.6 Use of Adaptive Filters for Echo Cancellation

Adaptive filters can be effectively used to remove the acoustic echoes which are very common now a day in telecommunication systems [21]. Acoustic echoes are produced when an input and the output both operate in full duplex mode. The received signal has outlet through the loudspeaker and an audio signal is reflected through physical surroundings is picked up by microphone. This reflection of an audio signal through physical environment causes time delay and affects the original audio signal which results in reduced speech quality. The optimal output is itself an echoed signal that accurately surpasses the unwanted echo signal. This signal is then used to minimize the echo in return signal. A successful cancellation is one in which the echo is reduced effectively by the filter [24]. Various types of adaptive filtering algorithms are available for echo cancellation. These are listed below.

5.6.1 Normalized Least Mean Square Algorithm (NLMS)

Normalized Least Mean Square algorithm is widely used due to its computational simplicity. It can be used for applications such as echo cancellation and channel equalization. NLMS also has an advantage of high convergence rate and minimum steady state error. Despite of having these advantages the disadvantages of Normalized Least Mean Square algorithm cannot be ignored [25]. It requires comparatively more number of computations for

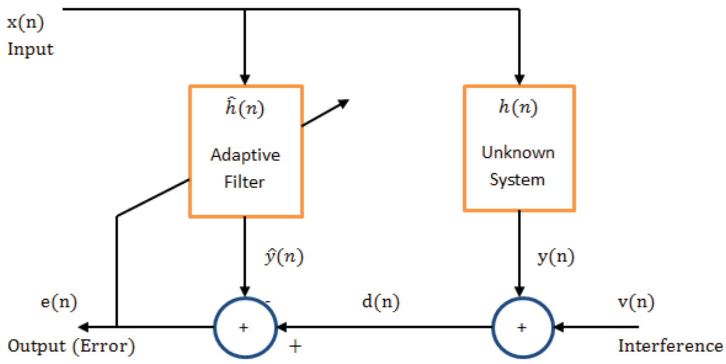

Figure 5.5: General block diagram for various adaptive filters

evaluation purpose than LMS algorithm. Also in case of NLMS the number of multiplications required is 3N+1 which is N more than LMS. The formula for convergence factor has been modified and is given as.

$$\mu = \frac{\beta}{||x(n)||^2} \tag{5.1}$$

Where, $\mu(n)$= step size.
β = Normalized step size ($0 < \beta < 2$).
And also the value of the weight factor can be derived from the equations given below.

$$\mathbf{w}(n+1) = \mathbf{w}(n) + \mu(n)e(n)\mathbf{x}(n) \tag{5.2}$$

or

$$\mathbf{w}(n+1) = \mathbf{w}(n) + \beta\frac{\mathbf{x}(n)}{||\mathbf{x}(n)||^2}e(n) \tag{5.3}$$

Use of the Normalized least mean square algorithm in a general adaptive filtering scheme is shown in figure 5.5.

The input $x(n)$ is fed to an adaptive filter (n) and simultaneously to an unknown system $\boldsymbol{h}(n)$. The output $y(n)$ of an unknown system is added with an interference $v(n)$ results with a function $d(n)$ and then it is subtracted from the output $\hat{y}(n)$ of an adaptive filter $\hat{h}(n)$. This difference is known as final output or an error $e(n)$.

5.6.2 Recursive Least Square Algorithm (RLS)

The recursive least square algorithm has an advantage of fast convergence rate and is also widely used in speech enhancement, echo cancellation and channel equalization. It is a simple adaptive and also time update version of pre-existing Weiner filter. In non-stationary environment the performance of RLS is not good as that of LMS and also due to being sensitive to the round-off error it leads to instability. In RLS algorithm the filter parameters are continuously updated with new data set without solving the matrix inversion. Recursive Least Square algorithm also has greater computational complexity as compared to the Normalized Least Mean Square algorithm as each iterations of the RLS algorithm requires $4N^2$ multiplications and $3N^2$ additions which makes the implementation of recursive least square algorithm very costly. The complexity and convergence delay in both NLMS and RLS algorithms are dealt effectively by another algorithm known as the Affine Projection Algorithm (APA). The main idea behind the Recursive Least Square filter is to minimize the value of cost function by properly estimating the filter coefficients $\boldsymbol{w}_n$ and also updating the filter as new data or new values are found. The error signal is $e(n)$ and the desired signal is $d(n)$. The RLS algorithm can be explained from general adaptive filtering scheme as shown in figure 5.5.

The error signal is the difference of desired signal and estimate $\hat{d}(n)$.

$$e(n) = d(n) - \hat{d}(n) \tag{5.4}$$

The computational data for the recursive least square algorithm is given below.
λ= Exponential weighting factor.
δ= Value used to start the inverse of Auto correlation at n=0.
i.e., $\mathbf{P}(0) = \delta^{-1}\mathbf{I}$

The calculation of estimation error is done by the equation given below x.
$\boldsymbol{I}$= Identity matrix.
$\boldsymbol{P}(n)$=Inverse of the Auto correlation matrix $\boldsymbol{R}_x(n)$, $\boldsymbol{g}(n)$=gain vector.

$$\mathbf{R}_x(n) = \begin{bmatrix} r_{xx}(0) & r_{xx}(1) & \cdots & r_{xx}(N-1) \\ r_{xx}(1) & r_{xx}(0) & \cdots & r_{xx}(N-2) \\ \vdots & \vdots & \ddots & \vdots \\ r_{xx}(N-1) & r_{xx}(N-2) & \cdots & r_{xx}(0) \end{bmatrix} \tag{5.5}$$

$$\mathbf{g}(n) = [g(0), g(1), .g(N-1)]^T \tag{5.6}$$

$$\mathbf{z}(n) = \mathbf{P}(n-1)\mathbf{x}(n) \tag{5.7}$$

$$\mathbf{g}(n) = \frac{1}{\lambda + x^T(n)z(n))}\mathbf{z}(n) \tag{5.8}$$

The calculation of an estimation error is done by the equation given below.

$$e(n) = d(n) - \mathbf{w}^T(n)\mathbf{x}(n) \tag{5.9}$$

Now the calculations of an adaptive filter coefficient and also the coefficients of an auto correlation matrix can be made by the following equations.

$$\mathbf{w}(n) = \mathbf{w}(n-1) + e(n)\mathbf{g}(n) \tag{5.10}$$

$$\mathbf{P}(n) = 1/\lambda[\mathbf{P}(n-1) - \mathbf{g}(n)\mathbf{z}^T(n)] \tag{5.11}$$

5.7 Comparison between various Adaptive Filtering Algorithms

Several algorithms like Least Mean Square (LMS), Normalized Least Mean Square (NLMS), and Recursive Least Square algorithms (RLS) are available to predict the filter coefficients for an adaptive filtering techniques but each algorithm has its own advantages and disadvantages. The Least Mean Square algorithm has a problem of stability while Recursive Least Square has greater complexity. A comparison of these algorithms is shown in table 5.2 [21] which compares these algorithms on various parameters such as Mean Square Error, Complexity and Stability.

S.No.	Algorithms	Mean Square Error	Complexity	Stability
1	LMS	$1.5 * 10^{-2}$	2N+1	Less Stable
2	NLMS	$9.0 * 10^{-3}$	3N+1	Stable
3	RLS	$6.2 * 10^{-3}$	$4N^2$	High Stable

Table 5.1: Comparison of various algorithms

For LMS algorithm the Stability is less compare to RLS. In LMS algorithm the complexity is 2N+1 for which Mean Square error will me more and comparatively a very high to RLS algorithm. For this the LMS is not preferred and the RLS algorithm has High Stability and has less Mean Square Error. The values for which each of the adaptive algorithms are tables as shown in table 5.1.

A comparison of different algorithms like Least Mean Square algorithm, Normalized Least Mean Square algorithm and Recursive Least Square algorithm has been presented in the table 5.2 [21]. The comparison is now based on signal to noise (SNR) improvement in dB. The Recursive Least Square and the Normalized Least Mean Square have maximum values of SNR for any value of noise variance. The sampling rate has been kept same for all the three algorithms and all three noise variance.

Noise Variance	Sampling Rate (kHz)	SNR Improvement LMS	SNR Improvement(dB) NLMS	SNR Improvement(dB) RLS
0.02	1.5	8.85	9.85	9.91
0.05	1.5	7.55	8.62	8.89
0.10	1.5	5.12	6.38	7.02

Table 5.2: Comparison of various algorithms based on SNR

The SNR Signal to Noise Ratio of NLMS and RLS are high for which we consider the RLS and NLMS algorithms are more preferable, because the more the SNR the better error free in any filtering techniques. When the Noise variance varies as shown in Table 5.2, the SNR of LMS is less compare to NLMS and RLS. For Noise Variance of 0.02 with equal sampling rate of 1.5 kHz, the SNR of LMS is low as 8.85 dB and for NLMS it is 9.85 dB also for RLS the SNR is 9.91 dB. RLS and NLMS are almost same, for this we have preferred NLMS and RLS are better adaptive filtering techniques to remove Echo cancellations.

The main disadvantage of LMS algorithm lies in a fact that it is very sensitive to scaling of its inputs. The Normalized Least Mean Square algorithm provides a solution to these problems by actively normalizing with the power of an input signal, this problem and this algorithm was popularly known as the Normalized Least Mean Square algorithm and both NLMS and LMS algorithms have good SNR. Thus they must be compared on some other parameters and they were compared on the basis of mean square error [22] as shown in table 5.3.

Algorithm	Iterations	Filter Order	Mean Square Error	Average Attenuation
LMS	7500	1025	0.001	-11.2435
NLMS	7500	1025	0.0004	-13.6812

Table 5.3: Comparison of LMS and NLMS algorithm

As compared to LMS and NLMS algorithms with same number of iterations 7500 and filter order 1025 the Mean square error and the Attenuation factor for the selected adaptive algorithms are varying with 0.001 and 0.0004 for which the NLMS has very less Mean Square error. Thus the NLMS is a better adaptive algorithm with good SNR and Mean Square Error.

Affine Projection Algorithm and its Advantages (APA)

Chapter 6

Affine Projection Algorithm and its Advantages (APA)

6.1 Affine Projection Algorithm (APA)

Affine Projection algorithm is the generalized form of Normalized Least Mean Square (NLMS) adaptive filtering algorithm. The Affine Projection algorithm is sought as the intermediate algorithm between NLMS and RLS algorithms. The complexity and performance of the Affine Projection algorithm lies between Normalized Least Mean square (NLMS) algorithm and Recursive Least Square (RLS) algorithm [26]. The APA employs multi- dimensional view of projections. Fast convergence rate is provided to high projection order in Affine Projection algorithm while slow convergence rate is observed for low projections. However, the Affine Projection algorithm has the same complexities as in case of Least Mean Square (LMS). The major advantage of Affine Projection algorithm is that it does not causes any delay in the input or output signals, thus making the affine projection algorithm suitable for acoustic echo cancellation. The Affine Projection algorithm is known to have less amount of computational complexity without compromising the convergence rate. The Affine Projection algorithm has come up as a better alternative to NLMS in applications where the input is highly correlated.

In general form an Affine projection algorithm can be defined by the following two equations.

$$\mathbf{h}(n) = \mathbf{h}(n-1) + \mu \mathbf{x}^T(n)[\mathbf{x}(n)\mathbf{x}^T(n)]^{-1}e(n) \tag{6.1}$$

$$e(n) = y(n) - \mathbf{x}^T(n)\mathbf{h}(n-1) \tag{6.2}$$

Here the transpose is denoted by superscript T and other parameters are as follows.

- $\mathbf{x}(n)$ is an input or an excitation vector signal and n is the time index.

- $\mathbf{x}(n) = [x(n), x(n-1), x(n-2), ..., x(n-L+1)]^T$ is the L length excitation vector at n^{th} time instant.
- $\mathbf{a}(n) = [x(n), x(n-1), x(n-2), ..., x(n-N+1)]$ is the N length excitation vector and each vector contains L samples.

$$\mathbf{h}(n) = [h_0, h_1, h_2.., h_(L-1)]^T \tag{6.3}$$

Where $\mathbf{h}(n)$ contains L samples in each.

6.2 Two channel improved APA

Apart from single band simple Affine Projection algorithm a channel improved Affine Projection algorithm is available. The Affine Projection algorithm has lower complexity than Recursive Least square and also the convergence speed of Affine Projection algorithm is less than RLS algorithm. In an improved Affine Projection algorithm an error signal is the difference of an output and desired signal. This error signal is common for both echo paths in one channel.

$$\mathbf{e}_{ap}(n) = \mathbf{d}_{ap}(n) - \mathbf{y}_{ap}(n) \tag{6.4}$$

And,

$$\mathbf{e}_{ap}(n) = \begin{bmatrix} e_{1ap} \\ e_{2ap} \end{bmatrix} \tag{6.5}$$

$$\mathbf{d}_{ap}(n) = \begin{bmatrix} d_{1ap} \\ d_{2ap} \end{bmatrix} \tag{6.6}$$

$$\mathbf{x}_{ap}(n) = [\mathbf{x}_1(n), \mathbf{x}_2(n)] \tag{6.7}$$

$$\mathbf{x}_i(n) = [x_i(n), x_i(n-1), x_i(n-2), , x_i(n-L+1]^T hasLsamples. \tag{6.8}$$

Where L is the dimensionality of the input vector. The Affine Projection algorithm is derived by first requiring an error being zero. i.e.,

$$\mathbf{e}_{ap} = 0 \tag{6.9}$$

And it implies that,

$$\mathbf{d}_{ap}(n) - \mathbf{x}_{ap}^T(n)\mathbf{w}(n) = 0 \tag{6.10}$$

This means that an affine projection maintains the next coefficient vector $\mathbf{w}(n+1)$ as near as possible to the current vector $\mathbf{w}(n)$ which is known as minimal distance procedure. The prior error $\mathbf{e}_{ap}(n)$ can be defined as

$$\mathbf{e}_{ap}(n) = \mathbf{d}_{ap}(n) - \mathbf{x}_{ap}^T(n)\mathbf{w}(n) \tag{6.11}$$

From equations 6.10 and 6.11,

$$\mathbf{x}_{ap}^{T}(n)[\mathbf{w}(n+1) - \mathbf{w}(n)] = \mathbf{e}(n) \tag{6.12}$$

For the calculation of cross correlation between the two channels

$$\mathbf{x}_1^T(n)\Delta\mathbf{w}_1(n) = 0_{px1} \tag{6.13}$$

and,

$$\mathbf{x}_2^T(n)\Delta\mathbf{w}_2(n) = 0_{px1} \tag{6.14}$$

The objective of APA is minimizing the error according to

$$\frac{1}{2}||\mathbf{w}(n) - \mathbf{w}(n-1)||^2 \tag{6.15}$$

From equations 6.13 and 6.14,

$$\begin{bmatrix} \mathbf{x}_1^T(n) & \mathbf{x}_2^T(n) \\ \mathbf{x}_1^T(n) & 0_{px1} \\ 0_{px1} & \mathbf{x}_1^T(n) \end{bmatrix} \begin{bmatrix} \Delta\mathbf{w}_1(n) \\ \Delta\mathbf{w}_2(n) \end{bmatrix} = \begin{bmatrix} \mathbf{e}(n) \\ 0_{px1} \\ 0_{px1} \end{bmatrix} \tag{6.16}$$

The minimum affine projection algorithm can be found out by minimum normal solution given in the equation below.

$$\mathbf{w}_i(n+1) = \mathbf{w}_i(n) + \mu\mathbf{x}_j(n)(\mathbf{x}_j^H(n)\mathbf{x}_j(n))^{-1}\mathbf{e}(n) \tag{6.17}$$

where i,j=1,2,.

6.3 Advantages of Employing Affine Projection Algorithm (APA)

Affine Projection has its own superiority from other adaptive filtering algorithms such as Normalized Least Mean Square (NLMS) algorithm and Recursive Least Square (RLS) algorithm. Affine Projection algorithm has faster and better tracking abilities than Normalized Least Mean Square (NLMS) algorithm [26]. Affine Projection algorithm has an advantage of better performance in steady state mean square error or the transient response as compared with other algorithms. When compared separately with Normalized Least Mean Square algorithm, the Affine Projection algorithm comes up with better performance and when compared with the recursive least square, the APA has better complexity.

Affine Projection also has an advantage of producing no delay in the input as well as the output signal and this factor makes Affine Projection algorithm suitable for adaptive filter in echo cancellation scenario. The Affine Projection algorithm has sought a way for less computational complexity without compromising the convergence rate. In case of highly correlated inputs the Affine Projection has come up as a better alternative to the Normalized Least Mean Square (NLMS) algorithm. The Affine Projection algorithm has low memory requirements as compared to RLS and NLMS algorithm [27]. The Affine Projection algorithm is also easily regularized and thus resulting in its robust performance.

In Affine Projection algorithm the projections are not made in a single dimension, but in multiple dimensions as the projection dimension increases, the convergence rate also increases. The Affine Projection algorithm provides better performance for high projection by providing high convergence rate. However, for low projection the convergence rate is slow[28]. Thus the Affine Projection algorithm is preferred for high projections so that it gives high convergence rate.

Thus due to having better convergence rate and also better complexity than other algorithms such as NLMS and RLS the Affine Projection algorithm (APA) is highly employed as an adaptive filtering algorithm for echo cancellation scheme.

ERLE and Misalignment

Chapter 7

ERLE and Misalignment

7.1 Echo Return Loss Enhancement (ERLE)

Echo return loss enhancement is the ratio of expected value at the microphone output squared to the value of an error signal squared. The echo return loss enhancement is measured in db. Higher the value of ERLE better is the echo cancellation thus the quality of stereo echo cancellation scheme is evaluated as a measure of ERLE [29]. It also measures the loss that is caused by an adaptive filter and also echo suppression. The ERLE measurement helps to estimate a value of echo loss done by an adaptive algorithm. Echo return loss enhancement is dependent on an algorithm design of the filter and also size of an adaptive filter. In other words the echo return loss enhancement can be seen as the attenuation of an echo signal as it passes through the path of an echo cancellation system. ERLE can be calculated from the equation given below.

$$ERLE = 10 \log_{10}(\frac{P_d}{P_e}) \tag{7.1}$$

Where P_d the signal is power and P_e is the power of an error signal after echo cancellation.

7.2 Misalignment

It is used to measure the mismatch between the original and the estimated impulse response of a room. It also measures how accurately an adaptive filter converges to the impulse response of a system that needs to be identified [11].

Apart from echo return loss enhancement (ERLE) and misalignment there are other parameters which should be considered at the time of determining the performance of filters.

- **Convergence rate**: - It is the number of iteration required by an algorithm to converge to its steady state mean square error, hence the convergence rate should be fast for a desired filter [30].

- **Complexity**: - It is the measure of numbers of arithmetic operations such as addition, multiplication, division for different adaptive filter algorithms.

- **Signal to noise ratio (SNR)**: - Signal to noise ratio is defined as the ratio of signal power to the noise power and is mostly expressed in decibels (dB). It compares the level of a desired signal to the background noise present in the signal.

- **Echo return loss**: - It is the difference between the echo signal level and the speech signal level. It is usually expressed in decibels (dB).

Fullband and Subband Adaptive Filtering Techniques

Chapter 8

Fullband and Subband Adaptive Filtering Techniques

8.1 Subband Adaptive Filtering

Subband adaptive filtering has become one of the most efficient techniques for reduction of computational complexity and at the same time improving the convergence rate of algorithms in digital signal processing system. The other additional feature of subband adaptive filtering is this technique useful in many real life applications. Subband is mainly used where the system is required to identify very long impulse responses. Subband also has an advantage of parallelization of processing tasks.

Subband adaptive filtering despite of having a numerous advantages also suffers from some limitations. The convergence rate is affected by low energy at frequency band edges. The model accuracy and the minimum mean square error are affected by aliasing introduced in the discrimination stage and also due to the distortion of overall filter bank. Subband filter bank often introduce some degradation in the signal. Some of the degradation introduced is due to the structure of filter and some degradation is due to filter bank parameters such as the coefficients of analysis and synthesis filter.

The filter bank of subband adaptive filters needs to be designed in such a way that the performance degradation is minimum and also reducing transmission delay introduced due to the filters in subband filtering technique [31]. The aliasing problem in subband adaptive filtering technique can be reduced by imposing oversampling in subband at the place of using critical sampling. The decomposition of signal into subband and adaptively filtering each subband by using separate adaptive filters is mainly to overcome the limitations of fullband adaptive filtering.

The subband adaptive filtering model for a system modeling is shown in figure 8.1. The desired output signal and desired input signal represented by y and x respectively is split into M subband signals. This splitting is done with the help of analysis filters namely $h_0, \cdots, h_{M-1}$.The difference between desired subband signal y_m and output of subband adaptive filters is denoted by e_m. The separated subband is downsampled by a factor D. The subband error signal may be used directly or indirectly to construct a fullband output. The reconstruction is done by upsampling by a factor D and then filtering with synthesis filters $g_0, \cdots, g_{M-1}$.

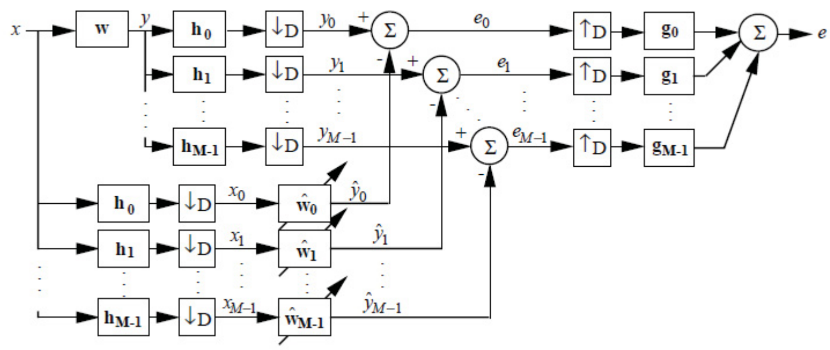

Figure 8.1: Block diagram of subband adaptive filtering system

The foremost benefit of above mentioned system is the fact that subband adaptive filters are shorter in length than conventional fullband adaptive filter and operate at downsampled rate. The subband adaptive filter also has an advantage of maximum noise attenuation. The second advantage is that the length of all the filters need not be same and the length can be adjusted for better matching the signal characteristics in that particular band. The subband system decompose the signal into smaller subbands thus by decomposing the signal, each filter can work on smaller band width and can take an advantage of this decomposition.

8.2 Fullband Adaptive Filtering

For long time, the adaptive filters are being used for fullband adaptive filtering in which there is only one band and no subbands or multiband is present. Though having less convergence speed the fullband adaptive filtering has the advantage of no inter-bands aliasing and also the computational efforts employed are less as compared to the subband or the multiband filtering techniques. The fullband adaptive filtering system block diagram is shown in figure 8.2.

A digital input signal *x(k)* is fed in to the adaptive filter that gives an output signal *y(k)*. The parameters of the adaptive filters are adjustable to

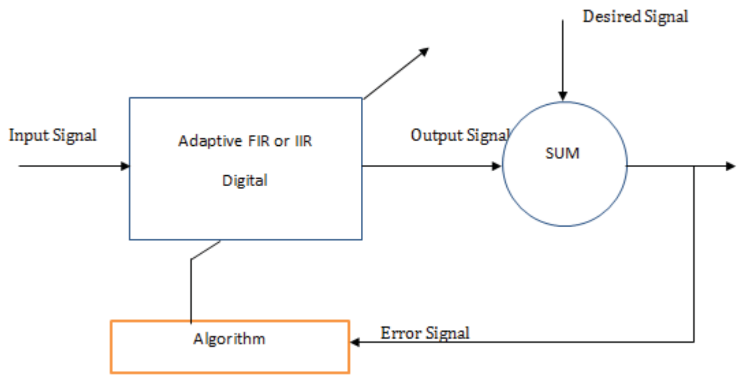

Figure 8.2: Block diagram of fullband adaptive filtering system

calculate the value of the output signal *y(k)*. This output is compared with the desired signal *d(k)* and the difference of these two signals is the error signal *e(k)= d(k)-y(k)*.This error signal is fed to an algorithm, which help the parameters of the filter are updated. With updating, the output of the adaptive filter is made equal to the desired signal such that the error signal becomes less. The most important part is the self-learning or the adaptation algorithm selected for updating the filter coefficients. The perfect adaptation is achieved when the error signal becomes zero but this is ideal result and is not feasible in the real world.

8.3 Fullband and Subband a Comparison

In the subband adaptive filtering, the input is decomposed into a number of subband signals and also the adaptive filtering is applied on each of the subband and it has the potential for faster convergence and also a lower computational complexity as compared to a fullband adaptive filtering scheme. The subband scheme suffers from two drawbacks. Firstly, inter-band aliasing which is introduced by the downsampling process to reduce the data rate degrades the performance of the filter and also cannot be avoided and also the filter bank introduces system delay and additional computation. In the subband scheme, an analysis filter and a synthesis filter is also required for decomposing and combining the subband signals. The fullband adaptive filtering scheme is free from these types of filter requirements. In the subband scheme the signals in each band can be treated separately and can be manipulated as a separate unit but in case of the fullband, signal cannot be divided into subbands and thus each signal can be processed as a single unit only and can be manipulated as a whole.

Results

Chapter 9

Results

The simulation of subband adaptive was done in Matlab and outputs are studied. The coefficients of an adaptive filter and also the echo cancellation of subband adaptive schemes employing Affine Projection algorithm was compared with output of single band adaptive filtering scheme employing the use of Normalized Least Mean Square (NLMS) and Recursive Least Square (RLS) algorithm. Here are the results after simulation in Matlab.

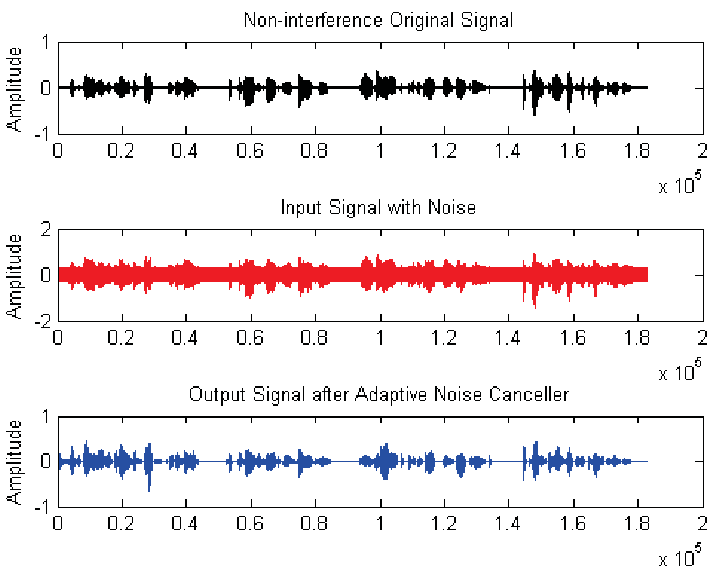

Figure 9.1: Echo cancellation by subband adaptive filtering

The original sound signal is shown in amplitude and the input signal mixed with noise signal is fed as input to an adaptive filter and at the output of filter we get noise removed signal which is similar to original signal. The Affine Projection algorithm is used in an adaptive filter. It serves as an effective algorithm for echo removal as the output after noise removal is quite same as original signal and hence the error is less as compared to conventional NLMS or RLS algorithm being used for echo cancellation.

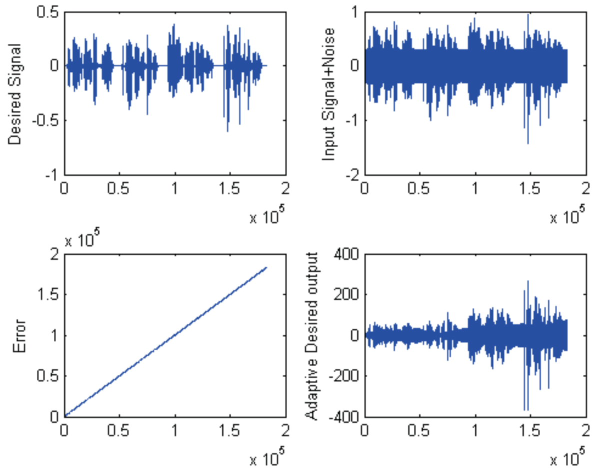

Figure 9.2: Adaptive filtering employing the RLS algorithm

Here the results of echo cancellation by using a conventional Recursive Least Square (RLS) algorithm are shown in figure 18. The desired signal along with an input signal added with noise is shown. The error in the system increases linearly, so the adaptive desired output is not same as desired input and the echo cancellation is not as effective as it should be.

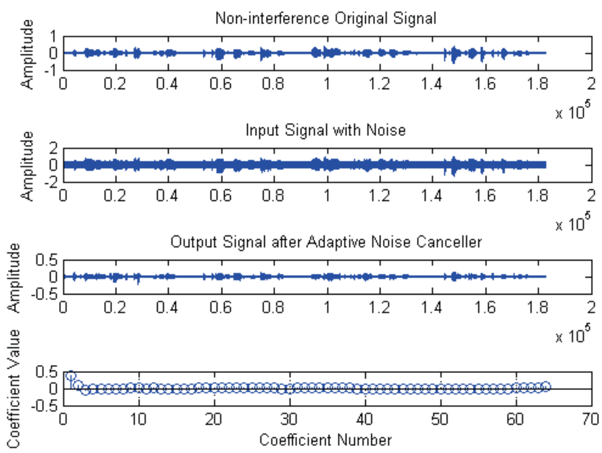

Figure 9.3: Echo cancellation employing the NLMS algorithm

The figure 9.3 shows the use of conventional Normalized Least Mean

Square algorithm (NLMS) for echo cancellation. The original signal free from any noise is shown first and then the noise introduced signal is displayed in amplitude. Then NLMS algorithm is employed in an adaptive filtering and the output of an adaptive filter is noise free signal, also the coefficients number of an adaptive filter are shown in last part of the figure 9.3.

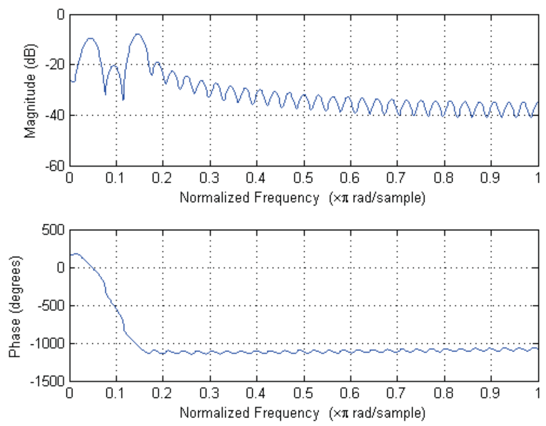

Figure 9.4: Frequency and phase response of subband filter

The frequency and phase response of the filters in subbands are shown in figure 9.4. The response for frequency is against magnitude and normalized frequency. The frequency response helps to understand the performance and characteristic of a filter in subband. The phase response is plotted against phase angle in degrees and normalized frequency. The phase is to have a maximum value in degrees for frequency of '0'rad and as normalized frequency increases phase angle (in degrees) decreases.

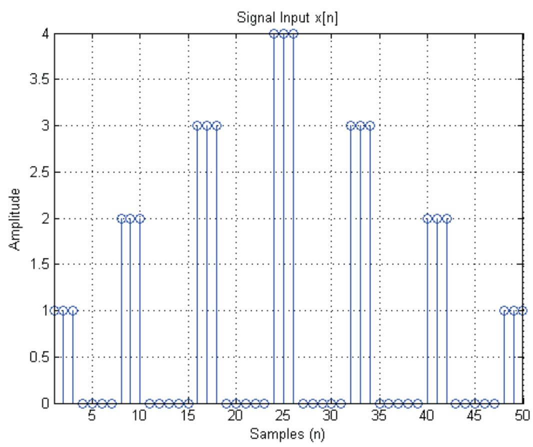

Figure 9.5: Amplitude of the input signal for various numbers of samples

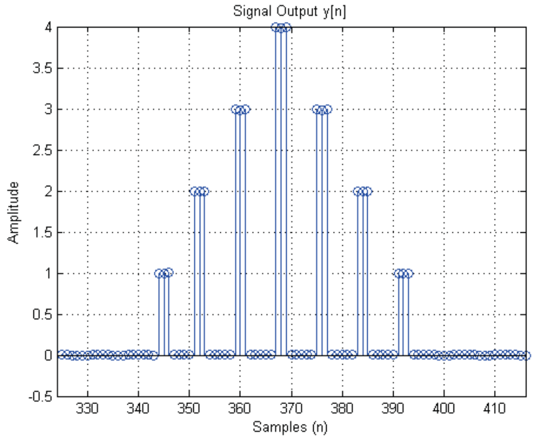

Figure 9.6: Amplitude of the output signal for various numbers of samples

The difference between the output and input is an error signal and this error signal has shown in figure 9.5 and figure 9.6 for corresponding number of samples.

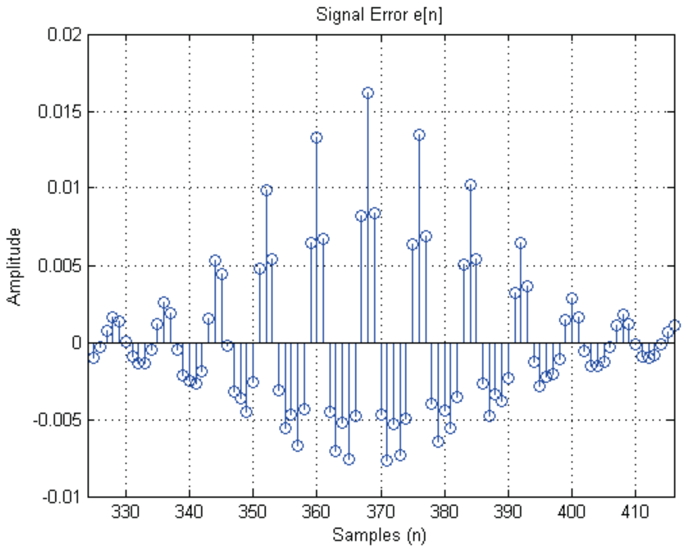

Figure 9.7: Error signals for various samples

The figure 9.7 shows the error signal which is the difference of output and input of the synthesis filters in subband and thus the error in subband is studied. The figure 9.8 shows the coefficients of analysis and synthesis filter bank in an adaptive filter. The filter 1 shows the synthesis filter and the filter 2 shows the analysis filter bank. The coefficients are shown with respect to amplitude of a signal in the filter.

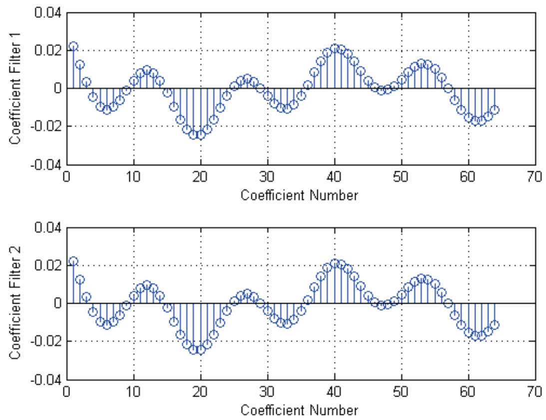

Figure 9.8: Coefficients of synthesis and analysis filters

The results of introduction or addition of noise in a signal and its removal or separation has shown in figure 9.9. The first part shows original signal which is free from noise. The second part shows input signal mixed with noise and this signal is the combination of input signal and a delayed version in the input which serves as the noise. The third part shows noise signal which has been removed by subband filtering.

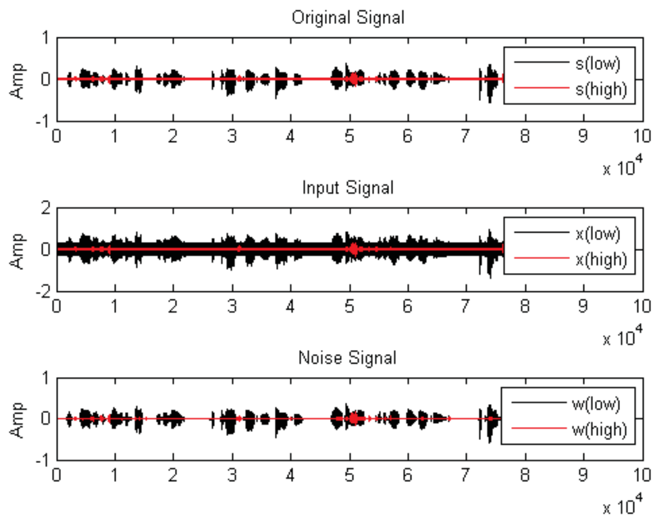

Figure 9.9: Input signal, Original signal and Noise signal in filtering process

The signal from low order filter and the signal from high order filter are shown separately by different colors.

Conclusion and Future Work

Chapter 10

Conclusion and Future Work

10.1 Conclusion

We propose the application of subband adaptive filtering to equalization problem as in case of fullband. The proposed subband filtering gives a significant reduction in computations and improved convergence rate over the fullband scheme. Use of multiple adaptive filters in a filter bank can reduce the computational complexity of a high order adaptive filter. The oversampled subband system offers much reduction in noise when compared to fullband system. The over sampled subband provides the best method of adaptive filtering, both in terms of convergence and computational efficiency. Also we have proposed to prefer Affine Projection Algorithm (APA) as Adaptive filtering algorithm at the place of Normalized Least Mean Square (NLMS) and Recursive Least Square (RLS) algorithm. The Affine Projection Algorithm provides better convergence rate and less computational complexity as compared to other adaptive filtering algorithm.

10.2 Future Work

In future, the number of channels for analysis and synthesis filter bank in a subband modeling can be increased from eight so that the reconstruction is more effective and other algorithms can be employed which may help to reduce the error and enhance the performance of a subband system.

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