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THE LOSS OF PERCEIVED BINAURAL SEPARATION IN ADAPTIVE INTERFERENCE CANCELLATION

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Abstract

This paper considers some of the advantages of binaural audio signal processing over monaural processing when the extraction of a signal from interference is required. For the hearing aid user it is argued that binaural separation should be preserved as far as possible along the signal processing path, and preferably should extend as far as the listener. In this way full advantage may be taken of the human auditory system to further enhance the binaural signal. Based on a successful adaptive filter structure with monaural output a modified filter structure with two inputs and two outputs is discussed and its binaural properties are analysed. It is shown that a system employing additive cancellation has no intrinsic phase preservation properties, rendering it unsuitable for direct application in a binaural processor.

1. INTRODUCTION

There are many instances in which a better estimate for a signal contaminated by interference is required and there exist many ways of obtaining improved estimates. The best approach in any particular situation will depend upon the nature of the interference to be removed or attenuated, the characteristics of the target signal being extracted and the use to which the enhanced signal is to be put. The problem addressed by this paper is the extraction of a single acoustic source (the target), usually, but not exclusively, speech, from a variety of acoustic environments where the target is partially obscured by additive interfering sounds. A particularly successful two-microphone approach has been reported by Peterson, Durlach, Rabinowitz and Zurek [1], which has yielded considerable improvements in the intelligibility of a speech target in a number of realistically simulated acoustic environments. In this paper we present a mathematical analysis of aspects of their system and modify it to create a binaural adaptive filter structure.

2. DEVELOPMENT OF THE BINAURAL ADAPTIVE FILTER

Methods of extracting a target signal from interference break down into a number of categories. Processing algorithms are either essentially time-domain or frequency-domain in their approach. Sensor arrangements are either single channel or multichannel. *A priori* knowledge of the target signal may or may not be assumed. In this paper we choose to concentrate on the time-domain approach and investigate the properties of a binaural adaptive filter structure based upon previous time-domain systems.

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2.1 Multichannel Interference Reduction Techniques.

The problem of how to isolate the signals from spatially diverse sources may be partially solved by using two or more microphones [2, 3]. Since a binaural advantage has been demonstrated for the human auditory system [4, 5] for both sound localisation and sound discrimination a two microphone system would seem to be a sensible choice.

3. THE BINAURAL ADAPTIVE FILTER STRUCTURE

Ultimately, preservation of a binaural signal as far as the listener would allow the human binaural auditory system, where this remains functional, to contribute to the signal enhancement process. As a first step in extending binaural processing of the input signals it was decided to investigate the consequences of modifying the adaptive beamformer structure of Peterson *et al* [1] to create two distinct left and right output channels. It is logically untenable to expect that the output from a fully adapted system could preserve the precise locations of attenuated interference sources, otherwise, by feeding the output into a similar structure, further attenuation of the interference would be obtainable. If the process were repeated enough times the interference would be totally removed, which is clearly an impossibility. However, it was felt worthwhile to study the issues surrounding the problem, with the aim of producing an artificial binaural separation, which might improve intelligibility of the target signal, even if the precise location of interference sources was not preserved.

The assumption is made that the target signal source is always located directly in front of the listener (in the median plane). The physical arrangement is shown in Figure 1. In this example a single interference source is situated to the right side of the listener and subtends an angle α with the median plane. The modified adaptive filter structure, shown in Figure 2, consists of two similar adaptive filters. By taking the difference between the left and right channel signals under ideal anechoic and balanced conditions, an interference-

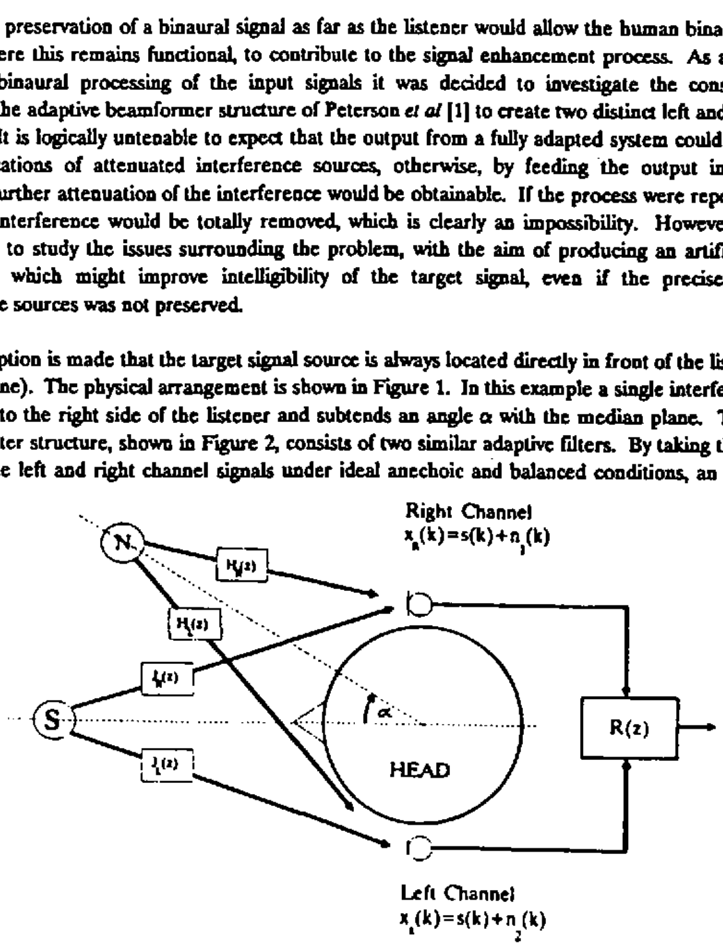

Figure 1: The arrangement of the simulated acoustic environment for the binaural adaptive filter structure.

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nly reference signal, $r(k)$, is obtained. The two filters share the same interference reference channel, but he output of each filter is used to apply separate interference cancellation to the left and right input hannels. The effect of this system is to place a steerable null in the direction of the interference. In the resence of spectrally overlapping interference from more than one direction cancellation will not be omplete, but will nevertheless approach a minimum energy solution.

n the simplest case where cancellation is complete, the output from this system is no different from the riginal monaural system. However, a difference between the two output channels may be expected when ancellation is incomplete, due, for example, to partial adaption to a single interference source, or full daption to multiple sources. Since incomplete cancellation will often occur in practice, some binaural eparation of the residual interference from the target may appear plausible. We now examine this ossibility in greater detail.

4. ANALYSIS OF INCOMPLETE CANCELLATION IN THE BINAURAL BEAMFORMER

Errors leading to incomplete cancellation arise at several points in the beamformer structure. These are iscussed in turn and their overall effect on the output signals is also considered.

.1 The Interference Reference Channel

Considering only the effects of interaural time delay the transfer function, $R(z)$, from an interference source o the interference reference channel is given approximately by

$$R(z) = 1 - z^{-L} \tag{1}$$

where L represents the difference in the time of arrival of the interference signal at the two microphones.

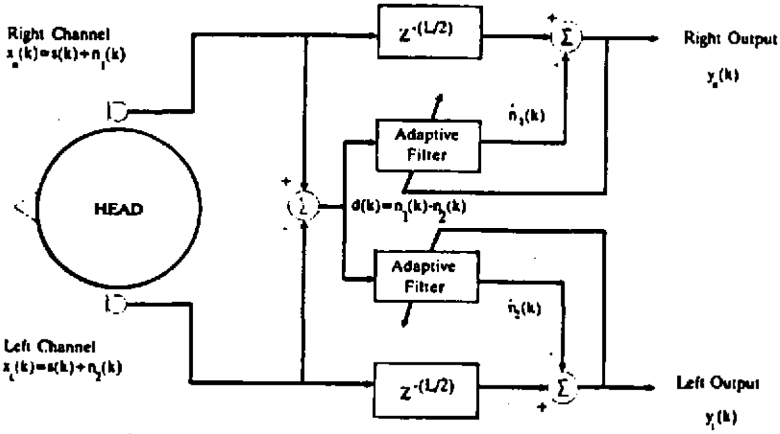

Figure 2: The block structure of the binaural adaptive filter.

R(z) is the transfer function of a comb filter. The number and positions of notches in the response depend upon the direction of the interference. In the region of these harmonically related notch frequencies little or no energy from the interference is present in the interference reference channel.

With reference to Figure 2 it may be shown that the adaptive filter transfer function, W(z), adapts to compensate for the effects of R(z), such that

$$W_R(z) = \frac{H_R(z)}{R(z)} \qquad\qquad W_L(z) = \frac{H_L(z)}{R(z)} \tag{2}$$

For white noise interference in the direction α the theoretical and experimentally obtained curves for W(z) are depicted in Figure 3. It may be seen that the error in the estimate of $R(z)^{-1}$ worsens close to the frequency of a null. This results in some leakage of interference which is highly distorted. The frequencies at which incomplete cancellation occurs are precisely those for which the relative phase of the interference arriving at each microphone is zero. This means that the residual interference at the output of both channels is in phase, and so appears to originate in the median plane, coincident with the target signal. Thus, a loss of binaural information in the leakage signal has occurred. However, deep notches outside the median plane only arise in R(z) for the simplified circumstances described above and are reduced in depth when head shadow effects are taken into account. To demonstrate this we have crudely modelled head shadow by introducing a 3dB attenuation of the interference signal received by the left channel. Figure 4 shows that the adaptive filter magnitude response is now smoother and non-zero at all frequencies, resulting in reduced interference leakage. The target-to-interference power ratio displays considerable improvement, increasing from about 12dB to about 20dB when the target and interference sources are of approximately equal power.

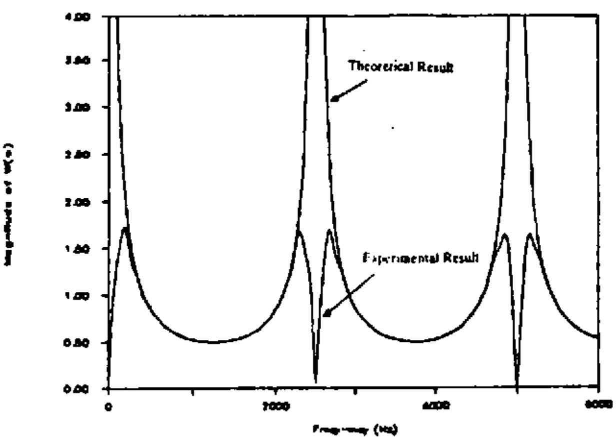

Figure 3: Theoretical and experimentally obtained frequency responses of one adaptive filter for a white noise interference source.

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4.2 The Adaptive Filters

It may be shown that the adaptive filter structure depicted in Figure 2 will cancel the interference source, N, by creating a notch in the polar response of the system in the direction of the source. Provided that the interference signal and target signal are uncorrelated perfect adaption is possible. In practice, some correlation between the two signals will exist and this will prevent full cancellation of the interference and also create some distortion of the target signal. The result is, once again, inaccuracy in the interference signal estimates, $\hat{n}_1(k)$ and $\hat{n}_2(k)$. Inaccuracies in estimating both the amplitude and the phase of the interference signal components will contribute to this error.

By way of illustration, let the component of the interference signal at frequency ω in the primary channel be $n_\omega(t)$, and the corresponding component of the interference estimate signal be $\hat{n}_\omega(t)$ such that

$$n_\omega(t) = \sin \omega t \tag{2}$$

and

$$\hat{n}_\omega(t) = K \sin(\omega t + \theta) \tag{3}$$

where K represents an amplitude error in the estimate and θ represents a phase error. Taking the difference $n_\omega(t) - \hat{n}_\omega(t)$ yields the cancellation error signal, $e_\omega(t)$, which may be expressed as

$$e_\omega(t) = (K^2 - 2K\cos\theta + 1)^{1/2} \sin\left\{ \omega t + \tan^{-1}\left[\frac{K \sin\theta}{(K\cos\theta - 1)}\right]\right\} \tag{4}$$

$e_\omega(t)$ represents the leakage signal at this frequency and quickly leads to the result that for a given peak error, $e_{\omega pk}$, the values of K and θ must lie within the limits

$$1 - e_{\omega pk} \leq K \leq 1 + e_{\omega pk} \tag{5}$$

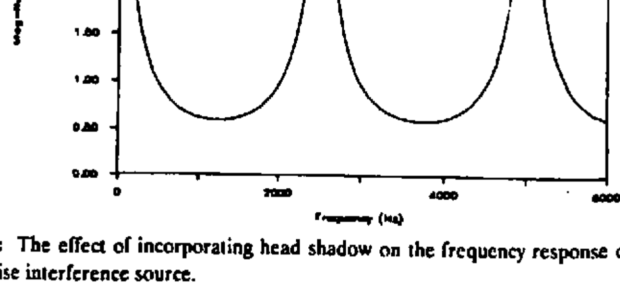

Figure 4: The effect of incorporating head shadow on the frequency response of one adaptive filter for a white noise interference source.

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and

$$-\cos^{-1}[(1-e_{\omega\,pk}^{2})^{1/2}] \leq \theta \leq \cos^{-1}[(1-e_{\omega\,pk}^{2})^{1/2}] \tag{6}$$

This creates a region of possible values for K and θ bounded by the contour corresponding to $e_{\omega pk}$, of the form shown in Figure 5.

For binaural separation between the target and the interference signals to be preserved the relative phase of the leakage components from each channel must be maintained. To determine this phase relationship we first relate the phase of the interference signal, $n_{\omega}(t)$, in one primary channel to the phase of the corresponding leakage component, $e_{\omega}(t)$, in the same channel. This may be obtained directly from equation (4).

$$\varphi = \tan^{-1}\left[\frac{K\sin\theta}{(K\cos\theta - 1)}\right] \tag{7}$$

This is plotted as a family of curves relating θ, K and φ in Figure 6. The figure shows that in any practical situation, that is with θ not exactly zero and K not exactly unity, the primary interference signal may undergo a phase shift, φ, of any value. The actual value of φ will depend upon the adaption history of the filter. Generally this results in arbitrary phase shifts at different frequencies, and a random-like phase relationship between the leakage signals from each channel. Hence a complete disruption of directional information based on phase may be anticipated from this filter structure.

4.3 Recovering Binaural Information

In principle the spread in the resultant phase shift of the partially cancelled interference signal may be reduced by deliberately introducing a small bias in both the phase and amplitude of the noise estimate. The shaded area of Figure 7(a) shows the possible values which may be assumed by θ and K. The effect of

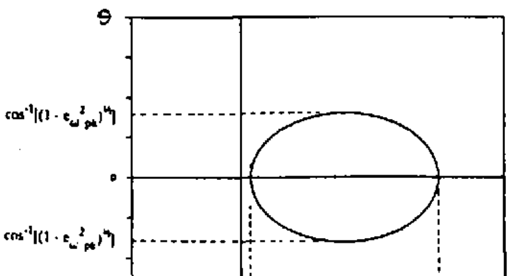

Figure 5: The contour of $e_{\omega pk}$, which bounds the possible values for K and θ.

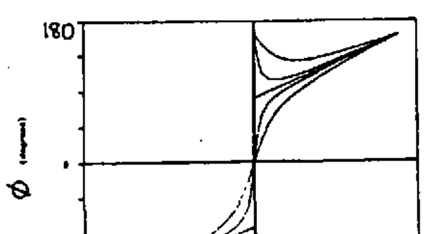

Figure 6: Family of curves relating cancellation amplitude and phase errors to leakage interference phase shift.

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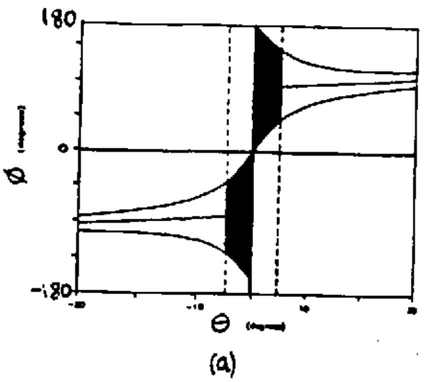
applying an appropriate phase offset is to limit the resultant phase shift spread in φ to 180 degrees, as shown in Figure 7(b). Reducing the amplitude of the noise estimate may further reduce the phase shift spread to no more than 90 degrees, as depicted in Figure 7(c). However, the cost of deliberately introducing cancellation error is increased interference leakage.

4.4 Implementation of the Filter Structure

The binaural adaptive filter has been computer modelled in non-realtime on a 33MHz PC AT-compatible machine with a 386 maths coprocessor. White noise, simulated speech and real speech have all been used as source materials. The algorithm itself has been coded using the C programming language and waveform analysis and listening tests have been conducted using the public domain SFS speech file manipulation and display package. Ten-second lengths of source material with a bandwidth of 4.5kHz were sampled at 10kHz.

5. EVALUATION OF PERFORMANCE

A number of informal listening tests have been carried out using the filter structure and confirm the predicted loss of binaural separation. In general, as the filters adapt the interference appears to move towards the median plane, even though the residual interference waveforms in each channel remain different.



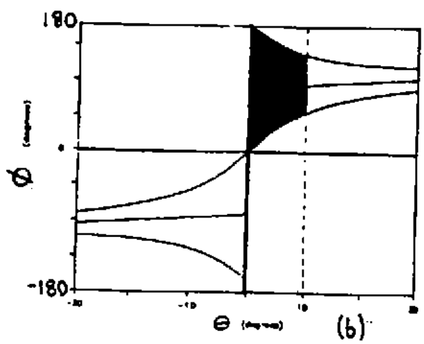
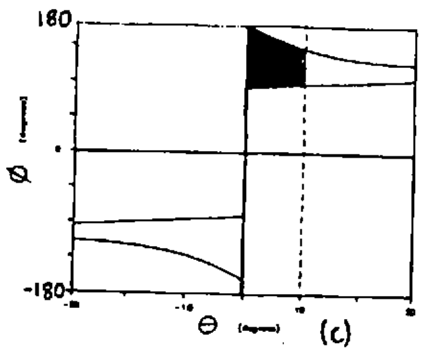


Figure 7: The maximum variation in leakage interference phase shift, φ, for no offset (a), phase error offset (b) and both phase and amplitude offset (c).



6. DISCUSSION

Our analysis and results show that interference reduction using beamforming is fundamentally incompatible with the aim of preserving binaural separation between attenuated interference sources and a target signal. We have suggested one means of trading interference attenuation for binaural separation. However, no tests have yet been undertaken to determine whether the benefits of this binaural separation are completely negated by the corresponding reduction in interference suppression. Other filter structures, which do not suffer from this fundamental difficulty are currently under consideration.

7. CONCLUSION

The binaural adaptive beamformer retains the characteristics of the adaptive beamformer described by Peterson *et al* and in particular increases the target to interference power ratio without distortion of the target signal. We have shown that incomplete cancellation of a single interference source is partially due to the comb filter response of the interference reference channel, and have explained why this is no longer a serious problem when head shadow is taken into account. A modified structure has been used to study the possibility of generating binaural output signals from the beamformer. Although subtractive cancellation destroys the directional phase characteristics of the residual interference we have gone on to show that binaural separation may be partially recovered at the expense of increasing the level of the attenuated interference. Other binaural filter structures are currently under consideration.

8. ACKNOWLEDGEMENTS

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9. REFERENCES

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