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PARALLEL ADAPTIVE NOISE CANCELLATION FILTER STRUCTURES

submitted by

R. BRUCE WALLACE B.E.Sc. P.Eng.

A thesis submitted to
the Faculty of Graduate Studies and Research
in partial fulfilment of
the requirements for the degree of

Master's of Engineering

Department of Systems and Computer Engineering

Carleton University

Ottawa, Ontario

January 7, 1991

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acceptance of the thesis

**PARALLEL ADAPTIVE NOISE CANCELLATION FILTER
STRUCTURES**

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R. BRUCE WALLACE B.E.Sc. P.Eng.

in partial fulfilment of the requirements
for the degree of Master's of Engineering

Thesis Supervisor

Chair, Department of Computer and Systems Engineering

Carleton University

February 16, 1991

ABSTRACT

Acoustically coupled noise to the microphone of a handsfree cellular telephone reduces the SNR of the transmitted signal and as a result reduces the performance of the units. This work presents the application of adaptive noise cancellation filters to this problem to reduce the noise in the transmitted signal.

This work presents parallel adaptive filter structures based on the adaptive noise canceler approach proposed by Widrow using a reference microphone. It is shown that in the multisource, multipath, non-stationary noise environment within an automobile, the structure of the noise cancellation filter will greatly affect the performance of the system.

The filters proposed use noise cancellation on subbands of the reference noise signals and multiple reference microphones for the various noise sources in addition to structure of the parallel adaptive filter to improve the ability of the canceler to track the non-stationarities in the noise processes and transfer functions achieving noise cancellation over the complete telephony audio band.

A two stage multireference subbanded beamforming canceler is shown to provide the best performance with a total noise power reduction of 16 dB and a maximum noise spectrum reduction of 28dB whereas the multiple reference filter proposed by Widrow is shown to provide only 6dB in total noise power reduction.

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TABLE OF SYMBOLS

Superscripted Symbols

H, h		FIR Filter transfer function tap weights
β		Adaption rate coefficient for LMS algorithm
e		Error in LMS filter output
S, R, P, d		Input signals to filter
n		Noise signal

Superscripted Symbols

T		Transpose of vector or matrix.
-----	-------	--------------------------------

Accented Symbols

$\bar{}$ (Overbar)		Column Vector
$\hat{}$		Estimate of a signal

Subscripted Symbols

t		Current sample of a scalar or vector delay line containing current sample.
$t+1$		Updated scalar or vector for use in next iteration.

1. INTRODUCTION

The increasing use of cellular telephones in automobile has caused a dramatic increase in the use of communication systems in high noise environments. Other examples include communication systems in the cockpits of planes, tanks, other military vehicles and high noise industrial examples.

The communication systems used in these situations should ideally be handsfree units so that the users' hands are available for the other tasks required. This is both a safety concern, in the case of an automobile and a necessity, in the case of a fighter aircraft. The requirement for handsfree operation also extends to the need for speech recognition systems to give the user added freedom. Speech recognition could be used for voice command dialing of a cellular telephone in addition to controlling systems in the vehicle.

The handsfree systems must provide the required performance for these applications and as a result the Signal to Noise Ratio (SNR) of the systems must be improved so that the required speech intelligibility is obtained. This will allow the far end listener to understand the speech more easily, especially when the performance of speech compression schemes such as linear predictive coding are enhanced by the reduced noise level, or so that a speech recognition system can perform adequately.

These requirements for improved quality of speech signals has led to research into noise cancellation and specifically adaptive noise cancellation so that the filter can optimally adapt for a particular situation and follow changes in noise characteristics that a fixed filter can't achieve.

Previous work on noise cancellation has shown that a two microphone adaptive noise canceler can be used to reduce the noise in the desired signal. One microphone is used to obtain the noisy desired signal and a second microphone is used to obtain a

reference noise signal. The results obtained show that noise can be reduced in the lower portion of the frequency spectrum that has the dominant noise energy at the expense of enhancement of the noise in the upper portion of the band.

The enhancement of the noise is an undesired effect that must be eliminated and cancellation of the noise in the upper band should be achieved.

This work presents an approach to adaptive noise cancellation using multiple reference microphones and parallel adaptive filter structures to obtain noise cancellation for a typical automotive handsfree cellular system. The proposed structure provides noise reduction over the complete audio band, eliminating the noise enhancement problem.

This report is organized as follows. The second chapter discusses the noise cancellation problem in an automobile and also reviews previous work on noise cancellation and related topics. The third chapter describes background material on the experimental methods used to obtain noise data and its subsequent processing and use in simulations that are discussed in chapter four and five. The fourth chapter details experimental simulations on adaptive filter structures leading to the design of an optimal filter structure. The fifth chapter describes simulations using the adaptive filter structure in the presence of speech. The last chapter details areas requiring further work in addition to conclusions obtained in this research.

Included in the appendices are listings of the routines used for the simulations and summaries of the simulation results obtained in this project. The listings of some simulation programs and the complete results plots of the simulations have been omitted as over 880 simulations were run to obtain the results discussed in this report and included in the summaries.

2. LITERATURE REVIEW

This chapter discusses previous work on adaptive noise cancellation and related topics as listed below:

- The characteristics of the noise cancellation problem in an automobile.
- Adaptive noise cancelers including a number of different approaches that have been proposed by various authors.
- Previous research on adaptive noise cancellation in an automobile.
- Adaptive beamforming and its applications to noise cancellation.
- Other methods for noise cancellation (Past methods and new approaches).
- Least Means Squared (LMS) algorithm for adaptive filters.

2.1. Description of Problem

The transmitted speech from a handsfree cellular telephone is impaired by the various noise sources in an automobile. Typically the microphone for the telephone is positioned a significant distance from the driver's mouth (20-30cm) so that the microphone does not impair the vision or motion of the driver. The microphone is typically placed on the steering wheel, the sun visor, or on the dashboard.

The placement of the microphone remote from the driver causes the driver's speech to be impaired by noise sources in the automobile. Examples of typical noise sources include engine noise, ventilation fan noise, air flow and road noise as shown in Figure 2.1. These noise sources can be divided into two distinct groups.

The first group is noise processes where the source can be considered as a point since the source of the noise is focused to a small region. Examples of these sources are

Figure 2.1 Typical Automobile Noise Sources.

fan and engine noise. The second group is processes where the source is distributed, such as air flow noise which originates over the exterior of the cabin.

The diagram shown in Figure 2.1 has been modelled in Figure 2.2 where the various noise sources are coupled to the handsfree microphone through their respective transfer functions. Typically the transfer functions are non-linear and non-stationary with the functions for the distributed noise sources being further complicated by the multipath nature of the system.

The characteristics of noise in an automobile has been analyzed extensively by Goubran [1,2] and Hebert [3,4]. The work by Hebert and Goubran involved the recording and analysis of the noise in an automobile under various driving conditions. The microphone placement for these experiments is shown in Figure 2.3.

The noise was found to be non-stationary, although it can be considered medium term stationary, which will allow the use of adaptive filters that are able to track the non-stationarities in the noise.

It was also shown that the cabin acts as a low pass filter with a cut-off frequency around 1000Hz. The exact location of this cut-off frequency is dependent on the particular car model under consideration. This low pass filtering effect causes the dominant portion of the noise energy to be located in the frequency band below 1000Hz.

The work by Goulding [5, 60] and Hebert [3] shows that the noise in microphones 1, 2, and 3 in Figure 2.3 is highly correlated with the noise in microphone 4 at frequencies below about 2000Hz. Correlation is also present above 2000Hz although it is much reduced.

This work leads to the conclusion that an adaptive noise cancellation system can be used to reduce the noise in the handsfree microphone (microphone 4) using adaptive

Figure 2.2 Transfer Function Coupling of Noise Sources to Primary Microphone.

Figure 2.3 Microphone Placement Used for Automobile Noise Recordings by Goubran [1] and Hebert [3,4].

filters to subtract the noise from this signal that is correlated with the noise in microphones 1, 2 and 3.

2.2. Adaptive Noise Cancellation Systems

A number of systems have been proposed for noise cancellation of speech signals. Each of these systems can be characterized by the diagram shown in Figure 2.4. This diagram shows a source for a desired signal in addition to noise sources. These sources are coupled to 1 or more microphones by unknown transfer functions.

The adaptive noise canceler receives the inputs from the microphone(s) and processes these signals to remove the noise so that an estimate of the desired signal is output from the filter. The adaptive noise canceler systems that have been proposed can be divided into two main groups. The first group uses a single microphone, while the second group uses more than one microphone.

A single microphone system is proposed by Sambur [6] and it uses the semi-periodic short term stationarity of speech to reduce the noise in a speech signal. The system proposed is shown in Figure 2.5 and it uses a pitch period detector along with an adaptive filter to estimate the current period of speech from the preceding period, which will reduce white noise which is uncorrelated between speech periods. The adaptive filter is designed to minimize the difference between the estimate and the actual period.

The results presented show that the SNR can be improved by up to 7dB with this system, with the drawback that the non-stationarity of speech will cause the system to introduce distortion and reverberation into the speech.

A second one microphone system was proposed by both Parson [7] and Frazier [8]. This system again takes advantage of the semi-periodic nature of speech and the resulting comb-like spectrum. The proposed system uses a pitch detector to

Figure 2.4 System Transfer Function Coupling Between Sources and Microphones (Primary and References) in a Noise Cancellation System.

Figure 2.5 Single Microphone Adaptive Noise Canceler Using Pitch Period Delay on Speech. Sambur [6]

determine the period of the incoming speech. This period is then used to form a comb filter that is used to pass the portions of the spectrum containing the speech while rejecting the bands containing only noise.

This system requires that the pitch period of the speech slowly change otherwise the comb filter will introduce distortion into the speech. Parson shows with subjective results that speech can be improved by this system although no objective results are provided.

Adaptive noise cancellation using more than one microphone was proposed by Widrow [9] when the structure shown in Figure 2.6 was presented. This structure uses a primary microphone to obtain the desired speech signal and a second microphone placed near the noise source to obtain a reference noise signal that can be adaptively filtered to remove the noise in the primary signal.

This structure makes the following assumptions based on the two microphone form of the generalized case shown in Figure 2.4:

- 1) The filtering of the desired speech signal between the source and the primary microphone is assumed to be negligible.
- 2) The reference microphone must be placed such that no speech is coupled to it.
- 3) The reference microphone is placed close to the noise source such that the filtering of the noise between the source and the microphone can be neglected.

Figure 2.6 Widrow [9] Single Reference Adaptive Noise Canceler

The corresponding effects on the transfer functions shown in Figure 2.4 are:

$$A_{11} = 1 \quad \{2.1\}$$

$$A_{12} = 0 \quad \{2.2\}$$

$$A_{21} = H \quad \{2.3\}$$

$$A_{22} = 1 \quad \{2.4\}$$

The system input signals are:

$$\text{Primary} = S + n_1 + e_1 \quad \{2.5\}$$

$$\text{Reference} = n_2 + e_2 \quad \{2.6\}$$

where S : Speech signal

n_1 : Noise in primary

n_2 : Noise in reference (correlated with n_1)

e_1 : Uncorrelated noise in Primary

e_2 : Uncorrelated noise in Reference

Except as noted these signals are all mutually uncorrelated.

The signals are also considered to be zero mean.

The resulting system equation is:

$$\hat{S} = S + n_1 + e_1 - \hat{H}(n_2 + e_2) \quad \{2.7\}$$

Taking expectations on both sides to find the power in the output signal:

$$E[\hat{S}^2] = E[(S + n_1 + e_1 - \hat{H}(n_2 + e_2))^2] \quad \{2.8\}$$

$$= E[S^2] + E[(n_1 - \hat{H}n_2)^2] + E[(e_1 - \hat{H}e_2)^2] \quad \{2.9\}$$

The output error can then be found as:

$$E[e^2] = E[\hat{S}^2] - E[S^2] \quad \{2.10\}$$

$$= E[(n_1 - \hat{H} * n_2)^2] + E[(e_1 - \hat{H} * e_2)^2] \quad \{2.11\}$$

Substituting Equation 2.3 leads to:

$$E[e^2] = E[((H - \hat{H}) * n_2)^2] + E[e_1^2 + (\hat{H} * e_2)^2] \quad \{2.12\}$$

Hence, the error in the output signal is a combination of two components:

The first component is noise that the canceler was unable to remove because of the differences between the estimated transfer function and the actual transfer function. This error can be caused by non-optimal convergence of the canceler due to uncorrelated noise in the system (e_1 , e_2), speech (S), and non-stationarity of the noise processes and transfer functions. Limits on the adaptive filter such as the number of taps in the FIR filter will limit the ability of the canceler to cancel long term portions of the transfer function between the primary and reference microphone.

The second component is the combination of noises in the output due to uncorrelated noises coupled into the system. Examples of these are thermal noise and quantization noise that can be reduced through appropriate system design. A second and more critical example is noise processes that are present in one microphone that are not present in the other microphone.

The above analysis shows that to obtain optimal performance from the canceler, it must provide the following features:

- 1) Adequate filter length to cancel the significant portion of the tails in the transfer function.

- 2) Appropriate adaptive structure so the canceler can adapt and maintain convergence to the non-stationarities in the noise processes.
- 3) The system is designed to minimize the thermal and quantization noise.
- 4) All noise processes in the primary microphone must have an associated noise process in a reference microphone.
- 5) Noise processes must not be present in the references that are not present in the primary.

The first two items in this list imply opposite constraints on the system as the first requires the use of longer filters while the second requires the use of shorter filters that can adapt faster while remaining stable. Both of these constraints indicate that the system should be designed to obtain a reference noise signal that is as closely correlated to the noise in the primary signal as possible. This will reduce the requirements on the filter function by reducing delays and also the length of the impulse response that the filter must model.

The third requirement involves the design of the system A/D converters and other logic and is beyond the scope of this work.

The fourth and fifth requirements imply that the system should use multiple reference microphones with appropriate placement to minimize speech and noises not present in the primary to obtain the various noise sources in the car adequately. This paper will show how a multireference adaptive noise canceler will outperform a single reference canceler.

The application of the Widrow canceler has been discussed by a number of authors. Boll [10] and Pulsipher [11] present a canceler with 15 taps that provides 20dB in noise reduction under simulated stationary noise conditions. The long filter was

required to properly model the delay and response between the primary and reference microphone.

This shows how in adaptive noise cancellation it is desirable to have the primary and reference microphone located as close together as possible to reduce the delays and differences between the noise in the primary and reference microphones. This requirement is contrary to the requirement given in Equation 2.2 that the speech signal not be present in the reference microphone, as close placement of the reference microphone to the speech source will introduce coupling of the speech to the reference.

Harrison [12,13] presents the application of adaptive noise cancellation to fighter cockpit communications. His results show an 11dB noise reduction using a filter with only 50 taps. The shorter filter was possible because the two microphones were located on the oxygen mask with the mask providing adequate protection of the reference microphone from the speech because of the high ambient noise in a fighter.

Gardner [14-20], Al-Kindi [21] and Dunlop [22] have presented a number of structures to protect the adaptive noise canceler from speech coupled to the reference microphones. A example of one of these structures is shown in Figure 2.7. These structures use two or more adaptive filter stages. The first stage of the structure is a standard adaptive noise canceler with the second stage adaptively filtering the output of the first stage to eliminate differences between the original primary signal and the estimate which will improve the desired signal since this is the only signal present in both inputs to the second stage. These results show that the structures tend to reduce distortion introduced into the speech by the first stage. A speech detector is used so that the first stage adapts during non-speech segments, while the second stage adapts during speech segments.

Figure 2.7 Two Stage, Single Reference Adaptive Noise Canceler proposed by Gardner [14] with Second Stage to Reduce Distortion of Speech Caused by the First Stage.

Mikhael [23,24] proposed a structure extending the Gardner structure using two reference microphones and including an initial stage to decorrelate the references with respect to each other. The structure presented requires a very complicated adaption procedure requiring that each filter be allowed to adapt individually and that the system must remain stationary so that the frozen filters do not require updating. This complicated adaption scheme makes this structure of limited practical use.

Multiple reference adaptive noise cancelers were also proposed by Widrow [9] and the proposed structure is shown in Figure 2.8. This structure is an extension of the single reference canceler shown in Figure 2.6. No results were presented regarding the performance of this canceler.

Feder [25-27] has presented a new filter based on a maximum likelihood criteria that assumes that the reference signals contain some speech. The algorithm uses estimation followed by maximization to estimate complete statistics for the system from the incomplete input data based on the algorithm presented by Dempster [28]. The results presented show that this algorithm provides improved performance over the LMS algorithm at the expense of increased complexity. The performance improvement was shown in subjective listening tests comparing the affects of the filters on the speech signal.

Other methods have been proposed for the cancellation of noise from speech in addition to the adaptive noise canceler discussed above. Boll [29] presented a method based on spectral subtraction where the spectrum of the noise is estimated during periods without speech and this spectrum estimate is then subtracted from the spectrum of the periods that contain speech to remove the noise. This system requires that the noise spectrum remain stationary between speech and non-speech periods and it also requires

Figure 2.8 Multiple Reference Adaptive Noise Canceler Proposed by Widrow [9].

that FFT's and IFFT's be performed in real time which is a drawback over the simpler time domain solution.

Amin [30] proposed a spectral domain adaptive noise canceler similar to the Widrow canceler with the exception that the system estimates the spectrum of the primary and reference signals. The system then adaptively filters the reference signal and subtracts the result from the primary signal spectrum to reduce the noise in the primary signal.

A number of researchers have discussed the problem of cancellation of the noise in an automobile. Goubran [1,2], Hebert [3,4], Goulding [5, 60], and Lecomte [31] have discussed methods for the reduction of noise in an automobile.

Goubran shows the results for the single reference noise canceler shown in Figure 2.6 that provides up to 12dB in noise cancellation. This cancellation is in the lower frequency band below 1200Hz while the canceler has the drawback that it enhances the noise in the band above 1200Hz as shown in Figure 2.9. The figure shows the noise cancellation performance using the engine compartment microphone as the single reference.

Goulding has pointed out that the effectiveness of this cancellation is very limited since the reduction of the noise in the lower frequency band is offset by the enhancement of the noise in the upper frequency band that contains most of the speech information.

All four authors have discussed the possibility of non-causality in the noise signals reaching the canceler. The cause is noise sources that are located closer to the primary microphone than to the reference microphone. This will cause the noise to appear in the primary microphone before the associated noise in the reference microphone. Reference microphone placement can be used to minimize this effect in

Figure 2.9 Typical Performance of Single Reference Noise Canceler Shown in Figure 2.6. Results Presented Use a Reference Microphone in the Engine Compartment with Noise Recorded at 60km hr, Ventilation Fan Off.

addition to the inclusion of delays on the primary speech path in the canceler to compensate for the effects. These delays must be used with care as they will also affect the performance of the telephone as the delays will affect the ease of conversation on the phone.

Goubran has also presented results showing the use of subband cancellation to eliminate the enhancement of noise in the upper band. The canceler shown in Figure 2.10 uses a low pass filter on the reference and the primary signal before they are input into the adaptive noise canceler. The output of the filter in the canceler is then subtracted from the unfiltered primary signal with an appropriate delay to compensate for filtering. The results show that this canceler will provide noise cancellation in the lower frequency band without affecting the noise level in the upper band. This provides a distinct advantage over the complete band canceler since no noise enhancement has been introduced by the canceler.

Subband cancelers have also been proposed by Gilloire [32,33] and Shynk [34] where the reference is filtered into a number of subbands each of which are passed through a separate FIR filter. Gilloire's work filters both the reference and primary signals and each of the sub-band signals is passed through an independent filter, each generating an error signal used for the adaption of the associated filter.

Gillioire application is echo cancellation and the algorithm takes advantage of the ability to subsample the bandlimited sub-bands to obtain increased computational efficiency.

Shynk proposes a canceller structure that bandpass filters the reference signal passing each band through a separate adaptive pole-zero filter (1 pole, 1 zero). The filtered reference signal is then reconstructed and subtracted from the primary signal forming a single reference used to adapt all the filters. This proposal has the advantage

Figure 2.10 Single Reference, Single Subband Adaptive Noise Canceler Proposed by Goubran [1].

that the subband filtering requirement has been cut in half but has the disadvantage that the single error will degrade performance as a misadaptation on one filter and will affect all of the sub-bands.

Hebert has extended the principle of subband cancellation to the upper band and has shown that noise reduction in the upper band can be achieved when subband cancellation is used. This leads to the conclusion that cancellation over the complete band may be achieved through the combination of subband cancelers.

Goubran has also discussed the use of multiple reference microphones. The results presented show that noise cancellation of up to 15dB may be obtained when three reference microphones are used in a complete band canceler. Again noise enhancement occurred in the upper frequency band.

The results presented above lead to the conclusion that for optimal performance of the adaptive noise canceler, multiple reference microphones and subband adaptive filters should be used.

2.3. Beamforming Microphone Arrays

A beamforming microphone array was proposed by Frost [35] and is shown in Figure 2.11. The principle of the beamformer is that a linear array of microphones can be arranged perpendicular to the path taken by the signal from the desired source. The signal from the source will then reach each of the microphones simultaneously. Each of the microphones in the array is then passed through an adaptive FIR filter. The filter array is constrained such that the total of the tap weights for each of the delays is constant and the total forms the desired filter for the input signal. This provides a constant filter for the desired signal. For a system with N microphones and M taps per filter, the

remaining $M(N-1)$ degrees of freedom in the adaptive array can be used to reduce undesired signals.

The undesired signals are assumed to originate from positions other than perpendicular to the array so that the signal from these microphones will reach the microphones in the array at different times. The filter array can then be adapted to remove the undesired signal since the undesired signal will reach the various microphones in the array at different times allowing taps in the delay lines at different delays to be adapted to remove the undesired signal.

Adaptive beamforming is used in many applications such as antenna arrays and sonar systems in addition to adaptive noise cancellation.

Andresdottir [36] presents experimental results for noise cancellation obtained for a four and five microphone array using Frost's constrained adaptive beamformer in comparison to a nonadaptive beamformer and an adaptive noise canceler. The results show that adaptive beamformer provides better results, with the improvement provided at lower frequencies being greater than that obtained at higher frequencies. The results also show that the adaptive noise canceler (Figure 2.6) provides results that are better than the non-adaptive beamformer and that the constrained adaptive beamformer outperforms both of these systems. These results again indicate that adaptive noise cancelers using more than one reference microphone should perform better than the single reference situation.

A combination of beamforming and adaptive noise cancellation in aids for the hearing impaired has been presented by various authors to help reduce the cocktail party effect. The cocktail party effect is the inability of hearing impaired people to distinguish a desired speaker from background noise and other speakers because of the lack of binaural cues (Plomp [37]). This is especially significant in monaural hearing impaired.

Figure 2.11 Beamforming Microphone Array Presented by Frost [29]

Farassopoulos [38] discusses a two microphone hearing aid system that assumes that the desired speaker is positioned directly in front of the user so that the desired signal received in each of the microphones is the same. The difference of the two microphones is then found with the result used as a noise reference signal. One of the microphones is then used as a primary signal and a Widrow adaptive noise canceler is used to remove the noise from the primary signal.

The results show that up to 28dB in reduction may be obtained in a situation with a main speaker and a single noise source located at 45 degrees to the forward looking direction. The amount of cancellation was reduced to 10dB when two noise sources were present at 45 degrees and 90 degrees.

Strube [33] also shows results for a similar system except that the primary and reference signal are formed by a weighted sum and difference of the short term FFT of the two input microphones. The weights are obtained through training with only the desired signal present and calculating optimum weights which are then fixed forming a two output fixed beamformer. The Widrow adaptive noise canceler (frequency domain implementation) is then used to obtain the desired signal. The results show that with three or four speakers improvements of around 6dB may be obtained depending on speaker location, text, and sex (voice pitch). The pretraining of the beamformer limits the practicality of this proposal.

Sondhi [34] discusses an eight microphone beamforming noise canceler using a softer adaption constraint than that proposed by Frost. Sondhi takes advantage of the fact that speech is relatively insensitive to phase shifting and as a result the constraint used only contains the magnitude information of the spectrum of the desired response. The results show that a null of up to 33dB may be obtained with a single noise source using filters with only eight taps per microphone.

2.4. Other Methods for Noise Reduction

A number of other methods for the reduction of noise and the improvement of the intelligibility of speech. Wathen-Dunn [41] presents results showing that the intelligibility of noisy speech is improved by amplification of the speech to clipping and Pollack [42] presents results on the use of fixed form filters to remove the noise from speech. Thomas [43] presents results for a combination of these two processes. Although each of these results show some improvement in speech intelligibility, the results are minimal and the drawbacks of the systems are significant as the methods significantly distort the speech. The fixed form filter affects the speech in addition to the noise so that the reduction of portions of the spectrum will affect the quality of the speech. While clipping the speech has even more significant affects on the speech quality. The improvements in the intelligibility of the speech provided by these methods is offset by the significant reductions in quality making these system ineffective.

Viswanathan [44-47] has presented a number of studies on the best choice of microphone type for noisy situations. He has compared the performance of a number of microphones including condenser, electret, pressure gradient, second-order gradient, and accelerometer microphones. The results show that a combination of second order gradient microphone at low frequencies and a first order gradient at high frequencies provides the best overall performance. The results presented by Viswanathan can be used to augment the performance of a noise canceler by reducing the noise in the primary signal.

Anderson [48] presents results using neural networks to reduce noise in chaotic signals. The filter uses the principle that signals that appear to be random are truly chaotic signals which can be modeled as deterministic signals providing the model for the

signal includes all the state equations for the process generating the signal. The number of state equations required is known as the dimension of the signal.

The neural network presented uses a single input to the network that contains the desired signal which is of finite dimension with a known bound and a second noise process known to be of a large dimension (infinite). The neural net must be trained using supervised learning. These results show promise for future application to the reduction of noise from speech signals although significant work is required into the applicability of chaotic signal modeling to speech and the performance of the resulting system with various speakers under various noise conditions.

Hoyt [49] details the application of a backpropagation neural network as the filter in the Widrow adaptive noise canceler shown in Figure 2.6 and the results show that the neural net provides better performance than the linear filter. The better results must be offset against the slower learning of the neural net and possible requirement for off-line learning which will make the network incapable of following non-stationarities.

The application of neural networks to noise cancellation will improve the performance of the cancelers as the modelling of non-linearities can be improved. This improvement is currently offset by the slow or off-line learning requirements of neural networks.

2.5. LMS Adaption Algorithm

The adaptive filters proposed in this work are based on the use of Finite Impulse Response (FIR) filters that are adapted through the use of the Least Means Square (LMS) adaption algorithm. The adaptive FIR filter is discussed in detail by Widrow [50] and Haykin [51] and is shown in Figure 2.12.

The principle of the adaptive FIR filter is to pass a signal S through a filter $H(n)$ to obtain an estimate of a second signal d . The difference between the actual signal and the estimate is then used as an error signal that is fed back and used to adapt the filter coefficients to improve the estimate.

The output error from the filter is given by:

$$e_t = \hat{d}_t - d_t \quad \{2.13\}$$

Where d is the estimate of the desired signal given by:

$$d_t = \bar{h}_t^T \bar{s}_t \quad \{2.14\}$$

The filter coefficients are then updated through:

$$\bar{h}_{t+1} = \bar{h}_t + \beta e \bar{s}_t \quad \{2.15\}$$

Where β is an adaption rate given by the following:

$$\beta = a / (b + \bar{s}_t^T \bar{s}_t) \quad \{2.16\}$$

Where:

- 1) d is the desired signal
- 2) $\bar{s}$ is a vector containing the last m samples of S
- 3) $\bar{h}$ is a vector of length m containing the FIR filter tap weights
- 4) $\hat{d}$ is an estimate of d
- 5) e is the error in the estimate of d
- 6) β is the adaption rate constant.
- 7) a, b are limiting constants on β
- 8) m is the length of the FIR filter

Figure 2.12 LMS Finite Impulse Response Adaptive Filter.

The coefficient updating Equation 2.14 sometimes includes a constant of 2 in front of the adaptive update term. This constant has been included as part of “a” in Equation 2.16.

The adaption rate constant β must be constrained to ensure that the filter remains stable. Haykin [51] shows that for stability β must meet the following criteria:

$$\beta < 2 / (\bar{s}_t^T \bar{s}_t) \quad \{2.17\}$$

The constant “a” in Equation 2.16 is restricted by the stability bound given by Equation 2.17 and as a result this constant must be less than two.

The constant “b” in Equation 16 is used to ensure that β is constrained in situations where the input power to the filter is low. This ensures that the filter remains stable.

The LMS algorithm has been applied widely in the literature including echo cancellation (Park [52], Brophy [53]) and line equalization (Eleftheriou [54]) in addition to the adaptive noise cancelers discussed previously.

2.6. Conclusions from Literature Review

This review of previous work on adaptive noise cancellation shows that a number of problems with current cancelers must be addressed. Specifically the noise enhancement in the upper frequency band caused by a complete band canceler must be eliminated.

The results showing that subband cancellation will allow for this effect to be eliminated, indicate a solution to this problem.

The improved performance of noise cancelers using more than one reference microphone indicates that this should also be considered in the design of an adaptive noise canceler.

The completion of this work will describe how the principles of subbanding and multiple references along with the structure of the adaptive filter can be used to solve the noise cancellation problem in an automobile. Chapter three describes the capture of the automobile noise data that was used in the simulations. It also describes the design of the non-adaptive filters that were used to create the subbands used the structures in chapter four and five. It also describes the various methods used to evaluate the performance of the cancelers.

The evaluation of the filter structures is described in chapter four. The first section of the chapter describes the various multiple reference cancelers not including subbands. These filters are evaluated using computer generated data in addition to the automobile data. The principle of subbanding is then introduced through simulations using both cancelers with a single reference and also by looking at the performance of the multiple reference cancelers within individual subbands. The multiple reference cancelers are then combined with subbanding and simulations are performed to evaluate the complete cancelers.

Chapter five describes additional simulations that were performed with speech added to the noise signals and chapter six discusses the results of the work.

3. SIMULATION AND DATA ACQUISITION METHODS

This chapter details the methods and procedures used to capture the automobile noise data and the subsequent processing of the data in preparation for simulation. The sections on data capture and preprocessing methods are detailed more completely in Hebert [3] and Belaire [55].

3.1. Data Acquisition Method

The noise data that was used in the simulations was obtained by Hebert [3] using the recording equipment detailed by Belaire [55]. The data was obtained by recording the noise in a compact car with four microphones under various driving conditions.

The microphones were placed as shown in Figure 3.1, with the primary microphone located on the steering wheel directed at the driver. Reference microphones were placed in each of the following locations; under the hood against the fire wall, in the glove box under the steering wheel, under the floor mat beneath the driver's seat. The microphones were mounted with padding to prevent the pick-up of mechanical vibration.

The signal from each of the microphones was then recorded on an multitrack tape recorder allowing each microphone to have a separate track. The tape recorder and associated preamplifiers were powered using a 12V car battery and a 120V power inverter. This equipment was all located, as shown in Figure 3.1, in the back seat of the car. A separate 12V battery was used instead of the main car battery to prevent electrical noise due to the alternator and other electrical systems present in the engine battery signal from affecting the recordings.

Figure 3.1 Recording Set-up for Automobile Noise Recording by Hebert [3].

The driving conditions that were recorded by Hebert [3] were as follows:

- Car Off, Ventilation Fan On
- Car Idling, Ventilation Fan Off
- Car Idling, Ventilation Fan On
- 60km/hr, Ventilation Fan Off
- 60km/hr, Ventilation Fan On
- 100km/hr, Ventilation Fan Off
- 100km/hr, Ventilation Fan On

The recorded noise signals were then sampled as shown in Figure 3.2 at 10kHz using a 12bit A/D converter on a PC compatible computer. The files were 53248 samples in length and an analog anti-aliasing filter was used to prefilter the data with a 3500Hz cut-off frequency. To ensure that the four microphone signals were sampled synchronously with respect to each other by the A/D converter a synchronization pulse was added to a spare track on the recorder.

The spare track was connected directly to the A/D converter and the recording program called AD (Belaire [55]) monitored the sync pulse line until the pulse was detected. Once the pulse was detected the sampling was started on the main channel. This process was repeated for each track using the same pulse so that the recordings of each track was synchronized to a common reference.

These files were converted to ASCII format files using a tool developed by Belaire [55] called XLAT and copied to a SUN workstation. The data was loaded into the Matlab [56] simulation package and adjusted to have zero mean so that any offset in the recording/processing system was removed. Hebert [122] details analysis that was performed on the data and discussed in the previous chapter.

Figure 3.2 Block Diagram of Analog to Digital Conversion System.

3.2. Simulation Environment

The data obtained by Hebert [3] was copied to an 80386 PC for simulations using the Matlab [56,57] package.

The Matlab package provided both an interactive and programming environment for the manipulation of matrix data. The package includes many built in functions as detailed in the user guides and allows the user to program additional functions. Functions within the Matlab package are similar to subroutines in most high level languages where local variables and parameters are passed into and out of the functions. The package also supports routines called script files that function as main programs.

Appendix A lists a number of function routines that were created to augment those provided with the package. These tools provide functions such as LMS filter building blocks allowing the simulation files to be simpler and easier to debug and understand.

Appendix B lists simulation code files that implement the various noise cancelers that were used in the simulations detailed in Chapters 4 and 5. Some code files have been omitted due to similarity to included files.

The simulation files in appendix B are functions files and not script files as the function statement at the start of the file provides the ability to pass all appropriate variables into the simulation file allowing a batch file to be created that repeatedly called the simulation files for various filter structures and input parameters. Batch processing allowed for simulations to be run unattended maintaining maximum throughput from the computer.

The results of simulations were analyzed using a number of different methods. These included spectral analysis, where the noise spectrum of the input data was

compared with the output data; time based analysis, where the output noise signal was plotted and analyzed through power measurements; and subjective listening tests, where the data was played through a D/A converter.

3.2.1. Spectral Analysis

The Spectral Analysis was performed using the Welch algorithm for calculating the power spectrum of a signal in the presence of noise as discussed by Oppenheim [58]. The algorithm averages the spectrum of the signal calculated over a number of windows to remove effects of uncorrelated noise from the spectrum. The Welch algorithm was used to calculate the spectra on 512 data points. The data points were divided into 128 point sections overlapping by 64 points and the Welch algorithm was then used to calculate the average spectra based on these input vectors.

The spectral analysis was performed on the last 512 points of the simulation as shown in the call to the Plotter function in the files in Appendix B.

The Plotter function is included in Appendix A and it provides a complete routine for the analysis of the power spectra for a simulation in addition to plotting the spectral results. Both the input primary signal and the output signal from the filter are provided to the plotter routine so that it can generate comparison plots between the input and output noise spectra for the simulation.

The spectra calculated by the plotter function and shown in the figures in the following chapters are shown with a decibel scale. The reference for the calculation of the decibel scale is a white noise signal with a noise spectrum of unity magnitude. This common reference allows for the input and output spectra for a simulation to be compared directly and also allows for comparison of the output spectra for simulations using the same data case.

The Plotter routine allowed for simulations to automatically generate plots and related spectral analysis data including maximum, average and minimum noise reduction provided by the filter under simulation.

The results generated by the plot function will be used as objective measures to compare the performance of the various filters. Although the maximum, minimum and mean difference between two spectra have little physical meaning they do provide objective measures for comparison.

These measures were augmented by other measures including total power measurements of the noise power before and after cancelation and subjective listening tests discussed below.

3.2.2. Telephony Bandwidth Filtering

The noise data captured by Hebert [3] was limited to the 0 - 3500Hz frequency band by the anti-aliasing filter on the A/D converter. The data was subsequently filtered by a twelfth order elliptic IIR bandpass filter to simulate the telephony audio band. The filter had corner frequencies at 300Hz and 3500Hz and the frequency response is shown in Figure 3.3.

All simulation results presented in the following chapters are either based on the original sampled data or use telephony bandlimited data obtained using the above mentioned filter. The IIR filter was designed using the Elliptic IIR filter design function provided by the Matlab package [57].

3.2.3. FIR Subband Filtering

Finite impulse response filters were chosen to implement the filters to form subband data in the various structures. The FIR filters were designed using the FIR filter design tools provided by the Matlab package [57].

Figure 3.3 Frequency Response of Twelfth Order Elliptical Band Pass Filter Used to Simulate the Telephony Audio Band.

Figure 3.4 FIR Subband Filter for 0-1200Hz Subband. 129 Tap Filter.

Figure 3.5 FIR Subband Filter for 1200-3500Hz Subband. 129 Tap Filter.

Figure 3.6 FIR Subband Filter for 1200-2400Hz Subband. 129 Tap Filter.

Figure 3.7 FIR Subband Filter for 2400-3500Hz Subband. 129 Tap Filter.

The tools calculated linear phase FIR filters based on corner frequency specifications. Linear phase FIR filters are desirable over IIR filters since the FIR filters have constant delay and as a result the filtered signal can be realigned with non-filtered signals purely through the insertion of a single delay, as all FIR filters can be designed with the same delay.

The filters were used in the simulation in two manners; Initial simulations on the subbands were performed on single bands and as a result it was more efficient to create data files for the various subbands allowing a single simulation routine to be used for all the subbands. Simulations using multiple subbands were implemented with telephony bandwidth input data and the formation of the subbands was included as part of the simulation routine as would be required by any real time implementation of the system.

All FIR filters were designed with 129 taps and a resulting delay of 64 samples. The frequency response for each of the FIR filters is shown in Figures 3.4, 3.5, 3.6 and 3.7. The length of these filters can be reduced in real-time implementations, reducing the effects of the delays on the system performance.

3.3. Simulation Playback

The subjective analysis of simulations results was performed using the procedure shown in Figure 3.8 and the system shown in Figure 3.9. The ability to playback the results of simulations and especially the ability to do comparisons of two files allowed for subjective evaluations of the performance of filters. Specifically, it allowed speech distortion and reverberation effects introduced by the filters to be evaluated.

The procedure shown in Figure 3.8 describes the step by step process required to convert vectors from Matlab for playback and it uses a file conversion tool called TALX developed by the author and the RLS program given by Belaire [55]. The TALX

program converts vectors stored by the Matlab program to 12 bit audio files compatible with the RLS program. TALX includes a modified version of a FORTRAN subroutine provided as part of the Matlab package [56]. The TALX program will calculate the scale factor required so that the 12 bit range is used maximally or it allows the user to provide a scale factor so that multiple vectors can be converted with the same factor for concatenation and direct comparison.

The TALX program is currently limited to reading vectors stored by MS-DOS Matlab and is not compatible with SUN Matlab files. The cause of this incompatibility is that the MS-DOS files are stored using Intel binary format with the most significant byte stored first while SUN files are stored with the least significant byte first. The TALX program is designed to read Intel binary format only. SUN files may be converted to Intel format by loading the Matlab files into MS-DOS Matlab and saving them again on the PC.

The output files from TALX can be concatenated using the MS-DOS [59] COPY command as shown in Figure 3.8 with the "/b" option to specify binary format and the standard MS-DOS copy merge command format. This allows playback files with different vector combinations to be formed easily allowing for A/B comparisons to be easily performed.

The RLS program is used to play the resulting files through a 12 bit D/A converter using the system shown in Figure 3.9. The output of the converter is filtered to prevent aliasing and amplified for listening. The RLS and AQR programs given by Belaire [55] were originally written for a IBM PC XT compatible computer. The A/D-D/A interface was installed in the 386 PC used for simulations and as a result these programs were ported to the new machine by the author.

Figure 3.8 Block Diagram of Processing Steps for Digital to Analog Playback of Audio Files.

Figure 3.9 Block Diagram of Digital to Analog Playback System.

4. COMPARISON OF PARALLEL ADAPTIVE FILTER STRUCTURES

Analysis of previous research on adaptive noise cancelers discussed in Chapter 2 led to the conclusion that the performance of adaptive filters must be improved before they will provide quality improvements sufficient for practical use in the automotive handsfree environment.

The performance problems with adaptive noise cancelers that must be corrected include the noise enhancement problem discussed by Goubran [1] and also the relatively poor noise reduction provided by the cancelers as the best noise reduction results show only 15dB in reduction. This value is the maximum reduction of the spectrum and does not represent an overall power reduction as noise enhancement and lower reduction levels are present at other frequencies in the spectrum.

The results presented by Hebert [3] show the performance enhancements that the use of multiple reference microphones and subband adaption can provide. In this work, these principles are extended and combined to determine an optimal structure using both of these methods to provide improved noise reduction.

Figure 4.1 shows the block diagram of a generalized two stage multiple reference canceler that will be investigated. A number of variations of this structure are possible and they include, for the first stage; adaption for each reference independently or joint adaptive filtering of the references. Examples of possible second stages include; using the first stage output without additional filtering, a decision algorithm to select reference(s) or subband(s) to use or additional adaptive filtering. Both stages can use filters based on subbands of the original signals or the complete band.

A second structure that will be investigated is a cascade structure where each reference microphone is used consecutively. Variations that are possible with this

Figure 4.1 General Two Stage Multireference Adaptive Noise Canceler.

structure are the reference microphone order and the algorithm used to adapt the filters such as parallel adaption or sequential adaption.

4.1. Multiple Reference Filter Structures (Non-Subbanded)

The generalized block diagram discussed above was used to create a number of parallel filters as shown in Figures 4.2, 4.3, 4.4 and 4.5 with each of these corresponding to a multiple reference adaptive noise canceler without subband adaption. Subbanded adaption is not included initially so that the effects of the filter structure can be studied. Structures including subband adaption are discussed starting in Section 4.6.

The filter structure shown in Figure 4.2 is the single error adaptive filter structure with multiple reference microphones given by Widrow [9] as an extension to the single reference canceler. This canceler will be called the Widrow multiple reference canceler subsequently. This canceler has the advantage of it's simple structure, but has the drawback that the adaption of the taps for each reference is affected by the other references, possibly causing reduced performance because of misadaption due to the non-stationarities in the processes.

The mathematical formulation of Figure 4.2 is shown below and it is the multiple reference extension of the LMS filter described in Section 2.5.

The output of the filter is given by:

$$\hat{S} = e_t = P_t - \hat{P}_t \quad \{4.1\}$$

Where the estimate of the noise in P is given by:

$$\hat{P}_t = \bar{h}_t^T \bar{s}_t \quad \{4.2\}$$

Figure 4.2 Widrow Multiple Reference Adaptive Filter Structure [9] (Structure 1).

Where $\mathbf{R}$ is the vector formed from the three input microphones:

$$\bar{\mathbf{R}}_t = \{\bar{\mathbf{R}}_{1t}^T, \bar{\mathbf{R}}_{2t}^T, \bar{\mathbf{R}}_{3t}^T\}^T \quad \{4.3\}$$

Where $\mathbf{R}_1, \mathbf{R}_2, \mathbf{R}_3$ are the vector delay lines for each of the reference inputs. The filter coefficients are updated by:

$$\bar{\mathbf{h}}_{t+1} = \bar{\mathbf{h}}_t + \beta e \bar{\mathbf{R}}_t \quad \{4.4\}$$

and β is calculated with:

$$\beta = a / (b + \bar{\mathbf{R}}_t^T \bar{\mathbf{R}}_t) \quad \{4.5\}$$

The problems with the structure in Figure 4.2 are corrected in the second adaptive noise canceler shown in Figure 4.3. This canceler is a two stage canceler where the first stage is formed from single reference adaptive filters and the second stage is a beamformer that adaptively combines the outputs of the first stage. The first stage allows each reference to be adaptively filtered to exploit the correlation between each reference and the primary signal. The beamforming second stage then combines the correlated references produced by the first stage to cancel the noise in the primary signal. The formulation of the beamforming filter is a combination of the independent single reference LMS cancelers as detailed in Section 2.5 for the first stage filters and a second stage using the Widrow canceler formulation given above.

The second stage was tested in a number of forms depending on the number of taps used on each of the references. The variations that were tested are 1 tap per input, 5 taps per input in a sparse FIR formation where the taps are spaced with 4 zeros separating each tap and 25 taps per input in a complete FIR formation. This structure is a new

Figure 4.3 Two Stage Multireference Beamforming Adaptive Noise Canceler (Structure 4, 5, 6).

adaptive filter structure proposed in this work that will be shown to be the best structure once subband filtering has been included.

The structure shown in Figure 4.4 is formed by a cascade of single reference adaptive filters. This structure has the disadvantage that it is non-symmetrical with respect to the reference inputs, whereas the structures shown in Figures 4.2, 4.3 and 4.5 are symmetrical. This disadvantage is manifested in the requirement for the order of the reference microphones to be either fixed, which may lead to a non-optimal arrangement or adaptively rearranged, which requires undue algorithm complexity.

The problems with the cascade structure will be discussed further when the simulation results are detailed. The mathematical formulation of each of the filters in the cascade structure is the single reference canceler described in Section 2.5.

The structure shown in Figure 4.5 is a form of the structure shown in Figure 4.3 except that the second stage is a decision algorithm that chooses the output of the filter based on the output of the first stage single reference canceler with the minimum power over the previous 100 iterations. Strictly speaking this canceler is a multiple single reference structure with dynamic output selection. The formulation of each of the first stage cancelers is again as shown in Section 2.5.

This canceler has the drawback that the reference microphone switching will possibly introduce switching noise in the primary signal. This canceller will be used to provide a measure of the best single reference noise cancellation performance possible for comparison and also to investigate the relative performance of the individual references under the various noise situations.

Figure 4.4 Multireference Cascade Adaptive Noise Canceler (Structure 2).

Figure 4.5 Minimum Power Output Parallel Single Reference Adaptive Noise Canceler (Structure 3).

4.2. Simulated Data

The initial simulations of the filter structures shown in Figures 4.2, 4.3, 4.4 and 4.5 were run using noise data that was generated with a random number generator. This allowed the data to be formed with known characteristics so that some of the performance characteristics of the filters could be identified and tested under stationary conditions before using actual automobile noise.

The noise data was created by forming seven independent zero mean, unity variance Gaussian noise vectors, each 3000 points long. These vectors were formed with the random number generator in the Matlab package [56]. These vectors were divided into two sets of vectors, with one set of three vectors (x_1 , x_2 , x_3) used as the three reference noise sources and the second set of four vectors (nx_1 , nx_2 , nx_3 , nx_4) used as uncorrelated noise in the reference and primary inputs.

The three reference noise sources were combined to form a primary signal using IIR filters as shown in Equation 4.6. The IIR filters were designed as low pass filters with corner frequencies at 200, 175, and 250Hz respectively for the references x_1 , x_2 , x_3 when the sampling rate is 10kHz. The low pass filters were chosen due to the low pass characteristics of the car noise data that has been reported previously by Hebert [122].

$$x_4 = 0.5 * fx_1 + 0.6 * fx_2 + 0.6245 * fx_3 \quad \{4.6\}$$

where: fx_1 is reference x_1 IIR filtered
 fx_2 is reference x_2 IIR filtered
 fx_3 is reference x_3 IIR filtered

The resulting primary microphone signal is also zero mean and unity variance. The cross correlation values between each of the references and the primary signal are given by the coefficients in Equation 4.6. The coefficients in Equations 4.6 and 4.7

below were chosen to maintain unity variance noise signals and to ensure that the three reference noise signals were combined with unequal weights.

The reference signals were also processed to form correlated references with the following equation:

$$\begin{bmatrix} cx1 \\ cx2 \\ cx3 \end{bmatrix} = \begin{bmatrix} 0.8000 & 0.5196 & 0.3000 \\ 0.3873 & 0.7000 & 0.6000 \\ 0.9000 & 0.4000 & 0.1732 \end{bmatrix} * \begin{bmatrix} x1 \\ x2 \\ x3 \end{bmatrix} \quad \{4.7\}$$

These correlated references were used to simulate the mixing of references noise microphones that the microphones in an automobile will provide, such as engine noise being present in most of the microphones.

The cross correlation data for the references produced by Equation 4.7 are given as follows:

$$R_{cx} = E \begin{bmatrix} cx1cx1 & cx1cx2 & cx1cx3 \\ cx2cx1 & cx2cx2 & cx2cx3 \\ cx3cx1 & cx3cx2 & cx3cx3 \end{bmatrix} = \begin{bmatrix} 1.0000 & 0.8536 & 0.9798 \\ 0.8536 & 1.0000 & 0.7325 \\ 0.9798 & 0.7325 & 1.0000 \end{bmatrix} \quad \{4.8\}$$

The primary signal discussed above with each of the two sets of reference signals ($x1$, $x2$, $x3$ and $cx1$, $cx2$, $cx3$) form two complete sets of noise files for use in the simulations. Four additional sets of noise files were created by adding the noise vectors ($nx1$, $nx2$, $nx3$, $nx4$) to the corresponding reference/primary signal at both 5dB and 30dB SNR to form a total of 6 noise reference files.

Each of the noise files was simulated with each of the four filter structures discussed previously with both 100 tap and 250 tap adaptive filters. The adaptive filter algorithm used is as discussed in Section 2.5 and the 100 tap case used an adaption rate constant " a " = 1.0 for Equation 2.16. The 250 tap case was simulated for both " a " = 0.5

and $\alpha = 1.0$. The simulation files used are given in Appendices A and B where Appendix A contains a number of common functions used by the main simulation files and Appendix B lists the main simulation programs. The summary of the results for the simulations is given in Table C.2 in Appendix C. In all simulations, all filters are adapting continuously unless noted otherwise.

Additional simulations were also run using the simulated data with the results shown in Table C.3. These simulations used additional values of the α parameter in addition to increasing the simulation length to 3000 iterations from the 1500 iterations used in the first simulation set.

4.3. Simulated Data Results

The results show a number of the characteristics of the filter structures. The Widrow multireference canceler (Structure 1) has the best overall performance in the results in Table C.2 which show the maximum and minimum difference between the spectrum of the primary input signal and the filter output. The results show that the canceller provides the best maximum noise cancellation for almost all of the cases and in addition it also shows positive noise cancellation for the complete frequency band as shown in Figure 4.6.

The cascade canceler (Structure 2), shown in Figure 4.5, shows convergence problems as the filter provides little or no noise cancellation and significant noise enhancement in some of the cases as shown by the maximum noise reduction in the results. The convergence problem is also shown in Figure 4.7 which shows the noise reduction provided by the cascade structure is much lower than that shown in Figure 4.6 for the Widrow structure. The poor results for the cascade structure are believed to be caused by convergence difficulties in the stages of the cascade structure, since the convergence noise from each stages affects the convergence of the subsequent stages.

Figure 4.6 Typical Noise Reduction Performance for Widrow Multireference Canceler Shown in Figure 4.2 Using Simulated Random Data. Results are Shown for Pure Noise Data Case with 100 Tap FIR Filters.

Figure 4.7 Typical Results for Multireference Cascade Canceller Shown in Figure 4.4 with the Simulated Random Data. Results are Shown for Pure Noise Data with 250 Tap Filters and $\alpha=0.5$ in β Calculation.

The results for the structure shown in Figure 4.5, which is the parallel single reference canceler with a minimum power output selection (Structure 3) are comparable to the Widrow multireference adaptive noise canceler in the maximum noise cancelation provided, but the minimum reduction results show more noise enhancement in most cases. The additional noise enhancement in the minimum noise reduction data is caused by the use of only a single reference so that noise from all noise sources can not be cancelled.

The beamforming adaptive noise canceler (Structure 4) in Figure 4.4 was simulated with 1 tap for each reference in the second stage and shows noise reduction in all cases. The performance of this canceler is comparable to that provided by the minimum power selection structure and the Widrow multireference adaptive noise canceler and an example of the results are shown in Figure 4.8.

The additional results shown in Table C.2 show that the performance of the filters is improved when the simulation length is increased to 3000 iterations as additional convergence time is allowed. The results also show that the best performance is achieved when the "a" parameter is in the 0.5 to 1.0 range. Increasing the parameter to 2.0 caused a reduction in performance and 2.5 caused the filters to be non-convergent. The non-convergence is expected as Haykin [51] specifies that the convergence bound for β when the LMS filter is as specified in Section 2.5 requires that "a" be less than 2 as shown by Equation 2.17.

The results of these simulations show that the cascade structure has convergence problems and that a choice of the "a" parameter used in the calculation of β should be selected around 1.0. The results also show that the best performance for all of the structures is observed when the noise references show some cross correlation and all the noise signals have low level uncorrelated noise.

Figure 4.8 Typical Noise Reduction Performance for Multireference Beamforming Canceler Shown in Figure 4.3 Using Simulated Random Data. Results are Shown for Pure Noise Data Case with 250 Tap FIR Filters.

This case corresponds to the typical characteristic of the observed data from the automobile as the references will typically be cross correlated and will also include uncorrelated noise. These results are contrary to the design goal used by Mikhael [23,24] in a two reference canceler where the first stage of the canceler proposed was used to decorrelate the references.

4.4. Minimum Power Output Decision Simulations

The simulations with the simulated data discussed above showed that the structure of a multireference noise canceler can significantly affect the noise reduction provided by the filter. Before the performance of the various filters shown in Figure 4.2, 4.3, 4.4 and 4.5 was tested with the automobile noise data that had been collected by Hebert [3], simulations were performed to show that the use of multiple reference cancelers would enhance the operation of single reference cancelers.

Work by Goubran [11] has shown that a multiple reference canceler out performs a single reference canceler. These initial results were then extended by running simulations on the automobile noise data using the structure shown in Figure 4.5. The simulation results are shown in Figure 4.9 where the reference microphone chosen by the decision block is plotted against the iteration in the simulation. The input data for the simulations was complete band data (0-5000Hz bandwidth) automobile noise as described in Chapter 3 and Appendix C.

These plots give a measure for the relative performance of the three single reference cancelers over the duration of the simulations. Where the reference microphone placement is as shown in Figure 4.10 with microphone 4 used as the primary microphone.

Figure 4.9a shows the results for the 60km/hr, ventilation fan on case which shows that microphone 2 provides the minimum output power over the dominant portion of the simulation. This is expected since fan noise will dominate the cockpit noise and microphone 2 is located in the dashboard and as a result provides a good reference to this noise source and it also includes engine noise allowing it to be canceled.

The situation shown in Figure 4.9b is the same as that for 4.9a with the exception that the ventilation fan has now been shut-off. The results show that the dominance of the dashboard reference (microphone 2) has been reduced as expected since the other reference microphones provide references for the wind and road noise which will be more dominant in this situation.

Figure 4.9c shows the results for 100km/hr, ventilation fan off. These results extend the results in Figure 4.9b as the dominance of reference microphone 2 has been almost completely eliminated with the canceler using each of the three references almost equally. This result is expected as the dominance of the wind and road noise for this situation will be even greater, while microphone 2 will only provide a good reference to the engine noise after filtering by the dashboard.

The results of these simulations confirm the previous results given on the use and placement of reference microphones as the results show that the optimal placement of the reference microphone depends heavily on the driving conditions as shown by Figure 4.9 where in one case microphone 2 appears to be the ideal placement while the third case shows that the same location does not provide the best performance.

The results also show that the various reference microphones provide similar noise reduction based on average noise power output since each reference microphone was selected by the decision algorithm and as a result the combination of the references in a multiple reference canceler should provide improved performance.

Figure 4.9 Plots Showing Reference Microphone Selection at Each Iteration by the Minimum Power Decision Noise Canceler Shown in Figure 4.5.
 a) Shows 60km/hr, Ventilation Fan On Case, 100 Taps, $a=1.0$
 b) Shows 60km/hr, Ventilation Fan Off Case, 100 Taps, $a=1.0$
 c) Shows 100km/hr, Ventilation Fan Off Case, 100 Taps, $a=1.0$

The results do show that the relative performance of the various microphones will vary depending on the noise situation and as a result the relative weighting of the various references will vary. For example in the situation shown in Figure 4.9a microphone 2 will be the dominant microphone while in the situation shown in Figure 4.9c each microphone will be weighted approximately equally.

These results tend to indicate that a canceler of the form shown in Figure 4.3 should provide good performance since this allows for the relative weight of each reference microphone to be directly adapted by the second stage of the filter.

4.5. Automobile Noise Simulations (Wideband)

The results given by the minimum power output decision structure detailed above led to the conclusion that multiple reference adaptive noise cancelers will provide improved performance over single reference cancelers. Simulations were run to test the performance of each of the cancelers shown in Figures 4.2, 4.3, 4.4 and 4.5 using automobile noise data.

Table C.5 lists seven noise cases, discussed in Chapter 3, that were used for the simulations were run for the structures. Table C.4 in Appendix C describes the noise input data that was used for all the automobile noise simulations. Each of the simulations was run with delays on the primary microphone of 0, 5 and 10msec. This delay is shown as the adjustable delay in the figures and is included to compensate for any non-causality in the noise signals in the primary microphone with respect to the references due to noise originating closer to the primary microphone than the reference microphone.

The initial set of simulations were 1500 iterations in length and the results of these simulations are shown in Table C.6. These simulations were followed by 4000

Figure 4.10 Positions of Primary and Reference Microphones in an Automobile.

iteration simulations with the results of these given in Table C.7. The 1500 iteration simulations used the structures as described in Section 4.1 while the 4000 iteration simulations used additional formulations that are described below. The second stage of the beamforming canceler used in these simulations used only a single tap for each of the first stage canceler.

The results for the 1500 Iteration simulations show that the beamforming adaptive noise canceler shown in Figure 4.3 outperforms both the Widrow multireference adaptive noise canceler in Figure 4.2 and the minimum output power decision canceler in Figure 4.5 based on maximum noise spectrum reduction. Figures 4.11 and 4.12 show typical noise cancelation results for the first two structures where Figure 4.12 shows the beamforming results and 4.11 shows the results for the Widrow canceler.

The superiority of the beamforming canceler is shown in the significantly better performance of the canceler in the band below 1000Hz over that shown by the Widrow canceler. The improved performance of the beamforming canceler in the lower frequency band is offset by reduced performance in the form of increased noise enhancement in the band above 1000Hz. This noise enhancement has been described by Hebert [3] and he has proposed the use of subband cancelation to overcome the problem. The superiority of the beamforming canceler shows it's better ability to track the non-stationarities in the noise processes and transfer functions which was not observed previously in the stationary simulated data simulations.

As a result, the performance degradation caused by the noise enhancement is not a significant problem with the beamforming structure as the addition of subband cancellation to the structure should address the fault. The possibility for cancellation in the upper band is also shown in Figure 4.11 as the canceler was able to reduce the noise in the peak in the spectrum at 3000Hz.

Figure 4.11 Typical Noise Reduction Performance for Widrow Multireference Canceller Shown in Figure 4.2 Using Automobile Noise. Results are Shown for 60km/hr, Ventilation Fan On Case with 250 Tap FIR Filters.

Figure 4.12 Typical Noise Reduction Performance for Multireference Beamforming Canceler Shown in Figure 4.3 Using Automobile Noise. Results are shown for 60km/hr, Ventilation Fan Off Case with 100 Tap FIR Filters.

The results given in Table C.6 for the cascade canceler show that this canceler provides the best maximum spectrum reduction in a number of the cases. These results are offset by the result shown in Figure 4.13 where the canceler has failed to converge and the noise has been enhanced at all frequencies in the spectrum. This performance from the cascade canceler was observed previously in the simulated data simulations and shows a significant fault with the cascade structure as the failure to converge indicates the requirement for the microphone order in the cascade to be adapted dynamically which introduces undue complication into the filter.

The results for the 4000 iteration simulations given in Table C.7 show similar results to those discussed above for the 1500 iteration simulations. The beamforming canceler again has provided better performance than the Widrow and minimum power decision canceler.

The 4000 iteration simulations also included some additional simulations of the cascade structure. These simulations include the following variations on the original cascade structure. The original canceler is called Structure 2 in Table C.7.

- 1) The order for the reference microphones was reversed so that microphone 3 was used first followed by 2 and then 1, whereas, the original canceler used the 1, 2, 3 order. This structure is called 2A in Table C.7.
- 2) The references were used in the same order as for Structure 2 except that sequential adaption of the stages was used where each stage was allowed to adapt for 500 iterations and frozen for 1000 iterations while the other were adapting. Called Structure 2C in Table C.7.
- 3) Uses the same reference order as Structure 2A and the adaption algorithm described for Structure 2B. Called Structure 2C in Table C.7.

Figure 4.13 Typical Noise Reduction Performance for Cascade Canceler Shown in Figure 4.4 using Automobile Noise. Results are Shown for 100km/hr, Ventilation Fan On Case with 100 Tap FIR Filters.

The results for the various forms of the cascade canceler given, show that although this canceler does provide better maximum noise cancelation in some of the cases over that provided by the three non-cascade structures, the cascade canceler does show convergence problems in all the formations.

The convergence problems are shown by results for the two cascade structures using continuous adaption (2,2A) for the simulations using the MI10 data file. This case corresponds to 100km/hr, ventilation fan on case and the results show that the cascade structure has failed to provide any significant cancelation at any frequency in the band.

The effect of the microphone order on the performance of the canceler is shown by the MI5 case with a 10msec delay in the table, which is the 60km/hr, ventilation fan off case. The results show that over 6dB poorer performance is obtained by the Structure 2A version of the canceler over that provided by the Structure 2 version. The only difference between the two simulations is the order of the reference microphones.

The failure of the cascade structure to converge in all cases is believed to be caused by the adaption of one stage affecting the subsequent stages. Based on this assumption the Structure 2B and 2C versions of the cascade filter were tested and these show little or no cancellation. The reason for this is believed to be caused by the non-stationarities in the automobile noise data causing the cancelers to be unable to track the noise processes when frozen.

The adaption problems with the cascade structure given above lead to the requirements for an optimal implementation of the cascade to have sequential adaption at an interval based on the non-stationarities of the noise in addition to an algorithm to dynamically order the microphones in an optimal fashion. These requirements make the implementation of this filter impractical and as a result this structure will not be considered further.

Although the minimum output power decision canceler provides good performance when compared with the Widrow multireference adaptive noise canceler, the switching in the output from this canceler will reduce the performance and this canceler uses only one reference at a time and hence the multiple reference beamforming canceler outperforms it. As a result this canceler will not be simulated further.

The characteristics of the beamforming canceler were further tested with the simulations of two additional forms of the canceler. The original form of the canceler used in the 1500 iteration simulations (called Structure 4) used β as given by Equation 2.16. Since the second stage of the beamforming canceler based the calculation of β on only the current output of each of the first stage filters (one sample of 3 signals), the effects of the variation of the β given by Equation 4.5 were tested through the creation of two additional methods for calculating β .

The first method uses an FIR filter on β where Equation 4.5 is used in each iteration to calculate a β for that iteration. The actual β used in Equation 4.4 was calculated as the average of the previous 10 β values given by Equation 4.5. This structure is called 4A in Table C.7.

A second method that was used to calculate β based on IIR filtering of β . The formulation of the filter is given by Equation 4.9 below:

$$\beta_a = 0.1\beta_a + 0.9\beta_t \quad \{4.9\}$$

Where B_t is given by Equation 4.10 and B_a is used in Equation 4.4 in the adaption of the coefficients.

Both of these new methods for the calculation of β represent a low pass filter on the β values calculated in the original beamforming structure.

Figure 4.14 Noise Reduction Performance for Multireference Beamforming Canceler Shown in Figure 4.3 Using Automobile Noise When β Calculated with Equation 4.10. Results are Shown for 60km/hr, Ventilation Fan On Case with 100 Tap FIR Filters.

Figure 4.15 Noise Reduction Performance for Multireference Beamforming Canceler Shown in Figure 4.3 using Automobile Noise When β calculated with Equation 4.10 and Averaged Over Previous 10 Iterations. Results are Shown for 60km/hr, Ventilation Fan On Case With 100 Tap FIR Filters.

Figure 4.16 Noise Reduction Performance for Multireference Beamforming Canceler Shown in Figure 4.3 Using Automobile Noise When β Calculated With Equations 4.10 and 4.4. Results are Shown for 60km/hr, Ventilation Fan On Case With 100 Tap FIR Filters.

2

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The results in Table C.7 show that the structure using the FIR calculation of β provides the best maximum noise cancelation results with the IIR calculation providing results slightly better than the original formulation. These results are offset by reduced performance in the upper frequency band as shown in Figures 4.14, 4.15 and 4.16 which show the results for the 60km/hr, ventilation fan off case for each of the β calculation methods.

Further testing of the FIR calculation method showed that when the averaging was extended over 100 samples the filter failed to converge and caused noise enhancement over the complete band. The cause of the increased noise enhancement is believed to be the reduction of the effects of the high frequency noise energy on the value of β . The noise power due to frequencies in the lower portion of the frequency band in consecutive samples is more constant than the noise power due to the higher frequencies. The averaging will enhance the lower frequency energy while reducing the effects of the higher frequency energy resulting in a larger value of β than desired and causing convergence problems in the filters at higher frequencies.

4.6. Telephony Bandlimited

Since the noise canceler under consideration in this work was being designed for application as a handsfree cellular telephone, simulations were performed with the input data limited to the telephony frequency band of 300 to 3500Hz. Table C.8 shows the results of the simulations for the structures shown in Figure 4.2 and 4.3. and all subsequent simulations used data that was limited to the telephony band.

The filter used to create the telephony bandlimited data is discussed in Section 3.2 and the filter is an IIR bandpass filter. The structures that were simulated include four filters that were used in the previous simulations, the Widrow multireference adaptive noise canceler shown in Figure 4.2 and the three beamforming adaptive noise cancelers

using a single tap for each of the inputs to the second stage with the various methods for the calculation of β .

In addition two new forms of the beamforming canceler were included. The first new form of the canceler uses the structure shown in Figure 4.4 except that each of the inputs to the second stage has five adaptive taps where the taps form a sparse FIR filter. The five taps for each reference have 4 zero weight taps between them to form the sparse FIR filter. This filter is called structure 5 in the tables in Appendix C. The second new form of the beamforming canceler uses 25 taps on each input to the second stage. These taps form a standard FIR filter. This structure is called structure 6 in the tables in Appendix C.

The results for the simulations for the six different multiple reference cancelers are given in Table C.8. All of the beamforming cancelers provide comparable performance when the maximum noise reduction values are compared and they outperform the results shown for the Widrow canceler.

The results for the FIR and IIR filtered β calculation structures show similar results to those shown in the previous Section where the two filters provide added noise reduction in the lower frequency band at the expense of increased noise enhancement in the upper frequency band.

Since the performance enhancement in the lower band is not significant when compared to the performance reduction in the upper band for the methods of calculating β using filters, Equation 4.5 will be the only method used to calculate β subsequently.

The significant differences in the performance between the Widrow canceler and the beamforming cancelers using the three variations on the number of taps in the second stage are shown by the data in Table C.8 and also by Figures 4.17, 4.18, 4.19 and 4.20

Figure 4.17 Noise Reduction Performance for Widrow Multireference Canceled Shown in Figure 4.2 Using Telephony Bandlimited Automobile Noise. Results are Shown for Engine Idling, Ventilation Fan Off Case with 100 Tap FIR Filters.

Figure 4.18 Noise Reduction Performance for Multireference Beamforming Canceler Shown in Figure 4.3 Using Telephony Bandlimited Automobile Noise with 1 Tap for Each Reference in the Second Stage. Results are Shown for Engine Idling, Ventilation Fan Off Case with 100 Tap FIR Filters.

Figure 4.19 Noise Reduction Performance for Multireference Beamforming Canceler Shown in Figure 4.3 Using Telephony Bandlimited Automobile Noise with 5 Taps (Sparse FIR) for Each Reference in the Second Stage. Results are Shown for Engine Idling, Ventilation Fan Off Case with 100 Tap FIR Filters.

Figure 4.20 Noise Reduction Performance for Multireference Beamforming Canceler Shown in Figure 4.3 Using Telephony Bandlimited Automobile Noise with 25 Taps for Each Reference in the Second Stage. Results are Shown for Engine Idling, Ventilation Fan Off Case with 100 Tap FIR Filters.

where each of these plots shows the results for the car idling, ventilation fan off case from Table C.8.

The figures and table show that the maximum cancelation provided by the beamforming cancelers is significantly larger than that provided by the Widrow canceler and this is clearly shown by the differences between the input and output spectra in the lower frequency band of figures.

The main performance difference between the beamforming cancelers with additional taps in the second stage filter is that the additional taps have provided the filters with additional degrees of freedom that have allowed for the average and minimum noise reduction performance of the canceler over the band to be improved. This is shown by the ability for the canceler in Figure 4.20 to provide cancelation at all frequencies between 300 and 2700Hz while the other figures all show some noise enhancement below 2700Hz.

The data in Table C.8 also shows this improvement in performance where the filter with the sparse FIR filter outperforms the single tap filter when average noise cancelation data is compared and that the structure with the complete FIR filter outperforms both of the other beamforming structures.

4.7. Cancellation in Sub Frequency Bands

The noise canceler structures presented in the previous sections are all based on cancellation where the complete band of each noise reference is passed through an adaptive filter to obtain noise cancelation. The results have shown that due to the dominance of the low frequency band in the power spectral densities, the filters have converged to a state where the low frequency noise is cancelled while the higher frequency noise is enhanced.

If the filters were able to converge to the optimal solution, in the worst case the output of the canceler would have cancellation in the lower frequencies and pass the higher frequencies unchanged. Ideally the filter would provide cancellation at all frequencies in the band. The results presented so far have shown that neither of these two conditions are true in the adaptive noise cancelers analyzed.

The inability for the cancelers to converge such that they do not cause noise enhancement in the upper band is believed to be the effects of the filter tracking the non-stationary automobile noise. The adaptive filter must be designed such that it can adapt fast enough to track the non-stationarities and as a result short adaptive filters must be used to allow the canceler to be able to adapt fast enough while remaining stable.

This requirement for short filters is in direct conflict with the requirement to have longer filters so that the canceler has sufficient degrees of freedom to attain the ideal taps weights so that no noise enhancement is caused by the filter.

The trade off between these two requirements results in the filter not being able to converge in the ideal optimal sense where at worst the noise is passed without enhancement. Instead the filter must converge to a quasi-optimal solution where the filter is cancelling the dominant spectral components of the noise at the expense of noise enhancement. The result is a whitening of the spectrum of the noise as can be seen in the relatively flat characteristic of the output noise spectrum shown in Figure 4.20 compared with the input.

The inability of the cancelers to provide noise cancellation in the upper frequency bands can be caused by either the non-optimal convergence effect discussed above or it may be caused by poor correlation between the primary signal and the references.

Goulding [5] discusses the noise enhancement in the upper band and attributes the cause to a lack of correlation between the reference and the primary signals. The plots shown in Figure 4.20 and 4.11 show noise cancellation in the upper frequency bands. Specifically 4.11 shows that the filter was able to cancel the peak in the spectrum near 3000Hz and Figure 4.20 shows noise cancellation over all frequencies from 300 to 2700Hz which is most of the telephony audio band.

Based on these observations which are also shown by the results of the other simulations discussed previously, it is believed that the noise enhancement is caused by the non-optimal convergence effect discussed above and not by a lack of correlation between the noise processes at the higher frequencies.

Hebert [3] has shown that the noise processes do have some correlation in the upper frequency bands and he has also shown that noise cancellation may be attained in the upper band when a single reference canceler is set up so that the primary and reference signals are both high pass filtered before they are input into the adaptive noise canceler. The output of the adaptive noise canceler is then subtracted from a delayed version of the primary signal to account for filters delays in a similar fashion to Figure 2.9. The resulting output noise signal shows noise cancellation in the upper band while not affecting the lower frequencies.

Based on these results and conclusions, a series of simulations were performed to ensure that noise cancellation was possible in the various bands of the noise spectrum. The simulations used input audio data that had been previously FIR filtered to obtain the desired subband noise signals for each microphone. The subbanded noise was then used in simulations with the Widrow multireference adaptive noise canceler and the beamforming adaptive noise canceler with the three different tap configurations for the second stage as used in the telephony bandwidth simulations.

The FIR filters used to form the subbands are discussed in Section 3.2. FIR filters were chosen as they allow for the low pass and band pass filters for various bands to all be designed with the same filtering delay. The matching of the delay will be beneficial when the filters are used in parallel so that delays won't be required to align the outputs of the various filters.

The frequency bands that were chosen for analysis were 300-1200Hz, 1200-2400Hz, and 2400-3500Hz with 1200-3500Hz also included. The first three bands divide the 300-3500Hz telephony audio band into three bands that are relatively equal in size and allows the dominant noise energy in the band below 1000Hz to be contained within a single band. The 1200-3500Hz band allows for a two band system to be formed covering the complete telephony band with a narrow lower band containing the dominant noise energy and a larger upper band with the remaining noise energy.

The results of the simulations that were run with the subbanded data are shown in Tables C.9, C.10, C.11 and C.12 where each of these tables gives the results for one subband. Typical results plots for each of the filter structures for each of the subbands are shown in Figure 4.21 through 4.28.

Figures 4.21 and 4.22 show the results for the Widrow multireference adaptive noise canceler and the beamforming adaptive noise canceler with 25 taps per reference in the second stage respectively for the 300-1200Hz band. These are the same structures as discussed in the previous section and the results presented in the figures are for the car off, ventilation fan on case.

Figure 4.21 and Table C.9 show that the use of subbanded data has not caused the Widrow multireference adaptive noise canceler to obtain any significant improvement in the reduction of the low frequency noise. Specifically, the maximum noise cancelation results for the telephony data in Table C.8 and the subbanded results are very similar

Figure 4.21 Noise Reduction Performance for Widrow Multireference Canceled Shown in Figure 4.2 Using Subband Automobile Noise (300-1200Hz) Results are Shown for Engine Off, Ventilation Fan On case with 100 Tap FIR Filters.

Figure 4.22 Noise Reduction Performance for Multireference Beamforming Canceler Shown in Figure 4.3 Using Subband Automobile Noise (300-1200Hz) with 25 Taps on Each Input to the Second Stage. Results are Shown for Engine Off, Ventilation Fan On case with 100 Tap FIR Filters.

between the two tables with some cases showing better performance while others show poorer results when subbanded filtering is used.

The results shown in Figure 4.22 and Table C.9 for the various beamforming adaptive noise cancelers when compared with the results in Table C.8 for the telephony band all show that the use of subbands has allowed the filter to provide better or very similar maximum cancellation when the noise is subbanded.

The results again show that the filter with 25 taps in the second stage provides the best performance of the three with the filter with the sparse FIR showing better performance than the filter with the single tap in the second stage.

Figure 4.23 and 4.24 show the results for the 1200-3500Hz subband for two filters for the 60km/hr fan off case and the complete summary of the results for the subband are given in Table C.10.

The results presented in Figure 4.23 shows that the Widrow multireference adaptive noise canceler provides significant noise reduction of the peak in the noise spectrum at 3000Hz while it provides little improvement and some enhancement of the noise at frequencies elsewhere in the subband.

The results for the beamforming adaptive noise canceler in Figure 4.24 and Table C.10 show that this canceler was able to provide a reduction of the noise power at all frequencies in the subband with the filters with additional taps in the second stage providing better performance as before.

The Widrow canceler does show better performance than the beamforming cancelers at the 3000Hz peak, but this is offset by the poorer performance at other frequencies in the band. The beamforming cancelers results show that the filter has whitened the noise by flattening the spectrum, this characteristic can be seen by

Figure 4.23 Noise Reduction Performance for Widrow Multireference Canceled Shown in Figure 4.2 Using Subband Automobile Noise (1200-3500Hz) Results are Shown for 60km/hr. Ventilation Fan Off Case with 100 Tap FIR Filters.

Figure 4.24 Noise Reduction Performance for Multireference Beamforming Canceler Shown in Figure 4.3 Using Subband Automobile Noise (1200-3500Hz) with 25 Taps on Each Input to the Second Stage. Results are Shown for 60km/hr Ventilation Fan Off Case with 100 Tap FIR Filters.

comparing the output spectrum for the various beamforming canceler that show a flatter spectrum.

The relatively large cancelation that the Widrow canceler is able to provide around 3000Hz is only present in the two data cases that show a significant peak as shown by the large maximum noise cancellation data shown in Table C.10 for the two 60km/hr cases while the other cases show results that are much lower and also worse than the results for the beamforming cancelers for the same cases. The ability of the Widrow canceler to provide the better cancellation of the peak shows that the two stage beamforming canceler is not able to take advantage of all the correlation between the references and the primary at this frequency, but this is offset by the significantly better performance elsewhere in the band.

Figure 4.25 and 4.26 show the results for the 1200-2400Hz band for the two filters and the summary of the results for this band are given in Table C.11. The results shown in the figures are for the 60km/hr, fan off case which is the same case shown in Figure 4.23 and 4.24.

The results for the 1200-2400Hz band show that the beamforming cancelers provide better performance than the Widrow canceler for all of the data cases. This is also shown in Figure 4.25 where the spectrum of the output noise in the subband shows enhancement while the beamforming results in Figure 4.26 shows cancellation over the complete subband. The beamforming canceler provides a flatter spectrum than that provided by the Widrow canceler.

The results in Figure 4.23 when compared to the results in Figure 4.25 show that the Widrow canceler was unable to provide an improvement in the 1200-2400Hz range when using the 1200-3500Hz data while it did when only the 1200-2400Hz band was used. This shows how the Widrow multireference adaptive noise canceler has been

Figure 4.25 Noise Reduction Performance for Widrow Multireference Canceler Shown in Figure 4.2 Using Subband Automobile Noise (1200-2400Hz) Results are Shown for 60km/hr, Ventilation Fan Off Case with 100 Tap FIR Filters.

Figure 4.26 Noise Reduction Performance for Multireference Beamforming Canceler Shown in Figure 4.3 Using Subband Automobile Noise (1200-2400Hz) with 25 Taps on Each Input to the Second Stage. Results are Shown for 60km/hr, Ventilation Fan Off Case with 100 Tap FIR Filters.

unable to provide noise cancelation in the 1200-2400Hz band when using the larger band even though correlation between the references and the primary is present. The beamforming cancelers were able to provide cancelation.

The summary of the results for the 2400-3500Hz subband are shown in Table C.12 and example plots for the 100km/hr, ventilation fan on case are shown in Figure 4.27 and 4.28.

These results again show the superiority of the beamforming canceler over that of the Widrow canceler with the beamforming canceler with 25 taps per reference in the second stage providing the best results and the Widrow canceler showing the poorest results.

The only exception to this conclusion is the significant cancelation that the Widrow canceler was able to provide for data cases at 60km/hr. These are the data cases that show the peak in the spectrum at 3000Hz as shown in Figure 4.23. The better performance of the Widrow canceler was discussed previously and it shows that the beamforming canceler has not been able to take advantage of some of the correlation between the references and the primary.

This better performance of the Widrow canceler for the particular subband of two data cases is offset by the superior performance of the beamforming canceler in the same band in other data cases and also in the other subbands for all the data cases.

The results presented show that the noise in the upper band of the spectrum of the references is correlated with the noise in the primary as noise cancellation is possible when subbands of the data are used. These results show that the cause of the noise enhancement in the upper frequency band is due to inability of the canceler to converge

Figure 4.27 Noise Reduction Performance for Widrow Multireference Canceled Shown in Figure 4.2 Using Subband Automobile Noise (2400-3500Hz) Results are Shown for 100km/hr, Ventilation Fan On Case with 100 Tap FIR Filters.

Figure 4.28 Noise Reduction Performance for Multireference Beamforming Canceler Shown in Figure 4.3 Using Subband Automobile Noise (2400-3500Hz) with 25 taps on Each Input to the Second Stage. Results are Shown for 100km/hr Ventilation Fan on Case with 100 Tap FIR Filters.

to the optimal solution due to non-stationarities in the noise and not due to a lack of correlation in the noise data.

This leads to the conclusion that an adaptive noise canceler using a number of short adaptive filters on each reference signal with each of the filters acting on a separate subband should provide noise cancellation over the complete frequency band with little noise enhancement. The use of subbands is superior to the use of a longer filter since the subbanded filter can be shorter in length and as a result allows the filter to be able to track non-stationarities in the noise better.

4.8. Subband Single Reference Adaptive Noise Cancelers

The results of the previous section indicates that an adaptive noise canceler with the references broken down into subbands should provide better performance by providing cancellation in the upper bands of the spectrum. In order to test this assumption a series of simulations were run with a single reference subbanded noise canceler.

Figure 4.29 shows a block diagram of this canceler where each of the adaptive filters is governed by the LMS algorithm described in Section 2.5 and the fixed filter blocks are implemented as FIR filters using 300-1200Hz, 1200-2400Hz, and 2400-3500Hz bands.

The subbanded single reference filter operates by filtering both the reference and the primary signals to create the three subbands, which are input into three independent adaptive noise cancelers. The output of the filter in each of the cancelers, which is the estimate of the noise for the subband is filtered to ensure that it contains only the subband of interest and is subtracted from a delayed version of the primary signal. The delay is required to align the primary signal with the output of the noise cancelers because of the

Figure 4.29 Single Reference Subbanded Adaptive Noise Canceler Based on Canceller in Figure 2.6. (Structure 1T)

Figure 4.30 Noise Reduction Performance for Single Reference Subbanded Canceler Shown in Figure 4.29 with Engine Compartment Reference Microphone. Results are Shown for 60km/hr, Ventilation Fan Off Case with 100 Tap FIR Filters.

Figure 4.31 Noise Reduction Performance for Single Reference Subbanded Canceler Shown in Figure 4.29 with Dashboard Glove Compartment Reference Microphone. Results are Shown for 60km/hr, Ventilation Fan Off Case with 100 Tap FIR Filters.

Figure 4.32 Noise Reduction Performance for Single Reference Subbanded Canceler Shown in Figure 4.29 with Cabin Floor Reference Microphone. Results are Shown for 60km/hr, Ventilation Fan Off Case with 100 Tap FIR Filters.

delays in the FIR filters. The figure also includes an adjustable delay on the primary signal because of possible non-causality in the noise signals that has been discussed previously.

The results of the simulations that were run using this filter are given in Table C.13 and Figures 4.30 - 4.32 show plots of some of the results. The simulations were run using each of the three reference noise sources as the single reference for both of the data cases at 60km/hr. Not all data cases were simulated as these simulations were intended to show that subbanded cancelers can provide cancellation over the complete telephony band before subbanding is added to the multiple reference cancelers.

The results in the figures show that the subbanded single reference canceler has better performance in the upper band than the non-subbanded cancelers. This can be observed by comparing Figure 4.30 - 4.32 with results presented previously, such as the results Figure 4.12 which is a multireference canceler, single reference results shown in Figure 2.9 and also results presented by Hebert [3]. The summary data in Table C.13 along with the figures show that the performance of the canceler varies between the references and over the spectrum.

Specifically Figure 4.30 shows that the reference in the engine compartment is able to provide significant cancellation of the peak in the spectrum at 3000Hz while the other references provide less cancellation. This is expected since the peak is believed to be caused by air flow noise and the microphone in the engine compartment has been shown to contain the same peak in the spectrum (Hebert [3]).

The results for the reference microphone on the floor of the car shown in Figure 4.32 show that this microphone provides poorer performance than the other microphones in the band up to around 1000Hz, but provides better performance than both

of the other references between 1000 and 1200Hz. Similar comparisons can be made for the references at other points in the band.

The maximum noise cancellation results for each of the cancelers shown in Table C.13 if the peak at 3000Hz for the "x1" reference is ignored is around 12dB and this maximum is in the lower band below 1000Hz which is comparable to the results presented by Hebert for a single reference canceler without subbands.

The result of these simulations is the conclusion that the use of subbanded adaptive noise cancelers enhances the performance in the upper band of the spectrum without any loss in performance in the lower band. The results also show that the various references provide better cancellation in different regions of the spectrum indicating again that multiple reference cancelers will further improve the performance of the subbanded canceler.

4.9. Multiple Reference Subbanded Adaptive Noise Cancelers

The results presented in the previous sections showing the performance enhancements that can be obtained using multiple reference adaptive noise cancelers and subbanded cancellation led to the decision to combine both of these methods into a canceler.

The multiple reference results have shown that the beamforming canceler provides better performance than the Widrow canceler and the other structures that were proposed. The results also show that the beamforming canceler with 25 taps on each of the inputs to the second stage provides the best performance of all the beamforming results. Based on these results, the beamforming canceler with 25 taps on each input to the second stage was chosen for simulation with subbanded filters. The Widrow canceler was also chosen for comparison.

The choice of subbands was based on two criteria; which are the need to obtain the best performance from the canceler and the requirement to maintain the total number of bands to a reasonable limit so that the filters do not become too complex. Based on these two criteria, both two and three band cancelers were chosen for simulation with the two band cancelers using 300-1200Hz and 1200-3500Hz bands and the three band canceler using 300-1200Hz, 1200-2400Hz, and 2400-3500Hz bands. These are the same bands that were used in the simulations described in Sections 4.6 and 4.7.

The block diagram of the Widrow multireference subbanded adaptive noise cancelers using three subbands is shown in Figure 4.33. The canceler is an extension of the single reference subbanded canceler shown in Figure 4.29 where each of the single reference cancelers in Figure 4.29 have been replaced by a Widrow noise canceler as shown in Figure 4.2.

The addition of subbanding to the beamforming adaptive noise canceler led to two different structures depending on when the subbanded signals were recombined. The three band version of the first structure is shown in Figure 4.34. The canceler uses an initial stage of single reference subbanded noise cancelers for each of the references where the first stage cancelers are similar to the canceler shown in Figure 4.29. The second stage of the canceler, uses the recombined correlated reference noise signals created by the first stage as inputs into a Widrow canceler as shown in Figure 4.2.

The second method for the inclusion of subbands in the beamforming adaptive noise canceler is shown in Figures 4.35, where the figure shows a three band canceler. The canceler maintains the subbands of the references in the second stage and uses separate second stage beamformers for each subband. The canceler uses an initial stage of single reference cancelers for each band of each reference as used in the structure shown in Figure 4.34. The outputs of the first stage filters for each band are then input

Figure 4.33 Multiple Reference Subbanded Widrow Noise Canceller Based on Canceller Shown in Figure 4.2 with Three Subbands. (Structure 13)

Figure 4.34 Multiple Reference Beamforming Noise Canceller with the Second Stage Using Completer Cand Correlated References as Generated by the First Stage. Based on the Canceller Shown in Figure 4.3. (Structure 63)

Figure 4.35 Multiple Reference Beamforming Noise Canceller with Subbanded Second Stage Based on the Canceller Shown in Figure 4.3. (Structure 73)

into the subbanded second stage beamforming canceler. The outputs of the second stage of the canceler are then filtered and subtracted from a delayed version of the primary signal to complete the canceler.

In all of the subband filters presented above, the output of the subbanded cancelers is filtered by the appropriate subband filter to ensure that each of the subband canceler outputs contains only noise in the band of interest. This will remove any noise outside the appropriate band that the cancelers may introduce.

In addition to the three band cancelers shown in the figures, two band cancelers were also created and they are identical to the three band versions except that 300-1200Hz and 1200-3500Hz bands were used and the structures modified appropriately.

4.10. Multiple Reference Subbanded Adaptive Noise Cancelers Results

The results of the simulations that were run using the multireference subbanded adaptive noise cancelers are detailed in Tables C.14 and C.15 with the first table giving the results summary for the filters using two subbands and the second giving results for filters using three subbands. All three of the filter structures shown in the previous section were used in the simulations with both two and three subbands.

Figures 4.36, 4.37 and 4.38 show example results for each of the filters using three subbands. The results shown in the figures are for the 100km/hr, ventilation fan on case. The results in Figure 4.36 are for the Widrow canceler shown in Figure 4.33 and the plot shows that the performance of the canceler in the upper band has improved over that observed in the telephony bandwidth simulations in Table C.8. The improvement is shown in the results summary Tables C.14 and C.15 by the increases in the minimum and average noise reduction while maintaining the same maximum noise reduction. This

Figure 4.36 Noise Reduction Performance for Multireference Subbanded Widrow Canceler Shown in Figure 4.33. Results are Shown for 100km/hr, Ventilation Fan On Case with 100 Tap FIR Filters.

Figure 4.37 Noise Reduction Performance for Multireference Beamforming Subbanded Canceled Shown in Figure 4.34. Results are Shown for 100km/hr, Ventilation Fan On Case with 100 Tap FIR Filters.

Figure 4.38 Noise Reduction Performance for Multireference Beamforming Subbanded Canceler with Subbanded Second Stage Shown in Figure 4.35. Results are Shown for 100km/hr, Ventilation Fan On Case with 100 Tap FIR Filters.

indicates that the filters are performing better in the upper band by providing more cancellation and reduced enhancement of the noise.

The results shown in Figure 4.37 show how that the beamforming structure without a subbanded second stage in 4.34 has enhanced the performance of the Widrow canceler over the results in Figure 4.36. Specifically the improved performance over the complete band canceler is apparent as the canceler performs significantly better in the lower frequencies while still providing cancellation in the upper band. This improvement can also be observed in the results in Table C.15 that show the better performance of this filter.

The results for the canceler shown in Figure 4.35 with the subbanded second stage shows further improvement in performance with the canceler providing similar performance to the canceler shown in Figure 4.34 in the lower band but having better performance in the upper frequency band. This is also shown by comparing Figures 4.37 and 4.38.

The results shown in Table C.14 and the example in Figure 4.39 show the results for the two subband multireference filters. These results show that the performance of the various filters is slightly poorer when only two bands are used. The cancelers provide very comparable performance in the lower frequency band to the three band structures with the Widrow canceler and the subbanded second stage beamforming canceler showing identical results in the 300-1200 band for the two and three band cancelers. This is expected since the lower band portion of these two filters is identical in the two and three reference cancelers.

The main performance difference in the filters is in the upper frequency band where the use of the single larger upper band has caused the filter to have poorer

Figure 4.39 Noise Reduction Performance for Multireference Beamforming Subbanded Canceler with Two Subbands Based on the Canceler Shown in Figure 4.35. Results are Shown for 100km/hr, Ventilation Fan On Case with 100 Tap FIR Filters.

performance as shown in the summary table by the lower average and minimum noise reduction results for the two band filters versus the three band filters.

The results also show that the performance of the canceler in the various bands with the multireference subbanded cancelers is very similar to the results observed for the individual subbands and discussed in Section 4.6.

The results presented show that the multireference canceler using a subbanded second stage, shown in Figure 4.35 provides the best performance with the filter providing maximum noise reduction of up to 28dB for the 60km/hr ventilation fan off case whereas previous results for this same case have shown a maximum reduction of 15dB (Goubran [1]). The canceler has also provided a minimum cancellation over the 300-3500Hz telephony band of 3.1dB whereas previous results have shown either enhancement or no improvement in the upper band. The canceler provided an average 10.5dB reduction in the noise power over the telephony band. The results show the superiority of this canceler over the other cancelers presented here and by previous authors.

4.11. Comparison of Tracking Performance

The tracking performance of the multireference cancelers was compared by analyzing the output signal from the filter which is the overall error signal for the performance of the noise canceler. The error signals for four of the simulations discussed in pervious sections are shown in Figure 4.40 and they were calculated using equation 4.10 (Hajivandi [61]), which calculates a running average of the absolute value of the error signal from the canceler.

$$E_t = \frac{1}{64} \sum_{i=0}^{63} |\hat{S}_{t-i}| \quad (4.10)$$

The results shown in Figure 4.40 are for four simulations using the identical segment of noise data for the 100km/hr, ventilation fan on data case that are included in Tables C.8 and C.15 and correspond to the Widrow multireference canceler without subbands (Figure 4.2), with subbands (Figure 4.33), the two stage beamforming canceler without subbands (Figure 4.3), and with subbands (Figure 4.35).

The results show that the average output error from the two versions of the two stage beamforming canceler is significantly lower than that from both versions of the Widrow canceler. The addition of subbanding to the Widrow filter had little effect on the error performance compared with the canceler without subbanding while the addition of subbanding to the two stage beamforming canceler significantly improved the performance of the filter.

The better tracking performance of the two stage beamforming cancelers is shown by the ability of the canceler to adapt quickly to changes in the noise processes. This is shown in the figure by the spike in the error signal that is present in both of the Widrow filter error signals at the 2000 Iteration point, the change in the noise processes is also present in the two stage beamforming error signals but the figure shows how the filters were able to adapt faster to the change and as a result the spike is greatly reduced in amplitude. Similarly other peaks in the error signal for the Widrow cancelers are reduced in size and duration by the two stage beamforming cancelers.

These results show that the two stage beamforming canceler with subbands provides significantly better tracking performance of the non-stationarities in the noise processes and transfer functions without the requirement for a large adaption rate coefficient in the LMS adaption algorithm which will reduce the stability of the filters.

Figure 4.40 Average Error Output from Noise Canceler Showing Tracking Performance of Multireference Cancelers for 300-3500Hz Telephony Bandlimited Noise Data Recorded in an Automobile at 100km/hr, Ventilation Fan On.

- A: Widrow Canceler Without Subbands (Figure 4.2)
- B: Two Stage Beamforming Canceler Without Subbands (Figure 4.3)
- C: Widrow Canceler With Three Subbands (Figure 4.33)
- D: Two Stage Beamforming Canceler With Three Subbands (Figure 4.35)

4.12. Effects of Filter Length

The filter length in the adaptive filters used in all the automobile simulations in the previous sections has been fixed at 100 taps with the exception of the variations in the taps in the second stage of the beamforming cancelers.

All of the automobile noise results tables in Appendix C include simulation results for the 60km/hr, ventilation fan off case using 250 taps adaptive filters. This is the only difference between these results and the results shown directly above the entry for 100 taps in each of the tables. All other features of the filters are unchanged including the beamforming second stages.

A comparison of the results for the 250 tap filter shows that for the various simulations that were run using the non-subbanded filters, the difference between the results of the 250 and 100 tap filters is minimal. The results are shown in Tables C.6 - C.12. The maximum noise cancellation provided by the 250 tap filter for each of the cases is neither consistently better or worse than the 100 tap result. The split is approximately 50% better, 50% worse. If the same comparison is made of the average noise cancellation values, the 250 tap filter is consistently better with few exceptions.

This shows how the additional taps have allowed the filter to perform better over the complete spectrum although the extra taps don't consistently improve the maximum performance. Subbanding has been shown to provide a better improvement in the upper band than longer complete band filters have provided. This is shown by comparing the results of the beamforming canceler without subbands in Table C.8 that uses 250 taps/ref in stage 1 and 25 taps/ref in stage 2 (Structure 6) with the three subband beamforming canceler given in Table C.15 that has the equivalent of 300 taps/ref in stage 1 (3×100) and 25 taps/ref in stage 2 (Structure 63). The results show that even though the two cancelers

have similar structure and number of taps, the subbanded canceler provides better maximum, average and minimum noise spectrum reductions as it is able to track the non-stationarities in the processes better.

A closer look at the results for the multireferenced subbanded beamforming canceler in Table C.14 and C.15 shows that the average performance is only improved in the cases with a non-zero adjustable delay. Since the delay of 5msec represents fifty percent of the taps in the 100 tap filters and the 10msec delay represents all the taps in the 100 tap filter, the delay causes the 100 tap filter to have a significant reduction in the number of taps that are available to cancel causal processes with respect to the microphone inputs. Since it is desirable for the reference microphones to be placed as close to the noise sources as possible, it is desirable to have taps to cancel this type of noise. The longer filter allows for sufficient taps to be available to cancel the noise which is causal with respect to the microphones while providing taps for the non-causal processes.

Based on these observations the use of longer filters is only desirable to maintain taps to cancel the causal portion of the response and that the advantage of the longer filter is only obtained when the taps are used to provide taps to cancel non-causal noise processes while maintaining adequate taps to cancel the causal processes. The results show that the use of additional taps by separation of the references into subbands is more advantageous than using longer non-subbanded filters.

4.13. Effects of Delay

The results' summaries for each of the simulations using the automobile data in Table C.6 to C.15 include results for each of the cases with 0, 5, and 10 msec of delay added to the primary signal before it is input into the filter.

The possible need for this delay has been discussed by previous authors (Goulding [5], Hebert [3]) and it may be required if the source of a noise process is located closer to the primary microphone than any of the references. The result is that the noise process will reach the primary microphone before it reaches the reference and as a result the noise input into the filter primary input will be non-causal with respect to the filter inputs.

Two methods are available to prevent this from occurring and degrading the performance of the canceler. The first method is through appropriate microphone placement and the second is through the addition of delay on the primary path in the canceler. The addition of this delay will also degrade the overall performance of the canceller if the delay is noticeable to the user and affects the usability of the canceler.

The results shown in the tables in Appendix C show the effects of adding delay to the primary path while maintaining all other canceler parameters constant and no consistent improvement or degradation is provided by the addition of the delay.

Based on these results the inclusion of delay in the primary path is not warranted as the effects that this will have on the usability of the canceler are not offset by an improvement in performance as the delay on the speech path will affect the user in a similar fashion to the delays observed on satellite telephony trunks.

4.14. Summary

The results presented in this section show the systematic design of a multireference adaptive noise canceler through simulations of various cancelers with both noise data that has been created and noise data that has been recorded in an actual automobile.

A number of different filter structures have been simulated with the results showing that an adaptive noise canceler using multiple reference and subbanded filtering as shown in Figure 4.35 provides the best performance. The performance enhancement is achieved because the multistage structure is able to track the non-stationarities in the automobile noise environment better. It has also been shown that the first stage of this canceler should be made up with 100 tap adaptive filters while the second stage should have 25 tap filters for each input. The adjustable delay on the primary microphone path in the figure has also been shown to be 0msec for the best results based on usability criteria.

5. SYSTEM PERFORMANCE

The results in the previous chapter show that the subbanded beamforming noise canceler provides significantly better performance than the Widrow multireference adaptive noise canceler. To further test the performance of these two cancelers, additional simulations were run using the automobile noise data to compare the input noise power on the primary microphone to the output noise power. A second series of simulations investigated the effects of freezing the canceler which is required to prevent distortion of speech due to filter adaption as reported by Gardner [12]. The last series of simulations were run with speech added to the noise signals.

5.1. Performance Without Speech

Simulations were run using the data for the 60km/hr, ventilation fan off case with the subbanded beamforming adaptive noise canceler shown in Figure 4.35 and the subbanded Widrow multireference adaptive noise canceler shown in Figure 4.33 to compare the average input noise power on the primary microphone to the output noise power from the filter over a 30000 iteration simulation.

The noise power was measured using Equation 5.1 where the average was calculated over the last 25000 samples so that the filters were given 5000 iterations to converge ensuring that the results contain no start-up transients.

$$Pow = \frac{1}{N} \sum_{i=1}^N X_i^2 \quad \{5.1\}$$

Where X is the primary input signal to the filter or the output of the filter which is the noise canceled version of the input.

The results of these simulations show that with an average input power of 6280, the output power from the subbanded beamforming adaptive noise canceler was 143.4,

which is a noise reduction of 16.4dB. While the Widrow multireference adaptive noise canceler output power was 1456 which is only a reduction of 6.3dB, 10.1dB poorer than the beamforming canceler.

These results are presented in Table C.16 along with results that will be discussed in a later section. The powers are shown without dimensions as the results are based on the A/D converter output and not the actual input voltages to the converter or the sound energy of the noise input into the microphone of the recording system. Inclusion of the A/D converter gain and the recording system gains would allow the results to be presented with power dimensions but this conversion will not change the relative performance and as a result has been omitted.

The results in Table C.16 also show the noise spectrum reduction results for these simulations as were used in the previous chapter as a basis for comparison. The results when compared with the results in Table C.15 show that the performance of the cancelers during the long simulations is the same as those observed during the shorter simulations discussed previously.

Since the noise data case chosen for these simulations is one of the 60km/hr cases, the peak in the spectrum near 3000Hz is present in the data and as a result the spectral difference data shown in the Table C.16 shows a larger maximum noise reduction for the Widrow multireference adaptive noise canceler than the beamforming adaptive noise canceler. This better performance for this canceler near 3000Hz is offset by the significantly poorer performance elsewhere in the band as was discussed in the previous chapter and is shown by the better minimum and average noise cancellation results for the beamforming adaptive noise canceler and the overall better performance shown in the average power results.

These results again show the superior performance of the subbanded beamforming adaptive noise canceler over the subbanded Widrow multireference adaptive noise canceler.

5.2. Performance When Adaption Stopped

A series of simulations were performed to test the performance of both the Widrow multireference adaptive noise canceler and the beamforming adaptive noise canceler if the canceler was allowed to adapt for a period of time to allow the canceler to converge and then frozen for a period of time. This will allow for the performance of the noise cancelers while frozen to be evaluated.

The simulations were run using 8000 sample files for both of the data cases at 60km/hr. The results are presented for the 60km/hr ventilation fan on case and the results for the ventilation fan off case are similar. The filters were allowed to adapt for 0.2 sec (2000 iterations) and then frozen for 0.4 sec (4000 iterations), they were again allowed to adapt for an additional 0.2 sec (2000 iterations).

The output of the beamforming adaptive noise canceler and the Widrow multireference adaptive noise canceler are shown in Figures 5.1 and 5.2 respectively. These plots show the poor performance of the cancelers during the middle 0.4 seconds of the simulation. These results are further detailed in Figures 5.3, 5.4, 5.5 and 5.6 that show the input and output spectra of the cancelers during both the period where the cancelers are adapted and where they are frozen.

Figures 5.3 and 5.4 show the performance of the canceler while the canceler is adapting to follow the non-stationarities in the noise processes. These results are similar to those presented in the previous chapter and shows the superiority of the beamforming

Figure 5.1 Output Noise Signal from Subbanded Beamforming Multireference Noise Canceler Shown in Figure 4.35 for 60km/hr, Ventilation Fan Off Case When the Filter is Frozen at 0.2 Seconds and Allowed to Adapt at 0.6 Seconds.

Figure 5.2 Output Noise Signal from Subbanded Widrow Multireference Noise Canceler Shown in Figure 4.33 for 60km/hr, Ventilation Fan Off Case When the filter is Frozen at 0.2 Seconds and Allowed to Adapt at 0.6 Seconds.

Figure 5.3 Noise Cancellation Performance for Subbanded Beamforming Multireference Noise Canceller Shown in Figure 4.35 for 60km/hr, Ventilation Fan Off Case in Segment with the Filter Adapting.

Figure 5.4 Noise Cancellation Performance for Subbanded Widrow Multireference Noise Canceller Shown in Figure 4.33 for 60km/hr, Ventilation Fan Off Case in Segment with the Filters Adapting.

Figure 5.5 Noise Cancellation Performance for Subbanded Beamforming Multireference Noise Canceler Shown in Figure 4.35 for 60km/hr, Ventilation Fan Off Case in Segment with the Filters Frozen.

Figure 5.6 Noise Cancellation Performance for Subbanded Widrow Multireference Noise Canceller Shown in Figure 4.33 for 60km/hr, Ventilation Fan Off Case in Segment with the Filters Frozen.

Figure 5.7 Expanded View of a Portion of Figure 5.1 Showing the Performance of the Noise Canceler when the Filters are Frozen at 0.2 Seconds.

adaptive noise canceler. This superiority is also shown by comparing Figures 5.1 and 5.2 during the regions where the cancelers are adapting.

The results shown in Figures 5.5 and 5.6 show how both cancelers perform poorly when they are frozen and are no longer able to adapt to follow the non-stationarities in the system. Figure 5.7 shows an enlarged view of a portion of Figure 5.1 to detail the effects of freezing the canceler. The figure shows that the canceler was able to provide good cancellation for approximately 6 msec after the canceler was frozen at 0.2 seconds.

These results show that the noise processes in the automobile require that the canceler adapt continually to follow the non-stationarities in the processes. The cancelers may be frozen for periods up to 6 msec (60 iterations) with little effect on the performance of the canceler. These results are serious drawbacks on the performance of the canceler when the input signals include speech as will be discussed in the next section.

5.3. Performance With Speech

The performance of the subbanded beamforming adaptive noise canceler and subbanded Widrow multireference adaptive noise canceler was also evaluated through a series of simulations with speech signals added to the primary noise signal. These simulations allowed for the performance of the cancelers to be evaluated with subjective listening tests in addition to the objective measures previously used.

The summaries of the objective results of the various filter simulations that were run are detailed in Tables C.16, C.17 and C.18 where the simulations were run with a number of different conditions. The results in Table C.16 show results for simulations using the 60km/hr, ventilation fan off case with speech at a relatively high SNR of 16.9db and the 60km/hr, ventilation fan on case at an SNR of 7.2db using both of the

multireference subbanded structures in Figures 4.33 and 4.35 under a number of different conditions.

These simulations were performed on noise data containing only speech in the primary signal and also noise data with speech added to one of the reference microphones. The speech that was added to the reference microphone was a 2.5 msec delayed version of the speech in the primary signal. The speech in the reference microphone was also at a lower level as shown in the table. This allowed for the effects of speech coupling to the reference microphones to be observed.

A simple speech detector was also added to each of the filter structures to stop the adaption of the filters during speech segments. The speech detector used a power threshold criteria as shown in Figure 5.8. The speech detector allowed for comparison of the filter output when the cancelers were allowed to adapt continuously with the case when the cancelers were frozen during speech segments.

The resulting 16 simulations, of 30000 iterations each, are summarized in Table C.16. The noise cancellation spectral difference data that was used as the basis of comparison previously is shown in the table for each simulation. This data was calculated on a window of the data immediately following a speech period that contained no speech. The results show the better performance is that of the subbanded beamforming adaptive noise canceler as it has larger maximum, average and minimum noise reduction results compared to the subbanded Widrow multireference adaptive noise canceler for identical conditions. The results also show how the addition of a speech detector improves the performance of the canceler as this prevents the speech from causing the canceler to misadapt resulting in poorer noise reduction results.

The power data shown in the table is difficult to interpret as these signals are based on the residual noise power remaining after the speech signal has been subtracted

pow = (S' * S)/50

If pow > threshold then

Speech detected
Freeze filter
other actions if appropriate

Else

No Speech detected
Adapt filter

End IF

- Where S is a delay line containing the most recent 50 samples of the primary microphone signal
- Threshold is set as shown in Tables C.16 and C.17

Figure 5.8 Speech Detector Algorithm Used in Simulations

from the associated signal. The subtraction of the speech signal from the output signal assumes that the transfer function for the speech is pure delay which can be accounted for by a time shift.

Previous work (Gardner [12]) has shown that the transfer function on the speech is not a pure delay, due to the feedback of the output of the filters in the error signal used to adapt the filter coefficients. This effect introduces reverberation and cancellation of the speech which introduces additional power into the residual noise results. This obscures the noise cancellation effects even though the additional power may be of little subjective significance to the understandability of the speech.

The listening tests showed that when the cancelers were allowed to adapt continuously without a speech detector, significant speech cancellation and reverberation were observed with the beamforming adaptive noise canceler showing more cancellation. The listening tests on the files using the speech detector showed that the detector improved the performance of the cancelers although failure of the speech detector to detect speech allowed the canceler to introduce break-up into the speech as it would adapt for portions of speech segments while frozen for others. The listening tests showed that a more stringent speech detector is required so that the effects of the false non-detection of speech can be reduced.

These results were used in a second series of simulations with speech that were 18000 iterations in length. These simulations only used the 60km/hr, ventilation fan on case with an SNR of 5.9dB in the primary microphone. The summary of the results are detailed in Table C.17. These simulations repeated some of the test cases from the previous simulations using the same speech detector except the threshold was set lower to reduce the amount of false non-detection of speech.

The speech detector performance on the noisy speech is shown in Figure 5.9, where the speech detector state (0 = no speech, 3000 = speech) for each iteration has been plotted against the noisy speech signal. The speech detector state has also been plotted against the original noise free speech to allow the detector performance to be observed. The figure shows that the adjustment of the speech detector threshold was to reduce the false non-detection has introduced some false detection.

Additional simulations were run to assess the effects of the various methods that are available for freezing the filters. These include allowing some of the references to continue to adapt during speech segments such as the microphone in the engine compartment (x1) that will not contain any speech due to its isolation from the cabin and as a result speech cancellation due to speech in this reference will be insignificant.

The simulations also tested the effects of zeroing the output of some of the filters within the canceler during speech segments versus using the output of the cancelers based on the frozen state of the coefficients.

The various simulation cases discussed used noise data that contained speech in only the primary signal and data that contained speech in both the primary and one of the reference microphones. The noise in the reference microphone was delayed 2.5msec as in the previous simulations and at a lower level the primary as stated in Table C.17.

The objective data in Table C.17 again shows that the subbanded beamforming adaptive noise canceler has superior performance to the subbanded Widrow multireference adaptive noise canceler based on the spectral difference data that was calculated on a portion of the output that did not contain speech.

The power measurements again show the effects of residual speech power making comparison of these results difficult. The results do show that the residual power is

Figure 5.9 Plot Showing Speech Detector Performance for the Noisy Speech Plotted Against Both the Original Low Noise Speech and the Noisy Speech (5.9dB SNR). (0 = No Speech Detected, 3000 = Speech Detected)

reduced when the reference in the dashboard (x2) is zeroed instead of frozen during speech segments. This indicates that the non-stationarities of the noise processes in this reference are significant as zeroing the filter outperforms the results using the frozen filter indicating that the frozen coefficients are adding to the noise energy. The opposite is true for the floor microphone (x3) although the effect is much less pronounced as the zeroing of the filters outputs for this filter causes an increase in the noise power.

The effect of the speech in the reference signal is also shown by the increase in the residual noise power due to speech cancellation and the speech reverberation introduced into the canceler output because of the speech in the reference. This is shown by the significant increase in the residual noise power when the dashboard reference (x2) contains speech and the canceler only freezes the coefficients when speech is detected. The simulations that zeroed the output of the x2 filters during speech segments show little difference between the cases with speech in the reference and no speech as expected since zeroing the filters will eliminate the effects discussed above if the speech detector is ideal.

The listening tests that were performed on this data again showed that allowing cancelers to adapt during speech segments introduced significant speech cancellation and reverberation even if only a single input was allowed to adapt. The listening tests showed that the effect of zeroing the reference microphones had no observable effect as any effects that it may have had were obscured by the effects of the continuous adaption of the engine compartment microphone.

The listening results for the cases that froze all filters during speech segments showed that the more stringent speech detector improved the performance of the system by reducing the break-up of the speech due to false non-detection. Further improvement

to the speech detector would further improve its performance but this is beyond the scope of this work.

The tests did show that the reduction of the noise during the non-speech segments is significant with the subbanded beamforming adaptive noise canceler showing better performance than the subbanded Widrow multireference adaptive noise canceler. The canceler had little effect during speech segments but an overall improvement was observed because of lower noise during the silent segments and the initial 5-6msec of the speech.

The performance of the canceler will be superior to the performance observed with a gain switching system where the signal path is zeroed during non-speech periods and enabled during speech segments. This system will introduce noise pumping due the switching on/off of the signal path while the canceler has a slower and cleaner transition. Also the loss of speech segments due to switching and noise injection during non speech segments due to the speech detector operation will also be observed and affect the subjective performance. These effects can be observed in some handsfree telephones that use a gain switching algorithm to prevent feedback between the speaker and microphone in the set.

The results of simulations that were run using data containing speech with no speech detector where the SNR was low (-11.0dB) are detailed in Table C.18. The spectral difference data shown is similar to previous results for the Widrow and beamforming cancelers with the beamforming canceler providing the best results over the complete audio band and a significant reduction in the 300-1000Hz region of the band while the Widrow canceler has poorer performance in the lower portion of the band where the dominant noise is present and better performance at the peak in the spectrum at 3000Hz.

Listening tests on the data from these simulations show that the beamforming canceler provided much better noise reduction than the Widrow canceler and both cancelers also caused significant speech cancellation. The speech cancellation at such a high input noise SNR shows that the adaption rate of the filters is sufficiently fast to allow the feedback of the speech through the error path in the LMS algorithm to cancel the speech.

A possible solution to this problem would be the use of a much slower adaption rate through the reduction of β . The lower β could be introduced with the speech detector allowing the filters to continue to adapt during speech segments without causing significant speech cancellation while following the non-stationarities in the noise processes to maintain the noise cancellation.

5.4. Summary

The results shown in this chapter show that although the adaptive noise canceler that has been developed provides significant noise cancellation performance when continuously adapting. However, the non-stationarities in the noise data cause the canceler to operate poorly when frozen during speech segments.

Freezing of the cancelers is required to prevent the cancellation of the speech due the error feedback path into the canceler coefficients and also due to speech in the reference microphones. These effects require that a speech detector be used and that it must perform adequately to reduce false non-detections and false detections to a minimum.

These results show some areas for further work into the cancelers to try to improve their performance:

- 1) Improved speech detectors can be designed and used to reduce the false detections and non-detections.
- 2) The possible use of a slower adaption coefficient during speech segments that would allow the canceler to follow the non-stationarities in the noise processes while reducing the effects that the speech has in causing misadaption of the coefficients.

6. DISCUSSION OF RESULTS

6.1. Conclusions

The results that have been presented show the development of a parallel multireference subbanded adaptive noise canceler that provides noise cancellation over the complete telephony audio band of 300-3500Hz. The canceler uses a single microphone as the primary signal and a set of microphones (three) as reference noise sources that are adaptively filtered to cancel the noise in the primary signal.

It has been shown that the configuration of the adaptive noise canceler affects the performance of the system. The work shows that a two stage canceler in Figure 4.35 provides the best performance for the automobile noise cases that have been tested. This performance has been shown to be better than a single stage canceler as proposed by Widrow [9] and also better than a cascade of single reference cancelers. The work shows that the structure of an adaptive noise canceler significantly affects the ability of the canceler to track the non-stationarities in the multipath, multisource noise environment present in the automobile.

It has also been shown that splitting the noise signals into subbands improves performance of the canceler as it allows shorter filters to be used which are able to converge faster and track non-stationarities better than a single long filter. The use of subbands focuses the taps to specific frequency bands so that they can adapt to provide cancellation in the band of interest whereas the taps in the longer filters adapt so that they provide optimal performance over the wider band which may cause noise enhancement as the portions of the band with the largest noise energy have the greatest effect on the coefficients.

Investigation of the performance of the adaptive noise canceler with speech segments has shown that the canceler must be frozen or the adaption slowed during

speech segments, otherwise the speech signal fed back into the filter through the error signal used to adapt the coefficients causes the filter to introduce reverberation and cancellation of the speech.

The addition of a speech detector has shown that the canceler provides significant noise reduction during non-speech segments based on subjective listening tests. The performance during speech segments is degraded as the non-stationarities in the noise processes cause the filter taps to differ from the required weights for cancellation after approximately 6 msec. This causes the noise cancellation to be relatively good during the initial speech segment while the latter portions of the speech have no noise cancellation.

The speech signals were also degraded by the operation of the speech detector that was used. The detector used a power threshold decision algorithm to determine if speech was present or not. This simple detector introduced both false detections and false non-detections that reduced the performance of the system. The speech detector performance may be enhanced through the use of a more complex algorithm.

In summary, an adaptive noise canceler has been developed that extends the reference microphone approach originally proposed by Widrow [9] through the use of a two stage canceler, multiple reference microphones, and subbanded noise cancelers (Figure 4.35) to provide noise spectrum reduction of up to 28dB in the 300-1000Hz band that contains the dominant noise power. The minimum average reduction of the noise spectrum is 7.4dB for all the noise cases simulated with the canceler showing a reduction of the noise spectrum at all frequencies in the 300-3500Hz band for all the noises cases tested. These results are significantly better than the best previously published results that show a maximum reduction of only 15dB in the lower portion of the band with the upper portion of the band unchanged.

6.2. Recommendations for Further Work

The results presented in Chapter 4 show that the parallel multireference adaptive noise canceler using a two stage beamforming structure along with subbanded filters as shown in Figure 4.35 provides significantly better noise cancellation than has been previously reported.

The performance of this new canceler along with that of the Widrow multireference canceler structure shown in Figure 4.33 has been shown to be poor when the cancelers are frozen due to the non-stationarities in the automobile noise and the performance is also degraded due to the fast adaption of the filters when speech is present in the error signal used to adapt the filter coefficients.

The feedback of the speech causes the filters to introduce reverberation and to actually cancel the speech in the output signal. A simple speech detector was introduced into the noise cancelers and it was shown that it improves the performance by reducing the degradation of the speech caused by the adaption.

These results indicate that the further work should be pursued into the following problems:

- 1) **Use of a better speech detector:** A more complex speech detection algorithm should be added to the filters so that speech detection errors are reduced which will also reduce the speech cancelation and break-up effects that false non-detection causes. This detector will allow for a better subjective evaluation to be made of the performance of the cancelers when speech signals are used.
- 2) **Slow adaption during speech segments:** The possible performance enhancements of the canceler that may be obtained through the use of a slower adaption rate (smaller β) during speech segments should be investigated. The smaller β may

allow the canceler to follow the non-stationarities in the noise while reducing or eliminating the speech reverberation and cancellation effects that have been observed. The use of different β for the various microphones and subbands may be warranted. This may show that the filter is still able to track the non-stationarities while reducing the affects of speech on the adaption.

- 3) **Real time implementation:** Implementation of the adaptive noise canceler in real time will allow for the investigations of the performance of the canceler in the presence of speech to be evaluated and tested more easily. The design and implementation of the parallel processor hardware system required for the system will include a number of significant research design issues especially in the design of the data share and communication protocols between the parallel processors that will be required for the real time system.
- 4) **Application of the canceler to other systems:** The data that was used in the simulations for this work was all automobile noise. Other applications such a airplane cockpit noise, industrial noise, and any other systems requiring communication in a high noise environments should be investigated. The canceler may provide a significant improvement in these applications.
- 5) **Application of the canceler to medical applications:** Noise cancelers systems are currently used in medical equipment such as EEG and ECG. The performance enhancements that the two stage beamforming canceler may provide over the cancelers currently in use should be investigated. Also application of the multireference subbanded filter structure may allow for the improvement in the performance of hearing aids allowing for the reduction of the cocktail party effect that currently causes the difficulty for the hearing impaired in high noise environments.

- 6) **Further investigation of subband filters:** The use of subbanded adaptive filters has received relatively little treatment in the literature. The enhancements that this principle provides in the tracking performance of the canceler should be further investigated. The design of the fixed filters requires further evaluation to ensure that the specifications are sufficient to ensure that no edge effects occur at the seams between the subbands. The length of the filters should also be evaluated to minimize the effects of the delay in the filters.

This further work will allow the possible performance enhancements through improved noise reduction that the subbanded beamforming multireference adaptive noise canceler proposed in this work may provide over systems that currently use noise cancellation and also in applications that currently do not use noise cancellation.

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APPENDIX A:

M-FILE LIBRARY FUNCTIONS

The functions listed in this section are library functions used in the simulation programs listed in Appendix B. These files are functions created specifically for this work to augment the functions provided by Matlab [56,57].

The functions implement the processing and analysis building blocks required by the simulation files allowing the simulation programs to be more understandable and easier to debug.

A.1. Function Listings

A.1.1. INP - Inner Product Function

This function calculates the dot product of two vectors that are of equal length returning the scalar result. The function checks to see if both vectors are of equal length and returns an error message if they are not equal.

```
function y = inp(x,z)
% Function to calculate the dot product of two vectors
% Vectors are assumed to be rows
%
% format    inp(x,z)
%
%    inp(x,z) = x * z'
m = length(x);
n = length(z);
if m==n
    y= x * z';
else
    error('unequal length vectors in INP')
end
```

A.1.2. OUTP - Outer Product Function

This function is the complement of the INP function and calculates the outer product of two row vectors returning a square matrix. An error statement is generated if the vectors are of unequal length.

```
function y = outp(x,z)
% Funtion to calculate the dot product of two vectors
% Vectors are assumed to be rows
%
% format outp(x,z)
%
% outp(x,z) = x' * z
m = length(x);
n = length(z);
if m==n
    y= x' * z;
else
    error('unequal length vectors in OUTP')
end
```

A.1.3. LMSFIR - LMS Filter Routine

This function implements a complete LMS FIR filter. The function calculates the filter output and error signal in addition to calculating the adaption rate coefficient β and updating the filter tap weights. All of these results are returned as separate components in the results matrix.

This function uses the INP, BETAUP, and LMSU functions included in this appendix.

```
function [E,Y,W1,B]=lmsfir(X,D,W,a,b)
% Funtion to perform complete LMS filter update and calc.
%
% format [E,Y,W1,B]=lmsfir(X,D,W,a,b)
%
% E = error from filter (D-Y) - s
% Y = output of filter - s
% W1 = output filter taps - v
% X = input data to filter - v
% D = desired response - s
% W = input filter taps - v
% B = LMS step size - s
% a,b = constants to limit B - s
Y = inp(W,X);
E = D-Y;
B = betaup(X,a,b);
W1 = lmsu(W,X,B,E);
```

A.1.4. LMSU - LMS Tap Update Function

This function updates the taps of an FIR filter using the LMS adaption algorithm.

The function provides an error message if the input vectors are of unequal length.

```
function wn1 = lmsu(wn0,X,e,B)
% Function to update an LMS filter
% Vectors are assumed to be rows
%
% format lmsu(wn0,X,e,B)
%
% lmsu(wn1,X,e,B) = wn0 + B * e * X
% wn0 = filter taps
% B = LMS step size
% e = error
% X = Input vector
m = length(wn0);
n = length(X);
if m==n
    wn1 = wn0 + B * e * X;
else
    error('unequal length vectors in LMSU')
end
```

A.1.5. BETAUP - LMS β Function

This function calculates the adaption rate coefficient for the LMS filter based on the inverse of the power in the input signal to the filter. Two parameters a and b are provided to ensure that the value of β is limited under low power and start up transient conditions.

```
function B = betaup(X,a,b)
% Function to update LMS filter step size
%
%    format    betaup(X,a,b)
%
%    B = a/(b + X*X')
%    a,b = constants to limit B - s
%    X = input data - v
%    B = LMS step size - s
Y = inp(X,X);
B = a/(b+Y);
```

A.1.6. MOD - Modulus Function

This functions returns the modulus of a number for a given base.

```
function s = mod(x,modl)
% function to return the modulus of a number
%   s = mod(x,mod)
%
%   x = number
%   modl = modulus base
s = x - modl*(fix(x/modl));
end
```

A.1.7. FORMOUT - Minimum Power Comparison Function

This function returns the first location (current data value) from one of three vectors. The value returned is from the vector that has the minimum power. In addition the number of the vector that the result was selected from is returned.

This function was used in the simulations using the multiple single reference canceler with minimum output power selection to show that the when three separate cancelers were used, the canceler providing the minimum output power varies between the three reference microphones.

```
function [y,I]=formout(a,b,c)
% function to calculate the output of the struct3 filter
% the returned value is the value in the first location
% of the vector that has the lowest magnitude
ma=inp(a,a);
mb=inp(b,b);
mc=inp(c,c);
if ma>mb
    if mb>mc
        y=c(1);
        I = 3;
    else
        y=b(1);
        I = 2;
    end
else
    if ma>mc
        y=c(1);
        I = 3;
    else
        y=a(1);
        I = 1;
    end
end
end
```


A.1.8. SHFTR - Vector Shift Right Function

This function shifts a row vector one location to the right with zero fill. This implements a unit delay when the vector is a delay line.

```
function X1 = shftr(X)
% Function to backward shift a vector
%
% format shftr(x)
%
% shftr(X(i)) = X(i+1)
%
% includes zero fill
m = length(X);
X1(2:m) = X(1:(m-1));
end
```

A.1.9. SHFTL - Vector Shift Left Function

This function provides the converse to the SHFTR function implementing a left shift by one location of a vector with zero fill.

```
function X1 = shftl(X)  
% Function to forward shift a vector  
%  
% format shftl(x)  
%  
% shftl(X(i)) = X(i-1)  
%  
% includes zero fill  
m = length(X);  
    X1(1:(m-1)) = X(2:m);  
    X1(m) = 0;  
end
```

A.1.10. PLOTTER - Automatic Plotter Function

This function implements an automatic analysis tool for simulation results. The spectra of two input vectors are calculated using the Welch method (Oppenheim [58]). The resulting power spectra are converted to logarithmic scales (dB) and plotted. Two separate plots are generated; the first shows both curves while the second is a difference plot. The difference plot also includes text values stating the maximum, minimum and average difference between the two curves in addition to the standard deviation of the difference. These values are calculated on the telephony frequency band of 300-3500Hz.

The routine allows for a title for the plots to be passed into the function and also a filename that a Matlab Meta plot file is saved to.

```
function [Paa,Pbb]=plotter(a,b,fftl,noover,fname,plott)
% plotter
% function to calculate and plot spectral comparison
% plots of filter data
%
% Paa - Power spectral density of a vector
% Pbb - Power Spectral density of b vector
% a - input data vector - Primary
% b - filter output vector
% fftl - Length of fft to calculate Spectra
% noover - number of samples to overlap each spectra
% fname - name of file to save plots in
% plott - title to use on plots
    P = spectrum(a,b,fftl,noover);
    Paa = 10*log10(P(:,1));
    Pbb = 10*log10(P(:,2));
    mp=length(Paa);
    freq = linspace(0,5000,mp);
    Pdiff = Paa - Pbb;
    plot(freq,Paa(:,1),freq,Pbb(:,1));
    xlabel('frequency - Hz');
    ylabel('Spectrum Amplitude - dB');
    title(plott);
    grid
    eval(['meta ',fname])
```

```

    plot(freq,Pdiff(:,1));
    xlabel('frequency - Hz');
    ylabel('Spectrum Amplitude difference - dB');
    title(plott);
    grid
% Find limits in spectrum for telephony bandwidth
    limlow = fix(((mp-1)*3/50)+2);
    limhigh = fix(((mp-1)*35/50)+1);
    Pmax=max(Pdiff(limlow:limhigh,1));
    PMa=num2str(Pmax);
    Pmean=mean(Pdiff(limlow:limhigh,1));
    PMe=num2str(Pmean);
    Pmin=min(Pdiff(limlow:limhigh,1));
    PMi=num2str(Pmin);
    Pvar=std(Pdiff(limlow:limhigh,1));
    PVa=num2str(Pvar);
    text(.75,.85,'Max. =','sc')
    text(.75,.82,'Mean =','sc')
    text(.75,.79,'Min. =','sc')
    text(.75,.76,'Std. =','sc')
    text(.85,.85,PMa,'sc')
    text(.85,.82,PMe,'sc')
    text(.85,.79,PMi,'sc')
    text(.85,.76,PVa,'sc')
    meta
end

```

APPENDIX B: SIMULATION PROGRAMS

This appendix includes listings of most of the simulation script files that were used in simulations. Some files have been omitted to reduce the length of the appendix. The omitted files are minor variations on those included here, such as using different reference microphone assignments in the single reference and cascade cancelers.

Files that are variations on the two cancelers with speech detectors have been omitted. The difference between the omitted files and the included versions are the removal of the speech detector and the modification of the speech detector so that it was purely based on iteration number. These were used in the simulations with speech and no detector in the first case and for the simulations to see the effects of freezing the filter on noise without speech in the second case.

Also omitted were variations of the single band beamforming canceler (Struct4) that calculated β using various algorithms and a version of the two stage beamforming canceler with the complete beamforming second stage sparse FIR filters on the second stage which is a variation of Structure 4 and 6. The subbanded parallel multireference structures with two bands have also been omitted.

B.1. STRUCT1T - Single Reference Three-Band Canceled

This simulation code file implements the filter structure shown in Figure 4.29 which is a single reference three subband noise canceler. The function shown below uses the X1 reference as the single reference microphone. Two additional versions of this code file STRUCT2T and STRUCT3T have been omitted and they implement the same structure for the other two references.

```
function zz=struct1t(fdat,fmat,fmet,firl,a,b,T,title)
% function zz=struct1t(fdat,fmat,fmet,firl,a,b,T,title)
% M file for structure 1
% Single error feedback parallel structure
% with subband adaption (3 bands)
% special version for 3 band single reference
%
% add dat file to use on next line
eval(['load ',fdat])
load filt3
% X4 = primary
% X3 = floor
% X2 = glove
% X1 = engine
m = length(x4);
% total delay due to filters (FIR)
fdel = 128;
% initialize arrays for filter
N1 = zeros(1,firl);
N2 = zeros(1,firl);
N3 = zeros(1,firl);
w1 = zeros(1,(firl));
w2 = zeros(1,(firl));
w3 = zeros(1,(firl));
d = zeros(1,m);
Y1 = zeros(1,129);
Y2 = zeros(1,129);
Y3 = zeros(1,129);
% Only fill delay lines to length required by firl
% ie keep vectors to firl length even if delay is greater
Tmin=min([firl,T]);
for ii=T+1-Tmin:T
% prime delay lines for data on reference mics
    N1(T+1-ii) = x1b1(ii);
```

```

        N2(T+1-ii) = x1b2(ii);
        N3(T+1-ii) = x1b3(ii);
    end
    for i=T+1:m
        % shift arrays for time delay
        N1 = shft(N1);
        N2 = shft(N2);
        N3 = shft(N3);
        % get next sample
        s1 = x4b1(i-T);
        s2 = x4b2(i-T);
        s3 = x4b3(i-T);
        % assign each microphone to noise input on filter and get next sample
        N1(1) = x1b1(i);
        N2(1) = x1b2(i);
        N3(1) = x1b3(i);
        % calculate outputs of filters
        Y1 = shft(Y1);
        Y2 = shft(Y2);
        Y3 = shft(Y3);
        [e1,Y1(1),w1,B1]=lmsfir(N1,s1,w1,a,b);
        [e2,Y2(1),w2,B2]=lmsfir(N2,s2,w2,a,b);
        [e3,Y3(1),w3,B3]=lmsfir(N3,s3,w3,a,b);
        % calculate actual filter output
        % account for filtering delays and non-causal delays
        % note that first points of d(1:T+fdel) are garbage
        ll = max(1,(i-T-fdel));
        % filter and form output
        Z1 = inp(fb1,Y1);
        Z2 = inp(fb2,Y2);
        Z3 = inp(fb3,Y3);
        d(i)=x4(ll)-Z1-Z2-Z3;
    % end of for loop
    end
    [Pxx,Pdd]=plotter(x4(m-511-T-fdel:m-T-fdel),d(m-511:m),128,64,fmet,title);
    % input filename to save data (add/remove % to enable/disable)
    clear x1b1
    clear x2b1
    clear x3b1
    clear x4b1
    clear x1b2
    clear x2b2
    clear x3b2
    clear x4b2
    clear x1b3
    clear x2b3
    clear x3b3
    clear x4b3
    clear x4
    eval(['save ',fmat])
    % clean up workspace
    clear

```

B.2. STRUCT3A - Triple Single Reference Canceler - Minimum Power Output

This code file implements the minimum power decision output filter structure shown in Figure 4.5.

```
function zz=struct3a(fdat,fmat,fmet,firl,a,b,T,ptitle)
% M file for structure 3a
% multi error feedback parallel structure
%
% add dat file to use on next line
eval(['load ',fdat])
% X4 = primary
% X3 = floar
% X2 = glove
% X1 = engine
m = length(x4);
% number of samples to base power average output decision
n= 100;
% Input a,b constants for Beta limits
a1= a ;
a2= a ;
a3= a ;
b1= b ;
b2= b ;
b3= b ;
% initialize arrays for filter
n1 = zeros(1,firl);
n2 = zeros(1,firl);
n3 = zeros(1,firl);
w11 = zeros(1,firl);
w21 = zeros(1,firl);
w31 = zeros(1,firl);
e1 = zeros(1,m);
e2 = zeros(1,m);
e3 = zeros(1,m);
% Only fill delay lines to length required by firl
% ie keep vectors to firl length even if delay is greater
Tmin=min([firl,T]);
for ii=1:Tmin
% prime delay lines for data on reference mics
    n1(Tmin+1-ii) = x1(ii);
    n2(Tmin+1-ii) = x2(ii);
    n3(Tmin+1-ii) = x3(ii);
end
```



```

for i=1+T:m
% shift arrays for time delay
    n1 = shft(n1);
    n2 = shft(n2);
    n3 = shft(n3);
% get next sample
    s = x4(i-T);
% assign each microphone to noise input on filter and get next sample
% series order is n1,n2,n3
    n1(1) = x1(i);
    n2(1) = x2(i);
    n3(1) = x3(i);
% calculate each fir filter ( output, adapt beta etc.)
    w13=w12;
    w12=w11;
    w23=w22;
    w22=w21;
    w33=w32;
    w32=w31;
    [e1(i),Y1,w11,B1]=lmsfir(n1,s,w11,a1,b1);
    [e2(i),Y2,w21,B2]=lmsfir(n2,s,w21,a2,b2);
    [e3(i),Y3,w31,B3]=lmsfir(n3,s,w31,a3,b3);
% end of for loop
% find output of filter
    TT=max(1,(i-n+1));
    [d(i),ref(i)] = formout(e1(TT:i),e2(TT:i),e3(TT:i));
end
plot(ref)
xlabel('Iteration')
ylabel('Reference mic chosen')
title(ptitle)
eval(['meta ',fmet])
%[Pxx,Pdd]=plotter(x4(m-511-T:m-T),d(m-511:m),128,64,fmet,title);
% input filename to save data (add/remove % to enable/disable)
clear x1
clear x2
clear x3
clear x4
eval(['save ',fmat])
% clean up workspace
clear

```

B.3. STRUCT1 - Multiple Reference Widrow Canceler

This code file implements the Widrow multiple reference noise canceler shown in Figure 4.2.

```

function zz=struct1(fdat,fmat,fmet,firl,a,b,T,title)
% function zz=struct1(fdat,fmat,fmet,firl,a,b,T,title)
% M file for structure 1
% Single error feedback parallel structure
%
% add dat file to use on next line
eval(['load ',fdat])
% X4 = primary
% X3 = floor
% X2 = glove
% X1 = engine
m = length(x4);
% initialize arrays for filter
n1 = zeros(1,firl);
n2 = zeros(1,firl);
n3 = zeros(1,firl);
w11 = zeros(1,firl);
w21 = zeros(1,firl);
w31 = zeros(1,firl);
e = zeros(1,m);
% Only fill delay lines to length required by firl
% ie keep vectors to firl length even if delay is greater
Tmin=min([firl,T]);
for ii=1:Tmin
% prime delay lines for data on reference mics
    n1(Tmin+1-ii) = x1(ii);
    n2(Tmin+1-ii) = x2(ii);
    n3(Tmin+1-ii) = x3(ii);
end
for i=T+1:m
% shift arrays for time delay
    n1 = shft(n1);
    n2 = shft(n2);
    n3 = shft(n3);
% get next sample
    s = x4(i-T);
% assign each microphone to noise input on filter and get next sample
    n1(1) = x1(i);
    n2(1) = x2(i);
    n3(1) = x3(i);
% calculate outputs of filters

```

```

        nw1 = inp(n1,w11);
        nw2 = inp(n2,w21);
        nw3 = inp(n3,w31);
    % calculate error
        e(i) = s - nw1 - nw2 - nw3;
    % calculate betas
        N = [n1,n2,n3];
        B = betaup(N,a,b);
    % update filter coefficients
        w13=w12;
        w12=w11;
        w23=w22;
        w22=w21;
        w33=w32;
        w32=w31;
        w11 = lmsu(w11,n1,e(i),B);
        w21 = lmsu(w21,n2,e(i),B);
        w31 = lmsu(w31,n3,e(i),B);
    % end of for loop
end
% save filter output
d = e;
[Pxx,Pdd]=plotter(x4(m-511-T:m-T),d(m-511:m),128,64,fmet,title);
% input filename to save data (add/remove % to enable/disable)
clear x1
clear x2
clear x3
clear x4
eval(['save ',fmat])
% clean up workspace
clear

```

B.4. STRUCT13 - Three Subband Version of STRUCT1

This simulation code file implements the three subband version of the Widrow multiple reference noise canceler shown in Figure 4.33. A similar program with only two subbands has been omitted.

```
function zz=struct13(fdat,fmat,fmet,firl,a,b,T,title)
% function zz=struct13(fdat,fmat,fmet,firl,a,b,T,title)
% M file for structure 1
% Single error feedback parallel structure
% with subband adaption (3 bands)
%
% add dat file to use on next line
eval(['load ',fdat])
load filt3
% X4 = primary
% X3 = floar
% X2 = glove
% X1 = engine
m = length(x4);
% total delay due to filters (FIR)
fdel = 128;
% initialize arrays for filter
n11 = zeros(1,firl);
n21 = zeros(1,firl);
n31 = zeros(1,firl);
w1 = zeros(1,(3*firl));
n12 = zeros(1,firl);
n22 = zeros(1,firl);
n32 = zeros(1,firl);
w2 = zeros(1,(3*firl));
n13 = zeros(1,firl);
n23 = zeros(1,firl);
n33 = zeros(1,firl);
w3 = zeros(1,(3*firl));
d = zeros(1,m);
Y1 = zeros(1,129);
Y2 = zeros(1,129);
Y3 = zeros(1,129);
% Only fill delay lines to length required by firl
% ie keep vectors to firl length even if delay is greater
Tmin=min([firl,T]);
for ii=T+1-Tmin:T
% prime delay lines for data on reference mics
    n11(T+1-ii) = x1b1(ii);
```

3 of/de 3

MICROCOPY RESOLUTION TEST CHART
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STANDARD REFERENCE MATERIAL 1010a
(ANSI and ISO TEST CHART No. 2)

```

n21(T+1-ii) = x2b1(ii);
n31(T+1-ii) = x3b1(ii);
n12(T+1-ii) = x1b2(ii);
n22(T+1-ii) = x2b2(ii);
n32(T+1-ii) = x3b2(ii);
n13(T+1-ii) = x1b3(ii);
n23(T+1-ii) = x2b3(ii);
n33(T+1-ii) = x3b3(ii);
end
for i=T+1:m
% shift arrays for time delay
n11 = shft(n11);
n21 = shft(n21);
n31 = shft(n31);
n12 = shft(n12);
n22 = shft(n22);
n32 = shft(n32);
n13 = shft(n13);
n23 = shft(n23);
n33 = shft(n33);
% get next sample
s1 = x4b1(i-T);
s2 = x4b2(i-T);
s3 = x4b3(i-T);
% assign each microphone to noise input on filter and get next sample

n11(1) = x1b1(i);
n21(1) = x2b1(i);
n31(1) = x3b1(i);
n12(1) = x1b2(i);
n22(1) = x2b2(i);
n32(1) = x3b2(i);
n13(1) = x1b3(i);
n23(1) = x2b3(i);
n33(1) = x3b3(i);
% calculate outputs of filters
N1 = [n11,n21,n31];
N2 = [n12,n22,n32];
N3 = [n13,n23,n33];
Y1 = shft(Y1);
Y2 = shft(Y2);
Y3 = shft(Y3);
[e1,Y1(1),w1,B1]=lmsfir(N1,s1,w1,a,b);
[e2,Y2(1),w2,B2]=lmsfir(N2,s2,w2,a,b);
[e3,Y3(1),w3,B3]=lmsfir(N3,s3,w3,a,b);
% calculate actual filter output
% account for filtering delays and non-causal delays
% note that first points of d(1:T+fdel) are garbage
II = max(1,(i-T-fdel));
% filter and form output
Z1 = inp(fb1,Y1);
Z2 = inp(fb2,Y2);

```

```
        Z3 = inp(fb3,Y3);  
        d(i)=x4(II)-Z1-Z2-Z3;  
    % end of for loop  
end  
[Pxx,Pdd]=plotter(x4(m-511-T-fdel:m-T-fdel),d(m-511:m),128,64,fmet,title);  
% input filename to save data (add/remove % to enable/disable)  
clear x1b1  
clear x2b1  
clear x3b1  
clear x4b1  
clear x1b2  
clear x2b2  
clear x3b2  
clear x4b2  
clear x1b3  
clear x2b3  
clear x3b3  
clear x4b3  
clear x4  
eval(['save ',fmat])  
% clean up workspace  
clear
```

B.5. S13DLONG- Speech Detector Added to STRUCT13 - Long File

This is a modified version of the previous code file with the support code added so that simulations can be run on long files overcoming the 8196 vector size limit of Matlab. The routine also includes the speech detector.

```
function zz=s13dlong(fdat,fmat,firl,a,b,sdlevel,title)
% function zz=s13dlong(fdat,fmat,firl,a,b,sdlevel,title)
% M file for structure 1
% Single error feedback parallel structure
% with subband adaption (3 bands)
%
% add dat file to use on next line
load filt3
% Speech detect power threshold and detect length
dlength = 100;
thres = dlength * sdlevel;
% X4 = primary
% X3 = floar
% X2 = glove
% X1 = engine
% total delay due to filters (FIR)
fdel = 128;
% initialize arrays for filter
n1 = zeros(1,129);
n2 = zeros(1,129);
n3 = zeros(1,129);
s = zeros(1,max(129,dlength));
n11 = zeros(1,firl);
n21 = zeros(1,firl);
n31 = zeros(1,firl);
w1 = zeros(1,(3*firl));
n12 = zeros(1,firl);
n22 = zeros(1,firl);
n32 = zeros(1,firl);
w2 = zeros(1,(3*firl));
n13 = zeros(1,firl);
n23 = zeros(1,firl);
n33 = zeros(1,firl);
w3 = zeros(1,(3*firl));
d1 = zeros(2,2000);
d2 = zeros(2,2000);
d3 = zeros(2,2000);
d4 = zeros(2,2000);
d5 = zeros(2,2000);
```



```

d6 = zeros(2,2000);
d7 = zeros(2,2000);
d8 = zeros(2,2000);
d9 = zeros(2,2000);
d10 = zeros(2,2000);
d11 = zeros(2,2000);
d12 = zeros(2,2000);
d13 = zeros(2,2000);
d14 = zeros(2,2000);
d15 = zeros(2,2000);
Y1 = zeros(1,129);
Y2 = zeros(1,129);
Y3 = zeros(1,129);
for i1 = 0:14
for i2= 1:2000
i=i1*2000+i2;
% shift arrays for time delay
    s = shftr(s);
    n1 = shftr(n1);
    n2 = shftr(n2);
    n3 = shftr(n3);
    n11 = shftr(n11);
    n21 = shftr(n21);
    n31 = shftr(n31);
    n12 = shftr(n12);
    n22 = shftr(n22);
    n32 = shftr(n32);
    n13 = shftr(n13);
    n23 = shftr(n23);
    n33 = shftr(n33);
% get next sample
% assign each microphone to noise input on filter and get next sample
    if i > 28001
        n1(1)=x1f(i-28000);
        n2(1)=x2f(i-28000);
        n3(1)=x3f(i-28000);
        s(1)=x4f(i-28000);
    elseif i == 28001
        clear x1d
        clear x2d
        clear x3d
        clear x4d
        eval(['load ',fdat,'f'])
        n1(1)=x1f(1);
        n2(1)=x2f(1);
        n3(1)=x3f(1);
        s(1)=x4f(1);
    elseif i > 26001
        n1(1)=x1e(i-26000);
        n2(1)=x2e(i-26000);
        n3(1)=x3e(i-26000);
        s(1)=x4e(i-26000);

```

```

elseif i == 26001
    clear x1c
    clear x2c
    clear x3c
    clear x4c
    eval(['load ',fdat,'e'])
    n1(1)=x1e(1);
    n2(1)=x2e(1);
    n3(1)=x3e(1);
    s(1)=x4e(1);
elseif i > 24001
    n1(1)=x1d(i-24000);
    n2(1)=x2d(i-24000);
    n3(1)=x3d(i-24000);
    s(1)=x4d(i-24000);
elseif i == 24001
    clear x1b
    clear x2b
    clear x3b
    clear x4b
    eval(['load ',fdat,'d'])
    n1(1)=x1d(1);
    n2(1)=x2d(1);
    n3(1)=x3d(1);
    s(1)=x4d(1);
elseif i > 22001
    n1(1)=x1c(i-22000);
    n2(1)=x2c(i-22000);
    n3(1)=x3c(i-22000);
    s(1)=x4c(i-22000);
elseif i == 22001
    clear x1a
    clear x2a
    clear x3a
    clear x4a
    eval(['load ',fdat,'c'])
    n1(1)=x1c(1);
    n2(1)=x2c(1);
    n3(1)=x3c(1);
    s(1)=x4c(1);
elseif i > 20001
    n1(1)=x1b(i-20000);
    n2(1)=x2b(i-20000);
    n3(1)=x3b(i-20000);
    s(1)=x4b(i-20000);
elseif i == 20001
    clear x19
    clear x29
    clear x39
    clear x49
    eval(['load ',fdat,'b'])
    n1(1)=x1b(1);

```

```

        n2(1)=x2b(1);
        n3(1)=x3b(1);
        s(1)=x4b(1);
elseif i > 18001
    n1(1)=x1a(i-18000);
    n2(1)=x2a(i-18000);
    n3(1)=x3a(i-18000);
    s(1)=x4a(i-18000);
elseif i == 18001
    clear x18
    clear x28
    clear x38
    clear x48
    eval(['load ',fdat,'a'])
    n1(1)=x1a(1);
    n2(1)=x2a(1);
    n3(1)=x3a(1);
    s(1)=x4a(1);
elseif i > 16001
    n1(1)=x19(i-16000);
    n2(1)=x29(i-16000);
    n3(1)=x39(i-16000);
    s(1)=x49(i-16000);
elseif i == 16001
    clear x17
    clear x27
    clear x37
    clear x47
    eval(['load ',fdat,'9'])
    n1(1)=x19(1);
    n2(1)=x29(1);
    n3(1)=x39(1);
    s(1)=x49(1);
elseif i > 14001
    n1(1)=x18(i-14000);
    n2(1)=x28(i-14000);
    n3(1)=x38(i-14000);
    s(1)=x48(i-14000);
elseif i == 14001
    clear x16
    clear x26
    clear x36
    clear x46
    eval(['load ',fdat,'8'])
    n1(1)=x18(1);
    n2(1)=x28(1);
    n3(1)=x38(1);
    s(1)=x48(1);
elseif i > 12001
    n1(1)=x17(i-12000);
    n2(1)=x27(i-12000);
    n3(1)=x37(i-12000);

```

```

        s(1)=x47(i-12000);
elseif i == 12001
    clear x15
    clear x25
    clear x35
    clear x45
    eval(['load ',fdat,'7'])
    n1(1)=x17(1);
    n2(1)=x27(1);
    n3(1)=x37(1);
    s(1)=x47(1);
elseif i > 10001
    n1(1)=x16(i-10000);
    n2(1)=x26(i-10000);
    n3(1)=x36(i-10000);
    s(1)=x46(i-10000);
elseif i == 10001
    clear x14
    clear x24
    clear x34
    clear x44
    eval(['load ',fdat,'6'])
    n1(1)=x16(1);
    n2(1)=x26(1);
    n3(1)=x36(1);
    s(1)=x46(1);
elseif i > 8001
    n1(1)=x15(i-8000);
    n2(1)=x25(i-8000);
    n3(1)=x35(i-8000);
    s(1)=x45(i-8000);
elseif i == 8001
    clear x13
    clear x23
    clear x33
    clear x43
    eval(['load ',fdat,'5'])
    n1(1)=x15(1);
    n2(1)=x25(1);
    n3(1)=x35(1);
    s(1)=x45(1);
elseif i > 6001
    n1(1)=x14(i-6000);
    n2(1)=x24(i-6000);
    n3(1)=x34(i-6000);
    s(1)=x44(i-6000);
elseif i == 6001
    clear x12
    clear x22
    clear x32
    clear x42
    eval(['load ',fdat,'4'])

```

```

        n1(1)=x14(1);
        n2(1)=x24(1);
        n3(1)=x34(1);
        s(1)=x44(1);
    elseif i > 4001
        n1(1)=x13(i-4000);
        n2(1)=x23(i-4000);
        n3(1)=x33(i-4000);
        s(1)=x43(i-4000);
    elseif i == 4001
        clear x11
        clear x21
        clear x31
        clear x41
        eval(['load ',fdat,'3'])
        n1(1)=x13(1);
        n2(1)=x23(1);
        n3(1)=x33(1);
        s(1)=x43(1);
    elseif i > 2001
        n1(1)=x12(i-2000);
        n2(1)=x22(i-2000);
        n3(1)=x32(i-2000);
        s(1)=x42(i-2000);
    elseif i == 2001
        eval(['load ',fdat,'2'])
        n1(1)=x12(1);
        n2(1)=x22(1);
        n3(1)=x32(1);
        s(1)=x42(1);
    elseif i > 1
        n1(1)=x11(i);
        n2(1)=x21(i);
        n3(1)=x31(i);
        s(1)=x41(i);
    elseif i == 1
        eval(['load ',fdat,'1'])
        n1(1)=x11(1);
        n2(1)=x21(1);
        n3(1)=x31(1);
        s(1)=x41(1);
    end
    s1 = inp(fb1,s(1:129));
    s2 = inp(fb2,s(1:129));
    s3 = inp(fb3,s(1:129));
    n11(1) = inp(fb1,n1);
    n21(1) = inp(fb1,n2);
    n31(1) = inp(fb1,n3);
    n12(1) = inp(fb2,n1);
    n22(1) = inp(fb2,n2);
    n32(1) = inp(fb2,n3);
    n13(1) = inp(fb3,n1);

```

```

        n23(1) = inp(fb3,n2);
        n33(1) = inp(fb3,n3);
    % calculate outputs of filters
        N1 = [n11,n21,n31];
        N2 = [n12,n22,n32];
        N3 = [n13,n23,n33];
        Y1 = shftr(Y1);
        Y2 = shftr(Y2);
        Y3 = shftr(Y3);
        dindex=fix((i-1)/2000)+1;
        ij=i-2000*(dindex-1);
        dinda=int2str(dindex);
        if inp(s(1:50),s(1:50)) < thres
            eval(['d',dinda,'(2,ij)=1;']);
            % adapt filters
            [e1,Y1(1),w1,B1]=lmsfir(N1,s1,w1,a,b);
            [e2,Y2(1),w2,B2]=lmsfir(N2,s2,w2,a,b);
            [e3,Y3(1),w3,B3]=lmsfir(N3,s3,w3,a,b);
        else
            eval(['d',dinda,'(2,ij)=0;']);
            % Freeze filters
            Y1(1)= inp(N1,w1);
            Y2(1)= inp(N2,w2);
            Y3(1)= inp(N3,w3);
        end
    % calculate actual filter output
        Z1 = inp(fb1,Y1);
        Z2 = inp(fb2,Y2);
        Z3 = inp(fb3,Y3);
    % account for filtering delays and non-causal delays
    % note that first points of d(1:T+fdel) are garbage
    % filter and form output
        eval(['d',dinda,'(1,ij)=s(129)-Z1-Z2-Z3;']);
    % end of for loop
end
end
[Pxx,Pdd]=plotter(x4f((2000-511-fdel):(2000-fdel)),d15(1:(2000-511):2000),128,64,fmat,title);
clear x1e
clear x2e
clear x3e
clear x4e
clear x1f
clear x2f
clear x3f
clear x4f
eval(['save ',fmat])
% clean up workspace
clear

```

B.6. STRUCT2 - Cascade Canceler

This routine implements the cascade noise canceler shown in Figure 4.4.

Additional versions of the routine have been omitted. These are STRUCT2A,

STRUCT2B, AND STRUCT2C that implement the cascade canceler with different input orders or use sequential adaption.

```
function zz=struct2(fdat,fmat,fmet,firl,a,b,T,title)
% M file for structure 2
% multi error feedback series structure
%
% add dat file to use on next line
eval(['load ',fdat])
% X4 = primary
% X3 = floar
% X2 = glove
% X1 = engine
m = length(x4);
% Input a,b constants for Beta limits
a1= a ;
a2= a ;
a3= a ;
b1= b ;
b2= b ;
b3= b ;
% Delay factor on primary mic (+ve = delay -ve= advance)
% initialize arrays for filter
n1 = zeros(1,firl);
n2 = zeros(1,firl);
n3 = zeros(1,firl);
w11 = zeros(1,firl);
w21 = zeros(1,firl);
w31 = zeros(1,firl);
e1 = zeros(1,m);
e2 = zeros(1,m);
e3 = zeros(1,m);
% Only fill delay lines to length required by firl
% ie keep vectors to firl length even if delay is greater
Tmin=min([firl,T]);
for ii=1:Tmin
% prime delay lines for data on reference mics
    n1(Tmin+1-ii) = x1(ii);
    n2(Tmin+1-ii) = x2(ii);
    n3(Tmin+1-ii) = x3(ii);
```

```

end
for i=1+T:m
% shift arrays for time delay
    n1 = shftr(n1);
    n2 = shftr(n2);
    n3 = shftr(n3);
% get next sample
    s = x4(i-T);
% assign each microphone to noise input on filter and get next sample
% series order is n1,n2,n3
    n1(1) = x1(i);
    n2(1) = x2(i);
    n3(1) = x3(i);
% calculate each fir filter ( output, adapt beta etc.)
    w13=w12;
    w12=w11;
    w23=w22;
    w22=w21;
    w33=w32;
    w32=w31;
    [e1(i),Y1,w11,B1]=lmsfir(n1,s,w11,a1,b1);
    [e2(i),Y2,w21,B2]=lmsfir(n2,e1(n),w21,a2,b2);
    [e3(i),Y3,w31,B3]=lmsfir(n3,e2(n),w31,a3,b3);
% end of for loop
end
% save output of filter
d = e3;
[Pxx,Pdd]=plotter(x4(m-511-T:m-T),d(m-511:m),128,64,fmat,title);
% input filename to save data (add/remove % to enable/disable)
clear x1
clear x2
clear x3
clear x4
eval(['save ',fmat])
% clean up workspace
clear

```


B.7. STRUCT4 - Two Stage Multireference - One Tap per Reference in the Second Stage

This routine implements the two stage beamforming canceler shown in Figure 4.3 with the second stage filters consisting of only a single tap for each input. Additional versions of this file that have been omitted are STRUCT4A and STRUCT4B that use different methods for the calculation of β and STRUCT5 that uses a sparse FIR filter in the second stage.

```
function zz=struct4(fdat,fmat,fmet,firl,a,b,T,title)
% M file for structure 4
% multi error double feedback parallel structure
% add dat file to use on next line
eval(['load ',fdat])
% X4 = primary
% X3 = floar
% X2 = glove
% X1 = engine
m = length(x4);
% Input a,b constants for Beta limits
a1= a ;
a2= a ;
a3= a ;
ay= a ;
b1= b ;
b2= b ;
b3= b ;
by= 100000 ;
% initialize arrays for filter
n1 = zeros(1,firl);
n2 = zeros(1,firl);
n3 = zeros(1,firl);
w1i = zeros(1,firl);
w2i = zeros(1,firl);
w3i = zeros(1,firl);
wyl = zeros(1,3);
e1 = zeros(1,m);
e2 = zeros(1,m);
e3 = zeros(1,m);
ey = zeros(1,m);
% Only fill delay lines to length required by firl
% ie keep vectors to firl length even if delay is greater
```

```

Tmin=min([firl,T]);
for ii=1:Tmin
% prime delay lines for data on reference mics
    n1(Tmin+1-ii) = x1(ii);
    n2(Tmin+1-ii) = x2(ii);
    n3(Tmin+1-ii) = x3(ii);
end
for i=T+1:m
% shift arrays for time delay
    n1 = shftr(n1);
    n2 = shftr(n2);
    n3 = shftr(n3);
% get next sample
    s = x4(i-T);
% assign each microphone to noise input on filter and get next sample
% series order is n1,n2,n3
    n1(1) = x1(i);
    n2(1) = x2(i);
    n3(1) = x3(i);
% calculate each fir filter ( output, adapt beta etc.)
    w13=w12;
    w12=w11;
    w23=w22;
    w22=w21;
    w33=w32;
    w32=w31;
    [e1(i),Y(1),w11,B1]=lmsfir(n1,s,w11,a1,b1);
    [e2(i),Y(2),w21,B2]=lmsfir(n2,s,w21,a2,b2);
    [e3(i),Y(3),w31,B3]=lmsfir(n3,s,w31,a3,b3);
% end of for loop
% find output of filter
    wy10=wy9;
    wy9=wy8;
    wy8=wy7;
    wy7=wy6;
    wy6=wy5;
    wy5=wy4;
    wy4=wy3;
    wy3=wy2;
    wy2=wy1;
    [ey(i),Z,wy1,By]=lmsfir(Y,s,wy1,ay,by);
end
% save filter output
d = ey;
[Pxx,Pdd]=plotter(x4((m-T-511):(m-T)),d((m-511):m),128,64,fmet,title);
clear x1
clear x2
clear x3
clear x4
eval(['save ',fmat])
% clean up workspace
clear

```

B.8. STRUCT6 - Full Beamformer Second Stage Version of STRUCT4

This routine implements the two stage beamforming canceler shown in Figure 4.4 with 25 taps on each of the input in the second stage cancelers.

```

function zz=struct6(fdat,fmat,fmet,firl,a,b,T,title)
% M file for structure 6
% multi error double feedback parallel structure
% structure 4 with extra history used for second stage
%
% add dat file to use on next line
eval(['load ',fdat])
% X4 = primary
% X3 = floar
% X2 = glove
% X1 = engine
m = length(x4);
% Input a,b constants for Beta limits
a1= a ;
a2= a ;
a3= a ;
ay= a ;
b1= b ;
b2= b ;
b3= b ;
by= 100000 ;
% initialize arrays for filter
n1 = zeros(1,firl);
n2 = zeros(1,firl);
n3 = zeros(1,firl);
w11 = zeros(1,firl);
w21 = zeros(1,firl);
w31 = zeros(1,firl);
wy = zeros(1,75);
e1 = zeros(1,m);
e2 = zeros(1,m);
e3 = zeros(1,m);
ey = zeros(1,m);
Y = zeros(3,25);
% Only fill delay lines to length required by firl
% ie keep vectors to firl length even if delay is greater
Tmin=min([firl,T]);
for ii=1:Tmin
% prime delay lines for data on reference mics
    n1(Tmin+1-ii) = x1(ii);
    n2(Tmin+1-ii) = x2(ii);

```

```

        n3(Tmin+1-ii) = x3(ii);
    end
    for i=T+1:m
        % shift arrays for time delay
        n1 = shftr(n1);
        n2 = shftr(n2);
        n3 = shftr(n3);
        % get next sample
        s = x4(i-T);
        % assign each microphone to noise input on filter and get next sample
        % series order is n1,n2,n3
        n1(1) = x1(i);
        n2(1) = x2(i);
        n3(1) = x3(i);
        % calculate each fir filter ( output, adapt beta etc.)
        Y(1,:) = shftr(Y(1,:));
        Y(2,:) = shftr(Y(2,:));
        Y(3,:) = shftr(Y(3,:));
        [e1(i),Y(1,1),w11,B1]=lmsfir(n1,s,w11,a1,b1);
        [e2(i),Y(2,1),w21,B2]=lmsfir(n2,s,w21,a2,b2);
        [e3(i),Y(3,1),w31,B3]=lmsfir(n3,s,w31,a3,b3);
        % end of for loop
        % find output of filter
        Yy = [Y(1,:),Y(2,:),Y(3,:)];
        [ey(i),Z,wy,By]=lmsfir(Yy,s,wy,ay,by);
    end
    % save filter output
    d = ey;
    [Pxx,Pdd]=plotter(x4((m-T-511):(m-T)),d((m-511):m),128,64,fmet,title);
    clear x1
    clear x2
    clear x3
    clear x4
    eval(['save ',fmat])
    % clean up workspace
    clear

```

B.9. STRUCT63 - Three Subband STRUCT6 with Non-Subbanded Beamformer

This canceler implements the three band version of the multiple reference two stage canceler that recombines the references before the second stage as shown in Figure 4.34. The two subband version has been omitted.

```
function zz=struct63(fdat,fmat,fmet,firl,a,b,T,title)
% M file for structure 6
% multi error double feedback parallel structure
% structure 4 with extra history used for second stage
% subband adaption (3 bands)
%
% add dat file to use on next line
eval(['load ',fdat])
load filt3
% X4 = primary
% X3 = floar
% X2 = glove
% X1 = engine
m = length(x4);
fdel = 128;
% Input a,b constants for Beta limits
ay= a ;
by= 100000 ;
% initialize arrays for filter
n11 = zeros(1,firl);
n21 = zeros(1,firl);
n31 = zeros(1,firl);
w11 = zeros(1,firl);
w21 = zeros(1,firl);
w31 = zeros(1,firl);
n12 = zeros(1,firl);
n22 = zeros(1,firl);
n32 = zeros(1,firl);
w12 = zeros(1,firl);
w22 = zeros(1,firl);
w32 = zeros(1,firl);
n13 = zeros(1,firl);
n23 = zeros(1,firl);
n33 = zeros(1,firl);
w13 = zeros(1,firl);
w23 = zeros(1,firl);
w33 = zeros(1,firl);
```

```

wy = zeros(1,75);
Y11 = zeros(1,129);
Y21 = zeros(1,129);
Y31 = zeros(1,129);
Y12 = zeros(1,129);
Y22 = zeros(1,129);
Y32 = zeros(1,129);
Y13 = zeros(1,129);
Y23 = zeros(1,129);
Y33 = zeros(1,129);
d = zeros(1,m);
Y = zeros(3,25);
% Only fill delay lines to length required by fir1
% ie keep vectors to fir1 length even if delay is greater
Tmin=min([fir1,T]);
for ii=T+1-Tmin:T
% prime delay lines for data on reference mics
    n11(T+1-ii) = x1b1(ii);
    n21(T+1-ii) = x2b1(ii);
    n31(T+1-ii) = x3b1(ii);
    n12(T+1-ii) = x1b2(ii);
    n22(T+1-ii) = x2b2(ii);
    n32(T+1-ii) = x3b2(ii);
    n13(T+1-ii) = x1b3(ii);
    n23(T+1-ii) = x2b3(ii);
    n33(T+1-ii) = x3b3(ii);
end
for i=T+1:m
% shift arrays for time delay
    n11 = shftr(n11);
    n21 = shftr(n21);
    n31 = shftr(n31);
    n12 = shftr(n12);
    n22 = shftr(n22);
    n32 = shftr(n32);
    n13 = shftr(n13);
    n23 = shftr(n23);
    n33 = shftr(n33);
% get next sample
    s1 = x4b1(i-T);
    s2 = x4b2(i-T);
    s3 = x4b3(i-T);
% assign each microphone to noise input on filter and get next sample
% series order is n1,n2,n3
    n11(1) = x1b1(i);
    n21(1) = x2b1(i);
    n31(1) = x3b1(i);
    n12(1) = x1b2(i);
    n22(1) = x2b2(i);
    n32(1) = x3b2(i);
    n13(1) = x1b3(i);
    n23(1) = x2b3(i);

```

```

        n33(1) = x3b3(i);
% calculate each fir filter ( output, adapt beta etc.)
        Y11 = shftr(Y11);
        Y21 = shftr(Y21);
        Y31 = shftr(Y31);
        Y12 = shftr(Y12);
        Y22 = shftr(Y22);
        Y32 = shftr(Y32);
        Y13 = shftr(Y13);
        Y23 = shftr(Y23);
        Y33 = shftr(Y33);
        [e11,Y11(1),w11,B11]=lmsfir(n11,s1,w11,a,b);
        [e21,Y21(1),w21,B21]=lmsfir(n21,s1,w21,a,b);
        [e31,Y31(1),w31,B31]=lmsfir(n31,s1,w31,a,b);
        [e12,Y12(1),w12,B12]=lmsfir(n12,s2,w12,a,b);
        [e22,Y22(1),w22,B22]=lmsfir(n22,s2,w22,a,b);
        [e32,Y32(1),w32,B32]=lmsfir(n32,s2,w32,a,b);
        [e13,Y13(1),w13,B13]=lmsfir(n13,s3,w13,a,b);
        [e23,Y23(1),w23,B23]=lmsfir(n23,s3,w23,a,b);
        [e33,Y33(1),w33,B33]=lmsfir(n33,s3,w33,a,b);
% end of for loop
% find output of filter
        Y(1,:) = shftr(Y(1,:));
        Y(2,:) = shftr(Y(2,:));
        Y(3,:) = shftr(Y(3,:));
% add filtering here if needed
        Y(1,1) = inp(fb1,Y11) + inp(fb2,Y12) + inp(fb3,Y13);
        Y(2,1) = inp(fb1,Y21) + inp(fb2,Y22) + inp(fb3,Y23);
        Y(3,1) = inp(fb1,Y31) + inp(fb2,Y32) + inp(fb3,Y33);
        Yy = [Y(1,:),Y(2,:),Y(3,:)];
        II = i-T-fdel;
        if II > 0
                [d(i),Z,wy,By]=lmsfir(Yy,x4(II),wy,ay,by);
        end
end
[Pxx,Pdd]=plotter(x4((m-T-511-fdel):(m-T-fdel)),d((m-511):m),128,64,fmet,title);
clear x1b1
clear x2b1
clear x3b1
clear x4b1
clear x1b2
clear x2b2
clear x3b2
clear x4b2
clear x1b3
clear x2b3
clear x3b3
clear x4b3
clear x4
eval(['save ',fmat])
% clean up workspace
clear

```

B.10. STRUCT73 - Three Band STRUCT6 with Subbanded Beamformer

This canceler implements the three band version of the multiple reference two stage canceler that uses a subbanded second stage as shown in Figure 4.35. The two subband version of the canceler has been omitted.

```
function zz=struct73(fdat,fmat,fmet,firl,a,b,T,title)
% M file for structure 7
% multi error double feedback parallel structure
% structure 4 with extra history used for second stage
% subband adaption (3 bands)
%
% add dat file to use on next line
eval(['load ',fdat])
load filt3
% X4 = primary
% X3 = floar
% X2 = glove
% X1 = engine
m = length(x4);
fdel = 128;
% Input a,b constants for Beta limits
ay= a ;
by= 100000 ;
% initialize arrays for filter
n11 = zeros(1,firl);
n21 = zeros(1,firl);
n31 = zeros(1,firl);
w11 = zeros(1,firl);
w21 = zeros(1,firl);
w31 = zeros(1,firl);
n12 = zeros(1,firl);
n22 = zeros(1,firl);
n32 = zeros(1,firl);
w12 = zeros(1,firl);
w22 = zeros(1,firl);
w32 = zeros(1,firl);
n13 = zeros(1,firl);
n23 = zeros(1,firl);
n33 = zeros(1,firl);
w13 = zeros(1,firl);
w23 = zeros(1,firl);
w33 = zeros(1,firl);
```



```

wy1 = zeros(1,75);
wy2 = zeros(1,75);
wy3 = zeros(1,75);
Y1 = zeros(3,25);
Y2 = zeros(3,25);
Y3 = zeros(3,25);
Z1 = zeros(1,129);
Z2 = zeros(1,129);
Z3 = zeros(1,129);
d = zeros(1,m);
% Only fill delay lines to length required by fir1
% ie keep vectors to fir1 length even if delay is greater
Tmin=min([fir1,T]);
for ii=T+1-Tmin:T
    % prime delay lines for data on reference mics
        n11(T+1-ii) = x1b1(ii);
        n21(T+1-ii) = x2b1(ii);
        n31(T+1-ii) = x3b1(ii);
        n12(T+1-ii) = x1b2(ii);
        n22(T+1-ii) = x2b2(ii);
        n32(T+1-ii) = x3b2(ii);
        n13(T+1-ii) = x1b3(ii);
        n23(T+1-ii) = x2b3(ii);
        n33(T+1-ii) = x3b3(ii);
    end
    for i=T+1:m
        % shift arrays for time delay
            n11 = shftr(n11);
            n21 = shftr(n21);
            n31 = shftr(n31);
            n12 = shftr(n12);
            n22 = shftr(n22);
            n32 = shftr(n32);
            n13 = shftr(n13);
            n23 = shftr(n23);
            n33 = shftr(n33);
        % get next sample
            s1 = x4b1(i-T);
            s2 = x4b2(i-T);
            s3 = x4b3(i-T);
        % assign each microphone to noise input on filter and get next sample
        % series order is n1,n2,n3
            n11(1) = x1b1(i);
            n21(1) = x2b1(i);
            n31(1) = x3b1(i);
            n12(1) = x1b2(i);
            n22(1) = x2b2(i);
            n32(1) = x3b2(i);
            n13(1) = x1b3(i);
            n23(1) = x2b3(i);
            n33(1) = x3b3(i);
        % calculate each fir filter ( output, adapt beta etc.)

```

```

Y1(1,:) = shftr(Y1(1,:));
Y1(2,:) = shftr(Y1(2,:));
Y1(3,:) = shftr(Y1(3,:));
Y2(1,:) = shftr(Y2(1,:));
Y2(2,:) = shftr(Y2(2,:));
Y2(3,:) = shftr(Y2(3,:));
Y3(1,:) = shftr(Y3(1,:));
Y3(2,:) = shftr(Y3(2,:));
Y3(3,:) = shftr(Y3(3,:));
[e11,Y1(1,1),w11,B11]=lmsfir(n11,s1,w11,a,b);
[e21,Y1(2,1),w21,B21]=lmsfir(n21,s1,w21,a,b);
[e31,Y1(3,1),w31,B31]=lmsfir(n31,s1,w31,a,b);
[e12,Y2(1,1),w12,B12]=lmsfir(n12,s2,w12,a,b);
[e22,Y2(2,1),w22,B22]=lmsfir(n22,s2,w22,a,b);
[e32,Y2(3,1),w32,B32]=lmsfir(n32,s2,w32,a,b);
[e13,Y3(1,1),w13,B13]=lmsfir(n13,s3,w13,a,b);
[e23,Y3(2,1),w23,B23]=lmsfir(n23,s3,w23,a,b);
[e33,Y3(3,1),w33,B33]=lmsfir(n33,s3,w33,a,b);
% end of for loop
% find output of filter
% add filtering here if needed
Yy1 = [Y1(1,:),Y1(2,:),Y1(3,:)];
Yy2 = [Y2(1,:),Y2(2,:),Y2(3,:)];
Yy3 = [Y3(1,:),Y3(2,:),Y3(3,:)];
Z1 = shftr(Z1);
Z2 = shftr(Z2);
Z3 = shftr(Z3);
[ey1,Z1(1),wy1,By]=lmsfir(Yy1,s1,wy1,ay,by);
[ey2,Z2(1),wy2,By]=lmsfir(Yy2,s2,wy2,ay,by);
[ey3,Z3(1),wy3,By]=lmsfir(Yy3,s3,wy3,ay,by);
II = max(1,(i-T-fdel));
d(i) = x4(II) - inp(fb1,Z1) - inp(fb2,Z2) - inp(fb3,Z3);
end
[Pxx,Pdd]=plotter(x4((m-T-511-fdel):(m-T-fdel)),d((m-511):m),128,64,fmet,title);
clear x1b1
clear x2b1
clear x3b1
clear x4b1
clear x1b2
clear x2b2
clear x3b2
clear x4b2
clear x1b3
clear x2b3
clear x3b3
clear x4b3
clear x4
eval(['save ',fmat])
% clean up workspace
clear

```

B.11. S73DLONG - Speech Detector Added to STRUCT73 - Long File

This routine implements the two stage beamforming canceler with three subbands and a subbanded second stage as shown in Figure 4.35 with the addition of support for long simulations and a speech detector.

```
function zz=s73dlong(fdat,fmat,firl,a,b,sdlevel,title)
% M file for structure 7
% multi error double feedback parallel structure
% structure 4 with extra history used for second stage
% subband adaption (3 bands)
%
% add dat file to use on next line
load firl3
% Speech detect power threshold and detect length
dlength = 100;
thres = dlength * sdlevel;
% X4 = primary
% X3 = floar
% X2 = glove
% X1 = engine
fdel = 128;
% Input a,b constants for Beta limits
ay= a ;
by= 100000 ;
% initialize arrays for filter
n1 = zeros(1,129);
n2 = zeros(1,129);
n3 = zeros(1,129);
s = zeros(1,max(129,dlength));
n11 = zeros(1,firl);
n21 = zeros(1,firl);
n31 = zeros(1,firl);
w11 = zeros(1,firl);
w21 = zeros(1,firl);
w31 = zeros(1,firl);
n12 = zeros(1,firl);
n22 = zeros(1,firl);
n32 = zeros(1,firl);
w12 = zeros(1,firl);
w22 = zeros(1,firl);
w32 = zeros(1,firl);
n13 = zeros(1,firl);
n23 = zeros(1,firl);
n33 = zeros(1,firl);
```

```

w13 = zeros(1,fir1);
w23 = zeros(1,fir1);
w33 = zeros(1,fir1);
wy1 = zeros(1,75);
wy2 = zeros(1,75);
wy3 = zeros(1,75);
Y1 = zeros(3,25);
Y2 = zeros(3,25);
Y3 = zeros(3,25);
Z1 = zeros(1,129);
Z2 = zeros(1,129);
Z3 = zeros(1,129);
d1 = zeros(2,2000);
d2 = zeros(2,2000);
d3 = zeros(2,2000);
d4 = zeros(2,2000);
d5 = zeros(2,2000);
d6 = zeros(2,2000);
d7 = zeros(2,2000);
d8 = zeros(2,2000);
d9 = zeros(2,2000);
d10 = zeros(2,2000);
d11 = zeros(2,2000);
d12 = zeros(2,2000);
d13 = zeros(2,2000);
d14 = zeros(2,2000);
d15 = zeros(2,2000);
for i1 = 0:14
for i2= 1:2000
i=i1*2000+i2;
% shift arrays for time delay
    s = shftr(s);
    n1 = shftr(n1);
    n2 = shftr(n2);
    n3 = shftr(n3);
    n11 = shftr(n11);
    n21 = shftr(n21);
    n31 = shftr(n31);
    n12 = shftr(n12);
    n22 = shftr(n22);
    n32 = shftr(n32);
    n13 = shftr(n13);
    n23 = shftr(n23);
    n33 = shftr(n33);
% get next sample
% assign each microphone to noise input on filter and get next sample
    if i > 28001
        n1(1)=x1f(i-28000);
        n2(1)=x2f(i-28000);
        n3(1)=x3f(i-28000);
        s(1)=x4f(i-28000);
    elseif i == 28001

```

```

clear x1d
clear x2d
clear x3d
clear x4d
eval(['load ',fdat,'f'])
n1(1)=x1f(1);
n2(1)=x2f(1);
n3(1)=x3f(1);
s(1)=x4f(1);
elseif i > 26001
n1(1)=x1e(i-26000);
n2(1)=x2e(i-26000);
n3(1)=x3e(i-26000);
s(1)=x4e(i-26000);
elseif i == 26001
clear x1c
clear x2c
clear x3c
clear x4c
eval(['load ',fdat,'e'])
n1(1)=x1e(1);
n2(1)=x2e(1);
n3(1)=x3e(1);
s(1)=x4e(1);
elseif i > 24001
n1(1)=x1d(i-24000);
n2(1)=x2d(i-24000);
n3(1)=x3d(i-24000);
s(1)=x4d(i-24000);
elseif i == 24001
clear x1b
clear x2b
clear x3b
clear x4b
eval(['load ',fdat,'d'])
n1(1)=x1d(1);
n2(1)=x2d(1);
n3(1)=x3d(1);
s(1)=x4d(1);
elseif i > 22001
n1(1)=x1c(i-22000);
n2(1)=x2c(i-22000);
n3(1)=x3c(i-22000);
s(1)=x4c(i-22000);
elseif i == 22001
clear x1a
clear x2a
clear x3a
clear x4a
eval(['load ',fdat,'c'])
n1(1)=x1c(1);
n2(1)=x2c(1);

```

```

        n3(1)=x3c(1);
        s(1)=x4c(1);
elseif i > 20001
    n1(1)=x1b(i-20000);
    n2(1)=x2b(i-20000);
    n3(1)=x3b(i-20000);
    s(1)=x4b(i-20000);
elseif i == 20001
    clear x19
    clear x29
    clear x39
    clear x49
    eval(['load ',fdat,'b'])
    n1(1)=x1b(1);
    n2(1)=x2b(1);
    n3(1)=x3b(1);
    s(1)=x4b(1);
elseif i > 18001
    n1(1)=x1a(i-18000);
    n2(1)=x2a(i-18000);
    n3(1)=x3a(i-18000);
    s(1)=x4a(i-18000);
elseif i == 18001
    clear x18
    clear x28
    clear x38
    clear x48
    eval(['load ',fdat,'a'])
    n1(1)=x1a(1);
    n2(1)=x2a(1);
    n3(1)=x3a(1);
    s(1)=x4a(1);
elseif i > 16001
    n1(1)=x19(i-16000);
    n2(1)=x29(i-16000);
    n3(1)=x39(i-16000);
    s(1)=x49(i-16000);
elseif i == 16001
    clear x17
    clear x27
    clear x37
    clear x47
    eval(['load ',fdat,'9'])
    n1(1)=x19(1);
    n2(1)=x29(1);
    n3(1)=x39(1);
    s(1)=x49(1);
elseif i > 14001
    n1(1)=x18(i-14000);
    n2(1)=x28(i-14000);
    n3(1)=x38(i-14000);
    s(1)=x48(i-14000);

```

```

elseif i == 14001
    clear x16
    clear x26
    clear x36
    clear x46
    eval(['load ',fdat,'8'])
    n1(1)=x18(1);
    n2(1)=x28(1);
    n3(1)=x38(1);
    s(1)=x48(1);
elseif i > 12001
    n1(1)=x17(i-12000);
    n2(1)=x27(i-12000);
    n3(1)=x37(i-12000);
    s(1)=x47(i-12000);
elseif i == 12001
    clear x15
    clear x25
    clear x35
    clear x45
    eval(['load ',fdat,'7'])
    n1(1)=x17(1);
    n2(1)=x27(1);
    n3(1)=x37(1);
    s(1)=x47(1);
elseif i > 10001
    n1(1)=x16(i-10000);
    n2(1)=x26(i-10000);
    n3(1)=x36(i-10000);
    s(1)=x46(i-10000);
elseif i == 10001
    clear x14
    clear x24
    clear x34
    clear x44
    eval(['load ',fdat,'6'])
    n1(1)=x16(1);
    n2(1)=x26(1);
    n3(1)=x36(1);
    s(1)=x46(1);
elseif i > 8001
    n1(1)=x15(i-8000);
    n2(1)=x25(i-8000);
    n3(1)=x35(i-8000);
    s(1)=x45(i-8000);
elseif i == 8001
    clear x13
    clear x23
    clear x33
    clear x43
    eval(['load ',fdat,'5'])
    n1(1)=x15(1);

```

```

        n2(1)=x25(1);
        n3(1)=x35(1);
        s(1)=x45(1);
elseif i > 6001
    n1(1)=x14(i-6000);
    n2(1)=x24(i-6000);
    n3(1)=x34(i-6000);
    s(1)=x44(i-6000);
elseif i == 6001
    clear x12
    clear x22
    clear x32
    clear x42
    eval(['load ',fdat,'4'])
    n1(1)=x14(1);
    n2(1)=x24(1);
    n3(1)=x34(1);
    s(1)=x44(1);
elseif i > 4001
    n1(1)=x13(i-4000);
    n2(1)=x23(i-4000);
    n3(1)=x33(i-4000);
    s(1)=x43(i-4000);
elseif i == 4001
    clear x11
    clear x21
    clear x31
    clear x41
    eval(['load ',fdat,'3'])
    n1(1)=x13(1);
    n2(1)=x23(1);
    n3(1)=x33(1);
    s(1)=x43(1);
elseif i > 2001
    n1(1)=x12(i-2000);
    n2(1)=x22(i-2000);
    n3(1)=x32(i-2000);
    s(1)=x42(i-2000);
elseif i == 2001
    eval(['load ',fdat,'2'])
    n1(1)=x12(1);
    n2(1)=x22(1);
    n3(1)=x32(1);
    s(1)=x42(1);
elseif i > 1
    n1(1)=x11(i);
    n2(1)=x21(i);
    n3(1)=x31(i);
    s(1)=x41(i);
elseif i == 1
    eval(['load ',fdat,'1'])
    n1(1)=x11(1);

```



```

        n2(1)=x21(1);
        n3(1)=x31(1);
        s(1)=x41(1);
    end
    s1 = inp(fb1,s(1:129));
    s2 = inp(fb2,s(1:129));
    s3 = inp(fb3,s(1:129));
    n11(1) = inp(fb1,n1);
    n21(1) = inp(fb1,n2);
    n31(1) = inp(fb1,n3);
    n12(1) = inp(fb2,n1);
    n22(1) = inp(fb2,n2);
    n32(1) = inp(fb2,n3);
    n13(1) = inp(fb3,n1);
    n23(1) = inp(fb3,n2);
    n33(1) = inp(fb3,n3);
    % calculate each fir filter ( output, adapt beta etc.)
    Y1(1,:) = shftr(Y1(1,:));
    Y1(2,:) = shftr(Y1(2,:));
    Y1(3,:) = shftr(Y1(3,:));
    Y2(1,:) = shftr(Y2(1,:));
    Y2(2,:) = shftr(Y2(2,:));
    Y2(3,:) = shftr(Y2(3,:));
    Y3(1,:) = shftr(Y3(1,:));
    Y3(2,:) = shftr(Y3(2,:));
    Y3(3,:) = shftr(Y3(3,:));
    dindex=fix((i-1)/2000)+1;
    ij=i-2000*(dindex-1);
    dinda=int2str(dindex);
    level = inp(s(1:50),s(1:50));
    if level < thres
        eval(['d',dinda,'(2,ij)=1:']);
        % adapt first stage filters
        [e11,Y1(1,1),w11,B11]=lmsfir(n11,s1,w11,a,b);
        [e21,Y1(2,1),w21,B21]=lmsfir(n21,s1,w21,a,b);
        [e31,Y1(3,1),w31,B31]=lmsfir(n31,s1,w31,a,b);
        [e12,Y2(1,1),w12,B12]=lmsfir(n12,s2,w12,a,b);
        [e22,Y2(2,1),w22,B22]=lmsfir(n22,s2,w22,a,b);
        [e32,Y2(3,1),w32,B32]=lmsfir(n32,s2,w32,a,b);
        [e13,Y3(1,1),w13,B13]=lmsfir(n13,s3,w13,a,b);
        [e23,Y3(2,1),w23,B23]=lmsfir(n23,s3,w23,a,b);
        [e33,Y3(3,1),w33,B33]=lmsfir(n33,s3,w33,a,b);
    else
        eval(['d',dinda,'(2,ij)=0:']);
        % freeze adaption
        Y1(1,1) = inp(n11,w11);
        Y1(2,1) = inp(n21,w21);
        Y1(3,1) = inp(n31,w31);
        Y2(1,1) = inp(n12,w12);
        Y2(2,1) = inp(n22,w22);
        Y2(3,1) = inp(n32,w32);
        Y3(1,1) = inp(n13,w13);

```

```

        Y3(2,1) = inp(n23,w23);
        Y3(3,1) = inp(n33,w33);
    end
% find output of filter
% add filtering here if needed
    Yy1 = [Y1(1,:),Y1(2,:),Y1(3,:)];
    Yy2 = [Y2(1,:),Y2(2,:),Y2(3,:)];
    Yy3 = [Y3(1,:),Y3(2,:),Y3(3,:)];
    Z1 = shftr(Z1);
    Z2 = shftr(Z2);
    Z3 = shftr(Z3);
    if level < thres
        % adapt first stage filters
        [ey1,Z1(1),wy1,By]=lmsfir(Yy1,s1,wy1,ay,by);
        [ey2,Z2(1),wy2,By]=lmsfir(Yy2,s2,wy2,ay,by);
        [ey3,Z3(1),wy3,By]=lmsfir(Yy3,s3,wy3,ay,by);
    else
        % freeze adaption
        Z1(1) = inp(Yy1,wy1);
        Z2(1) = inp(Yy2,wy2);
        Z3(1) = inp(Yy3,wy3);
    end
% calculate actual filter output
    ZZ1 = inp(fb1,Z1);
    ZZ2 = inp(fb2,Z2);
    ZZ3 = inp(fb3,Z3);
% account for filtering delays and non-causal delays
% note that first points of d(1:T+fdel) are garbage
% filter and form output
    eval(['d',dinda,'(1,ij)=s(129)-ZZ1-ZZ2-ZZ3;']);
% end of for loop
end
end
[Pxx,Pdd]=plotter(x4f((2000-511-fdel):(2000-fdel)),d15(1,(2000-511):2000),128,64,fmat,title);
clear x1e
clear x2e
clear x3e
clear x4e
clear x1f
clear x2f
clear x3f
clear x4f
eval(['save ',fmat])
% clean up workspace
clear

```

APPENDIX C: SIMULATION DATA SUMMARY

The following simulation summary tables detail the results of all simulations discussed in Chapters 4 and 5.

Table C.1 below shows a cross reference list between the structure number short forms used in the summary tables included in this appendix and the diagrams of the structure in Chapters 4 and 5.

Table C.1 Adaptive Filter Structure Cross Reference List

Structure	Figure	Notes
1	4.2	
12	4.33	Except using only two subbands
13	4.33	
2	4.4	R1 = X1 R2 = X2 R3 = X3 Parallel Apation
2A	4.4	R1 = X3 R2 = X2 R3 = X1 Parallel Adaption
2B	4.4	R1 = X1 R2 = X2 R3 = X3 Sequential adaption of R1,R2,R3 each for 500 iterations each
2C	4.4	R1 = X3 R2 = X2 R3 = X1 Sequential adaption of R1,R2,R3 each for 500 iterations each
3	4.5	
4	4.3	Second stage single tap per reference
4A	4.3	Second stage single tap per reference β calculation using FIR averaging
4B	4.3	Second stage single tap per reference β calculation using IIR averaging
5	4.3	Second stage sparse FIR 5 taps per reference
6	4.3	Second stage 25 taps per reference
62	4.34	Except using only two subbands
63	4.34	
72	4.35	Except using only two subbands
73	4.35	
T	4.29	

C.1. Simulation Results using Generated Random Data

Table C.2 Simulated Random Data Simulation

Simulation parameters

Delay = 0.0

Length = 1500 Iterations

- P = Uncorrelated references
 N = Uncorrelated references with noise added at 30dB SNR
 W = Uncorrelated references with noise added at 5dB SNR
 C = Correlated references
 CW = Correlated references with noise added at 30dB SNR
 CW = Correlated references with noise added at 5dB SNR

Structure	Data	100 taps a = 1.0		250 taps a = 1.0		250taps a = 0.5	
		Max	Min	Max	Min	Max	Min
		dB	dB	dB	dB	dB	dB
1	P	31.0	15.8	25.2	6.2	5.6	18.8
1	N	27.4	16.2	20.2	7.6	18.0	5.8
1	W	2.8	-6.8	5.2	-6.4	4.6	-3.6
1	C	5.4	-3.8	6.0	-3.2	6.4	-1.3
1	CN	21.6	3.2	16.2	4.4	15.0	4.1
1	CW	2.1	-6.9	3.4	-6.8	2.5	-4.0
2	P	0.0	-11.8	2.8	-10.0	10.2	-3.0
2	N	2.8	-10.2	4.5	-10.0	10.2	-4.0
2	W	-2.0	-16.0	-2.5	-13.8	3.2	-8.0
2	C	0.1	-13.2	-1.6	-13.6	3.9	-5.5
2	CN	6.2	-11.2	3.4	-9.5	10.0	0.0
2	CW	-2.8	-16.8	-4.0	-17.0	1.0	-9.2
3	P	9.2	-4.5	13.8	-5.0	23.0	-1.0
3	N	7.0	-5.8	5.0	-5.4	8.5	-3.6
3	W	2.6	-8.2	4.5	-7.0	4.5	-5.8
3	C	8.8	-5.4	9.6	-4.4	9.2	-2.0
3	CN	25.0	2.0	22.5	2.0	23.6	4.2
3	CW	3.8	-7.5	2.2	-8.2	3.6	-7.0
4	P	6.6	-3.3	7.8	-2.0	15.6	0.6
4	N	3.6	-4.9	5.8	-5.4	9.0	-1.4
4	W	0.6	-5.6	4.0	-5.4	4.4	-2.7
4	C	6.2	-4.6	6.4	-3.0	7.8	-2.1
4	CN	16.0	3.0	19.4	2.4	23.2	4.1
4	CW	2.4	-6.2	2.6	-5.3	1.0	-2.1

Table C.3 Additional Simulated Random Data Simulations**Simulation Parameters**

Structure = 1

Data File = P = Uncorrelated references

#Iterations	a	100 taps		100 taps	
		Max dB	Min dB	Max dB	Min dB
1500	0.5	-	-	18.8	5.6
3000	0.5	43.2	18.1	26.6	12.7
1500	1.0	31.0	15.8	25.2	6.2
3000	1.0	48.0	20.0	33.7	17.3
3000	2.0	11.6	-8.5	8.7	-6.8
1500	2.5	-15.2	-34.6	-2.5	-23.0
3000	2.5	-39.0	-65.1	-6.8	-35.3

Note: 100 tap, a=0.5, 1500 Iterations case was not run.
Some entries in this table are included in Table C.1.

C.2. Simulations Results for Simulations using Car Data

The tables included in the completion of this appendix detail the results of simulations that were run using actual noise data recorded in a car. The tables provide summary results in addition to specifying the parameters for each simulation.

For each simulation the input data used is specified. The input data files can be organized into a tree as shown in Figure C.1 where the files in each node of the tree were formed through processing on the previous node in the tree.

All files reside on the 386PC used for the simulations with the exception of the original files recorded by Hebert [3] that reside on the SUN network. Each file contains four vectors x_1 , x_2 , x_3 and x_4 , corresponding to microphones 1, 2, 3 and 4 as described in Chapter 2. Microphone 4 is the primary microphone and all others are considered to be references.

All data files are names with the following format:

Mxxyy

where **xx:** is one or more letters indicating characteristics of data in file as specified by Table C.4.

yy: is a number specifying the particular automobile noise case as specified in Table C.5.

Figure C.1 Automobile Noise File Name Tree Structure Showing Relationship Between Files.

Table C.4 File Name Letter Meaning (xx)

File Name (xx)	Length	Filtering
Sun File	53248	Files as recorded by Hebert [3] adjusted for 0 mean. On SUN network and called MAyy.
nul	1500	These files are 1500 point files from the Sun Files. Also used by Hebert [3].
A	30000	These files were formed by adding speech to the primary signal of the "C" files.(High SNR)
B	30000	These files were formed by adding speech to the x2 reference signal of the "A" files.
C	30000	These files are the first 30000 samples of the Sun Files filtered to the telephony band using the same filter as used to form the "F" files. These files are also segmented into short vectors to be compatible with the limitations of Matlab.
D	18000	These files were formed by adding speech to the primary signal of the "C" files. Only the first 18000 points of the "C" files were used. (Medium SNR)
E	18000	These files were formed by adding speech to the x2 reference signal of the "D" files.
F	4000	These files were formed by filtering the "I" files to obtain telephony bandwidth noise. The filter used is discussed in Chapter 3.
G	18000	These files were formed by adding speech to the primary signal of the "C" files. Only the first 18000 points of the "C" files were used. (Low SNR)
H	18000	These files were formed by adding speech to the x2 reference signal of the "G" files.
I	4000	These files are the first 4000 points of the "L" files with no additional processing.
L	8000	These files are the first 8000 points of the Original Sun Files. No other processing has been done.
S	4000	These files are subbanded versions of the "F" files. The data has been filtered using the filter discussed in Chapter 3 for the 300-1200Hz band.
T	4000	These files are subbanded versions of the "F" files. The data has been filtered using the filter discussed in Chapter 3 for the 1200-3500Hz band.

Table C.4 cont. File Name Letter Meaning (xx)

File Name (xx)	Length	Filtering
U	4000	These files are subbanded versions of the "F" files. The data has been filtered using the filter discussed in Chapter 3 for the 1200-2400Hz band.
V	4000	These files are subbanded versions of the "F" files. The data has been filtered using the filter discussed in Chapter 3 for the 2400-3500Hz band.
W	4000	These files were formed by combining the "S", "U" and "V" files along with the primary microphone from the "F" files. These files contain 13 vectors which are the 12 vectors from "S", "U" and "V" renamed to avoid conflict in addition to the original non-subbanded primary vector. These files were used in subband simulations so that the FIR filtering to form the subbands was not required in the simulations.
X	4000	These files were formed by combining the "S" and "T" files along with the primary microphone from the "F" files. These files contain 9 vectors which are the 8 vectors from "S" and "T" renamed to avoid conflict in addition to the original non-subbanded primary vector. These files were used in subband simulations so that the FIR filtering to form the subbands was not required in the simulations.

Table C.5 Noise Environment Cases for recording

File Name (yy)	Car Speed km/hr	Ventilation Fan	Notes
5	60	OFF	(Engine idling) (Engine idling)
6	60	ON	
7	0	OFF	
8	0	ON	
9	100	OFF	(Engine off)
10	100	ON	
15	0	ON	

Table C.6 Car Noise Data Simulation Run #1**Simulation Parameters**

Delay = 0.0 msec

Noise data = complete band (0-5000Hz)

Simulation Length = 1500 Iterations

Data Based on complete 0 - 5000 Hz Band

S	Data File	100 taps a = 1.0			250 taps a = 1.0			250 taps a = 0.5		
		Min dB	Mean dB	Max dB	Min dB	Mean dB	Max dB	Min dB	Mean dB	Max dB
1	M5	-7.0	-1.3	10.4	-6.6	-1.0	12.1	-2.8	-0.4	7.0
1	M6	-9.1	-1.7	11.4	-9.0	-1.2	17.1	-4.0	-0.5	12.4
1	M7	-10.5	-3.1	13.6	-7.8	-1.9	11.6	-4.2	-0.7	10.2
1	M8	-8.6	-1.8	13.6	-7.7	-1.5	11.2	-3.3	-1.0	8.3
1	M9	-14.3	-3.6	8.2	-16.2	-5.4	5.6	-12.7	-3.1	6.7
1	M10	-10.3	-2.4	9.6	-7.8	-3.0	7.8	-6.0	-1.8	4.6
1	M15	-6.6	-1.0	13.9	-6.7	-0.9	11.3	-2.8	-0.5	8.6
2	M5	-18.7	-5.9	18.6	-19.3	-5.4	16.5	-8.3	-2.3	14.2
2	M6	-22.6	-5.7	25.9	-23.5	-5.7	23.8	-10.2	-3.3	17.9
2	M7	-27.4	-9.0	17.8	-27.0	-6.2	21.0	-15.8	-3.9	18.9
2	M8	-23.8	-6.4	19.6	-23.2	-5.4	24.1	-9.7	-3.2	17.2
2	M9	-24.7	-10.3	6.6	-26.3	-12.5	5.8	-15.4	-6.8	7.2
2	M10	-47.6	-31.0	-6.4	-47.8	-31.1	-6.3	-30.8	-18.7	8.5
2	M15	-23.5	-8.0	-18.0	-23.0	-8.5	20.9	-10.8	-4.7	17.3
3	M5	-16.0	-4.0	9.6	-15.2	-3.0	8.1	-22.2	-6.0	9.2
3	M6	-24.5	-4.6	14.9	-13.7	-2.2	19.5	-8.6	-1.2	15.1
3	M7	-10.0	-4.0	9.4	-9.9	-1.3	9.0	-14.5	-2.2	6.2
3	M8	-19.7	-4.0	13.1	-15.7	-4.3	14.4	-19.9	-4.7	11.3
3	M9	-27.9	-9.1	5.4	-22.0	-8.2	7.4	-18.8	-4.5	5.7
3	M10	-9.7	-2.1	12.1	-7.9	-1.9	11.6	-24.2	-4.5	7.5
3	M15	-14.1	-2.4	11.8	-8.2	-0.3	14.9	-4.9	-0.8	13.4
4	M5	-24.8	-7.2	15.7	-30.1	-9.7	13.8	-18.2	-4.6	13.2
4	M6	-33.5	-9.1	19.0	-34.3	-10.6	19.3	-22.6	-6.0	12.1
4	M7	-21.4	-7.3	21.0	-18.3	-5.4	21.4	-8.5	-0.9	25.5
4	M8	-28.2	-9.1	16.4	-31.6	-8.8	18.3	-13.9	-3.7	14.1
4	M9	-37.3	-12.9	10.9	-34.1	-14.6	7.0	-21.3	-7.1	7.7
4	M10	-35.6	-10.5	18.8	-34.7	-12.2	15.3	-28.2	-10.3	11.7
4	M15	-27.3	-9.3	13.5	-21.0	-5.3	18.4	-11.7	-3.4	15.5

Table C.7 Car Noise Data Simulation Run #2**Simulation Parameters**

a = 1.0

Noise data = complete band (0-5000Hz)

Simulation Length = 4000 Iterations

Data Based on complete 0 - 5000 Hz band

S	Data File	Taps	Delay 0.0 msec			Delay 5.0 msec			Delay 10.0 msec		
			Min dB	Mean dB	Max dB	Min dB	Mean dB	Max dB	Min dB	Mean dB	Max dB
1	MI5	100	-12.3	-2.2	8.9	-10.3	-1.6	10.2	-6.7	-1.3	8.9
1	MI5	250	-8.1	-0.6	10.7	-6.3	-0.7	11.4	-6.4	-0.4	14.7
1	MI6	100	-9.3	-2.1	7.6	-8.5	-2.4	8.8	-6.8	-0.9	11.8
1	MI7	100	-9.1	-2.9	8.1	-6.7	-1.8	7.3	-7.4	-2.9	5.7
1	MI8	100	-7.4	-1.6	8.1	-6.3	-1.8	8.9	-8.0	-1.8	12.7
1	MI9	100	-10.3	-2.3	6.4	-6.5	-1.2	7.8	-8.0	-1.2	13.6
1	MI10	100	-9.4	-2.8	8.3	-6.5	-1.0	16.7	-7.3	-0.8	11.0
1	MI15	100	-6.3	-0.8	16.0	-6.1	-0.7	15.5	-6.3	-0.7	14.7
3	MI5	100	-12.2	-2.2	17.7	-15.3	-2.8	15.4	-21.6	-3.3	19.1
3	MI5	250	-17.0	-4.1	15.2	-18.5	-3.2	16.0	-15.9	-0.9	18.2
3	MI6	100	-16.4	-3.1	13.7	-17.8	-3.8	12.9	-7.9	-0.3	12.6
3	MI7	100	-17.3	-5.4	15.0	-13.3	-2.6	9.6	-10.7	-1.6	12.2
3	MI8	100	-26.2	-5.2	13.8	-28.8	-5.5	12.0	-25.1	-7.1	12.1
3	MI9	100	-21.1	-6.0	12.7	-15.1	-2.3	12.5	-9.8	-1.2	13.6
3	MI10	100	-23.1	-6.7	9.3	-18.8	-2.9	11.4	-24.7	-4.1	13.1
3	MI15	100	-21.7	-4.6	15.7	-12.9	-1.3	18.5	-10.5	-2.6	14.1
2	MI5	100	-26.2	-6.2	20.2	-21.8	-4.2	20.9	-22.8	-4.4	24.3
2	MI5	250	-19.1	-3.4	21.8	-19.7	-3.3	23.8	-20.6	-3.8	25.2
2	MI6	100	-23.5	-6.6	14.6	-19.4	-5.4	18.5	-19.7	-5.3	19.4
2	MI7	100	-25.9	-9.2	26.8	-24.8	-8.3	19.5	-24.0	-6.6	17.5
2	MI8	100	-20.8	-5.6	21.5	-22.0	-5.8	29.1	-19.9	-6.1	21.4
2	MI9	100	-21.8	-7.4	20.9	-20.3	-5.5	19.7	-20.8	-4.7	29.6
2	MI10	100	-45.1	-28.7	-2.3	-18.4	-3.7	21.2	-20.6	-4.7	22.5
2	MI15	100	-21.8	-5.6	23.4	-18.3	-4.5	23.2	-18.2	-5.4	19.3
2A	MI5	100	-19.5	-5.0	20.5	-18.7	-3.9	23.5	-18.6	-4.9	18.2
2A	MI5	250	-20.4	-5.6	16.9	-19.3	-3.6	21.4	-18.1	-3.0	21.4
2A	MI6	100	-23.5	-5.9	16.6	-19.1	-6.0	14.1	-18.2	-5.0	21.0
2A	MI7	100	-21.1	-8.2	14.3	-19.3	-7.2	12.8	-19.8	-5.9	16.7
2A	MI8	100	-20.5	-5.1	18.6	-18.3	-5.5	17.1	-19.1	-4.8	22.5
2A	MI9	100	-22.1	-6.9	19.2	-19.3	-5.8	15.8	-18.7	-5.4	15.2
2A	MI10	100	-47.8	-29.3	1.1	-18.7	-4.4	24.0	-18.7	-4.0	23.1
2A	MI15	100	-24.9	-7.9	17.5	-18.3	-4.0	20.8	-18.7	-4.3	21.2

Table C.7 cont. Car Noise Data Simulation Run #2

S	Data File	Taps	Delay 0.0 msec			Delay 5.0 msec			Delay 10.0 msec		
			Min dB	Mean dB	Max dB	Min dB	Mean dB	Max dB	Min dB	Mean dB	Max dB
2B	MI5	100	-29.8	-7.9	-2.1						
2B	MI5	250	-27.5	-11.0	0.5						
2B	MI6	100	-27.3	-13.4	-4.4						
2B	MI7	100	-33.1	-20.6	-9.4						
2B	MI8	100	-25.1	-12.3	-3.6						
2B	MI9	100	-42.2	-24.3	-10.2						
2B	MI10	100	-40.0	-25.0	-10.3						
2B	MI15	100	-28.0	-12.9	-2.5						
2C	MI5	100	-32.6	-13.7	-3.9						
2C	MI5	250	-27.9	-13.7	-4.9						
2C	MI6	100	-30.9	-16.0	-8.7						
2C	MI7	100	-42.7	-32.8	-23.6						
2C	MI8	100	-32.2	-13.4	-2.9						
2C	MI9	100	-35.9	-16.6	-6.8						
2C	MI10	100	-36.6	-19.3	-6.5						
2C	MI15	100	-30.7	15.3	-3.0						
4	MI5	100	-11.2	-1.9	24.0	-10.4	-2.1	23.6	-10.4	-1.4	26.1
4	MI5	250	-9.9	-1.3	19.9	-10.5	-2.0	22.8	-11.9	-1.4	27.2
4	MI6	100	-16.7	-5.4	13.6	-17.6	-4.6	16.9	-16.1	-4.2	14.5
4	MI7	100	-9.1	-2.6	13.9	-9.9	-2.1	13.1	-9.6	-1.4	10.2
4	MI8	100	-19.8	-4.7	16.3	-18.8	-4.4	14.8	-14.1	-3.8	14.9
4	MI9	100	-13.7	-3.8	16.9	-15.6	-3.7	14.4	-13.4	-3.3	16.5
4	MI10	100	-23.5	-7.2	15.8	-15.0	-4.1	16.8	-16.8	-4.4	17.0
4	MI15	100	-12.0	-3.4	21.4	-10.7	-2.4	23.3	-11.5	-3.3	24.3
4A	MI5	100	-27.8	-16.2	25.6	-51.6	-15.0	26.1	-50.3	-12.0	26.3
4A	MI5	250	-56.5	-17.2	24.4	-55.8	-17.4	24.7	-52.0	-12.7	28.3
4A	MI6	100	-59.6	-18.6	14.4	-60.3	-18.0	18.2	-37.2	-8.5	15.2
4A	MI7	100	-40.5	-9.1	14.4	-27.4	-5.4	19.9	-22.2	-3.7	13.1
4A	MI8	100	57.4	-18.2	16.4	-61.2	-22.1	16.7	-56.0	-17.6	14.2
4A	MI9	100	-48.0	-13.8	16.1	-48.9	-13.0	15.0	-58.3	-20.6	17.3
4A	MI10	100	-55.2	-17.8	15.7	-70.3	-27.2	16.5	-73.4	-30.3	14.4
4A	MI15	100	71.3	-28.8	20.1	-54.9	-15.6	25.8	-44.0	-11.4	24.4
4B	MI5	100	-13.5	-2.1	24.4	-12.0	-2.2	23.8	-11.9	-1.5	26.0
4B	MI5	250	-11.7	-1.5	20.1	-11.8	-2.3	22.4	-13.7	-1.6	27.5
4B	MI6	100	-26.6	-7.2	13.9	-27.7	-6.1	17.6	-20.9	-5.1	15.0
4B	MI7	100	-9.3	-2.6	13.9	-10.4	-2.2	13.1	-9.9	-1.5	10.3
4B	MI8	100	-32.0	-7.0	16.6	-23.8	-5.6	15.3	-21.0	-4.9	15.4
4B	MI9	100	-17.0	-4.8	17.0	-17.9	-4.8	14.6	-17.9	-4.4	16.8
4B	MI10	100	-40.1	-10.9	16.3	-34.0	-7.6	17.2	-31.9	-6.8	17.0
4B	MI15	100	-16.5	-3.9	21.4	-14.9	-2.7	23.5	-13.2	-3.5	24.6

Table C.8 Car Noise Data Simulation Run #3**Simulation Parameters**

a = 1.0

Noise data = telephony band (300-3500Hz)

Simulation Length = 4000 Iterations

Data Based on complete 0 - 5000 Hz band (Structure 1,4,4A,4B)
telephony 300 - 3500 Hz band (Structure 5,6)

S	Data File	Taps	Delay 0.0 msec			Delay 5.0 msec			Delay 10.0 msec		
			Min dB	Mean dB	Max dB	Min dB	Mean dB	Max dB	Min dB	Mean dB	Max dB
1	MF5	100	-34.7	-5.5	9.9	-33.4	-5.7	11.3	-31.7	-6.5	11.2
1	MF5	250	-24.8	-2.6	11.3	-20.4	-2.3	12.5	-24.1	-2.9	11.1
1	MF6	100	-22.6	-4.6	15.1	-29.3	-6.2	12.0	-28.1	-5.6	15.3
1	MF7	100	-38.5	-6.7	9.9	-33.2	-4.7	10.6	-40.9	-6.3	13.7
1	MF8	100	-27.1	-3.5	13.5	-23.5	-3.5	13.4	-24.1	-3.2	10.8
1	MF9	100	-33.0	-5.2	8.7	-33.9	-3.7	11.1	-33.2	-5.0	9.1
1	MF10	100	-22.4	-2.4	12.9	-22.8	-3.1	11.9	-17.0	-2.6	11.2
1	MF15	100	-13.7	-1.2	12.5	-15.6	-1.0	12.6	-15.8	-1.8	11.0
4	MF5	100	-50.5	-8.6	15.8	-50.8	-8.9	13.1	-48.6	-9.3	14.7
4	MF5	250	-51.1	-8.8	15.0	-48.5	-8.8	14.2	-50.4	-9.3	15.4
4	MF6	100	-56.8	-12.7	20.4	-63.7	-13.6	21.0	-63.4	-13.7	21.8
4	MF7	100	-34.0	-4.3	11.8	-38.0	-4.5	13.5	-41.8	-5.1	15.4
4	MF8	100	-63.8	-12.6	21.6	-62.8	-13.4	21.1	-62.9	-14.0	18.9
4	MF9	100	-57.6	-12.5	15.8	-61.5	-12.3	15.1	-55.8	-11.1	22.8
4	MF10	100	-61.8	-13.2	20.4	-60.6	-13.4	20.3	-60.5	-14.2	18.0
4	MF15	100	-55.7	-11.6	20.0	-57.5	-10.5	20.7	-55.0	-10.4	19.6
4A	MF5	100	-69.0	-14.0	17.7	-70.8	-15.5	15.0	-76.0	-17.0	16.2
4A	MF5	250	-71.7	-15.5	19.0	-72.6	-15.6	18.5	-72.4	-15.7	16.1
4A	MF6	100	-94.3	-24.8	20.6	-96.3	-26.0	18.5	-92.2	-22.6	20.0
4A	MF7	100	-55.2	-9.8	13.6	-59.5	-10.9	16.7	-59.3	-10.1	16.1
4A	MF8	100	-97.0	-24.6	21.8	-92.5	-23.0	23.7	-98.8	-26.4	18.1
4A	MF9	100	-86.1	-21.2	19.0	-92.2	-22.8	15.2	-93.7	-24.7	24.3
4A	MF10	100	-106.0	-30.3	21.1	-106.1	-34.2	17.4	-113.3	-44.0	14.5
4A	MF15	100	-94.7	-22.4	20.8	-84.2	-17.1	22.1	-88.0	-19.8	20.8
4B	MF5	100	-52.4	-8.9	15.8	-51.8	-9.1	13.1	-51.3	-9.9	14.7
4B	MF5	250	-53.4	-9.4	15.1	-50.3	-9.2	14.2	-51.8	-9.6	15.5
4B	MF6	100	-65.5	-14.7	20.7	-70.8	-16.2	21.2	-71.4	-16.2	21.8
4B	MF7	100	-34.2	-4.4	11.8	-38.3	-4.6	13.6	-42.1	-5.2	15.4
4B	MF8	100	-73.0	-15.2	21.9	-71.8	-15.4	21.4	-70.6	-16.5	18.9
4B	MF9	100	-61.3	-13.6	15.7	-65.0	-13.2	15.1	-59.7	-12.1	22.9
4B	MF10	100	-85.2	-18.8	20.9	-81.0	-18.9	20.5	-78.3	-18.0	18.9
4B	MF15	100	-59.9	-12.8	20.1	-61.8	-11.9	21.3	-61.6	-12.2	19.4

Table C.8 cont. Car Noise Data Simulation Run #3

S	Data File	Taps	Delay 0.0 msec			Delay 5.0 msec			Delay 10.0 msec		
			Min dB	Mean dB	Max dB	Min dB	Mean dB	Max dB	Min dB	Mean dB	Max dB
5	MF5	100	-8.6	1.8	21.8	-7.3	1.5	18.1	-8.5	2.0	21.8
5	MF5	250	-8.7	1.3	18.7	-7.6	2.0	20.6	-7.7	2.4	24.5
5	MF6	100	-12.7	-0.3	20.7	-10.4	-0.3	19.0	-10.3	-0.1	21.9
5	MF7	100	-4.2	2.5	14.4	-4.5	2.2	13.9	-4.8	2.7	15.6
5	MF8	100	-10.0	1.3	23.9	-11.6	0.7	24.8	-9.9	0.4	19.2
5	MF9	100	-10.4	-0.5	15.1	-9.5	0.4	21.0	-8.0	0.7	18.6
5	MF10	100	-10.8	0.4	24.4	-11.2	-1.3	29.5	-10.1	-0.1	21.2
5	MF15	100	-8.7	0.6	22.1	-9.1	1.6	23.3	-9.3	0.8	21.7
6	MF5	100	-9.5	3.6	19.7	-8.2	3.8	18.1	-8.5	3.5	21.2
6	MF5	250	-8.9	3.7	19.2	-8.9	4.1	21.8	-8.8	4.0	21.5
6	MF6	100	-10.6	1.5	18.8	-7.7	2.6	16.0	-8.3	2.3	23.4
6	MF7	100	-4.4	3.4	17.1	-4.1	3.8	15.3	-6.8	4.3	20.2
6	MF8	100	-9.0	2.1	18.6	-9.1	2.8	25.9	-8.0	1.5	19.4
6	MF9	100	-8.0	2.1	14.2	-5.9	2.8	19.1	-6.9	3.3	20.8
6	MF10	100	-7.9	2.2	23.8	-7.7	3.0	25.9	-7.6	2.2	18.9
6	MF15	100	-11.0	1.7	19.2	-9.3	2.9	20.2	-9.7	2.7	18.7

Table C.9 Car Noise Data Simulation Run #4**Simulation Parameters**

a = 1.0

Noise data = lower subband (300-1200Hz)

Simulation Length = 4000 Iterations

Data Based on telephony 300 - 3500 Hz Band

S	Data File	Taps	Delay 0.0 msec			Delay 5.0 msec			Delay 10.0 msec		
			Min dB	Mean dB	Max dB	Min dB	Mean dB	Max dB	Min dB	Mean dB	Max dB
1	MS5	100	-23.0	-5.5	10.3	-23.8	-7.4	10.7	-26.1	-8.8	12.0
1	MS5	250	-12.8	-0.1	13.4	-10.9	-0.6	14.5	-11.3	-0.9	14.2
1	MS6	100	-19.2	-4.1	9.3	-22.1	-7.4	9.3	-15.4	-3.6	11.1
1	MS7	100	-21.0	-7.6	11.9	-22.0	-8.4	11.5	-22.8	-7.4	13.8
1	MS8	100	-15.3	-1.7	11.3	-15.4	-1.8	11.8	-16.5	-2.5	10.0
1	MS9	100	-27.8	-9.4	10.7	-21.0	-5.9	12.3	-24.5	-8.6	10.5
1	MS10	100	-18.8	-3.1	13.5	-16.8	-3.9	11.7	-20.3	-4.9	11.1
1	MS15	100	-20.1	-2.4	12.4	-17.4	-1.8	12.8	-15.9	-1.6	11.0
4	MS5	100	-44.6	-22.7	17.0	-46.0	-22.6	14.6	-49.8	-22.7	16.0
4	MS5	250	-44.2	-21.2	16.2	-45.4	-21.3	15.4	-48.1	-20.9	16.1
4	MS6	100	-55.4	-30.4	20.1	-60.5	-31.9	17.9	-57.0	-28.8	20.4
4	MS7	100	-30.8	-9.6	13.6	-31.5	-10.1	15.6	-28.3	-9.1	14.4
4	MS8	100	-57.1	-30.0	21.7	-53.1	-27.6	22.5	-64.6	-33.6	17.7
4	MS9	100	-54.1	-30.4	16.5	-55.2	-30.7	15.8	-52.0	-26.5	22.6
4	MS10	100	-61.2	-32.4	20.1	-61.5	-33.2	19.9	-62.3	-34.5	18.7
4	MS15	100	-54.0	-28.0	21.4	-47.4	-25.6	21.4	-46.7	-24.5	19.4
5	MS5	100	-34.7	-14.6	25.6	-36.0	-14.8	21.2	-34.0	-14.7	22.2
5	MS5	250	-36.8	-15.8	21.0	-34.2	-12.9	24.2	-31.4	-12.0	27.0
5	MS6	100	-42.4	-21.6	20.9	-45.3	-21.5	18.2	-37.9	-16.4	22.8
5	MS7	100	-26.5	-9.7	15.2	-30.1	-10.6	16.5	-30.0	-10.3	15.5
5	MS8	100	-33.3	-14.9	23.1	-36.7	-16.1	25.1	-43.3	-19.9	19.3
5	MS9	100	-46.5	-23.0	15.5	-44.9	-22.6	21.4	-41.1	-19.0	19.1
5	MS10	100	-45.2	-22.3	20.7	-34.8	-14.4	25.6	-48.5	-24.2	19.8
5	MS15	100	-38.9	18.4	23.5	-36.2	-14.8	22.5	-34.4	-16.8	22.1
6	MS5	100	-27.0	-6.3	26.7	-23.3	-6.4	24.4	-23.5	-7.5	24.3
6	MS5	250	-26.9	-6.5	22.8	-23.8	-5.8	28.2	-19.0	-4.2	24.6
6	MS6	100	-30.8	-12.8	19.7	-28.7	-12.2	17.2	-23.5	-7.5	24.3
6	MS7	100	-27.1	-8.1	18.2	-27.1	-8.1	18.8	-24.6	-6.1	19.4
6	MS8	100	-22.5	-5.8	22.0	-21.9	-4.9	25.7	-33.5	-11.5	19.5
6	MS9	100	-34.0	-13.8	16.7	-31.3	-11.4	20.2	-27.0	-8.2	20.9
6	MS10	100	-29.4	-9.7	22.2	-24.4	-6.5	26.5	-29.3	-11.3	18.9
6	MS15	100	-29.2	-8.6	22.8	-30.3	-10.1	22.4	-26.0	-8.5	21.3

Table C.10 Car Noise Data Simulation Run #5**Simulation Parameters**

a = 1.0

Noise data = large upper subband (1200-3500Hz)

Simulation Length = 4000 Iterations

Data Based on telephony 300 - 3500 Hz Band

S	Data File	Taps	Delay 0.0 msec			Delay 5.0 msec			Delay 10.0 msec		
			Min dB	Mean dB	Max dB	Min dB	Mean dB	Max dB	Min dB	Mean dB	Max dB
1	MT5	100	-18.4	-0.9	23.2	-19.6	-0.6	22.7	-14.0	0.2	23.4
1	MT5	250	-16.4	-0.4	22.4	-13.5	0.7	24.5	-7.4	2.0	23.5
1	MT6	100	-13.6	-1.3	24.2	-18.0	-1.0	23.0	-15.1	-0.5	29.8
1	MT7	100	-17.4	-1.8	7.6	-19.5	-2.8	7.6	-20.6	-2.3	6.3
1	MT8	100	-15.0	-2.2	4.1	-17.9	-2.3	4.2	-16.7	-3.0	3.3
1	MT9	100	-13.8	-2.5	9.2	-18.6	-2.5	10.2	-12.9	-1.5	10.4
1	MT10	100	-12.6	-0.9	6.3	-9.7	-1.6	6.5	-10.7	-1.3	5.1
1	MT15	100	-10.1	-1.2	8.5	-13.4	-2.4	7.6	15.3	-3.0	5.7
4	MT5	100	-12.6	-0.9	7.8	-13.3	1.2	6.6	-10.7	0.7	7.4
4	MT5	250	-7.9	-1.9	8.5	-8.7	2.3	8.4	-6.2	2.4	8.8
4	MT6	100	-12.1	-0.4	12.4	-12.7	0.1	4.7	-12.6	0.1	7.2
4	MT7	100	-13.5	1.1	10.6	-11.0	0.5	7.5	-15.2	-0.3	10.1
4	MT8	100	-14.6	0.1	7.0	-12.8	0.1	7.9	-13.5	-0.4	8.2
4	MT9	100	-12.5	-0.1	9.2	-15.1	0.0	9.6	-12.4	-0.1	8.7
4	MT10	100	-12.5	0.7	10.3	-10.0	0.3	9.7	-13.0	0.1	10.7
4	MT15	100	-17.3	-0.2	11.2	-13.9	-0.3	11.6	-11.7	-0.2	9.7
5	MT5	100	-15.8	0.7	11.3	-15.8	1.3	9.7	-13.4	0.7	9.8
5	MT5	250	-10.5	1.7	12.4	-10.3	2.2	10.9	-7.9	2.4	11.8
5	MT6	100	-12.7	0.3	12.1	-13.9	0.3	9.1	-14.7	0.1	13.3
5	MT7	100	-14.1	1.0	10.8	-11.2	0.6	8.6	-14.9	-0.2	10.0
5	MT8	100	-13.7	-0.5	7.1	-15.1	-0.1	8.5	-16.0	-0.6	11.8
5	MT9	100	-12.3	-0.5	10.8	-19.9	-0.4	9.1	-15.2	-0.2	9.8
5	MT10	100	-14.0	1.0	11.5	-12.5	0.3	10.8	-17.4	-0.6	11.7
5	MT15	100	-13.7	0.0	12.6	-15.1	-0.5	11.8	-13.7	-0.4	10.3
6	MT5	100	-15.4	1.0	11.7	-17.7	1.4	11.3	-13.4	1.1	11.9
6	MT5	250	-9.0	2.1	13.3	-11.9	2.4	11.2	-10.2	3.4	12.6
6	MT6	100	-15.5	1.7	13.3	-16.9	0.5	9.6	-17.4	0.8	13.1
6	MT7	100	-12.5	1.6	10.5	-12.5	1.2	9.5	-11.6	0.5	10.2
6	MT8	100	-16.2	0.4	8.6	-16.3	0.3	9.8	-19.2	0.5	13.3
6	MT9	100	-15.8	-0.4	12.8	-18.2	0.4	9.2	-20.1	0.6	12.0
6	MT10	100	-18.1	1.3	11.5	-16.2	1.3	12.6	-18.2	1.1	12.2
6	MT15	100	-14.8	0.6	13.5	-19.2	1.1	12.8	-16.0	1.2	12.4

Table C.11 Car Noise Data Simulation Run #6**Simulation Parameters****a = 1.0****Noise data = middle subband (1200-2400Hz)****Simulation Length = 4000 Iterations****Data Based on telephony 300 - 3500 Hz Band**

S	Data File	Taps	Delay 0.0 msec			Delay 5.0 msec			Delay 10.0 msec		
			Min dB	Mean dB	Max dB	Min dB	Mean dB	Max dB	Min dB	Mean dB	Max dB
1	MU5	100	-18.3	-3.2	9.3	-21.5	-4.4	9.2	-15.4	-2.9	13.5
1	MU5	250	-16.2	-2.0	10.4	-17.5	-1.6	11.0	-11.5	0.6	12.3
1	MU6	100	-20.3	-4.8	6.5	-19.2	-4.7	5.9	-19.9	-5.3	5.2
1	MU7	100	-23.3	-4.9	11.3	-20.1	-5.2	8.6	-23.0	-5.3	7.4
1	MU8	100	-20.8	-5.1	7.5	-20.8	-6.0	5.6	-25.5	-6.2	4.7
1	MU9	100	-16.8	-3.7	9.3	-16.1	-3.8	8.9	-19.1	-3.7	9.6
1	MU10	100	-16.4	-3.0	6.5	-15.8	-2.2	7.2	-12.6	-2.7	5.5
1	MU15	100	-14.3	-2.8	8.9	-13.1	-4.2	8.0	-14.6	-4.9	5.8
4	MU5	100	-13.2	-0.6	10.6	-15.1	-1.3	9.0	-13.0	-1.0	9.9
4	MU5	250	-10.8	0.6	11.6	-12.0	0.7	11.0	-8.6	2.0	13.0
4	MU6	100	-13.8	-1.4	13.0	-15.6	-2.2	8.1	-14.6	-1.8	8.9
4	MU7	100	-21.6	-2.6	12.2	-13.3	-1.4	8.8	-20.8	-2.5	12.1
4	MU8	100	-20.4	-3.4	8.7	-17.5	-3.5	8.4	-22.3	-3.8	9.2
4	MU9	100	-17.0	-1.8	11.2	-15.7	-1.3	11.5	-20.1	-3.2	12.2
4	MU10	100	-19.2	-2.9	12.2	-18.3	-2.0	12.4	-16.1	-2.4	12.8
4	MU15	100	-19.8	-3.4	12.5	-22.3	-3.4	13.3	-18.4	-3.2	10.7
5	MU5	100	-16.0	-1.4	9.8	-17.2	-1.7	9.0	-14.5	-1.6	9.4
5	MU5	250	-12.1	0.3	11.5	-13.3	0.5	11.8	-12.0	1.1	11.8
5	MU6	100	-18.0	-2.7	11.3	-16.5	-2.3	9.5	-21.0	-3.1	9.6
5	MU7	100	-20.4	-2.7	10.0	-12.4	-1.3	8.5	-19.7	-2.5	11.6
5	MU8	100	-20.1	-3.6	7.3	-20.8	-3.9	9.2	-21.9	-4.2	11.7
5	MU9	100	-18.7	-2.9	12.1	-16.9	-2.5	12.8	-20.3	-3.0	12.9
5	MU10	100	-20.9	-3.2	12.6	-22.7	-2.5	12.8	-18.7	-2.8	14.2
5	MU15	100	-20.8	-3.6	13.9	-20.4	-4.3	14.4	-18.5	-3.8	11.8
6	MU5	100	-16.0	-1.3	11.0	-16.9	-1.7	9.0	-14.0	-1.5	10.2
6	MU5	250	-12.0	-0.5	13.2	-14.6	-0.6	11.7	-8.5	0.8	12.3
6	MU6	100	-18.4	-1.9	12.3	-17.5	-2.5	11.9	-20.7	-2.4	14.5
6	MU7	100	-20.5	-2.4	10.5	-11.6	-1.0	10.0	-18.7	-2.0	11.7
6	MU8	100	-23.0	-4.5	8.0	-20.8	-3.6	14.3	-22.8	-4.0	12.7
6	MU9	100	-18.4	-2.5	13.6	-17.7	-2.2	13.4	-20.0	-3.7	13.1
6	MU10	100	-21.1	-2.8	12.9	-21.0	-2.0	15.5	-19.7	-3.5	15.5
6	MU15	100	-21.5	-3.4	14.7	-19.4	-2.8	15.3	-20.4	-3.1	14.9

Table C.12 Car Noise Data Simulation Run #7**Simulation Parameters**

a = 1.0

Noise data = upper subband (2400-3500Hz)

Simulation Length = 4000 Iterations

Data Based on telephony 300 - 3500 Hz Band

S	Data File	Taps	Delay 0.0 msec			Delay 5.0 msec			Delay 10.0 msec		
			Min dB	Mean dB	Max dB	Min dB	Mean dB	Max dB	Min dB	Mean dB	Max dB
1	MV5	100	-9.5	0.6	31.0	-9.7	0.8	32.6	-11.0	-0.6	30.7
1	MV5	250	-6.2	1.6	28.2	-6.3	2.2	31.3	-6.3	1.7	32.3
1	MV6	100	-14.3	-1.9	19.6	-10.7	0.0	28.3	-11.0	0.6	32.6
1	MV7	100	-22.3	-7.6	7.0	-24.2	-6.1	7.3	-22.3	-6.1	8.0
1	MV8	100	-18.9	-5.8	6.2	-14.7	-4.1	8.4	-18.9	-7.0	4.4
1	MV9	100	-16.6	-5.1	15.8	-18.5	-5.1	23.7	-17.2	-4.4	16.9
1	MV10	100	-15.0	-3.6	11.7	-16.9	-5.0	7.4	-16.9	-4.7	7.0
1	MV15	100	-13.3	-1.2	10.2	-15.7	-2.1	9.4	-16.7	-4.7	4.6
4	MV5	100	-12.4	-1.1	15.3	-10.1	0.4	18.8	-12.1	-0.6	18.0
4	MV5	250	-8.6	0.7	14.8	-7.5	1.7	15.6	-7.4	1.1	17.5
4	MV6	100	-11.6	-1.4	15.0	-11.6	-0.8	19.0	-12.4	-0.4	19.7
4	MV7	100	-12.6	-2.3	7.3	-16.0	-2.9	7.8	-12.9	-2.3	7.0
4	MV8	100	-15.4	-3.1	12.1	-14.9	-1.8	11.0	-14.0	-2.9	8.4
4	MV9	100	-15.3	-3.3	9.4	-15.6	-2.8	9.4	-15.4	-3.0	8.6
4	MV10	100	-13.9	-2.7	12.0	-13.9	-3.4	9.1	-12.4	-2.6	8.7
4	MV15	100	-15.0	-0.6	11.5	-13.6	-0.9	13.6	-13.8	-1.9	12.0
5	MV5	100	-13.5	-0.9	15.1	-10.5	0.6	18.0	-11.5	-0.3	17.3
5	MV5	250	-9.3	0.5	13.5	-8.2	1.2	15.5	-8.4	0.8	16.7
5	MV6	100	-14.5	-1.7	14.8	-15.1	-1.5	20.1	-16.9	-1.0	23.1
5	MV7	100	-13.8	-1.9	8.4	-16.2	-2.6	7.2	-14.0	-2.1	7.6
5	MV8	100	-16.7	-3.1	12.0	-14.8	-1.8	10.7	-14.0	-2.7	8.4
5	MV9	100	-18.0	-3.2	12.6	-14.1	-2.6	13.6	-14.8	-2.9	10.7
5	MV10	100	-14.6	-2.5	11.5	-15.8	-3.7	8.4	-15.7	-2.8	10.5
5	MV15	100	-16.4	-0.7	11.1	-14.8	-0.7	13.5	-13.0	-1.8	9.0
6	MV5	100	-6.8	0.6	16.6	-5.0	1.4	19.6	-6.5	0.7	18.9
6	MV5	250	-5.1	1.3	18.3	-4.4	1.9	17.1	-5.0	1.4	17.7
6	MV6	100	-14.3	-0.8	18.2	-12.1	-0.6	19.9	-11.8	-0.3	22.7
6	MV7	100	-7.4	-0.2	8.9	-9.8	-0.7	10.4	-7.8	-0.6	8.4
6	MV8	100	-16.2	-1.9	11.1	-13.2	-0.6	10.4	-12.4	-1.7	7.4
6	MV9	100	-11.1	-1.8	14.4	-12.4	-1.9	13.8	-14.8	-1.9	11.6
6	MV10	100	-13.0	-1.1	12.5	-13.2	-2.2	9.1	-10.9	-1.4	11.3
6	MV15	100	-7.5	0.7	11.4	-6.4	0.4	12.5	-9.3	-0.9	8.0

Table C.13 Car Noise Data Simulation Run #8**Simulation Parameters****a = 1.0****Noise data = telephony band (300-3500Hz)****Simulation Length = 4000 Iterations****Data Based on telephony 300 - 3500 Hz Band****Taps = 100****Single reference cancelers**
Structure using three subbands (300-1200Hz)
 (1200-2400Hz)
 (2400-3500Hz)

S	Data File	Taps	Delay 0.0 msec			Delay 5.0 msec			Delay 10.0 msec		
			Min dB	Mean dB	Max dB	Min dB	Mean dB	Max dB	Min dB	Mean dB	Max dB
T	MW5	x1	0.8	7.6	31.0	-2.8	7.3	32.5	-2.1	6.7	29.6
T	MW5	x2	0.5	4.9	12.5	-0.5	5.4	13.1	-0.6	4.1	14.4
T	MW5	x3	-3.6	3.5	11.7	-1.5	3.7	12.0	-1.4	4.0	10.4
T	MW6	x1	-7.8	2.2	15.3	-3.5	5.4	28.0	-7.3	4.8	30.3
T	MW6	x2	-5.5	3.4	11.8	-7.3	2.3	14.4	-4.8	3.6	13.0
T	MW6	x3	-2.2	4.0	11.9	-3.6	2.3	9.2	-1.7	3.0	6.4

Table C.14 Car Noise Data Simulation Run #9**Simulation Parameters**

a = 1.0

Noise data = telephony band (300-3500Hz)

Simulation Length = 4000 Iterations

Data Based on telephony 300 - 3500 Hz Band

Structure using two subbands (300-1200Hz)
(1200-3500Hz)

S	Data File	Taps	Delay 0.0 msec			Delay 5.0 msec			Delay 10.0 msec		
			Min dB	Mean dB	Max dB	Min dB	Mean dB	Max dB	Min dB	Mean dB	Max dB
12	MX5	100	-5.0	4.0	23.5	-3.8	4.1	24.0	-2.7	4.1	25.3
12	MX5	250	-4.4	4.4	22.4	-4.0	4.9	24.5	-3.4	5.5	24.3
12	MX6	100	-5.1	2.2	24.1	-6.0	1.4	22.2	-8.9	2.8	27.2
12	MX7	100	-3.5	2.5	11.6	-4.7	1.8	13.1	-3.8	2.3	12.9
12	MX8	100	-3.1	2.1	11.1	-3.8	1.8	10.8	-3.7	2.1	9.5
12	MX9	100	-5.8	0.9	11.1	-5.0	1.9	11.2	-5.1	1.8	11.8
12	MX10	100	-5.3	2.2	11.0	-5.0	1.2	7.7	-3.4	1.7	11.6
12	MX15	100	-4.8	2.5	11.3	-2.2	2.2	12.9	-3.7	1.9	10.6
62	MX5	100	-3.6	6.1	28.8	-3.1	6.4	26.4	-3.4	5.4	23.8
62	MX5	250	-3.8	5.7	23.8	-2.6	7.2	29.1	-4.1	7.2	25.8
62	MX6	100	-6.3	2.5	19.5	-3.6	3.8	17.6	-5.3	4.3	23.8
62	MX7	100	-0.1	5.4	18.8	-2.3	5.1	19.0	-1.3	5.4	18.6
62	MX8	100	-5.2	4.6	21.9	-4.6	4.5	24.7	-5.5	3.1	21.6
62	MX9	100	-6.1	3.8	16.4	-4.1	4.1	18.3	-3.2	5.6	21.5
62	MX10	100	-6.2	4.8	22.4	-5.8	5.6	28.2	-4.8	4.2	19.5
62	MX15	100	-7.4	4.4	22.3	-6.4	4.9	21.7	-6.4	4.4	21.4
72	MX5	100	-1.7	8.1	28.0	0.7	7.9	26.9	0.5	7.1	24.6
72	MX5	250	-1.2	7.8	23.5	-0.1	8.9	30.1	-1.7	8.7	27.2
72	MX6	100	-1.8	5.7	19.3	-2.2	5.6	17.5	-0.8	7.1	23.7
72	MX7	100	1.2	6.2	19.1	-0.5	5.9	19.5	0.4	6.2	18.6
72	MX8	100	-0.5	7.2	21.9	-0.1	6.9	24.6	-0.3	6.1	21.6
72	MX9	100	-2.9	5.6	16.5	-1.1	6.0	18.2	1.2	8.0	21.6
72	MX10	100	-0.9	7.7	22.0	1.4	8.7	28.8	0.7	7.2	19.4
72	MX15	100	-5.4	6.9	23.2	-2.5	7.4	22.2	0.0	7.3	22.2

Table C.15 Car Noise Data Simulation Run #10**Simulation Parameters**

a = 1.0

Noise data = telephony band (300-3500Hz)

Simulation Length = 4000 Iterations

Data Based on telephony 300 - 3500 Hz Band

Structure using three subbands
(300-1200Hz)
(1200-2400Hz)
(2400-3500Hz)

S	Data File	Taps	Delay 0.0 msec			Delay 5.0 msec			Delay 10.0 msec		
			Min dB	Mean dB	Max dB	Min dB	Mean dB	Max dB	Min dB	Mean dB	Max dB
13	MW5	100	0.1	7.8	31.0	-0.7	7.5	32.6	-0.4	7.0	29.6
13	MW5	250	1.9	7.8	28.2	1.1	8.2	31.2	2.2	8.7	31.0
13	MW6	100	-2.3	4.5	19.5	-2.2	4.8	28.4	-6.3	5.5	29.8
13	MW7	100	-3.5	3.6	11.6	-1.3	3.5	13.1	-3.3	3.8	12.9
13	MW8	100	-3.0	3.2	11.1	-2.4	2.9	10.8	-2.2	3.3	9.5
13	MW9	100	-3.2	2.3	11.1	-2.4	4.1	11.2	-2.7	4.0	11.8
13	MW10	100	-0.5	3.9	11.0	-4.7	2.7	8.3	-1.2	3.0	11.6
13	MW15	100	-1.4	5.0	11.3	-2.5	3.9	12.9	-3.0	2.8	10.6
63	MW5	100	-0.4	8.4	28.5	0.3	8.6	26.6	0.3	7.6	24.0
63	MW5	250	-0.8	7.7	23.7	0.8	9.4	28.9	0.6	9.9	26.4
63	MW6	100	-2.9	4.6	19.8	0.7	6.7	18.6	-2.9	6.2	24.2
63	MW7	100	1.9	7.1	19.0	-0.8	6.8	19.4	1.8	7.4	18.7
63	MW8	100	-3.5	6.2	22.4	-1.9	6.8	24.5	-2.7	4.8	21.8
63	MW9	100	-1.1	6.2	16.8	-0.6	6.7	18.1	-0.2	7.8	21.5
63	MW10	100	-2.5	6.6	22.5	-2.7	6.9	29.2	-0.7	5.8	19.5
63	MW15	100	-4.3	6.9	22.7	-3.0	6.9	21.8	-5.5	5.7	21.7
73	MW5	100	3.1	10.5	28.0	3.2	10.5	26.9	1.4	9.4	24.6
73	MW5	250	2.8	10.0	23.5	3.3	11.0	30.1	3.0	11.1	27.2
73	MW6	100	-1.8	8.6	19.3	2.8	9.2	19.4	0.3	10.1	23.7
73	MW7	100	2.3	7.4	19.1	1.2	7.4	19.5	2.5	7.8	18.6
73	MW8	100	1.8	8.4	21.9	2.0	9.6	24.6	1.5	7.7	21.6
73	MW9	100	1.8	8.2	16.5	2.4	8.6	18.2	2.6	9.8	21.6
73	MW10	100	3.2	9.9	22.0	2.3	10.1	28.8	0.7	8.7	19.4
73	MW15	100	1.5	9.5	23.2	2.9	9.6	22.2	-0.7	8.4	22.2

Table C.16 Car Noise Data Simulation Run #11**Simulation Parameters**

a = 1.0

Noise data = telephony band (300-3500Hz)

Simulation Length = 30000 Iterations

Taps = 100

Delay = 0.0 msec

Data Based on telephony 300 - 3500 Hz Band

S	Data File	Speech Prime	Ref(x2)	Detect	Cancellation			Primary Before power	Noise After power
					Min dB	Mean dB	Max dB		
13	MC5	N	N	N	-0.4	7.2	32.1	6280	1455.8
73	MC5	N	N	N	4.7	11.2	23.8	6280	143.4
13	MA5	Y	N	N	-21.7	-11.9	4.0	6280	322780
73	MA5	Y	N	N	-8.1	2.4	15.5	6280	311930
13	MA6	Y	N	N	-14.6	-5.6	9.2	58654	325630
73	MA6	Y	N	N	-5.3	6.3	22.5	58654	314010
13	MB5	Y	Y	N	-16.4	-8.1	9.3	6280	312350
73	MB5	Y	Y	N	-6.4	4.1	16.1	6280	311700
13	MB6	Y	Y	N	-14.8	-4.0	12.2	58654	324700
73	MB6	Y	Y	N	-6.2	6.1	19.8	58654	313330
13	MA5	Y	N	Y	-4.6	2.8	20.7	6280	47373
73	MA5	Y	N	Y	-2.0	6.6	15.5	6280	155650
13	MA6	Y	N	Y	-6.6	3.9	23.8	58654	260840
73	MA6	Y	N	Y	0.1	9.5	21.3	58654	148710
13	MB5	Y	Y	Y	-7.5	1.5	22.0	6280	60197
73	MB5	Y	Y	Y	-1.8	7.2	18.7	6280	796680
13	MB6	Y	Y	Y	-5.7	3.9	22.6	58654	370940
73	MB6	Y	Y	Y	0.9	9.7	21.2	58654	196200
Data File		SNR(dB)			Ref(x1)			Speech Detector Threshold	
		Prime	Ref(x1)		Ref(x2)		Ref(x3)		
MA5		16.9	na		na		na	100000	
MB5		16.9	na		-3.5		na	100000	
MC5		na	na		na		na	na	
MA6		7.2	na		na		na	160000	
MB6		7.2	na		-9.2		na	160000	

Note: Signal in x2 is 14dB lower than signal in Prime and also delayed by 2.5msec.

Table C.17 Car Noise Data Simulation Run #12**Simulation Parameters**

a = 1.0

Noise data = telephony band (300-3500Hz)

Simulation Length = 18000 Iterations

Delay = 0.0 msec

Taps = 100

Data Based on telephony 300 - 3500 Hz Band

S	Data File	Ref Spch x2	Detection Freeze						Cancellation			Primary Noise	
			x1	x2	x3	x1	x2	x3	Min dB	Mean dB	Max dB	Before power	After power
13	MD5	N	Y	Y	Y	N	N	N	-10.0	0.7	18.8	59005	245320
73	MD5	N	Y	Y	Y	N	N	N	1.8	8.9	22.4	59005	288880
13	ME5	Y	Y	Y	Y	N	N	N	-8.2	0.2	20.2	59005	251870
73	ME5	Y	Y	Y	Y	N	N	N	2.1	8.8	23.7	59005	331030
73	MD5	N	N	Y	Y	N	N	N	-0.5	7.6	20.7	59005	362020
73	ME5	Y	N	Y	Y	N	N	N	-0.7	7.4	19.7	59005	398740
73	MD5	N	N	Y	N	N	N	N	-2.7	7.7	19.6	59005	381190
73	ME5	Y	N	Y	N	N	N	N	-4.2	7.6	20.0	59005	341350
73	MD5	N	N	Y	Y	N	Y	Y	-0.5	7.6	20.7	59005	236250
73	ME5	Y	N	Y	Y	N	Y	Y	-0.7	7.4	19.7	59005	236110
73	MD5	N	N	Y	N	N	Y	N	-2.7	7.7	19.6	59005	226500
73	ME5	Y	N	Y	N	N	Y	N	-4.2	7.6	20.0	59005	226680
Data File		SNR(dB)		Ref(x1)		Ref(x2)		Ref(x3)		Speech Detector Threshold			
		Prime											
MD5		5.9		na		na		na		90000			
ME5		5.9		na		-13.3		na		90000			

Note: Signal in x2 is 16.9dB lower than signal in Prime and also delayed by 2.5msec.

Table C.18 Car Noise Data Simulation Run #13**Simulation Parameters**

a = 1.0

Noise data = telephony band (300-3500Hz)

Simulation Length = 18000 Iterations

Delay = 0.0 msec

Taps = 100

Data Based on telephony 300 - 3500 Hz Band

S	Data File	Speech Prime	Ref(x2)	Speech Detector	Cancellation		
					Min dB	Mean dB	Max dB
13	MG5	Y	N	N	-2.8	4.4	27.7
73	MG5	Y	N	N	-2.8	9.3	20.7
13	MH5	Y	Y	N	-2.9	4.5	27.3
73	MH5	Y	Y	N	-2.5	9.3	20.8
Data File		SNR(dB)					
		Prime	Ref(x1)	Ref(x2)	Ref(x3)		
MG5		-11.0	na	na	na		
MH5		-11.0	na	-25.3	na		

Note: Signal in x2 is 12dB lower than signal in Prime and also delayed by 2.5msec.

VITAE

RAYMOND BRUCE WALLACE B.E.Sc. P.Eng.

WORK EXPERIENCE

Contract Consultant - Part Time February 1990 - date
System Design Engineer
Northern Telecom Canada Ltd. (NAD Division)

Contract Consultant - Part Time September 1989 - February 1990
System Design Engineer
Bell Northern Research Ltd.

Member of Scientific Staff August 1987 - September 1989
Design Engineer
Bell Northern Research Ltd.

Summer Student May - August 1987
Bell Northern Research Ltd.

EDUCATION

Bachelor of Engineering Science 1983 - 1987
University of Western Ontario
Electrical Engineering

PROFESSIONAL CERTIFICATION

Member of the Association of Professional Engineers of Ontario 1990 - date

HONOURS and AWARDS

Graduate Scholarship 1989, 1990
Systems and Computer Engineering, Carleton University

BNR Postgraduate Scholarship 1988
Bell Northern Research Ltd. (Accepted 1989, 1990)

NSERC Postgraduate Scholarship 1987
Natural Sciences and Engineering Research Council (Accepted 1989, 1990)

APEO Gold Medal 1987
Association of Professional Engineers of Ontario
Faculty of Engineering Science, UWO

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