Active Voice Control: An Implementation of Active Noise Control for Canceling Speech

Except where reference is made to the work of others, the work described in this thesis is my own or was done in collaboration with my advisor. This thesis does not include proprietary or classified information.

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Thesis Abstract

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New developments in active noise control (ANC) have led to commercial products such as noise canceling headphones and low engine sound airplane cockpits. These products rely on adaptive filters to eliminate noise by playing the antinoise of the original sound. Similarly, this strategy can be applied to voice to cancel unwanted speech in the open air. In some cases, the speech must be preserved for the speaker to communicate and canceled in the surrounding environment for unwanted listeners. This research introduces active voice control (AVC) in canceling speech and discusses some of the difficulties in implementing AVC in an open air environment.

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Chapter 1

Introduction

One of the most researched subjects in signal processing and acoustics is active noise control. This system uses sensors, microphones, and digital signal processing boards to create the antinoise of a sound. The system can create the antinoise by manipulating the error signal with the adaptive filters of the digital signal processor, thereby compensating in any change in the plant of the system model. Depending on the application, both feedback control and feedforward control can be used in active noise control. Success has been seen in such applications as headsets and cockpits of aircraft to cancel low frequency noise. Success in the mid to upper range of audible frequency, however, has been limited to very small regions. Currently, passive systems are most often used to control sound in the upper frequency range.

By extending active noise control into higher frequencies, sound from speech can be actively controlled. This system, presented in this paper as active voice control (AVC), can be used in a range of applications to prevent unwanted vocal communication. In most of these applications, the speech signal itself must be preserved for communication to an intended audience while isolating it from unwanted listeners. Therefore, AVC seeks not to cancel the entire speech as ANC does for noise but to preserve some aspect of it.

This paper presents a AVC as a theoretical replacement of passive systems to control speech. A simulation of AVC has been developed which uses ANC to cancel unwanted speech. This simulation, however, does not take into consideration the geometry nor the actual conditions of the surrounding environment. The difficulties with implementing AVC in real systems, such as the limitations of ANC itself and the ethical responsibilities of silencing someone's speech, prevent AVC from being realized in a real system. Only after ANC has been extended into the bulk of vocal frequency range can AVC be successfully implemented.

Chapter 2

Literature Survey on Active Noise Control

2.1 Origins

Active control was first theorized by Paul Lueg in 1936 in U.S. Patent Number 2,043,416 [7]. His patent describes measuring the sound field with a microphone, electrically manipulating the resulting signal, and then feeding it to an electroacoustic secondary source [4]. As seen in Figure 2.1 of Lueg's patent, the sound is considered to travel as plane waves in a duct from a primary source A [4]. The microphone, M, detects the sound wave and supplies the excitation to the electronic controller, V, to drive the loudspeaker at L [4]. The loudspeaker produces a sound wave that is out of phase with the primary source's acoustic wave [4]. Destructive interference is created from the superposition of the acoustic wave of the loudspeaker and the wave of the original source [4]. This concept is the basis for today's active noise control; however, at that time Lueg could not practically demonstrate his patent.

Seventeen years later, Harry Olson and Everet May published another paper which describes another system for active noise control [4]. In contrast with Lueg's paper which used prior knowledge of the signal from the detecting microphone (feedforward control), Olson and May's strategy needed no prior knowledge of the sound field [4]. Instead, it used a feedback method to cancel sound by feeding back the signal from a much closer microphone to a second loudspeaker [4].

A few years later in 1956, William Conover was working on acoustic noise reduction from power distribution transformers at Generic Electric Company [3]. The magnetostriction in the transformer made it hum at even harmonics of the line frequency [3]. This

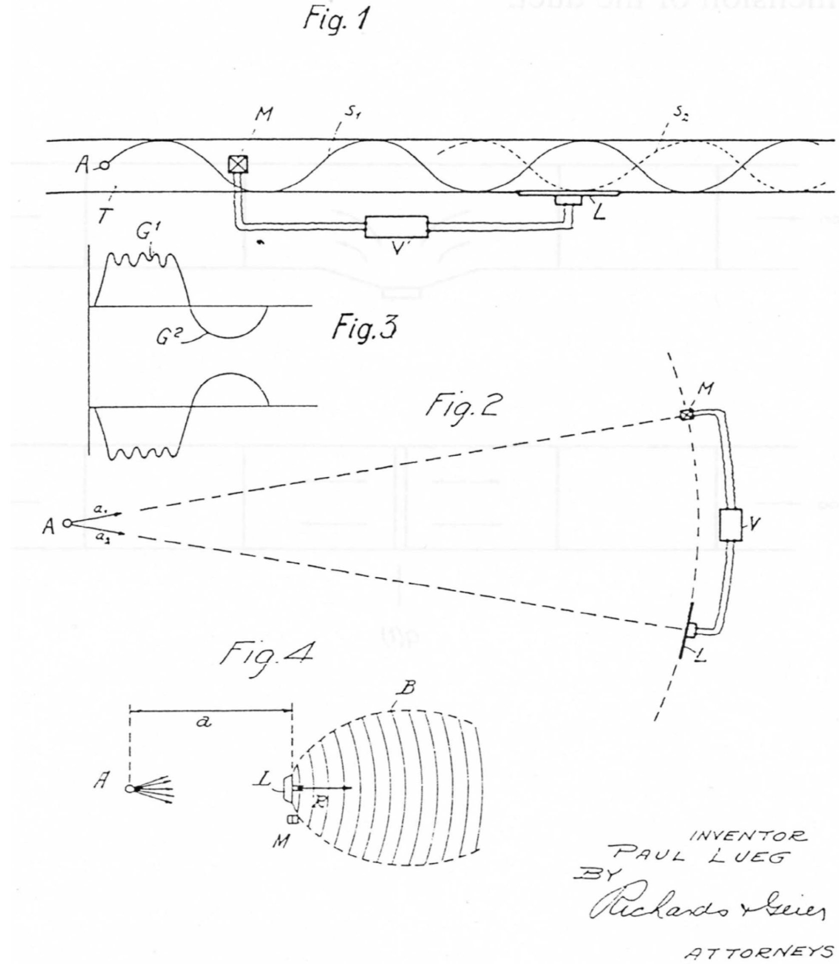

Figure 2.1: A figure from Paul Lueg's 1936 Patent for ANC [7].

periodic nature of the sound allowed Conover to avoid using a microphone and instead generate a reference signal which has the same frequency components as the primary noise. These reference signals were derived from full-wave rectified version of the line voltage and bandpass filtered to obtain the even harmonics [3]. Only the amplitude and phase of this reference signal need to be varied by the controller of the system [4]. The objective of his design was to cancel the pressure in a particular direction away from the transformer, such as a nearby house [4]. He carried out the cancellation using a manual controller, which he adjusted to compensate for winds and temperature changes [4]. The suggestions in his paper to use an automatic control system as well as multiple secondary loudspeakers and monitoring microphones were very important in the future development of ANC [4].

Due to the lack of capable technology in the 1930's and 1950's, ANC was not possible until modern computers became available. As such, the study of active noise control was silent until 1975, when Kido first used digital techniques to achieve the precise balance required for feedforward active control [4]. As digital computers became faster and more common, active noise control has become much more practical, and ANC became a mainstream research topic in the 1970's and 1980's [11]. In 1980, the well known filtered-x least mean squares algorithm was developed by Morgan and also independently by Widrow in 1981 and now figures prominently in recent active noise control research [12, 9] Today, researchers publish technical articles on active control at a rate of several hundred per year [11]. Dozens of companies now specialize in active control products such as headphones, and universities and government research laboratories are actively involved in active noise control research [11].

2.2 Physical Design of the ANC System

ANC systems consist primarily of four major parts: the plant, sensors, actuators, and controller. The plant is the physical system to be controlled, such as the compartment in a vehicle or the air traveling through a duct [11]. The sensors are the microphones, accelerometers, and other devices that sense the disturbance and monitor how well the control system is performing [11]. The actuators are the devices that physically change the system such as the speakers [11]. Finally, the controller is the digital signal processor that uses the information from the sensors to instruct the actuators to achieve noise cancellation [11]. Each of these components are the building blocks for a complete ANC system.

ANC itself works on the principle of destructive interference between the primary disturbance field heard as unwanted noise and the secondary field which is generated from control actuators [2]. In the simplest system, the disturbance field can be a simple sine wave, and the secondary field is the same sine wave but 180 degrees out of phase. The resulting superposition of the two waves is no sound at all. The human ear responds mainly to the mean square value of the pressure it registers. So the quantity that most active control systems are designed to minimize is the mean square value of this error signal [3]. Since most if not all systems have complex frequencies and waveforms, the secondary field is much more complex than merely 180 degrees out of phase. Moreover, the plant model usually changes dynamically, resulting in a system which requires an adaptive filter or many microphones to determine the current disturbance field.

An arrangement of two domains, one the disturbance and the other the cancellation noise, can be replaced by two sets of simple sources acting on the boundary between the two domains [14]. If a set of active sources is identical to the set of disturbance sources but in antiphase, or 180 degrees out of phase, then no sound will be present at the boundary [14]. Additionally, if the boundary is placed so that it isolates the disturbance domain and the cancellation domain from the silent domain, then the effects of the disturbances and active sources will not affect the silent domain [14]. This boundary is called the wall of silence and serves as a wall between the noise source and the silent domain [14]. The boundary would ideally have an infinite number of anti-phase sources to achieve this theoretical level of performance [14]. The number of actuators can make a significant impact on the performance and design of an ANC system. The next section describes a system with a single actuator, or microphone.

2.2.1 Single Sensor and Actuator Scheme

In a single sensor and actuator scheme, the primary source generates the unwanted sound field that propagates through the air [14]. The aforementioned duct ANC system was an implementation of a single sensor and actuator scheme. A sensor microphone senses the primary source and is placed in the vicinity of the primary source speaker [14]. The error microphone is placed at a location where sound cancellation is desired, and the sound wave propagates through the primary path until it reaches the error microphone [14]. The resulting sound field is an adaptively adjusted, three dimensional interference pattern, which is designed to have a node at the error microphone [14].

In the vicinity of the boundary, or zone of silence, between two waves, both sound fields are attenuated to the background sound level [14]. The zone actually extends in a wake away from the error microphone and the primary source [14]. This zone depends on the disturbance sound wavelength, and as such is very small for frequencies above 500 Hz. As seen in Fig. 2.2, the difference between the original sound and the canceled sound results in a complex field throughout the room. Several regions show constructive interference, specifically near the primary source [14]. Closer to the secondary source, destructive interference can been seen with reduced noise along with the zone of silence wake at the -7.3 dB curve along the boundary between the disturbance domain and the cancellation domain [14]. The tones used in the experiment to generate this figure were single tones of 80 Hz, which suggests that the results may be significantly different with more complex sound fields. The noise suppression was only capable of canceling to about 70 dB due to background noise, but any reduction in the noise suppression would result in cancellation to that level of reduction [14]. Also, the sound pressure level near the error microphone is erroneous due to reflections from the microphone stand [14].

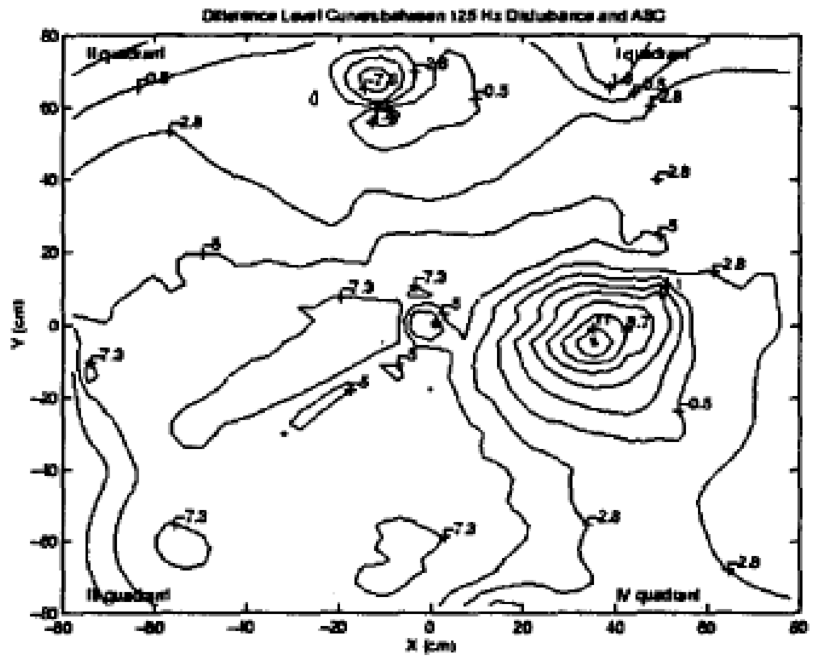

Figure 2.2: Difference between disturbance and ASC sound fields in an open air environment [14].

2.2.2 Multiple Sensors and Actuators Scheme

In more advanced ANC systems, multiple microphones are used create a more complete system. For enclosures, each acoustic mode has a specific frequency, and these natural frequencies prevent the secondary source from reducing the amplitude of one mode without increasing the amplitude of at least one of the other modes [3]. Similarly, for most frequencies above 200Hz or excitation frequencies between the natural frequencies, the secondary source is unable to control any one of these acoustic modes without increasing the excitation of a number of other modes, the optimum secondary source strength is reduced and little reduction in the total acoustic potential energy is achieved [3]. In order to control every mode, multiple secondary sources are needed. Increasing the number of secondary sources would increase the number of acoustic modes which could be actively controlled [3].

The number of significantly contributing acoustic modes in an enclosure increases at higher frequencies in approximate proportion to the cube of the excitation frequency [3].

In practice, it is impossible to measure the total acoustic potential energy in an enclosure without an infinite number of microphones [3]. Minimizing the sum of the squares of a finite number of microphones can significantly reduce the total acoustic potential energy in the system without requiring an exorbitant number of microphones [3]. Like secondary sources, the microphones must be placed in an enclosure so that they are all affected by the dominant acoustic modes [3], but this task has proved to be difficult for arbitrary enclosure geometry and wall impedance [14]. The sound pattern will be an interference pattern with nodes occurring at the error microphone positions [14]. Since the sound pressure amplitude cannot instantaneously change from zero, there will be some attenuation in the vicinity of the nodes [14]. With multiple microphones and actuators, it is possible to extend the aforementioned zone of silence to achieve a decoupling of the primary and secondary source sound fields from some observer space [14]. As frequency increases, node separation will need to be closer [14]. Typically, having twice as many microphones as secondary sources provides a reasonable compromise between complexity and performance [3].

2.3 Adaptive Algorithms

Adaptive algorithms are used in adaptive controllers to control the reference signals for driving the loudspeakers. This section describes several of these algorithms that have been successfully implemented in ANC including various finite impulse response (FIR) algorithms and infinite impulse response (IIR) algorithms.

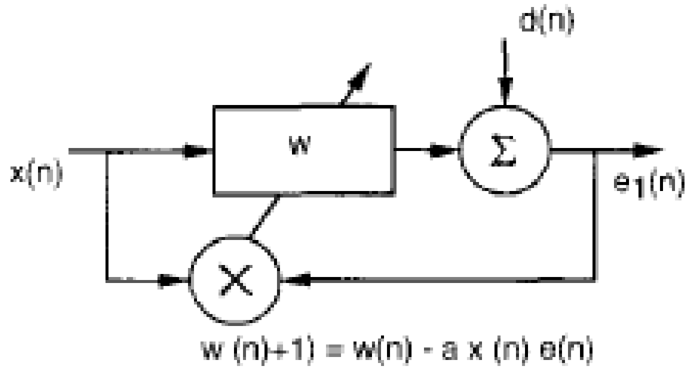

Figure 2.3: LMS block diagram [4].

2.3.1 Finite Impulse Response

Finite impulse response (FIR) filters are used predominantly in ANC because the impulse response of the system is very well damped and can be modeled easily by FIR filters [3]. The well known least mean square algorithm (LMS) adjusts the adaptive filter coefficients in adaptive noise cancellation applications to remove undesired components of a primary signal that are correlated with a given reference signal [9]. However, the canceling signal in active noise control systems cannot be applied to the primary disturbance signal due to the intervening transfer function C seen in Fig. 2.4 and the resulting possible instability [9]. This instability arises because the signal from the adaptive filter suffers a phase shift in passing through the secondary path, C [4]. The instantaneous measurement of the gradient of the mean square error with respect to the coefficient vector is no longer an unbiased estimate of the true gradient [4].

To compensate for the transfer function C, the filtered-x LMS (FxLMS) algorithm was developed which filters the reference signal by an estimate of the cancellation path transfer function $\hat{C}$ before summing with the error feedback signal [9]. The reference signal is no longer directly available and is replaced by a filtered version of it [12]. The convergence rate of this filtered version of the reference signal is slower than that of its LMS equivalent [12]. However, modifications to the algorithm can increase the convergence rate at the cost of the number of operations equal to the number of estimated filter coefficients [12]. Fortunately, many applications in noise control can be modeled by pure delay in the error path or very few coefficients [12]. In practice the FxLMS algorithm is stable even if the control filter coefficients change significantly in the time associated with the response of the secondary path [4]. Like all finite impulse response filters, the FxLMS algorithm is . The maximum convergence coefficient that can be used in the FxLMS algorithm has been found to be

$$\alpha_{max} = \frac{1}{\bar{r}^2(I + \delta)} \tag{2.1}$$

where $\bar{r}^2$ is the mean square value for the filtered reference signal, I is the number of filter coefficients, and δ is the overall delay in the secondary path in samples [4]. This delay, δ, forms the most significant part of the dynamic response of the system and reduces the maximum convergence coefficient [4]. For some applications, specifically those within an enclosed space such as a cockpit with dimensions of only a few meters, the delay is typically less than one second, and the initial convergence speed is rapid [4].

The LMS algorithms can exhibit slower modes of convergence whose time constants are determined by the eigenvalues of the reference signal autocorrelation matrix. The stability of the FxLMS algorithm is not limited to just these coefficients. The accuracy of the filter ($\hat{C}$ modeling the true secondary path C affects the stability of the system as well [4]. Fortunately, the gradient vector does not have to be exact and errors in $\hat{C}$ do not create a significant problem [4]. For example, pure tone reference signals require the phase at the excitation frequency only has to be within 90 degrees of $\hat{C}$ for the system to converge

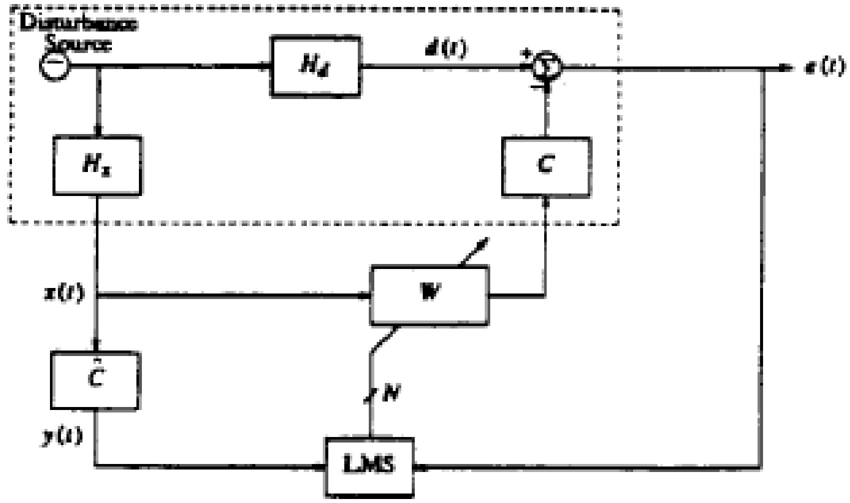

Figure 2.4: ANC block diagram [9].

slowly [4]. The most important attribute for stability is that its impulse response has at least as great a transport delay as the secondary path [4].

Additions to the FxLMS algorithm have led to an algorithm called the Modified Filtered-X Least Mean Squared Algorithm (MFxLMS) algorithm. As seen in Figure 2.5, the filter in the error path has been moved backward to the driving sequence $u(k)$ [12]. Under this system, the FxLMS algorithm behaves like the regular LMS algorithm but with the filtered input sequence [12]. However, if $\hat{C}(k)$ is time-varying, the systems will no longer be time equivalent for the plant C and a new error signal must be used [12]. The update for the MFxLMS algorithm is therefore

$$\hat{c}(k+1) = \hat{c}(k) + \mu(k)\bar{e}_f(k)u_f(k) \tag{2.2}$$

if the error signal can be calculated by known signals. The MFxLMS algorithm reduces the complexity of the FxLMS algorithm while keeping the behavior the same. If the filter order P of the system is small compared with the filter length M, then the complexity of the algorithms is approximately $2M$.

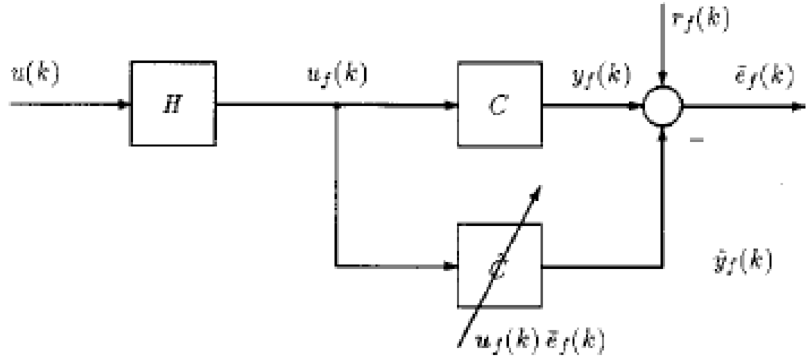

Figure 2.5: Modified filtered-X LMS block diagram [12].

2.3.2 Infinite Impulse Response

Infinite impulse response filters are more common in describing the vibration response of structures in active structural acoustic control due to the lightly damped quality of the structures but have still found application in ANC [3]. The recursive least squares (RLS) algorithm can quickly model the response of the system without the need for feedback cancellation [4]. However, unlike the FIR FxLMS filter described in the previous section, the RLS algorithm has a greater possibility of instability as an IIR filter. Extensions of stable RLS algorithms, such as the inverse QR-RLS algorithm (IQR-RLS) and the QR decomposition least-squares-lattice (QRD-LSL) have been developed [15]. These algorithms are combined with the modified version of the FxLMS algorithm whose delay-compensating structure eliminates the need to reduce the adaptive gain in the update equation of the adaptive filters because of the delay found in the error path between the actuators and the error sensors [15]. The modified version of the FxLMS structure, as it pertains to the following description of the IQR-RLS algorithm, is shown in Fig. 2.6.

The IQR-RLS algorithm is used in broadband multichannel ANC systems and must explicitly compute the time domain coefficients of the filters [15]. The block delay line structure is a requirement for fast versions of the RLS algorithms [15]. This block delay line

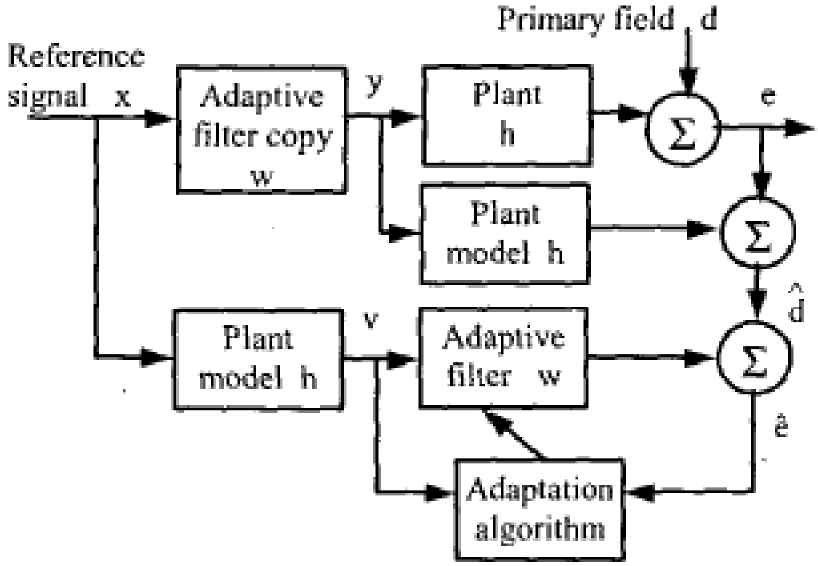

Figure 2.6: Modified filtered-X structure for RLS [12].

structure arises from the transform of the values of the filtered reference signal, $V(n)$ [15]. The samples in $V(n)$ are the samples of $V(n-1)$ that have been delayed by one sample except for the first block of samples (the new samples) in $V(n)$ and the last block of samples in $V(n-1)$ which are discarded [15]. The equations for the IQR-RLS algorithm are as follows:

$$y_j(n) = \sum_{i=1}^{I} w_{i,j}^T(n) x_i(n) \tag{2.3}$$

$$v_{i,j,k}(n) = h_{j,k}^T x_i(n) \tag{2.4}$$

$$\hat{d}_k(n) = e_k(n) - \sum_{j=k}^{J} h_{j,k}^T y_j(n) \tag{2.5}$$

$$e_k'(n) = d_k'(n) + \sum_{i=1}^{I} \sum_{j=1}^{J} w_{i,j}^T(n) v_{i,j,k}(n) \tag{2.6}$$

The computational load for the IQR-RLS algorithm for multichannel ANC systems has a computational load proportional to $(IJL)^2$ [15]. Therefore, the computational load increases with L^2 [15].

The QR decomposition least-squares-lattice (QRD-LSL) algorithm improves on the IQR-RLS algorithm by increasing the computational load by only L while still maintaining numerical stability [15]. However, it cannot provide the time-domain adaptive filters coefficients needed for the upper path of the ANC structure shown in Fig. 2.6 [15]. To compensate, the inverse transformation is required, but again, the computational load would approach $(IJL)^2$ [15]. As long as the period between updates is less than the time constant caused by the forgetting factor λ, common to all LMS algorithms, the convergence performance is not affected greatly [15]. Although the QRD-LSL algorithm requires less computations, the IQR-RLS algorithm is simpler to implement and can be more stable if low precision numerical representations(12 or 16 bits fixed point) or ill-conditioned systems (more actuators than error sensors) are used [15]. Both the QRD-LSL and the IQR-RLS algorithms are stable in 32 bit floating point environment.

Another IIR filter that uses the lattice structure is the gradient lattice algorithm (GL). Like all IIR filters, the GL algorithm is not unconditionally stable due to the possibility that some poles of the filters will move outside of the unit circle during the weights update [6]. Also, most IIR adaptive algorithms have a lower convergence speed and may converge to a local minimum [6]. The lattice structure for the GL algorithm possesses the advantage of inherent stability and greatly reduced sensitivity to the eigenvalue spread of the reference signal [6]. The GL algorithm makes full use of the orthogonalization property of the lattice structure and avoids the problem of slow convergence and possible instability while holding the benefits of an adaptive IIR filter [6].

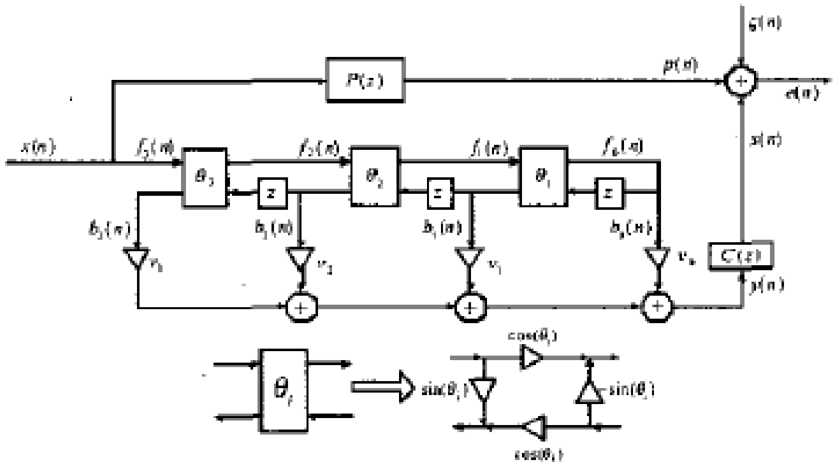

Figure 2.7: Gradient lattice block diagram [6].

M additional lattice filters are needed to obtain the filtered regressor signals $-\Delta\Theta_k(n)$ corresponding to the rotation parameters [6]. These filtered regressor signals are formed by filtering the input with the cancellation path transfer function and the lattice filter and can be obtained with an auxiliary lattice filter [6]. Since M additional lattice filters are needed, the complexity of the system is of the order M^2 for computation as well as storage [6]. Among IIR filters, this additional complexity is a disadvantage. However, by simplifying the computations, a gradient lattice algorithm can be obtained that is of order M [6]. The resulting algorithm is as follows [6]:

$$v_k(n+1) = v_k(n) - \mu e(n) B_k(z) C(z) x(n), \ \ k = 0, 1, ..., M \tag{2.7}$$

$$\Theta_k(n+1) = \Theta_k(n) + \mu e(n) \gamma_k z B_{k-1}(z) W(z) x(n), \ \ k = 1, 2, ..., M \tag{2.8}$$

The GL algorithm's performance can be measured with the commonly used adaptive IIR filters, the filtered-u LMS (FULMS) algorithm and the filtered-v LMS (FVLMS) algorithms, both of which suffer from instability and slow convergence. Compared with these direct form IIR filters, the convergence for the GL algorithm should be about the same for

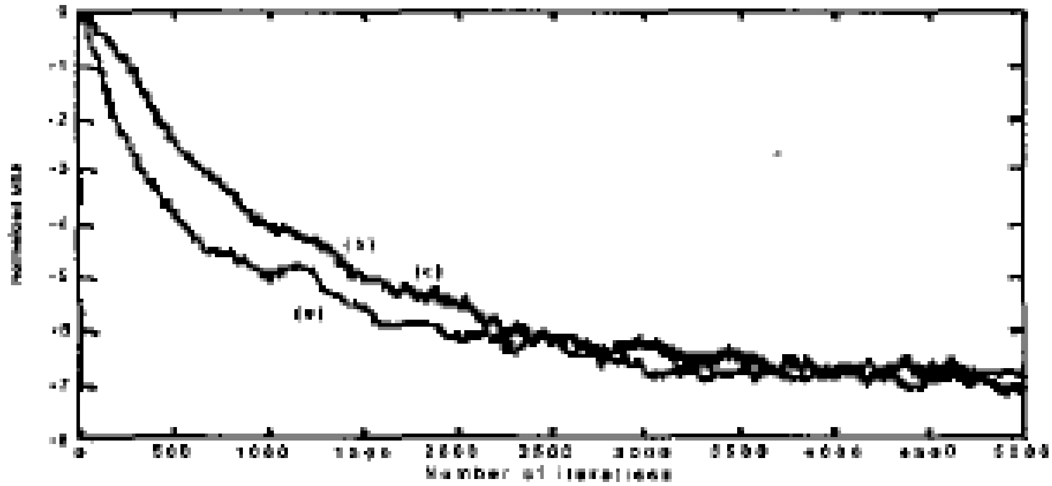

Figure 2.8: Convergence comparison between (a)GL, (b) FULMS, and (c) FVLMS algorithms [6].

a noise signal with a flat power spectrum, but since lattice form adaptive IIR filters are more stable, the convergence coefficient can be set larger to achieve a faster convergence speed [6]. As seen in Fig. 2.8, the comparison between the GL algorithm and other IIR algorithms shows that the GL algorithm converges faster and to a smaller mean squared error [6]. It is also far less sensitive to the cancellation path modeling error [6].

2.4 Limitations of Active Noise Control

Active noise control is not appropriate for high frequency cancellation, as seen in Fig. 2.9 from the consumer headset, Bose QuietComfort 2 ANC headphone. The spatial character of a sound field depends on wavelength and as such, frequency [11]. Destructive interference used by ANC is most efficient when the two sound fields can be aligned in space over an acoustic wavelength [3]. These low frequency sounds have wavelengths that are large compared to the zone in which the noise is canceled and are therefore more conducive to ANC than high frequency sounds [3]. For example, a sound wave with a frequency of 100 Hz, will have a wavelength of about 3.4 meters under normal conditions [4]. Sounds at

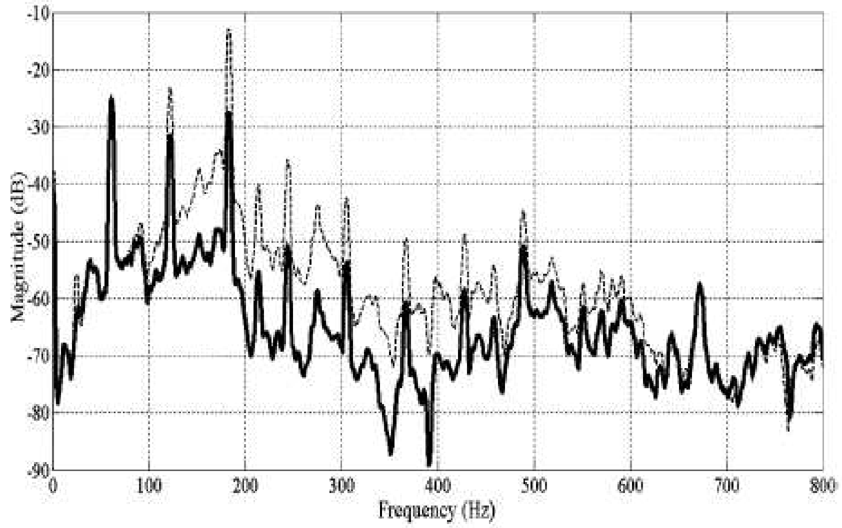

Figure 2.9: Power spectra for engine noise at 3700 rpm (dotted line) and the noise after ANC(solid line) [13].

around 100 Hz, such as that from a engine noise, can be canceled with ANC in a system much smaller than 3.4 meters, such as in personal headphones. Passive systems, such as heavy barriers and acoustic materials, work best for high frequency noise, where the acoustic wavelength is large compared to the thickness of the absorber [3]. As such, most ANC systems combine both passive and active control methods [11].

Active noise control is restricted to spatially simple systems [11]. Paul Lueg's experiment, described in Section 2.1, used a duct as its system, which is essentially a one dimensional problem [11]. Other systems have built upon his theories and expanded the system to more complex regions, but controlling a spatially complicated sound field is beyond today's technology [11]. Obviously, products are not on the market which cancel sound from a noisy party next door or trains passing by. The sound fields in these circumstances are hopelessly complex because of the high frequencies and complicated geometry of the objects in the surroundings [11].

Canceling sound in enclosed spaces at low frequencies, such as in a helicopter cockpit, has been successful because the wavelength of the noise is similar or longer than the dimension of the enclosure [11]. The sound field in an enclosure is typically created by standing waves and depends on the superposition of a number of acoustic modes [3]. These modes are characterized by the number of wavelengths that fit along one dimension of the enclosure [3]. Depending on the geometry of the enclosure, the mode will have a corresponding natural frequency. At an excitation frequency larger than this natural frequency, the secondary source will be unable to reduce the acoustic energy in the enclosure due to natural frequencies from other acoustic modes [3]. The natural frequencies from the other acoustic modes prevent the secondary source from reducing the amplitude of one mode without reducing the amplitude of at least one of the other modes [3]. Multiple secondary sources are then needed to control every mode [3]. When the number of acoustic modes contributing to the total energy at one frequency increases dramatically with excitation frequency, the modal overlap increases in proportion to the cube of the excitation frequency [3]. Doubling the upper frequency of active control requires eight times the number of loudspeakers at the original frequency [3]. It follows that broadband noise is much harder to cancel than narrowband or a single tone and harmonics [11]. Also, lightly damped systems are easier to control than heavily damped systems [11].

In wide open spaces, however, the reduction of sound in one localized region can actually amplify noise in another region [11]. Two components of sound pressure in phase will create constructive interference and actually increase the sound level [4]. Global noise cancellation only occurs for simple sound fields where impedance coupling is the primary mechanism [11]. Like enclosed spaces, more actuators are needed for complex sound fields to obtain global reductions, and higher frequencies complicate sound fields to the point that hundreds of actuators are needed for global control [11]. On the other hand, directional cancellation is possible at fairly high frequencies if the actuators and control system can accurately match the phase of the disturbance [11].

The control method used to cancel sound has a significant impact on how well ANC works for a system. If a disturbance can be measured before it reaches the region where sound attenuation is desired, feedforward control can be used to measure the actuator signal [11]. If not, then feedback control must be used to compute the actuator signal solely from error measurements [11]. For example, random noise in an aircraft due to air turbulence has no single observable source for an external reference signal in feedforward control [3]. Feedback control must be used in this situation. Under most conditions, however, feedback control is less stable than feedforward control and tends to be less effective at high frequencies [11]. The bandwidth over which attenuation is achieved is limited by the delay in the plant [3]. Since the bandwidth is proportional to the delay in seconds, the phase shift associated with the delay changes the sign of the feedback at higher frequencies, which results in a positive feedback system and more sound [3]. Therefore, the microphone must be close to the loudspeaker in order to reduce the acoustic propagation delay [3].

2.5 Commercial Products

Although ANC systems have shown in successful demonstrations to cancel noise, its applications are few [11]. A few applications that have been successful include canceling noise in enclosed spaces such as ducts, vehicle cabins, exhaust pipes, and headphones [11].

2.5.1 ANC Headphones

Noise canceling headphones are the most successful consumer product that uses ANC. These headphones cancel noise in the environment while allowing the user to listen to music. As seen in Fig. 2.10, the system consists of earmuffs containing speakers and one

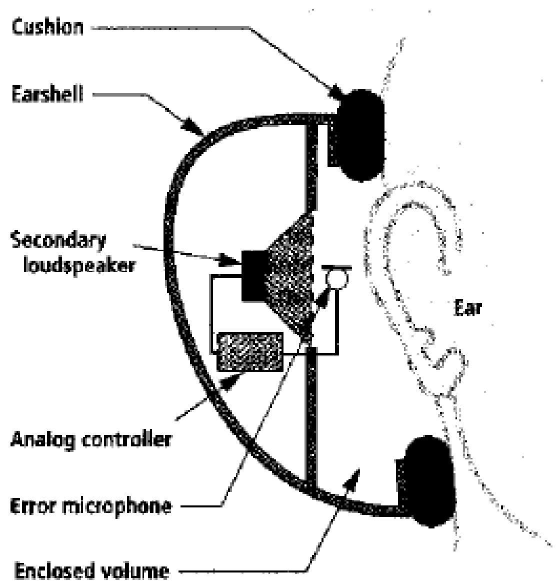

Figure 2.10: Noise canceling headphone using ANC [3].

or more small circuit boards [11]. Like most ANC commercial products, ANC headphones take advantage of an enclosure to carry out ANC. The microphone can be placed within one centimeter of the loudspeaker, thus reducing the propagation delay and enabling the feedback controller to cancel frequencies of about 1 kHz [3]. Additionally, the proximity to the ear canal provides a favorable acoustic environment in which the attenuation of the sound pressure at the microphone results in similar attenuation of the ear [3]. As of this writing, active control headsets are available for relatively cheap prices in standard electronics stores across the United States.

The two control systems for ANC, feedforward and feedback control, have both been implemented in ANC. Feedforward control, while the focus of research in communication and entertainment headsets, has significant stability and performance problems caused by nonstationary reference inputs, measurement noise, and acoustic feedback [5]. In feedforward control, the reference input is placed outside of the ear cup to pick up the primary noise $x(n)$ [13]. The reference signal $x(n)$ is processed by the ANC system to generate

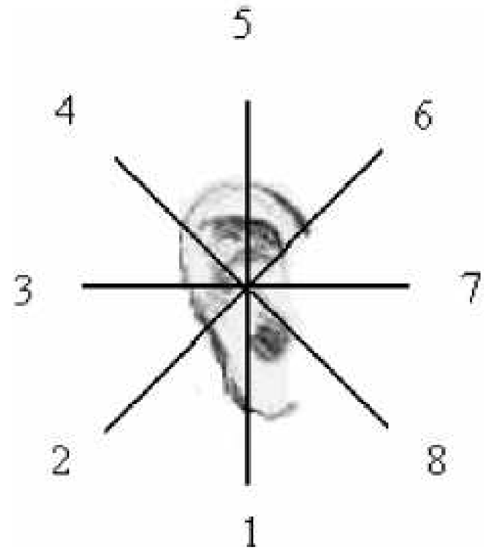

Figure 2.11: Locations for an error microphone for ANC headphones [13].

the control signal $y(n)$ to drive a secondary loudspeaker inside the ear cup for producing antinoise [13]. The FxLMS algorithm minimizes the residual noise $e(n)$, which is in turn measured by the error microphone [13]. The ideal location of the error microphone occurs where the frequency response of the system is flattest with the least number of dips and peaks [13]. This location is found experimentally in [13] to be near the external auditory meatus of the ear, which allows the microphone to be placed in the hollow in that location at location 8 of Fig. 2.11.

The secondary path $S(z)$ includes the digital-to-analog converter, reconstruction filter, power amplifier, loudspeaker, acoustic path from loudspeaker to error microphone, error microphone, preamplifier, antialiasing filter, and analog-to-digital converter [13]. The FxLMS algorithm places the secondary path estimate in the reference signal path to the weight update of the algorithm [13]. This model is usually estimated using white noise as the excitation signal, but for headphone applications, the white noise can be irritating [13]. In [13], music has been shown to perform as well as white and chirp noise for the excitation signal.

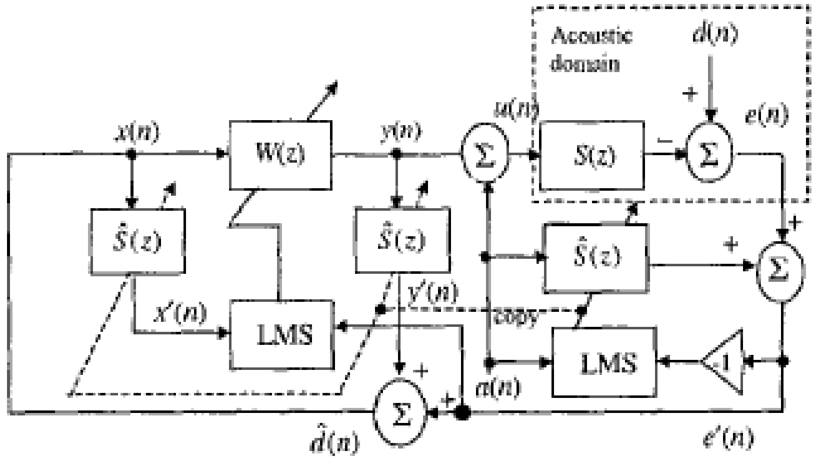

Figure 2.12: Integrated audio and ANC system block diagram [5].

Feedback control, however, provides more accurate noise cancellation since the microphone is inside the ear cup of the headset [5]. Specifically, the FxLMS algorithm, described in Section 2.4.1, is used in [5] to achieve a complete adaptive feedback ANC system. Together with the FxLMS system, the desired audio input can be integrated to form the integrated feedback active noise control (IFBANC) system [5]. This system uses the residual noise picked up by the error microphone to synthesize the primary noise $x(n)$ for updating the adaptive filter coefficients using the FxLMS algorithm [5]. The IFBANC is integrated with the existing audio playback system or communication headset to pick up both the residual noise and the desired audio signal as seen in Fig. 2.10 [5].

Figure 2.12 shows the block diagram of the combined audio and ANC system. The error sensor output signal, $e(n)$, contains both the residual noise and the desired audio signal [5]. The audio components in $e(n)$ are then removed by the audio interference cancellation filter $\hat{S}(z)$ using the audio signal $a(n)$ as the reference signal [5]. The adaptive noise control filter $W(z)$ is updated with the difference error signal $e'(n)$ which consists of the residual noise [5].

The optimal solution for the audio interference cancellation filter is

$$E'(z) = D(z) - Y(z)S(z) \tag{2.9}$$

which shows that the $E'(z)$ is reduced to the residual error of the ANC system where the primary noise $D(z)$ is canceled by the anti-noise $Y(z)S(z)$ [5].

Several advantages of the IFBANC system become apparent. First, the true residual noise $e'(n)$ can be estimated well without interfering with the audio signal $a(n)$ [5]. Second, a large step size can be used in adapting the cancellation filter $W(z)$ since the difference error signal $e'(n)$ used by the FxLMS algorithm is not corrupted by the high volume audio signal [5]. As stated earlier, the adaptive feedback ANC technique allows for more accurate noise cancellation since the microphone is placed inside the ear muff [5]. The system creates a cheap, compact, lower power consumption solution [5]. Finally, the audio signal can be used to drive the modeling of the secondary path transfer function [5].

Figure 2.13 shows the noise spectrum comparison between an ANC and non-ANC system undergoing a disturbance from a siren alternating between 775Hz and 965Hz [5]. The reduction in noise at the peaks is easily observed. The noise reduction by the IFBANC system is more than 40 dB and 30 dB at 775Hz and 965Hz, respectively [5]. Fig. 2.13 shows the spectrum for a disturbance of engine noise from a welding power generator at 3700 rpm [5]. The narrowband harmonics at 61Hz, 122Hz, and 183Hz, were canceled by more than 30 dB. These results show that ANC headphones can significantly reduce noise from industrial sources [5].

2.5.2 Other Products

Industries have used ANC on a limited basis in creating more efficient system components. Active mufflers for industrial engine exhaust stacks, commercial compressors, and

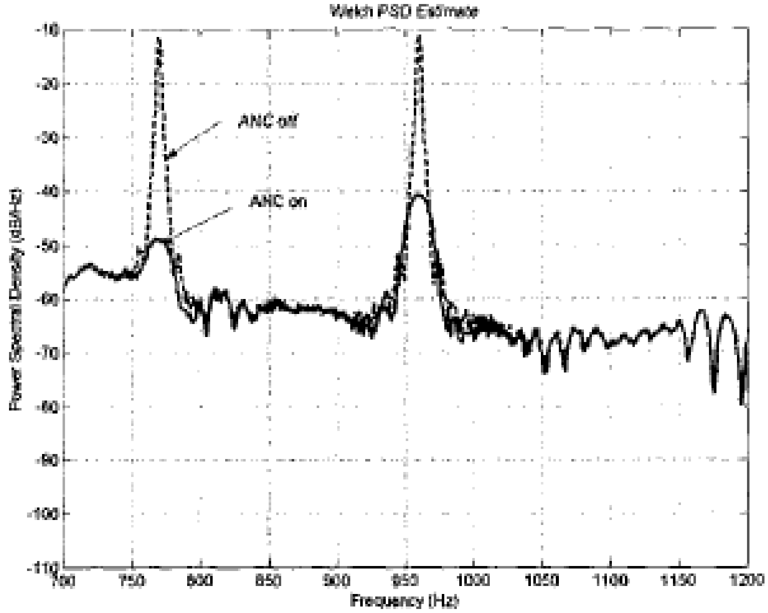

Figure 2.13: Noise spectrum comparison for ANC and non-ANC systems from siren disturbance [5].

generators reduce the noise output of these components [11]. Even automobile manufacturers are looking into ANC for active mufflers, although there are no current production automobiles with active mufflers available [11]. Industries have used ANC for their large industrial fans to reduce low-frequency noise downstream and upstream [11]. These ANC systems improve efficiency to such an extent that they pay for themselves within a year or two [11].

Adaptive feedforward controllers have been used to control the low-frequency engine noise in automobiles and cabin noise in propeller aircraft such as helicopters and airplanes [3]. Despite many successful demonstrations of ANC systems for both low frequency engine and road noise, fully active control systems are currently fitted to very few production vehicles [3]. At least one automobile in Japan offers ANC as a factory option [11].

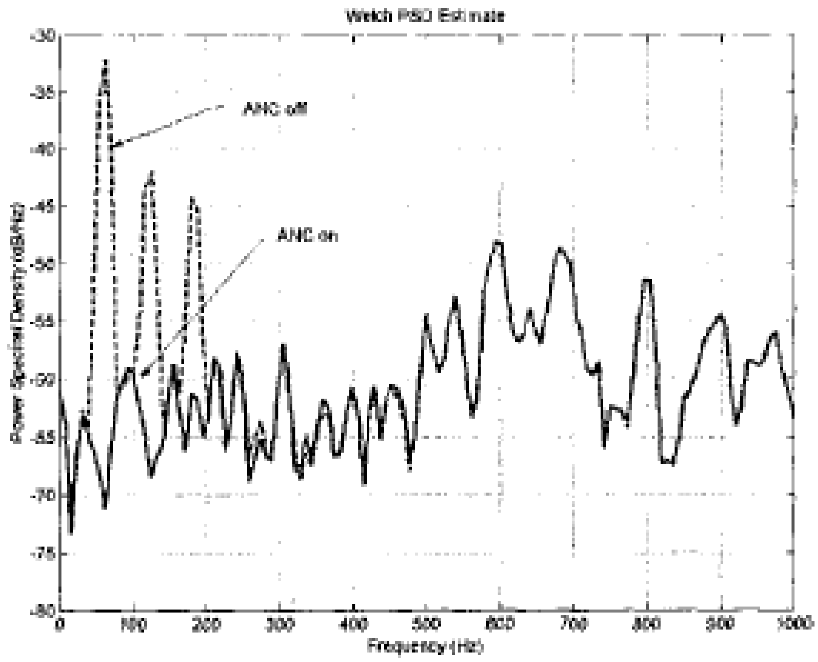

Figure 2.14: Noise spectrum comparison for ANC and non-ANC Systems from engine disturbance [5].

Most systems for automobiles involve superposing cancellation signals over the regular music signals in car stereo speakers to cancel muffler noise and engine noise [3]. However, these systems are expensive and not at all common [11].

In passenger propeller aircraft, ANC systems have been embedded inside the walls to cancel the propeller noise [3]. They primarily use rare-earth magnets to reduce their weight and generate high sound pressure levels at low frequencies to match that of the propeller noise [3]. Although this high sound pressure is not heard from cancellation, any high frequency components due to loudspeaker distortion will be clearly audible [3]. High quality components are used to reduce the likelihood of this happening [3]. A system made by Ultra Electronics, Cambridge, England for the Saab 2000 aircraft uses 37 loudspeakers and 72 microphones [3]. Ultra and Lord Corporation uses eight loudspeakers and 16 microphones in their ANC system for the smaller Beechcraft King Air aircraft [3]. The primary advantage to these systems is dramatic weight savings compared with using only passive systems [11].

Chapter 3

Active Voice Control

3.1 Differences between Active Noise Control and Active Voice Control

Active voice control (AVC) uses the strategies of ANC to cancel unwanted speech. The disturbance source of the AVC system is the voice of a speaker whose voice is to be canceled. The actuators and sensors, like in ANC, are distributed in the environment. The antinoise created by the adaptive filters is broadcast by the actuators to cancel speech. The purpose of canceling speech is not to reduce the volume of the sound like ANC does for noise since people do not normally speak or even shout at dangerous hearing levels. Instead, AVC cancels speech to prevent unwanted communication and annoying speech. Because of this, the antinoise of the sound does not need to be as loud as that for a system canceling sound in an industrial fan.

The properties of speech itself are different from the regular noise canceled by ANC systems. ANC systems usually cancel noise with frequency components under 500Hz. The speech frequency range, however, spans from about 80Hz to 3500Hz. There is a 420Hz overlay between ANC and AVC, which limits the number of people whose voice is affected by AVC. The noise canceled using ANC is usually periodic in nature like that from a propeller or from a duct. Human speech, however, can be thought of as random noise since there is no way to predict what someone will say in the near future. The speech that must be canceled in AVC is higher and wider in frequency as well as more random than noise that is canceled by ANC.

Depending on the application, however, the spatial arrangement of these actuators and sensors must compensate for the mobility of people. In ANC, the actuators and sensor locations are placed in set locations relative to the disturbance source, such as the walls of an airplane to cancel the propeller noise of the aircraft. In those circumstances, the sound fields, whose inherent variability is compensated by the adaptive filters, will not move away from the noise from the propellers for an ANC system. The speaker, however, can move between many different environments. The actuators and sensors, at least in an open air environment, will need to be placed with the speaker, or the speaker must stay in one location in order for their speech to be canceled.

The approach for the environment type (open or enclosed) for AVC differs from that of ANC. ANC can eliminate noise in enclosed spaces using multiple sensors and actuators. The use of AVC in enclosed spaces, however, is limited since the application of many sensors and actuators to cancel the speech of one person will not be viable for multiple people or for an enclosure where people commonly move in and out.

3.2 Difficulties with Voice Control

As mentioned before, vocal frequencies are higher than the noise canceled by ANC. The upper frequency limit of ANC is set by the size of the wavelengths in the noise. For vocal ranges, the wavelengths are much smaller than the corresponding wavelengths at low frequencies, and the space at which the high frequency sounds are canceled becomes too small for practical applications. The range of 420Hz at which the operating frequency range of ANC will work with speech is too small to reasonably suggest as a true implementation of AVC for all voice pitches. Typical strategies of ANC will cancel the fundamental tones of a typical male at 88-145Hz [1]. Nevertheless, the brain will create the missing fundamental using the harmonics of the voice, thereby masking the cancellation [10]. Therefore, the harmonics of the speech must be canceled as well.

The arrangement of actuators and sensors are dependent on the sound field properties of the system. For AVC, the speaker can move between many different environments. The components of AVC must therefore be on the speaker as he moves between environments, or the speaker must stay in the area of voice cancellation. These two options are naturally application dependent. The more complex of these strategies is the mobile AVC user moving between different environments. The antinoise from the actuators in this strategy must compensate for differences in the environment. As such, a multiple sensor and actuator arrangement similar to ANC is not viable due to its reliance on positioning in the environment itself. Instead, a multiple sensor and actuator arrangement may be successful if placed at various points on the speaker himself to achieve a full measurement of the sound field of the surrounding system. The strategy involving the user staying in a predetermined area of voice cancellation poses problems as well. In ANC systems using multiple sensor and actuators in the environment, the region for cancellation is generally in an enclosure where the sensors and actuators are spaced to suit the noise to be canceled. The voices for different people can require separate geometries for these sensors and actuators, making a generic arrangement of sensors and actuators impossible to build. Also, since the area of cancellation is an enclosure anyway, passive sound control will cancel speech like that of a speaker in a telephone booth mentioned in the previous section.

For just one actuator and sensor, the region for cancellation can be in both an open air environment or an enclosed environment. In an enclosed environment, the frequency range is limited to low frequencies as it is in ANC. For the open air environment, the cancellation can be expected to act similar to that presented in the Single Sensor and Actuator Scheme section above. When the disturbance reaches the secondary source and

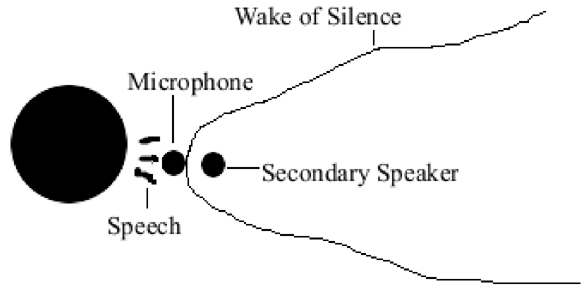

Figure 3.1: Overhead view of the preservation of speech due to the wake of silence.

antinoise, the unwanted voice attenuates in a wake of silence away from the source of the disturbance, as shown in Fig. 3.1. This region provides the system with the directional cancellation to cancel the voice from people nearby. The wake in current systems is small due to the high frequencies involved. The technology must be improved before the open air environment can be adequately used to cancel high frequency sounds.

Control for AVC systems will generally be regarded as feedback since the disturbance can not be measured before it reaches the region for sound attenuation. The adaptive filters, then, must rely solely on the feedback signal to calculate the antinoise of the system. Unfortunately, feedback control tends to be less stable and less effective at high frequencies, which is certainly necessary for voice attenuation.

The act of silencing a person's speech can extend into ethical issues as well. The basic freedom of speech can be broken by the use of AVC to cancel speech. Certainly, AVC should not be used as a means of silencing critics or keeping people from communicating with one another. Therefore, the user of AVC, who desires their own speech canceled, should be the only one whose voice is canceled to prevent communication. A situation in which the user destructively cancels the voice of a colleague or opponent is not ethically sound. Alternatively, AVC can be used to cancel residual speech as long as the AVC only cancels the voice in the immediate surroundings where the speech is undesired. The source of the speech should not be affected by AVC or perhaps even realize that their voice is being canceled. An example of this is speech between cubicles in an office environment. AVC can be employed from one cubicle to cancel the conversation from another cubicle as long as the two speakers can still communicate. Note that this is completely different to noise masking, where sound is played to drown out the noise of other people's speech and does not employ sound control.

The situation of canceling the user's speech creates another problem with using AVC - when would a user need their own voice canceled but still feel the need to speak? One application, elaborated in Section 3.3.1, uses a cell phone to speak to someone over the phone while canceling his speech to possible listeners. In all applications, the speech of the user would have to be preserved before being canceled in order for the user to communicate effectively but still prevent those in the surrounding environment from hearing the speech. The result of this is a vicious circle between preserving and canceling the speech. The voice coming from the mouth is canceled in the air, which results in no sound. At the same time, the voice from the air is picked up by the cell phone for communication to the listener on the other end. Naturally, the desired listener and the unwanted listener must be isolated from each other and must receive separate sounds.

Two possible resolutions occur for this contradiction. ANC itself never completely eliminates the noise. Instead, it generally reduces the sound from dangerous to safer levels by less than 100dB. The performance of AVC can be expected to match that of ANC given an idealized circumstance where AVC works as well as ANC. The attenuated voice, then, can be picked up by the cell phone for transmission. The other solution to the contradiction of preservation of voice to its cancellation is using the wall of silence between the boundaries of the region of suppressed voice and the region of the source of the voice. As given in [14], the region of silence after the secondary canceling source actually acts like a wake moving away from the disturbance source. This wake can act as directional cancellation away from the AVC user. The speech signal can then be picked up before it reaches the wall of sound and the resulting wave. Unfortunately, since AVC uses ANC methods, the wake will be very small due to the high frequency components of the speech signal [14].

3.3 Applications

3.3.1 Mobile Cellular Phone Booth

The mobile cellular phone booth is a theoretical application which uses AVC to cancel the voice of the speaker on a cell phone while preserving the speech itself for the recipient on the other end of the cell phone. The mobile cellular phone booth does not use the passive systems of typical telephone booths to attenuate noise. Instead, the AVC eliminates the sound in the surrounding environment. The benefits of this system are privacy for the speaker and for those people around the speaker, not having to listen to a one way conversation. The components of the system include a digital signal processing (DSP) microchip in the cell phone and the secondary speaker which is very close to the input microphone to reduce the delay in the plant. This delay actually limits the bandwidth of cancellation, so reducing the distance from the microphone and secondary source will help increase the upper bound of frequency for control. The microphone of the cell phone acts as both the receiving microphone for capturing the speech for the user on the other end of the cell phone, and as the error microphone for the adaptive filter for the DSP microchip. When the user speaks, his voice is picked up by the microphone and sent through the network to the listener on the other end of the line. At the same time, the speech signal is sent into the DSP microchip, which in turn creates the antinoise of the system. This antinoise then attenuates the speech. The control system for this type of system must be feedback due to the lack of a reference signal with which to better approximate the antinoise for a feedforward system. The attenuated noise can again be picked up by the system, broadcast to the other end, and used to drive the DSP chip. The attenuation of the speech occurs in a wake following the secondary source, which provides a somewhat directional trait to the layout of the cancellation. If necessary, additional secondary sources can be added to the cellular phone to provide cancellation in even more directions.

The mobile cellular phone booth can only exist as a theoretical application with today's technology. As with basic AVC itself, the frequency of voice is too high for ANC algorithms to work correctly. Also, the geometry of a constantly changing system is too complex for a single microphone to adequately cancel the voice, even if the voice could be attenuated thoroughly. Additional microphones are not feasible since they would not provide a complete understanding of the system. These additional microphones must be on the person instead of in the environment. As stated earlier, cancellation systems are restricted to spatially simple systems. Especially in open air environments, ANC and AVC can create constructive interference and actually amplify the disturbance noise in certain locations of the sound field. This constructive interference can be attributed to the path between the antinoise and the distance traveled away from the disturbance. The additional volume in the speech from the constructive interference will result in even more annoyance to other listeners than the regular speech on the cell phone and possibly amplify the speaker's speech that was thought to be private.

3.3.2 Residual Voice Cancellation

Residual voice cancellation (RVC) uses AVC to cancel voice without infringing on the wanted communication of the original speaker. One exception to using AVC in an enclosure that employs RVC is in a telephone booth. In this system, the residual voice is that speech that is heard outside of the telephone booth. This speech is not desired for both the observers outside of the telephone booth as well as the speaker, who may want his conversation kept confidential. The walls of the telephone booth act as its passive cancellation and cancels sound above 1000Hz. The addition of AVC in the telephone booth reduces the sound levels heard by outside observers by reducing the frequency range below 500Hz. The performance of the telephone booth in canceling sound in the mid-range, or 500Hz-1000Hz, is enhanced. Additionally, since the walls of the telephone are outside of the mouth for voice and the microphone for the telephone, only the AVC system will affect the voice volume levels going into the receiver of the telephone. Telephones employ usable frequency ranges from 300Hz to 3400Hz, which preserves enough harmonics for a discernible pitch of the voice by the listener on the other side of the telephone line. As such, only the lower frequency components of the voice will be canceled, and the observer outside of the telephone booth will hear reduced sound levels over all frequency ranges. However, AVC for this application is merely ANC under another name that cancels voice.

Another application for RVC which is much more difficult to implement is cancellation of voice in an open air environment such as the cubicle speech mentioned in Section 3.2. In this situation, the user of AVC wants to cancel the voices of the conversation in a nearby cubicle only in the immediate area so as not to cancel the conversation speech itself. Like the mobile cell phone booth, AVC must be able to reduce the entire frequency range of human speech to account for all ranges of voice. It must also account for any geometry of the surrounding objects and the resulting complex sound fields from the presence of those objects. ANC, and therefore AVC, is unable to complete either of these tasks, making the mobile cell phone booth and the RVC for the cubicle impossible with today's technology.

Chapter 4

Simulation

A simulation using MATLAB's Simulink was conducted to determine the validity of AVC by applying ANC with voice. The first experiment determined if a sinusoid at mid-range for voice frequencies could be canceled by inverting the sinusoid by 180 degrees. In Fig. 4.1, one sinusoidal source produces a frequency of 1710Hz, while the other source produces a sinusoid with a frequency of 1710Hz but at π radians out of phase. The speaker while the simulation is running produces no sound. As expected, this simulation proves that the principles of sound control work with vocal range sounds. The superposition of the two signals at opposite phase cancels both signals.

4.1 The Active Sound Control Simulink Diagram

Applying the current ANC strategies to cancel sound involves more than just adding two sound waves. As explained above, various adaptive filters are used to adjust for a dynamic plant to create the antinoise of the system. Using the adaptive filter tutorials from the Signal Processing Blockset on the MathWorks website [8], an acoustic environment and an ANC system was modified to implement AVC. The acoustic environment creates two signals - the original signal representing the vocal sound and the noisy signal representing the vocal sound after filtering and the desired output. These two signals are used by the AVC system as the inputs to the normalized LMS (NLMS) algorithm, where the original signal is the desired output of the NLMS algorithm and the filtered vocal and desired output is the input of the NLMS algorithm. The error output represents the resulting sound of

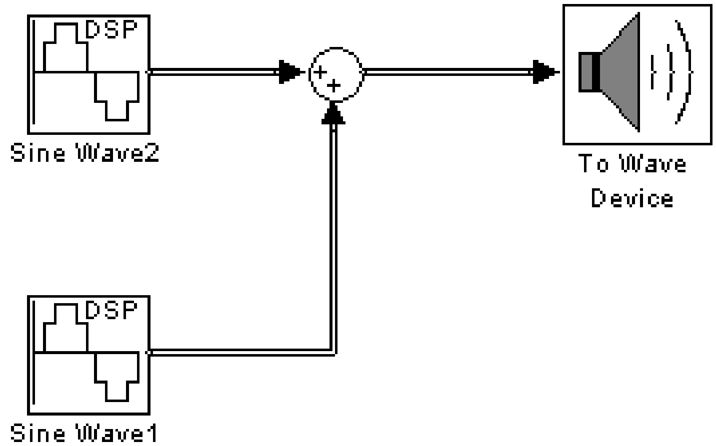

Figure 4.1: Simple Simulink diagram for antinoise cancellation of a sinusoid.

the system after cancellation. The three important signals of the AVC system are the original sound, desired sound and filtered original sound, and the error output of the NLMS algorithm. Each of these signals can be accessed through the blocks at the bottom of the Active Voice Control diagram, as seen in Fig. 4.2.

4.1.1 AVC with a Single Tone

The Acoustic Environment was modified to create a system to model a single tone AVC. This simulation desired to cancel a tonal noise at a higher frequency range, so the Acoustic Environment was modified to include the desired sound as a sine wave out of the frequency range of hearing and the unwanted sound as a sine wave at a frequency of 1710Hz. The NLMS algorithm takes both of these signals as its inputs, and its error signal is the resulting filtered signal. After running the simulation, the final filtered signal cannot be heard when playing the filtered signal using the Filtered Signal block in Fig. 4.2.

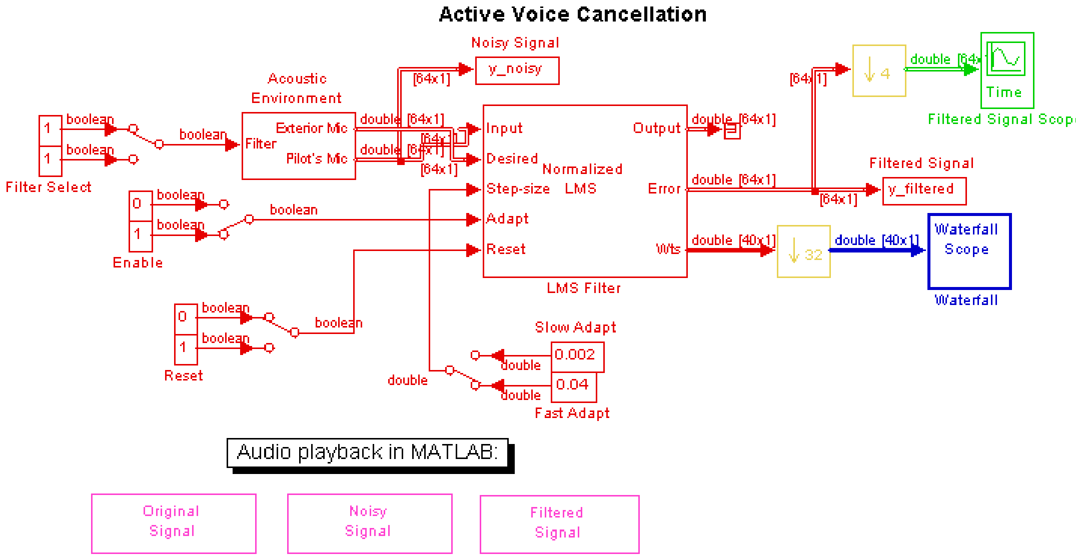

Figure 4.2: Active voice control Simulink diagram for single tone cancellation.

4.1.2 AVC with Recorded Speech

To better approximate the performance of the simulation, the Acoustic Environment was modified further to record speech from a microphone which is then played back as the filtered sound after the simulation is over. As seen in Fig. 4.5, the setup closely approximates that of the single tone simulation but with microphone inputs instead of sinusoid sources. As the simulation is running, the user speaks into a microphone. The signal from the microphone becomes the disturbance for the AVC system. The desired signal is set at 15kHz. The desired signal will effectively push the frequency of the disturbance to higher frequencies to reduce the sound. The filtered signal after the simulation has ended is attenuation of the speech. These results were found to vary among many people who participated in the simulation. Several stated that they could hear attenuation of their voice, while others could not. The different frequencies of the participants' voices as well as the complexity of their voices probably contributed significantly to this variation in results.

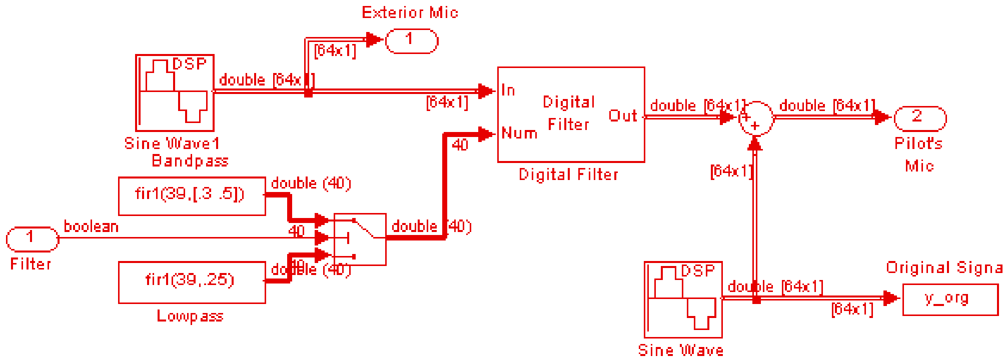

Figure 4.3: Acoustic environment for single tone cancellation.

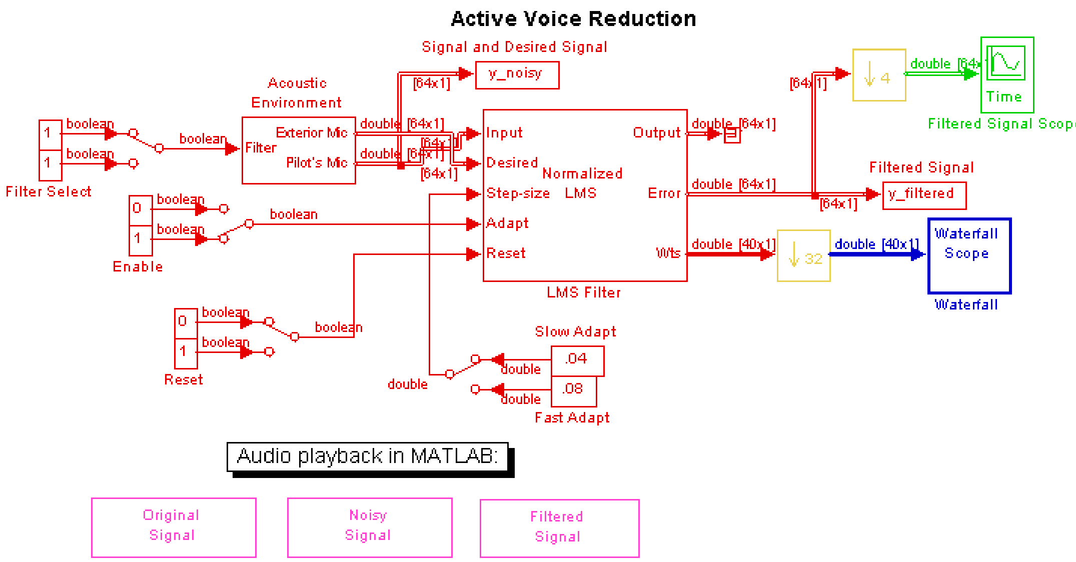

Figure 4.4: Active voice control Simulink diagram for recorded speech cancellation.

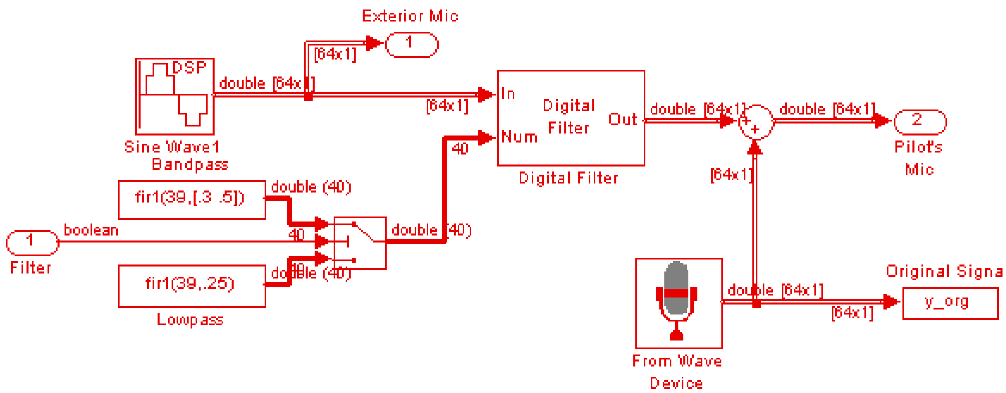

Figure 4.5: Acoustic environment Simulink diagram for recorded speech cancellation.

Chapter 5

Conclusion

The technology behind ANC must extend its range into higher frequencies before open cancellation of complex frequencies and sound fields can be achievable. If this technology is developed, cancellation of any type of sound will be possible, and passive techniques may be quickly eliminated. Unfortunately, as stated in [3]:

> . . . active noise control is restricted to relatively low frequency applications. This is because it is only at such low frequencies that the acoustic wavelength is large compared to the dimensions of the volume being controlled. Active control will thus never become a universal solution for all acoustic noise problems.

Even though ANC can be applied to higher frequencies to cancel voice as shown in the given simulations, the region of cancellation is too small for canceling a region large enough for a practical implementation. Even if the region was large enough, experiments show that ANC in open air environments can actually create more sound instead of canceling it. Despite these setbacks, AVC may be applicable within the operable range of ANC. Future work will be needed to take advantage of this potential area of research.

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Appendix

The following figures are the Simulink diagrams for ANC from the adaptive filter tutorial of the MathWorks website, found at [8]. Both of these diagrams were used in creating the AVC Simulink diagrams presented in Chapter 4.

C or C++ code can be developed from all of the Simulink figures by generating them from the Real-Time Workshop. This program is an addition to Simulink which allows the user to generate code directly from the Simulink diagrams, thereby avoiding the code writing stage of development. The Real-Time Workshop can generate code, an executable, and html files after configuring the block diagram. Additionally, an option is available for generating embedded programs on any microprocessor [8].

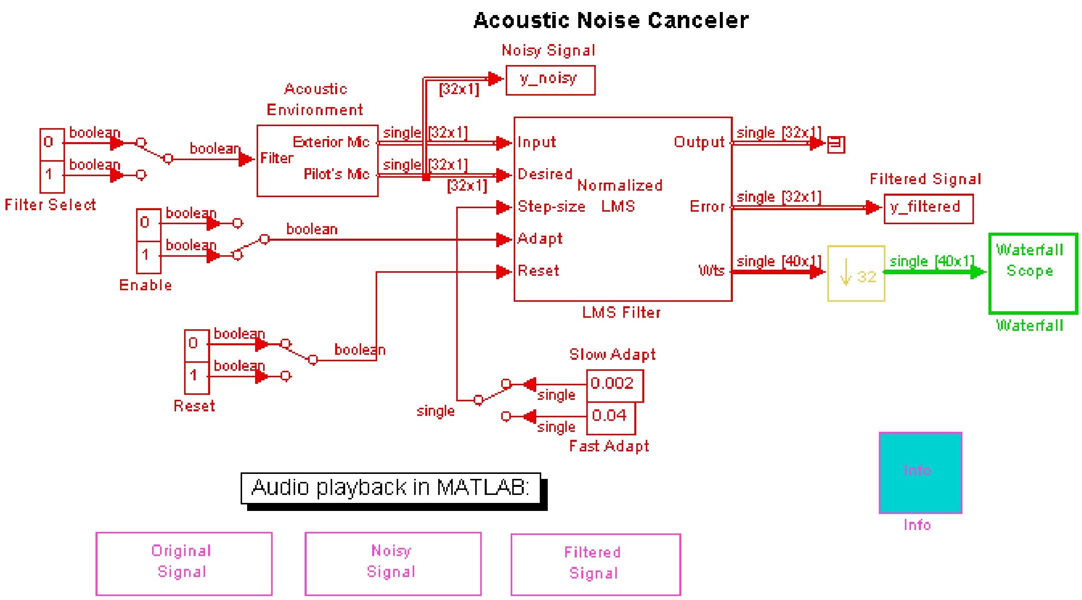

Figure 1: Acoustic Noise Cancellation Simulink Diagram [8]

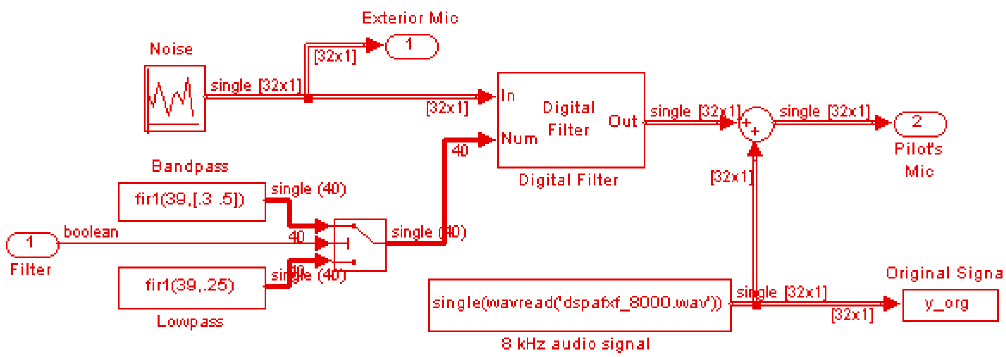

Figure 2: Acoustic Environment Simulink Diagram for Active Noise Control [8]