

**Active Control of Wind Tunnel Noise**

a Technical Report Submitted to

**NASA Ames Research Center  
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by

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## 1. Introduction

Unwanted sound or noise can be attenuated in two basic ways, either by passive control or by active control. The first method is prevalent in noise control problems. The principal reasons for the wide use of passive noise controllers are their low cost and their efficiency in medium and high frequency ranges. However, for low frequency noise, passive controllers become bulky, expensive and relatively ineffective when compared with active controllers. In cases where the unwanted noise is of low frequency and of high amplitude, an active noise control system can give desired attenuation at reasonable cost. One particular case of unwanted noise is wind tunnel fan drive noise.

A significant observation about sources in linear sound fields is that if two different source distributions can generate the same wave field and one source distribution is under control, then a simple change in sign makes the primary noise field subject to extinction by the presence of the secondary. Even though the source of anti-noise can be of completely different construction to that of primary field, the silence is, in principle, achievable everywhere outside the source distribution. The basic principle of active noise control (ANC) originates from when two pressure fields are arranged to overlap precisely with exactly opposite waveforms. Then they would destructively interfere to produce the constant pressure which is the condition of silence. This view of silence regards the null field as the superposition of sound and *anti-sound*. The term anti-sound is only used for those deliberately created waves that are produced by a controlled source such that they are superimposed on an existing noise field for the purpose of artificially creating a destructive interference.

Wind tunnel noise arising primarily from the motor drive is typically characterized by a number of discrete tones whose amplitude is significantly larger than the normal background noise of the wind tunnel. Elimination of this undesirable sound can greatly improve the capabilities of a wind tunnel for acoustic measurements. An active noise control system seems to offer an attractive means to achieve this noise reduction.

The proposed research is to construct a scale model of the motor drive section of an actual wind tunnel, in particular, the drive section of the Ames 80' × 120' wind tunnel. This model will be used to develop active control strategies for implementation in the actual wind tunnel. This will involve developing appropriate models for the wind tunnel and fan drives, designing controllers and identifiers together with sensor and actuator systems for implementation.

Completion of the proposed active control system will result in a very cost effective method to control noise over a wide range of operating parameters.

## 2. Research To Date

The research to date has primarily concentrated on selection of a suitable control strategy followed by extensive computer simulation and verification. Recently, an experimental model of a wind tunnel with an Active Noise Control (ANC) system has been fabricated and testing of the algorithm on a physical system has begun. From the experiments so far, some new associated problems have been identified for future research.

### 2.1 Theoretical Controller Design

A system almost always never performs as well as desired. Control systems are often added to improve and enhance system performance without the need to modify the system. There has been a wealth of knowledge and expertise built up in the area of linear control system design of continuous and discrete time lumped parameter systems. Basic features of such control systems include sensing devices to measure various system parameters and control devices which convert some error or reference signal into a form which can be input into the system. With the increased use of high speed computers in the control system and sensing devices has come expanded capability, increased sophistication in the controllers and the possibility of controlling more complicated systems.

Since the wind tunnel is subjected to changes in air velocity, ambient temperature, and sometimes to some nonlinearities in its sound field, the acoustic plant cannot be modeled by a fixed transfer function. Therefore, an adaptive control system is needed to take care of any changes in system transfer function or input noise.

#### 2.1.1 Selection of Adaptive Algorithm for the ANC System

Adaptive control techniques are very useful in a system identification problem. In most practical problems, the system is usually an unknown plant or the its behavior changes with time. In these cases, it is necessary to model the system using the output of the actual system and the output of the model. Obviously, the weights of the model have to be changed according to system behavior through some adaptive algorithm. Figure 2.1 shows an adaptive control system in a system identification framework illustrating its basic principle in terms of block diagrams. The figure shows an unknown dynamic system with a set of discrete-time measurements  $d(n)$  defining variations of the output signal of the system in response to a known stationary signal  $u(n)$  applied to the system input.

The system can be linear or nonlinear and the process can be stationary or time varying. For simplicity, let the system be linear and the process be stationary. In system identification, it is needed to develop a model for this system in the form of a transversal filter consisting of a set of delay line elements ( $z^{-1}$ ) and a corresponding set of adjustable coefficients ( $w_i$ ) as shown in the figure. At time  $n$ , the available signal consists of a set of samples  $u(n)$ ,  $u(n-1)$ ,  $\dots$ ,  $u(n-m+1)$ . These samples are multiplied by a corresponding set of adjustable tap weights  $w_1(i)$ ,  $w_2(i)$ ,  $\dots$ ,  $w_m(i)$ , to produce an output signal  $y(n)$ . If the actual signal output of the unknown system is denoted by  $d(n)$ , then the estimation error  $e(n)$ , is given by the difference between  $d(n)$  and  $y(n)$ . Usually  $e(n)$  is nonzero indicating the deviation of the model from the unknown system. In order to minimize this deviation, the estimation error  $e(n)$  is fed into an adaptive algorithm which controls the corrections applied to the individual tap weights in the transversal filter. Thus, the tap weights of the filter assume a new set of values for use on the next iteration and a new filter output along with a new value for the estimation error is produced at time  $n+1$ . The iteration process is continued for a number of times until estimation error  $e(n)$  becomes sufficiently small in some statistical sense. If the process is time-varying, the adaptive algorithm not only adjusts the filter coefficients, but also continually tracks the statistical variations in the process.

In a system identification problem, any unknown plant is usually modeled in z-domain either by an all-zero transfer function which has a finite impulse response (FIR) or by a pole-zero transfer function which has an infinite impulse response (IIR). The general form of an FIR filter in z-domain is given below:

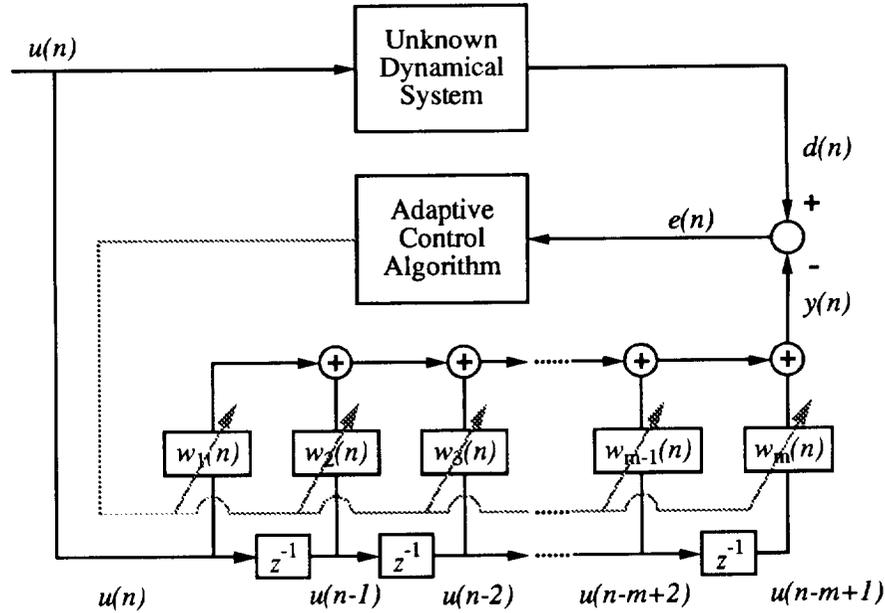


Figure 2.1. Adaptive control applied to a system identification problem.

$$H(z) = a_0 + a_1 z^{-1} + a_2 z^{-2} + \dots + a_m z^{-m}$$

where  $H(z)$  is the transfer function of the filter having a length  $m$ .

The general form of a IIR filter is given by

$$H(z) = \frac{a_0 + a_1 z^{-1} + a_2 z^{-2} + \dots + a_m z^{-m}}{b_0 + b_1 z^{-1} + b_2 z^{-2} + \dots + b_n z^{-n}}$$

Usually,  $b_0 = 1$  in this case.  $n$  indicates the order of a IIR filter.

A finite impulse response (FIR) filter is used to model the plant. FIR (finite impulse response) filters have certain advantages over IIR (infinite impulse response) filters. FIR filters are always stable even in a nonstationary environment. Adaptive IIR filters are computationally more involved than adaptive FIR filters and their stability is not guaranteed. Since the goal here is to use an adaptive filter which will be stable in all operating conditions and which will allow the use of a programmable controller to implement the adaptive filter, an FIR filter is used. Keeping in mind that the desired system model is an all-zero model, a suitable FIR adaptive algorithm has been selected for the model. The first choice is usually the LMS adaptive algorithm which is simple and easy to implement. The LMS algorithm can be modified to track the system with a little nonstationarity in the process. The LMS algorithm has been used for active noise control system by Burgess (1981), Poole and Leventhall (1976), and Eriksson(1987). Eriksson (1987) showed that if acoustic feedback is considered, the simulation of the system does not converge for a pseudo-random input. He attributed this to the fact that since acoustic feedback adds poles to the system, an IIR filter is needed in order for the algorithm to converge. It has been shown that the LMS algorithm converges for FIR filters in a Gaussian random process if a higher order filter is used with a very small convergence factor, although at a slow rate.

Considering these and others factors, Hollis and Anwar (1991), the LMS algorithm does not seem to be a robust one for the ANC system. The next adaptive algorithm that has been considered is the RLS algorithm which has the property of robustness and fast convergence rate, which come at the cost of computational complexity. Since RLS algorithms are deterministic and use all the past data of the process to adapt the filter coefficients, their

convergence properties are much better than those of LMS algorithms. Also with a slight change in the forgetting factor of the algorithm, the robustness of the filter can be preserved in a nonstationary environment. Since the exponentially weighted RLS algorithm with U-D factorization is numerically better suited for implementation, it seems a good choice for the ANC system. A brief description of the RLS algorithm is given in Hollis and Anwar (1991) and Anwar (1990).

### 2.1.2 Computer Simulation and Results

The ANC system is simulated to examine various algorithms and to examine the effects of different design parameters on the system dynamics both in the time and frequency domains. The basic assumptions that are made about the structural hardware of the ANC system are as follows:

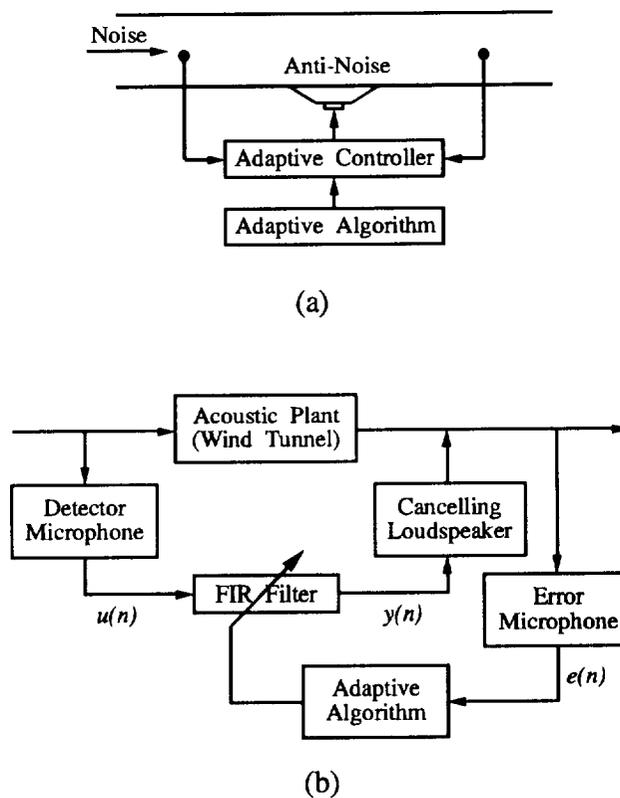


Figure 2.2 Model of the ANC system: (a) Basic structure of the hardware, (b) Block diagram representation.

The ANC system consists of the following equipment:

1. one detector microphone preferably unidirectional,
2. one noise canceling loudspeaker,
3. one error microphone (unidirectional or omnidirectional),
4. a programmable controller or digital computer with A/D, D/A converters, and anti-aliasing or low-pass filters.

Other assumptions made about the system itself are:

1. Only plane waves travel through the duct making the system one dimensional.
2. There is no delay in the error path. Usually there will be some delay in the error path, unless the error microphone is placed right on top of the canceling loudspeaker. However, for acoustic reasons, it is better to place the microphone at least some distance away from the canceling loudspeaker in the downstream direction.
3. The fundamental duct resonance frequency is higher than all operating frequencies. This is to ensure that the canceling loudspeaker produces only plane waves in the direction of the duct axis.
4. The acoustic plant is a pure delay plant. This assumption is reasonable if the noise terminates anechoically at the ends the duct. It was shown by Elliott and Nelson(1984) that the acoustic plant can be represented as a delay plant for a duct. If the end reflections are low in amplitude compared to the actual noise itself, the system can be assumed to be a delay plant for all practical purposes.

The basic structure of the hardware of the ANC system that is going to be used in the simulation is shown in Fig. 2.2(a).

The corresponding block diagram representation is shown in Fig. 2.2(b). Three types of input process have been considered in the simulation.

1. Discrete frequency noise containing 3 or more sinusoids of different frequencies.
2. Gaussian random noise with zero mean and unity standard deviation which is usually known as white noise.
3. Discrete frequency noise masked with white noise which is of broadband nature.

### 2.1.3 ANC System without Acoustic Feedback

Even though it is not practical to consider the system without acoustic feedback, it is done in this section to

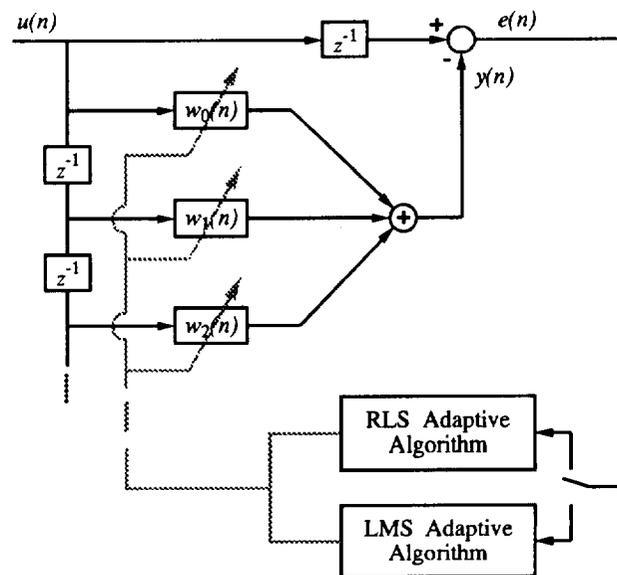


Figure 2.3. Simulation block diagram for the ANC system without acoustic feedback.

provide a comparison between LMS and RLS algorithms. The block diagram representation of the system where the acoustic plant is modeled as a single delay plant is shown in Fig. 2.3. Only a random input process is considered in this case. The FIR filter length was chosen to be 5.

The simulated responses of the systems in the time domain are shown in Fig. 2.4 for both RLS and LMS adaptive algorithms. For the RLS algorithm, the forgetting factor  $\lambda$  was chosen to be 1, meaning that the adaptation process has infinite memory. The convergence factor  $\mu$  of the LMS algorithm was chosen to be 0.025.

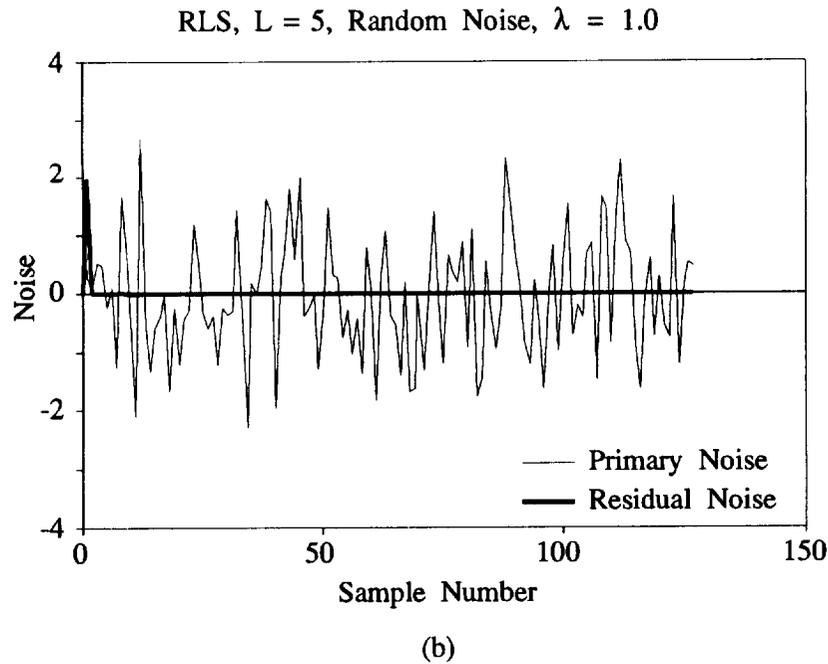
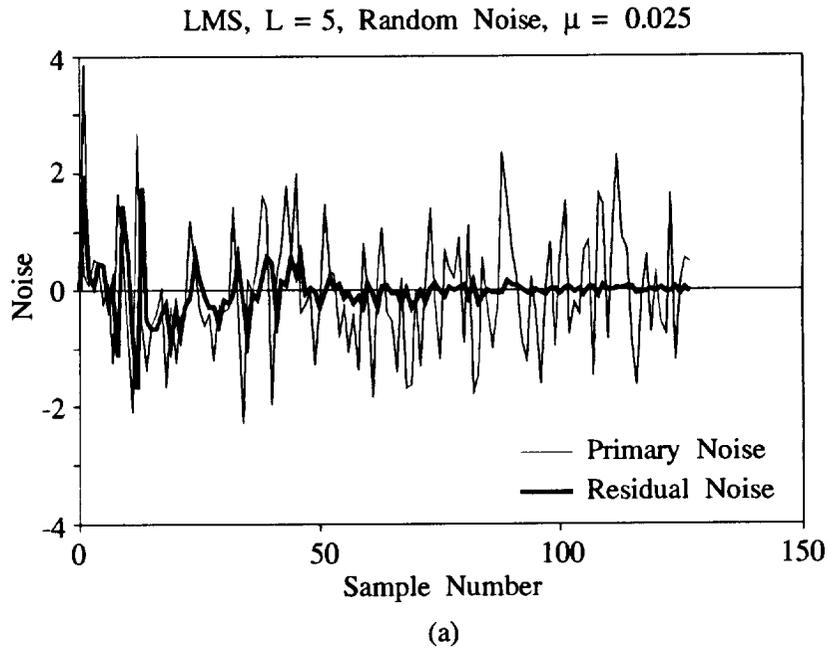
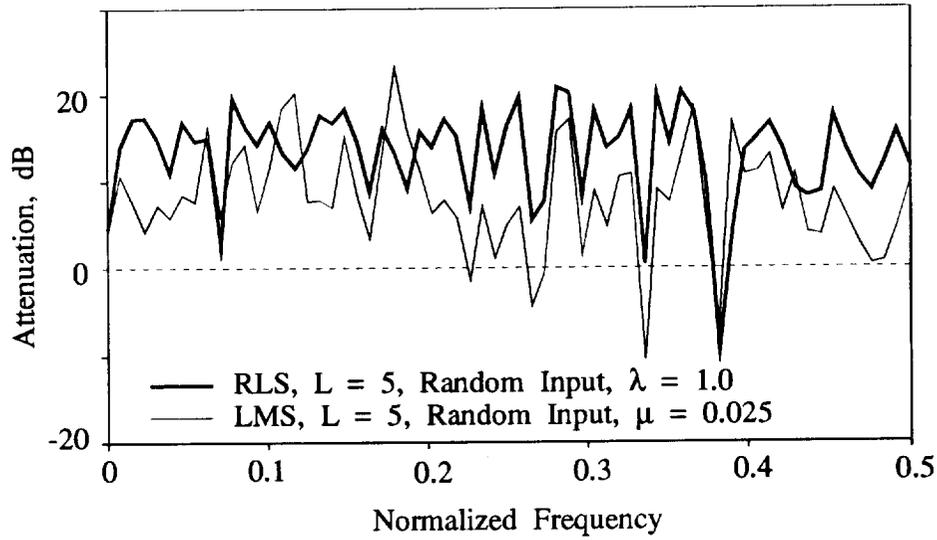


Figure 2.4. Time domain responses for the ANC system without acoustic feedback: (a) LMS and (b) RLS algorithms.

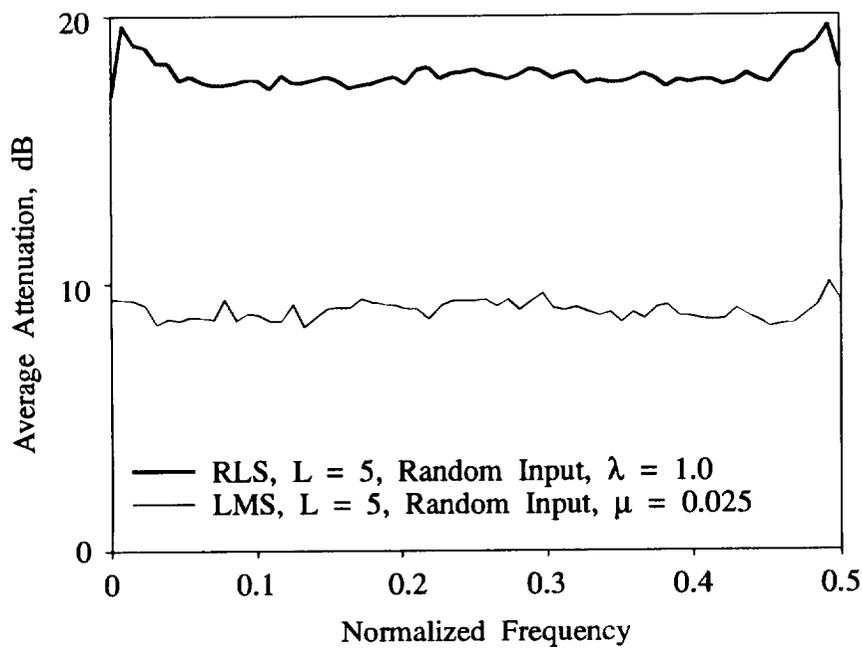
It can be seen from Fig. 2.4(b) that the residual noise output of the system with an RLS adaptive filter diminishes at a much faster rate than it does for the system with an LMS adaptive filter, Fig. 24(a).

Figure 2.5 shows the frequency domain attenuation. The attenuation is defined by the equation

$$\text{Attenuation, dB} = 20 \log_{10} UF - 20 \log_{10} EF$$



(a)



(b)

Figure 2. 5. Attenuation for the ANC system without acoustic feedback: (a) single simulated attenuation, and (b) average attenuation.

where

$UF$  = Fourier Spectra of the input noise

$EF$  = Fourier Spectra of the attenuated noise

The normalized frequency is defined by

$$\text{Normalized Frequency} = \frac{\text{Actual Frequency}}{\text{Sampling Frequency}}$$

Since the maximum frequency of FFT spectra is governed by the Nyquist frequency, the maximum value of the normalized frequency will be Nyquist Frequency/Sampling Frequency= 0.5. Figure 2.5(a) shows the attenuation for a particular random noise input. Intensification may occur at several frequencies for a particular random input over a particular time interval. If these attenuations are averaged over 100 cases of different random noise inputs, the average attenuation which can be expected is shown on Fig. 2.5(b). The average attenuation found for the RLS algorithm is 18 dB when averaged for 100 cases. For the LMS algorithm, over the same number of cases, the average attenuation is only 9 dB.

### 2.1.4 ANC System with Acoustic Feedback

The block diagram for the simulation of the ANC system with acoustic feedback is shown in Fig. 2.6. The acoustic plant is considered to be a single delay plant. The filter length was chosen to be 10.

For the RLS algorithm, the forgetting factor  $\lambda$  was chosen to be 1. The convergence factor  $\mu$  of the LMS algorithm was chosen to be 0.025. A simulation program was run for all three types of input processes. The results of the simulation are explained below.

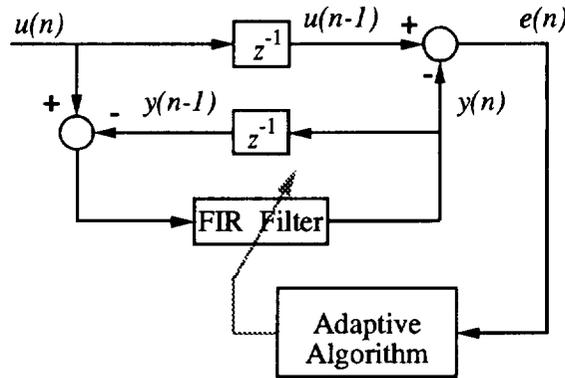


Figure 2.6. Simulation diagram of the ANC system with acoustic feedback.

### Discrete Frequency Noise

Three equally spaced frequencies were chosen as 100, 150, and 200 Hz which are in the low frequency range. The input noise was the summation of two sine waves (100 and 200 Hz) and one cosine wave (150 Hz). The amplitude of each of the sine and cosine functions was unity. Figure 2.7 (a) and (b) show the input and output noise for both RLS and LMS algorithms. From these, the convergence rate of the RLS algorithm appears faster than that of the LMS algorithm. The adaptation rate of the filter coefficients was much slower for the LMS algorithm than that of RLS algorithm. Figure 2.8 shows the frequency domain attenuation for the simulated ANC system. Attenuation occurs in all three discrete frequencies with a maximum of about 15 dB in the case of the system with the RLS adaptive filter. For the LMS system, even though it attenuates the input frequencies (a maximum of 27 dB), considerable intensification occurs at certain other frequencies. The use of a higher order filter may reduce this kind of intensification.

### White Noise

For white noise input, the convergence properties are much better for the system with the RLS adaptive filter than that with the LMS adaptive filter as can be seen from Fig. 2.9.

The RLS and LMS algorithms show considerable average attenuation over a wide range of frequencies, Fig. 2.10. The maximum simulated attenuation is about 18 dB at a normalized frequency of about 0.25. For the case of

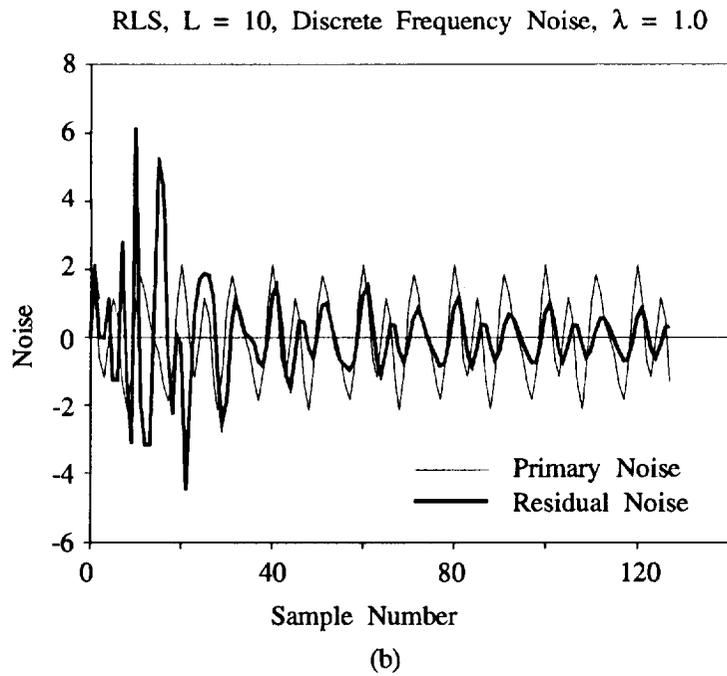
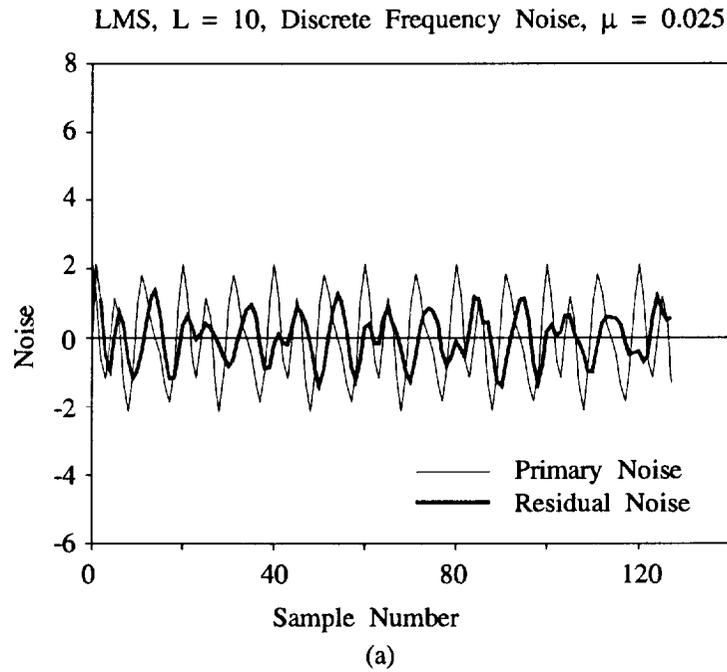


Figure 2. 7. Time response of the ANC system with discrete frequency noise: (a) LMS and (b) RLS algorithms.

white noise input, the maximum attainable average attenuation for both algorithms is about the same.

### Discrete Frequency Noise Masked with White Noise

Probably this kind of input is the most practical case that is encountered in noise control problems. As stated earlier, the noise spectra of a wind tunnel contains certain discrete characteristic frequencies masked with a hum which is of random nature. Time domain simulation results are shown in Fig. 2.11 (a). In the case of the system with the LMS algorithm, the system becomes unstable. Instead of diminishing, the residual noise keeps on increasing in each iteration. For this kind of noise, the robustness of RLS algorithm prevented the system from becoming unstable. As can be seen from Fig. 2.11, considerable average attenuation is obtained over the frequency range. Note that there can be intensification of various different frequencies for different random inputs.

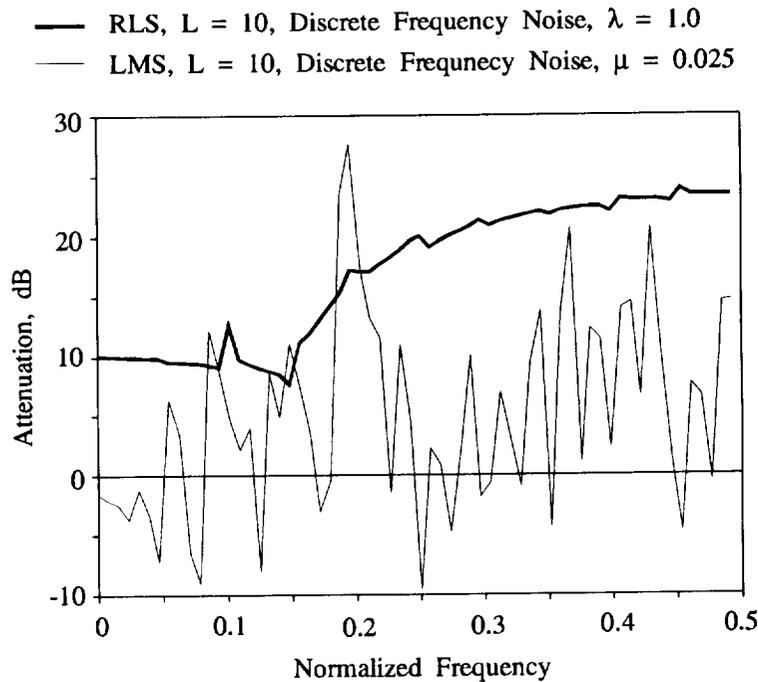


Figure 2.8. Average attenuation of the ANC system with discrete frequency noise.

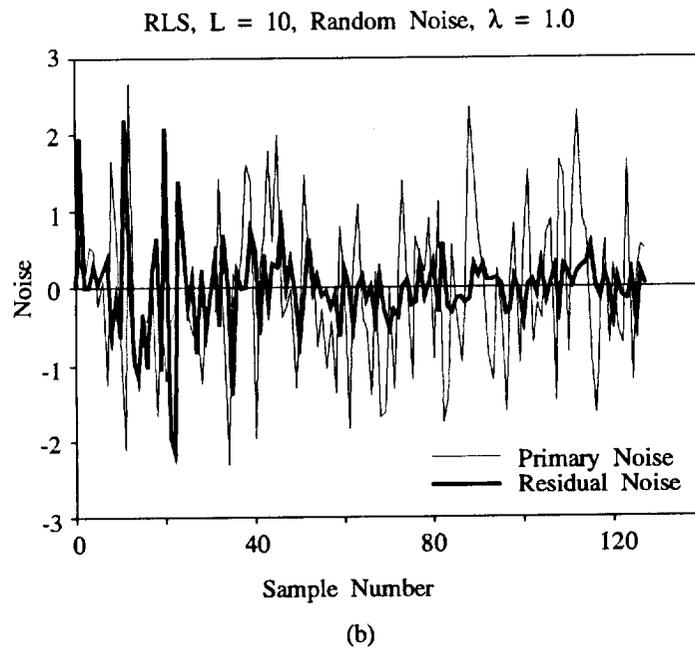
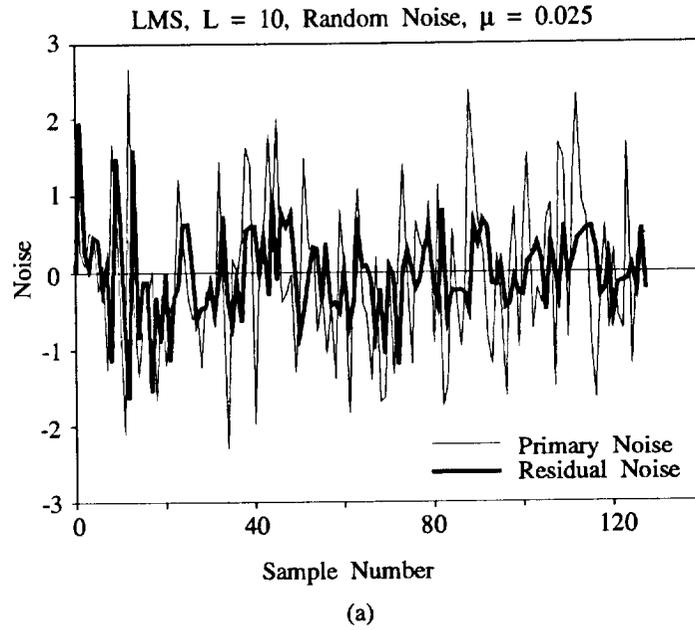


Figure 2.9. Time response of the ANC system with white noise: (a) LMS and (b) RLS algorithms.

### 2.1.5 Effect of Plant Model on the Simulation Results

In all the simulations presented here, the system model of the acoustic plant, i.e. the duct carrying the noise, is assumed to be a single delay plant. This assumption is reasonable to some extent as long as there are no end reflections of the transmitted noise through the duct. This is particularly true for a closed loop wind tunnel which

does not have any end. However, sudden changes in the cross-section of the wind tunnel will cause changes in the acoustic impedance resulting in reflections of sound at those sections. For this somewhat less realistic case, the acoustic plant was considered as a delay plant with 5 delay units. The only change found in the simulation was that the response of the system was delayed by the same amount. This case is of some importance, because the actual delay of the acoustic plant in time depends on the position of the detector microphone and that of canceling loudspeaker. But the delay in the z-domain will depend not only on the actual delay but also on the sampling period. The number of delays in the z-domain is given by  $\delta = \text{Actual Delay in Time}/\text{Sampling Period}$ . Therefore, it is more likely that the acoustic plant will be a delay plant with more than a single delay, depending on the sampling period given the position of the microphone and loudspeaker. For such a system, attenuation of noise will be obtained some time after the ANC system is turned on.

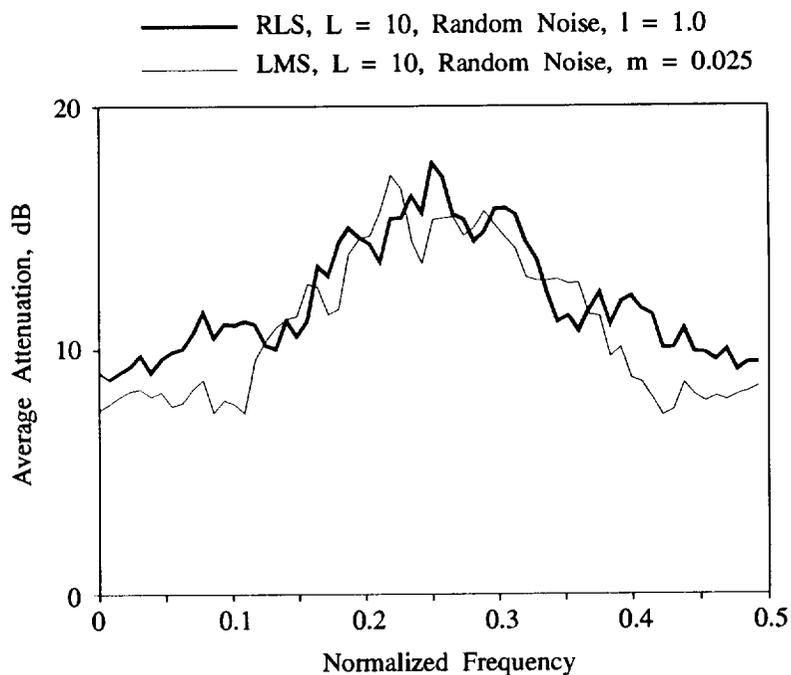
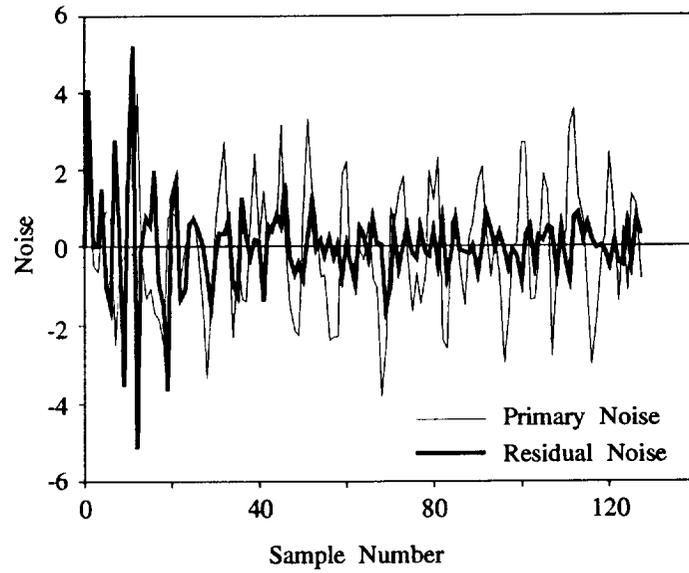


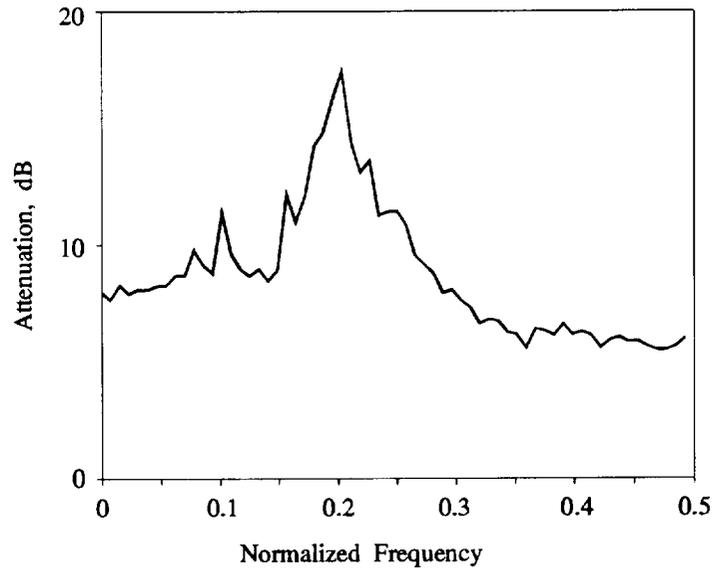
Figure 2.10. Average attenuation of the ANC system with random noise.

Next, a more realistic acoustic plant is considered which is not just a delay plant. For any air duct or closed loop duct with abrupt changes in the cross-section, reflections of the noise cannot be prevented. It is better to take these reflections into account when modeling the duct. As discussed earlier, these reflections will add poles to the acoustic plant indicating the need to consider a pole-zero model of the plant. Since the location of the poles depends on the amount of reflected noise, some value of the reflection coefficient of the duct has to be assumed. If it is assumed that 50 percent of the actual noise reaching the end of the duct is reflected back, then the reflection coefficient will be 0.5. This model of the acoustic plant is shown in Fig. 2.12(a). The simulation block diagram with the modified plant is shown in Fig. 2.12(b). Since it was demonstrated in the previous sections that the RLS algorithm has better convergence and attenuation properties than the LMS algorithm for an ANC system with acoustic feedback, henceforth simulation results using only the RLS algorithm will be presented. Only the white noise mixed with the discrete frequency noise input process is considered, since results with other kinds of input processes were found to show similar changes, Sohel (1990). The filter length considered here was 10 and the value of the forgetting factor was taken as unity. Figure 2.13 (a) shows the time domain response of the system. The rate of convergence is slower than that of the ANC system with single delay plant, Fig. 2.11. The reason for the slower convergence rate may be due to fact that the additional poles of the acoustic plant that are added to the total system cause the adaptation to take place at a slower pace which in turn reduces the rate of convergence of the ANC system. Figure 2.13(b) shows the average frequency domain attenuation for this simulation. As can be

RLS,  $L = 10$ , Discrete Frequency + Random Noise,  $\lambda = 1.0$



(a)



(b)

Figure 2.11. Time response and average attenuation for the ANC system with discrete frequency noise masked with white noise.

seen, some intensification of noise occurs over the lower and upper frequency ranges. Considerable attenuation is achieved over the middle range of frequencies with a maximum of 15 dB.

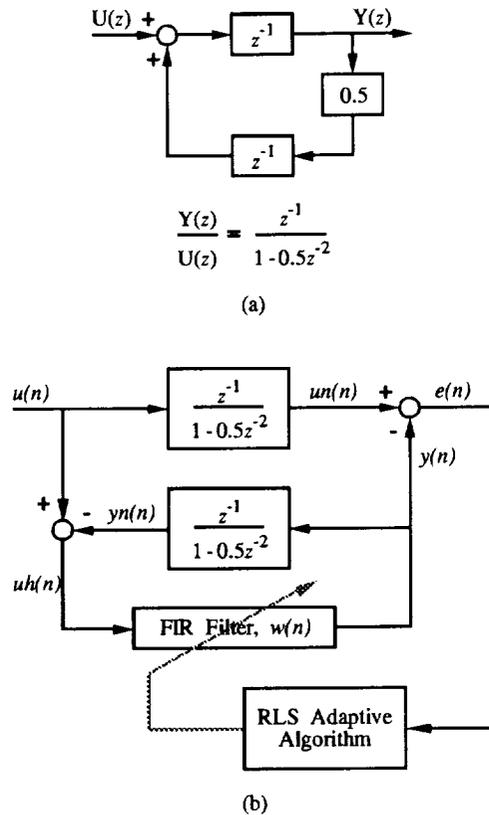
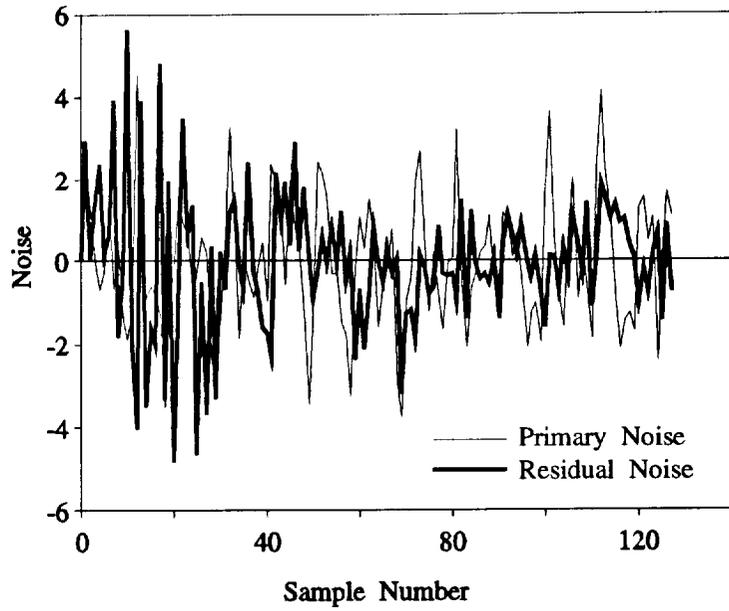


Figure 2.12. Modified acoustic plant and simulation diagram.

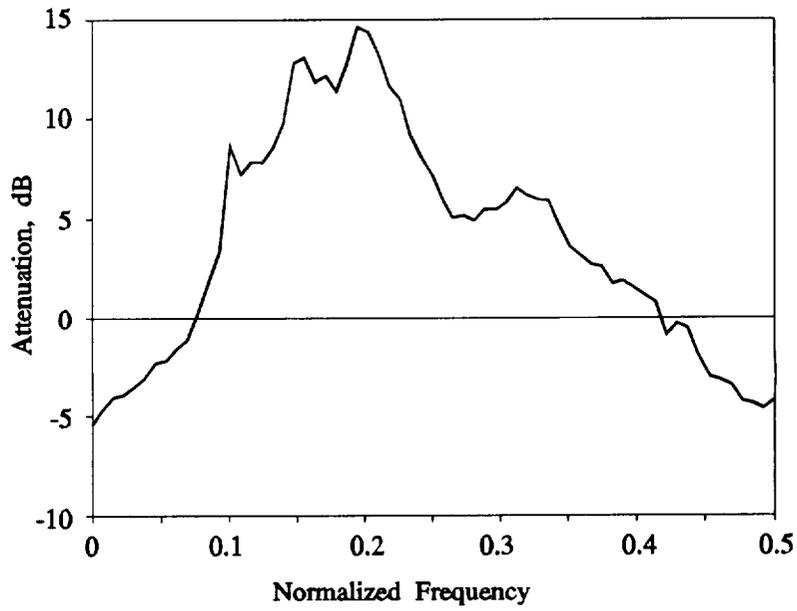
### 2.1.6 Effect of Filter Order on System Performance

In previous sections, it was demonstrated that the RLS algorithm is more stable than the LMS algorithm for an adaptive ANC system. The rate of convergence of the RLS algorithm was found to be faster than that of the LMS algorithm for the types of inputs considered. Frequency domain attenuation was better for the ANC system with the RLS adaptive filter than with the LMS adaptive filter in most cases, particularly in the case of discrete frequencies mixed with random noise. Considering all these factors, the RLS algorithm is adopted for the desired ANC system. Next it is necessary to determine the length of the filter. To see the effect of filter length on the convergence properties in the time domain and on the maximum attainable attenuation in the frequency domain of the ANC system, the simulation program was run for different filter lengths from 3 to 20. The results are shown in Fig. 2.14, Fig. 2.15, and Fig. 2.16 for the three types of noise inputs considered in this paper. The forgetting factor of the RLS algorithm was kept unity in all these cases. All the simulations done in this section assume the ANC system with modified acoustic plant as shown in Fig. 2.12. Figure 2.14(a) shows the variation of the rate of convergence of noise output with filter length  $L$  for discrete frequency noise. The rate of convergence has been defined by the inverse of the number of iterations needed for the squared system output noise averaged over 5 iterations to converge to a particular value  $\delta$ . The value of  $\delta$  was chosen to be 0.1 for all the cases presented here. Although the figure shows some abrupt changes, it can be concluded that the rate of convergence generally decreases with increasing filter length, i.e. more iterations are required for a desired convergence rate if a higher order filter is used.

RLS,  $L = 10$ , Discrete Frequency + Random Noise,  $\lambda = 1.0$

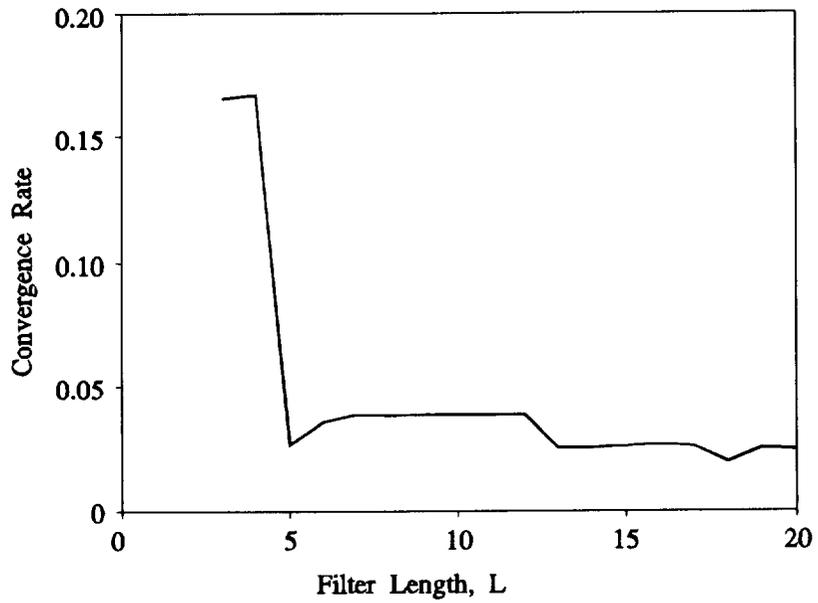


(a)

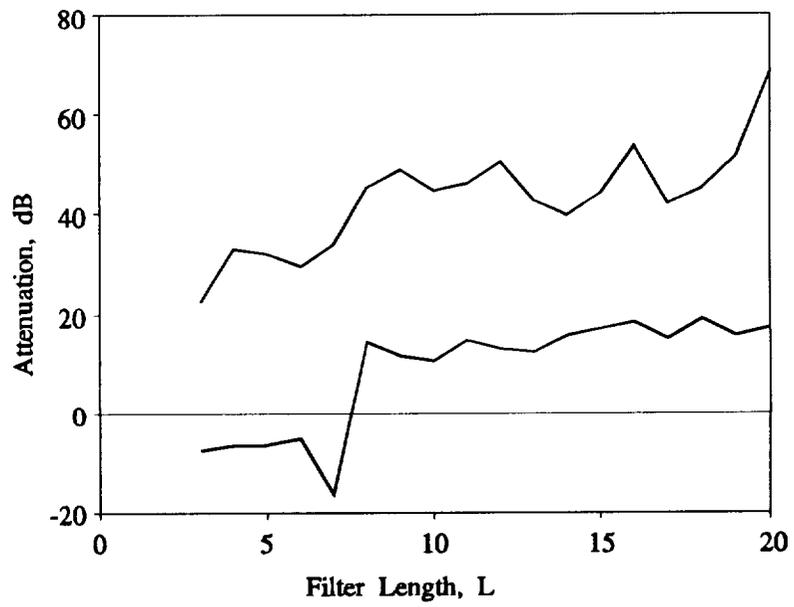


(b)

Figure 2.13. Time response and average attenuation for the ANC system with modified plant transfer function.

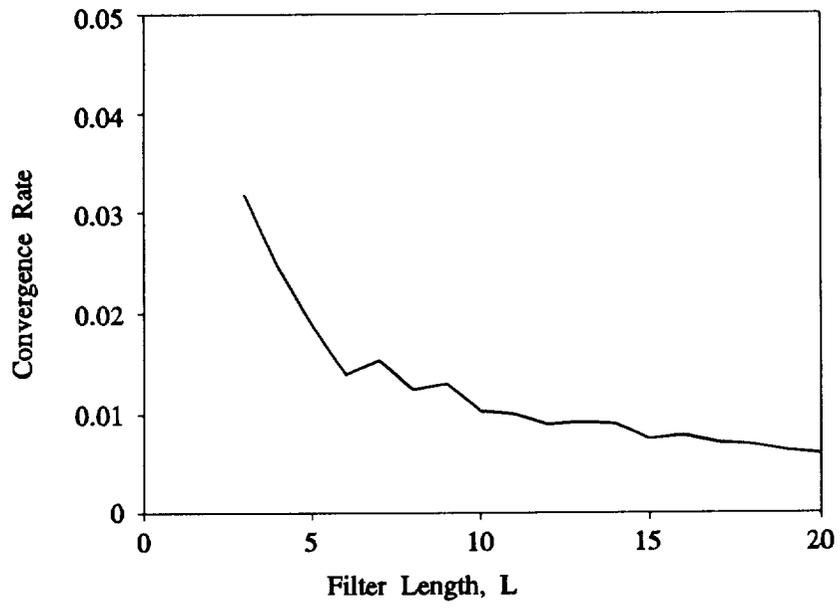


(a)

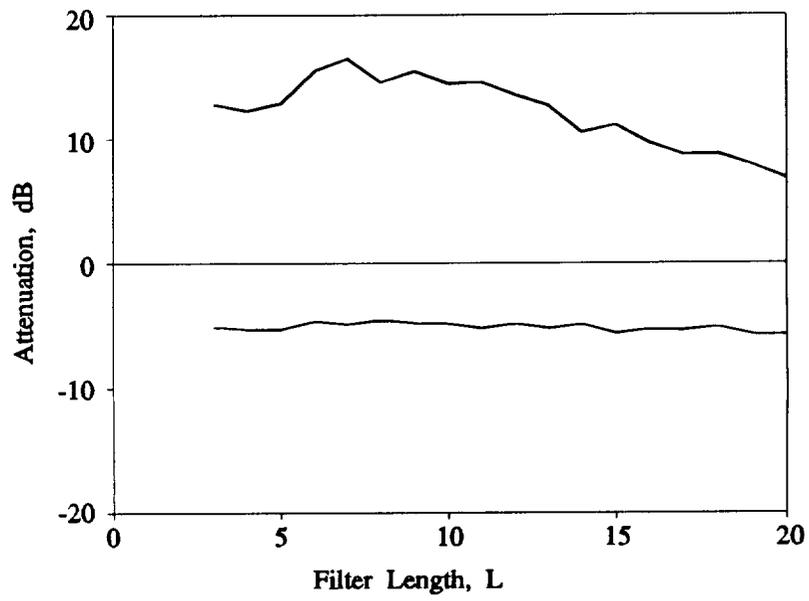


(b)

Figure 2.14. Effect of filter length for discrete frequency noise input.

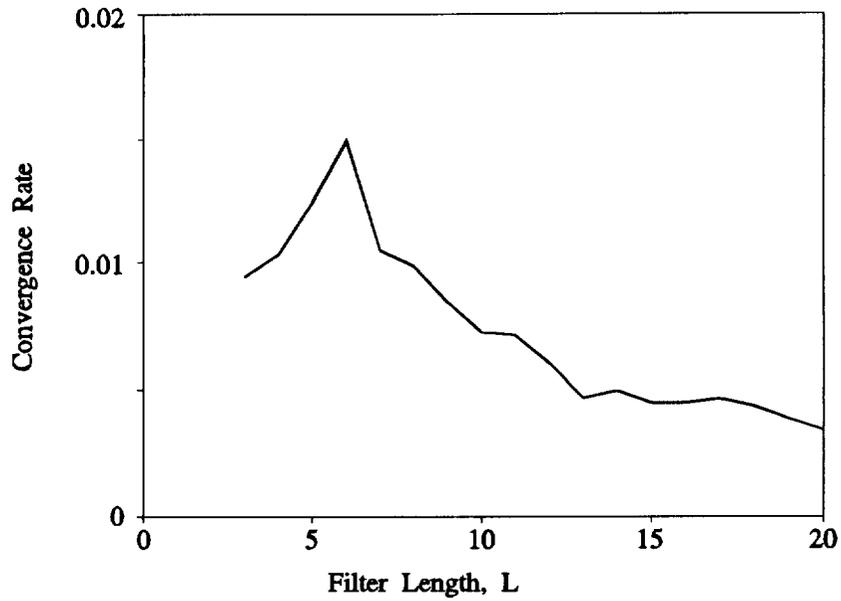


(a)

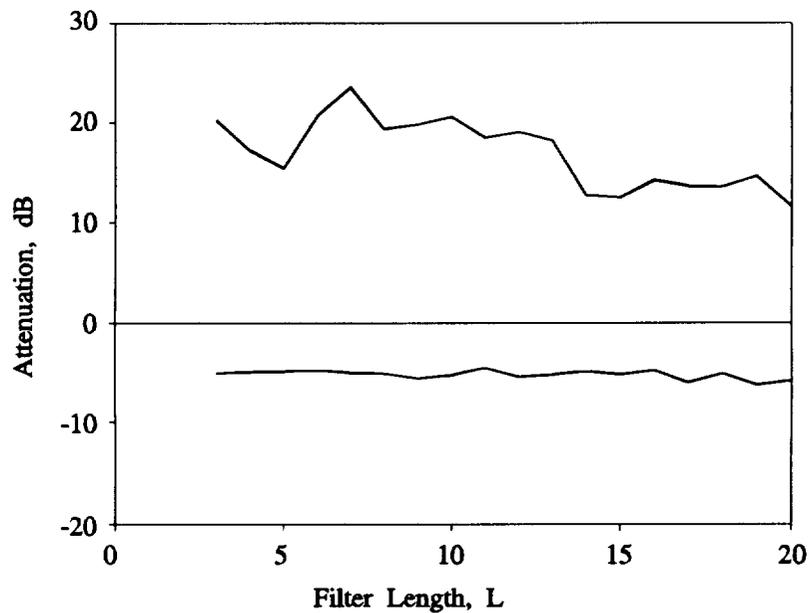


(b)

Figure 2.15. Effect of filter length for random noise input.



(a)



(b)

Figure 2.16. Effect of filter length for discrete frequencies mixed with random noise input.

This is likely to occur since the use of a higher order filter involves adaptation of a large number of filter coefficients. As a result, the overall adaptation rate will slow down because a longer period of time is needed for the algorithm to adapt the coefficients. Subsequently it results in a slower convergence rate. Figure 2.14(b) shows the change in maximum attainable average noise attenuation and maximum accompanying noise intensification in the frequency domain with filter length. It should be noticed that maximum attainable average attenuation and intensification increase with filter length. Hence the use of a higher order filter may not intensify noise at all for some value of the filter length,  $L$ . Although the maximum attainable attenuation increases with filter length, it may be desirable to use a lower order filter in some cases to achieve a particular rate of convergence.

Figure 2.15(a) shows the change in convergence rate with filter length for random noise input. In this case, higher order filters are again found to have a slower convergence rate. In the frequency domain, Fig. 2.15(b), the maximum average attenuation reaches a peak for a certain filter length, and then it decreases with filter length. In this case the maximum average intensification remains almost constant for all filter lengths examined. Similar results can be seen for the case of discrete frequency noise mixed with random noise, Fig. 2.16

Taking all three kinds of inputs into consideration, it was found that a filter order of 10 or less can give fast convergence as well as considerable attenuation without much intensification.

### 2.1.7 Conclusions

The need for an adaptive active control system was realized, since a wind tunnel is subjected to variations in air velocity, temperature, air turbulence, and some other factors such as nonlinearity. Among many adaptive algorithms, the LMS algorithm, which is the simplest one, has been used in an ANC system by some researchers. However, Eriksson's results, Eriksson (1985), showed instability in the ANC system with an FIR filter for random noise input. The RLS algorithm, although computationally more complex than the LMS algorithm, has better convergence and stability properties. The ANC system in the present work was simulated by using an FIR filter with an RLS algorithm for different inputs and for a number of plant models. Simulation results for the ANC system with acoustic feedback showed better robustness when used with the RLS algorithm than with the LMS algorithm for all types of inputs. Overall attenuation in the frequency domain was better in the case of the RLS adaptive algorithm.

Simulation results with a more realistic plant model and an RLS adaptive algorithm showed a slower convergence rate than the case with an acoustic plant as a delay plant. However the attenuation properties were satisfactory for the simulated system with the modified plant. The effect of filter length on the rate of convergence and attenuation was studied. It was found that the rate of convergence decreases with increase in filter length whereas the attenuation increases with increase in filter length. The final design of the ANC system was simulated and found to have a reasonable convergence rate and good attenuation properties for an input containing discrete frequencies and random noise. Further details and results can be found in Anwar (1990), Anwar and Hollis (1991), and Hollis and Anwar (1991).

## 2.2 Experimental Results

The experimental setup consists of a 3 m long tube of 0.3 m diameter, Fig. 2.17. A source speaker (simulating the fan drive) is located at one end of the tube and a control speaker is located at the midpoint of the tube. The other end of the tube is filled with an absorbent material since end reflections of an open tube cause the system to fail. A microphone is located near the signal source speaker and this serves as a detector microphone. A second microphone is located towards the other end of the tube and this is the error microphone. Both of these microphones have variable locations, in order to examine the effect of location on performance. The microphone outputs are amplified and input through A/D converters in the control computer (a Dell 333D) which uses the information to generate the control signal. The control signal is output through a waveform generator and amplified before the control speaker.

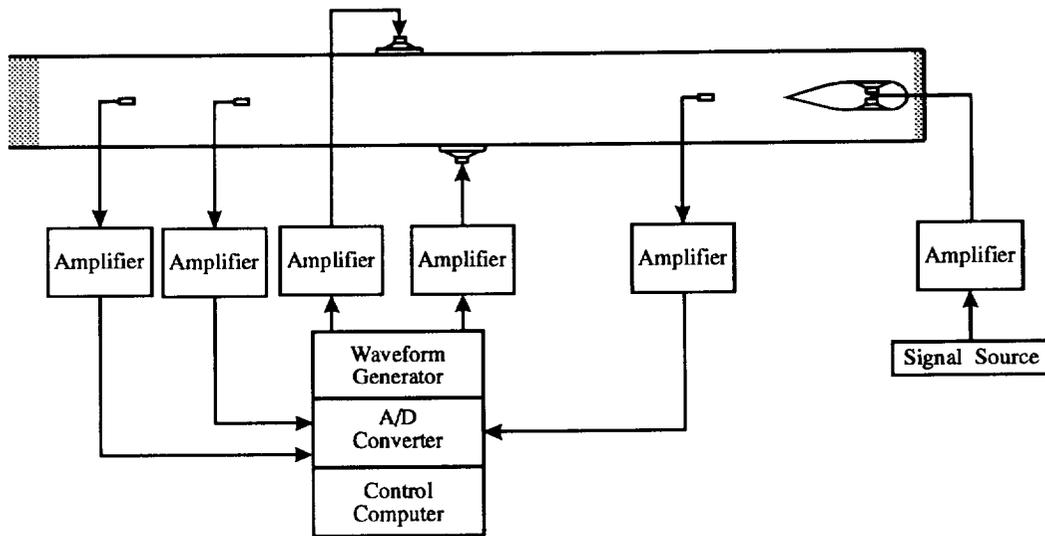


Figure 2.17. Schematic of Experimental Setup.

A single frequency sinusoid is used as the first type of input signal. The frequency is varied from a low of 50Hz up to about one-third the sampling frequency. The results for single frequency noise for various control filter lengths are shown below in Fig. 2.18 and Fig. 2.19. The lower limit of 50Hz is imposed by the speakers which fail to output a true sine-wave below 65Hz, and have an unusable output below 50Hz. It should be noted that the sampling frequency increases as filter length decreases. This is simply due to the fact that more calculations are involved per cycle for larger filter lengths. The shape of the single frequency curves resembles that of the simulated case for discrete frequency plus random noise, Fig. 2.13. This means that the controller is behaving as expected, except that the achieved attenuations are not as large as desired. Much of this could be due to the fact that the experimental setup is far from ideal. The control speaker location in the side of the tube means that the sound field generated will not match the source field for some distance down the pipe. Several speakers located strategically along and around the pipe may improve the situation.

Preliminary results indicate that microphone location, over the frequency range tried, has little effect on the results. Multiple frequency results show similar attenuations as single frequency cases, but results are inconclusive to date.

## 2.3 References

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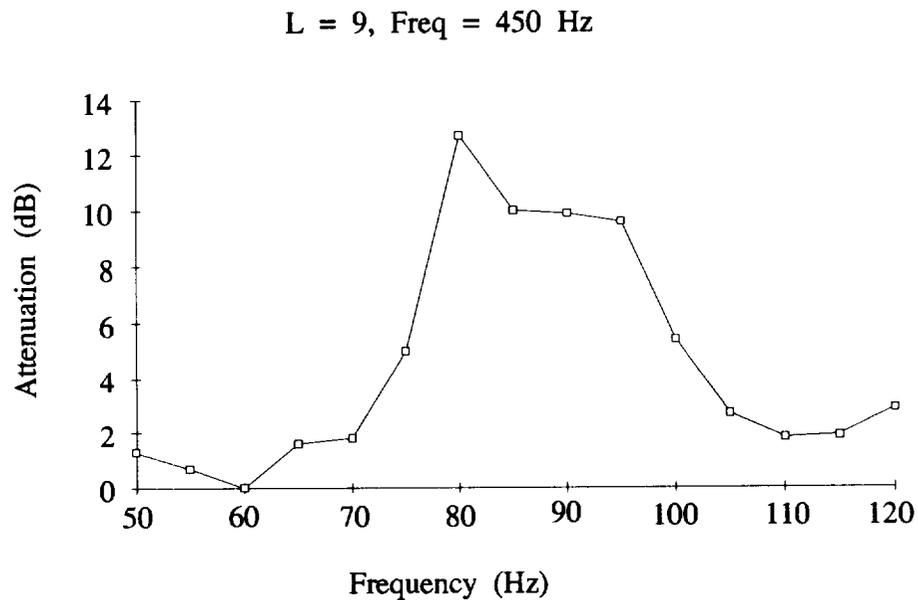
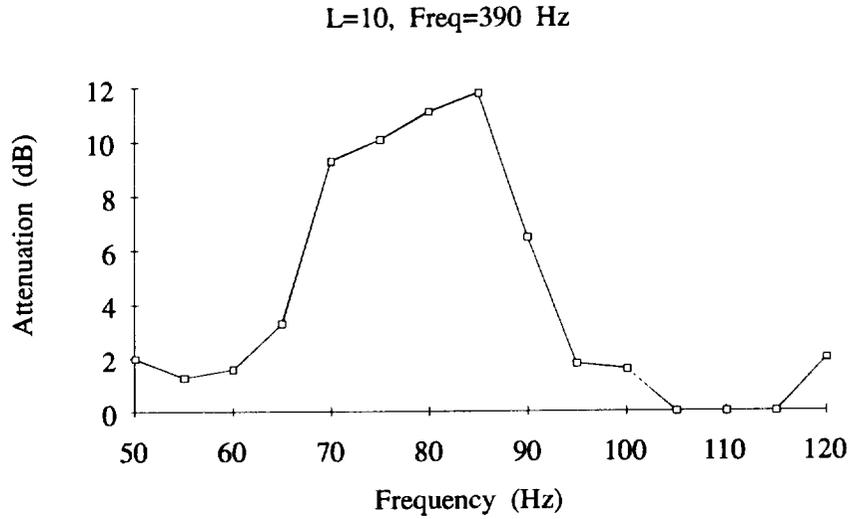


Figure 2.18. Experimental Results for Single Frequency Noise.

Eriksson, L. J. and Allie, M. C., and Greiner, R. A., 1987, "The selection and application of an IIR filter for use in active sound attenuation," *IEEE Transactions on ASSP*, ASSP-35(4), 433--437.

Elliot, S. J. and Nelson, P. A., 1984, "Models for describing active noise control in ducts," *I.S.V.R. Technical Report No. 127*.

Hollis, P. J. and Anwar, S., 1991, "An Adaptive Control System for Active Noise Attenuation", submitted to *J. Acoust. Soc. Am.*

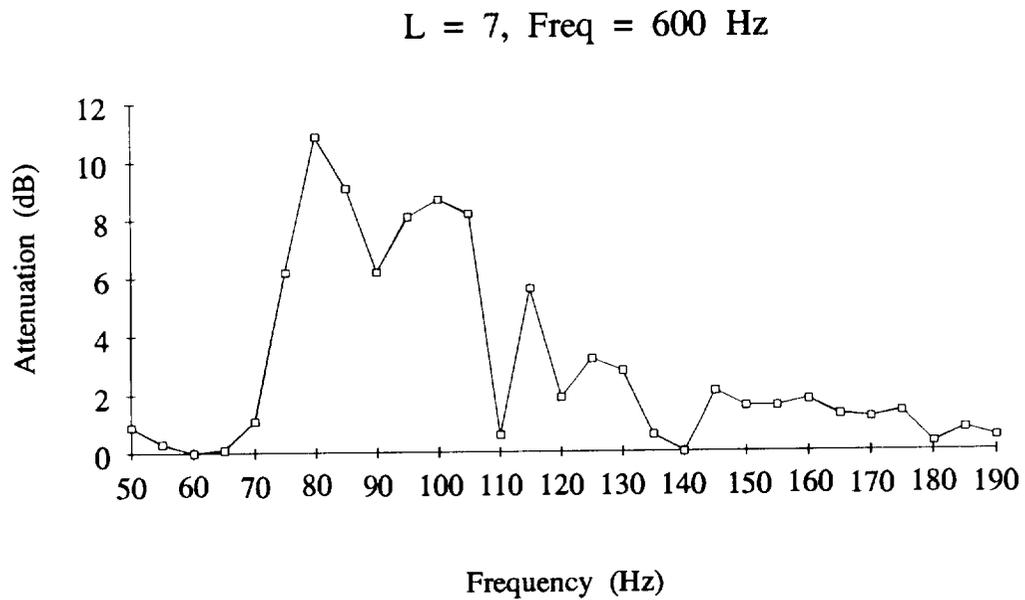
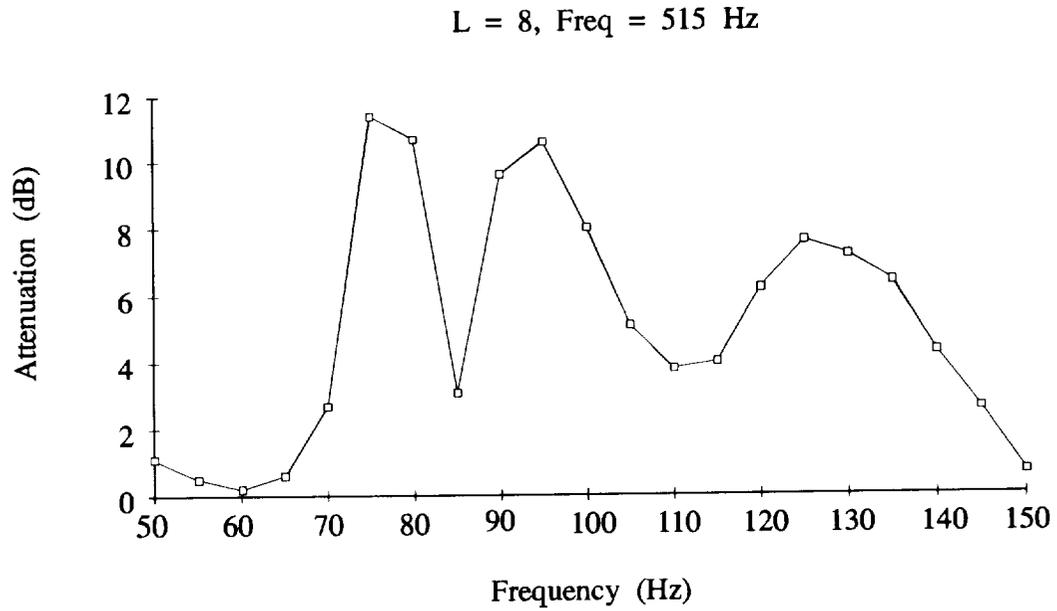


Figure 2.19. Experimental Results for Single Frequency Noise.

Poole, J. H. B. and Leventhall, H. G., 1976, "An experimental study of Swinbanks' method of active attenuation of sound in ducts," *J. Sound Vib.*, **49**(2), 257--266.

### 3. Proposed Research

#### 3.1 Experimental Configuration

This is a proposal to continue a feasibility study for applying active noise control to wind tunnel noise that arises primarily from the motor drive. This type of noise is typically characterized by a number of distinct discrete tones whose amplitude is much above the normal background noise of the wind tunnel, Fig. 3.1. The elimination of this unwanted sound can significantly improve the capabilities of a wind tunnel for acoustic measurements. The active noise control approach offers an attractive means to achieve this objective.

The concept and application of active noise control has been shown to be a feasible solution to the problem

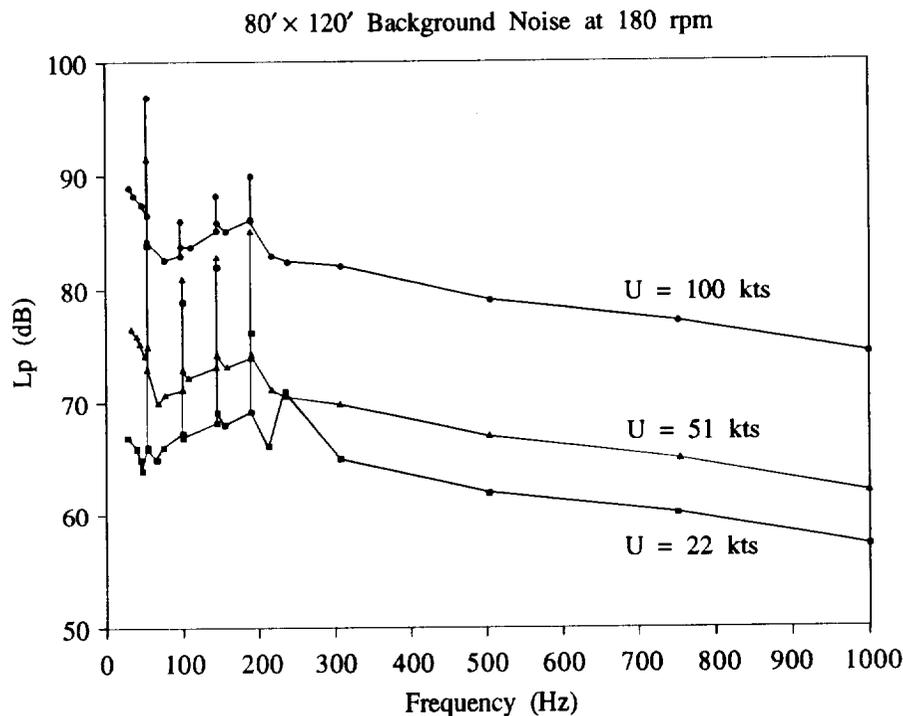


Figure 3.1. Frequency spectrum of a wind tunnel.

of unwanted wind tunnel noise. The next phase of the experimental investigation will involve fabricating a larger diameter duct to more closely resemble the Ames 80'x120' wind tunnel. Additional speakers and microphones will also be added to create a more realistic system. The basic wind tunnel model will be made up of short sections of approximately 2' length pipe joined together to make an overall duct length of about 20'. Metal or plastic rings fitted with speakers and microphones will be sandwiched between the other sections. This configuration will allow for easy access to and placement of sensors and speakers. The placement of these devices will be crucial to design of an effective control system. It is expected that this basic configuration will provide rules for positioning the microphones and speakers to obtain the desired results.

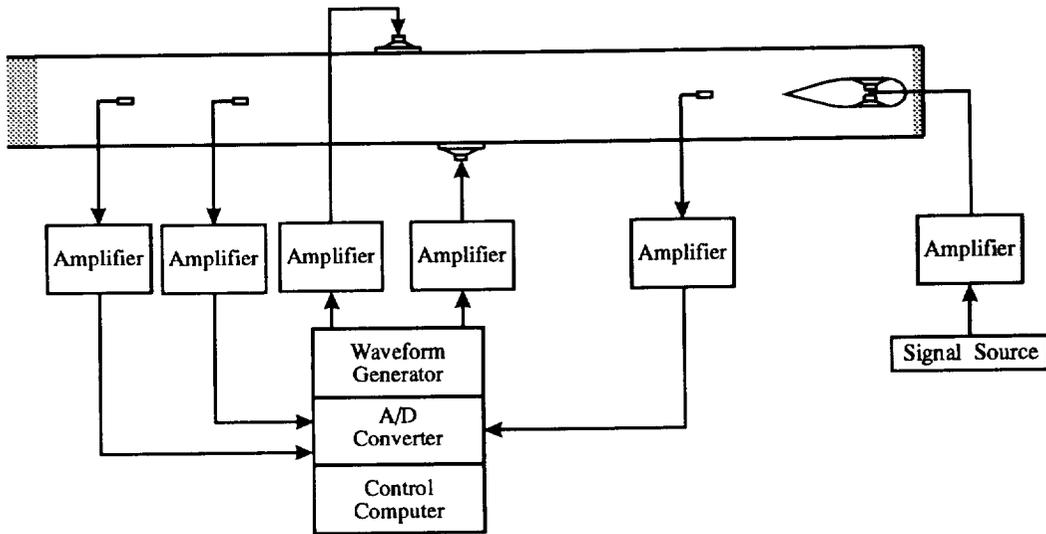


Figure 3.2. Experimental configuration with modified source.

The next study will replace the plane wave generator (single end speaker) with speakers arranged on a center body to more closely model the sound generated by the fan drive, Fig. 3.2. Additionally, the effects of various types of acoustic terminations will be investigated.

Finally, an actual model fan will be used as the noise source, Fig. 3.3. Together with the previously studied problems, the effects of nonconstant cross sections will be investigated. Since the Ames wind tunnel has a diverging section after the drive section, this feature will also be incorporated into the model to be studied. The results of the various experimental configurations should be a set of rules which will allow use of the control scheme in the full size system.

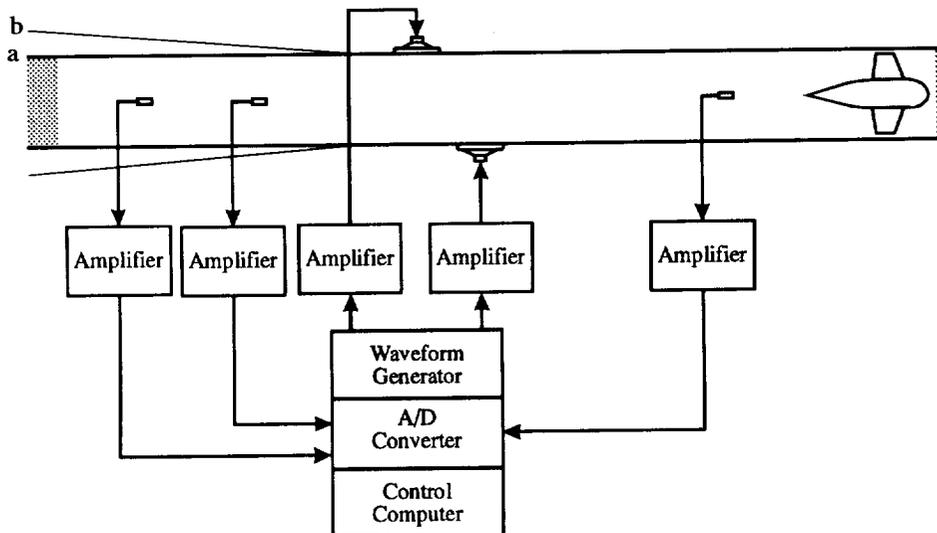


Figure 3.3. Experimental configuration with model fan.

#### **4. Schedule**

The remaining project period for a working control system for the NASA Ames wind tunnel is two years.

The first year will focus on the development of a modified test section which will have a more realistic speaker arrangement for simulating the fan drive noise sound source. Various acoustic terminations from an open end to an end closed with different materials will be investigated to model properly the effects of the remainder of the wind tunnel and the effects the control system will have on the fan drive system. It is expected that the control system may have some interaction with the noise source.

In the second year, a fan will be used as the sound source. This will allow fine tuning of the control system. Additionally, further work on including the effects of a diverging duct will be undertaken to make the model resemble as closely as possible the Ames wind tunnel.

## 5. Project Summary

The project has been broken down into three distinct yearly phases with realistic development goals for each year.

The first and most crucial year is the development of a working control system for a greatly simplified system. This has been accomplished, as demonstrated in the earlier sections. This system serves as the basis for the following two years work in developing a system suitable for use in the actual wind tunnel.

The advantage of the active control system approach is that it will be a cost effective method to eliminate or at least greatly reduce motor drive noise from the Ames wind tunnel. It is also expected that the approach taken will be sufficiently general that it might be applied to other similar noise reduction programs.