# AN ACTIVE HEADREST FOR PERSONAL AUDIO


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## ABSTRACT

This paper reports initial experiments on the practical implementation of a pair of seats with an active headrest designed to reproduce an acoustic signal in one seat, but attenuate it in the adjacent seat. This could be used to realise a "personal audio" system in each seat with reduced interference between seats. A self-contained unit consisting of a pair of loudspeakers has been incorporated into one wing of the headrest. The signal fed to the outer loudspeaker is filtered to maximise the acoustic contrast between the two seats, so that the sound in the adjacent seat is reduced to the greatest possible extent while maintaining the reproduced level in the first seat. The performance predicted through simulations is compared to the measured performance of a real-time implementation. Significant improvements in the acoustic contrast, greater than 20dB, are possible up to about 2kHz, above which the acoustic contrast between the seats is controlled by passive effects such as the directivity of the loudspeakers and the acoustic absorption of the headrest.


## 1. INTRODUCTION

Following on from previous work [4], where good performance was predicted from a simple theoretical model of an active noise control system for crosstalk cancellation between adjacent seats, this paper looks at the practical implementation of such a system and reports on the results achieved.
The focus of this work is the design and performance of the practical realisation of such a system, a diagram of which is shown in Figure 1, which used adjacent “primary” and “secondary” loudspeakers in one wing of a headrest on one seat. The secondary source signal is filtered by the filter *W*, which attempts to minimise sound pressure levels in one zone while maintaining constant levels in another zone. These two zones are termed here the ‘dark zone’ and the ‘bright zone’, respectively.

## 2. EXPERIMENTAL SETUP

The experimental apparatus consists of a pair of airline-style seats, each with headrests in which loudspeakers can be housed. One of the seats is designated to be the quiet seat (in the dark zone), and the adjacent seat to be in the bright zone. A diagram of the apparatus is shown in Figure 1, with the four microphones in the dark zone and four in the bright zone. A picture of the apparatus is also shown in *Figure 2*.

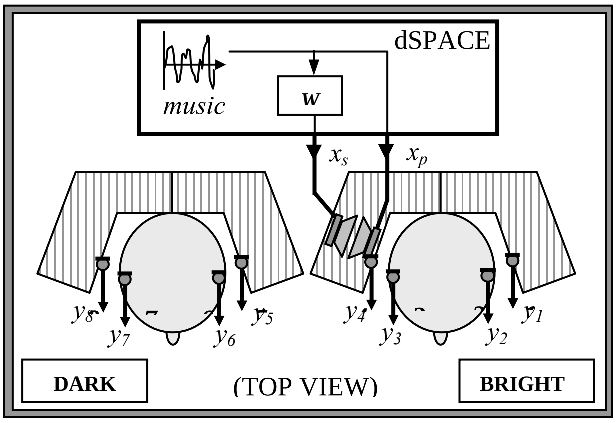


*Figure 1 – Diagram of experimental Setup*

*Figure 2 – Picture of experimental apparatus*

The eight microphones used in the experimental rig allow sampling of the sound field over a reasonable sized area, from which the measure of acoustic contrast can be found. Therefore contrast maximisation should lead to minimisation of sound levels over a *region* of space (the dark zone) as opposed to just a single point. Ideally this would mean the kind of reductions in sound level observed are experienced regardless of the position of the person’s head seated in the dark zone.

In this study, the back of one of the loudspeakers in the headrest of the seat in the bright zone is removed and another loudspeaker is clamped to the existing loudspeaker. The two 6" diameter loudspeakers are mounted face to face, but because they are each only 42mm thick, the thickness of the pair is thus 84mm, which can be readily incorporated into the headrest space. The additional loudspeaker in the headrest acts as the secondary source, which is required to be as close to the primary source as possible to yield the largest zone of silence. Only with such an arrangement does the task of minimising sound levels over all four microphones in the dark zone become feasible.

In order to predict the performance of such an active noise control system, frequency responses between both the primary and secondary loudspeakers and all eight microphones are measured. A simulation of the system using these frequency responses is then used to generate predicted spectra at all microphones from which the acoustic contrast can be calculated.

A real-time prototype of the 'active headrest' system has also been implemented, which can be configured to use both an on-line and an off-line approach. The off-line approach involves using the measured frequency responses to generate an ideal filter which results in maximum acoustic contrast when applied to the secondary source signal. The on-line approach uses an adaptive filter (adapted via the *filtered-x LMS algorithm*) where the error signal is designated to be microphone *6* (the left ear of the head in the dark zone). In this study the filtering was performed digitally in real-time by a dSPACE DS1103 DSP board. Such an arrangement has been chosen due to the ease of implementation, and the fact that there is great similarity in the predicted results for both approaches.

The results measured using the real-time system operating in a typical room environment are compared to the predictions using frequency responses measured in the anechoic chamber. This comparison serves to verify the measured results and also observe the effect that different environments have on the performance achieved.

## 3. THEORETICAL APPROACH

### 3.1. Acoustic Contrast

Here we have chosen to number the microphones in the bright zone as 1 to 4, and in the dark zone as 5 to 8 as shown in Figure 1. Our requirement to maximise acoustic contrast can be defined as:

$$C = \frac{\sum_{m=1}^{4} \left| y_m(j\omega) \right|^2}{\sum_{m=5}^{8} \left| y_m(j\omega) \right|^2} \tag{1}$$

where $y_m(j\omega)$ is the spectrum of the signal observed by microphone $m$.

The acoustic contrast is thus defined at each frequency as the sum of the modulus squared pressures at all microphones in the bright zone over the sum of the squared pressures in the dark zone.

We can represent the path (at a single frequency) from the primary loudspeaker to the microphones as $H_{pm}(j\omega)$ and from the secondary loudspeaker to these microphones as $H_{sm}(j\omega)$. We can lump together the responses from the bright zone as the matrix $\boldsymbol{H_b}$ and from the dark zone as $\boldsymbol{H_d}$. Likewise we consider the primary and secondary source signals in the frequency domain as $x_p(j\omega)$ and $x_s(j\omega)$ respectively, which are elements of the vector $\boldsymbol{x}$. The frequency domain representation of the signals observed at microphone $m$ are termed $y_m(j\omega)$, which can be grouped into vectors containing those in the bright zone, $\boldsymbol{y}_b(j\omega)$, and those in the dark zone, $\boldsymbol{y}_d(j\omega)$. Dropping the $(j\omega)$ term in the matrix representation for notational convenience, the relationship between the source signals and the observed signals can be written as:

$$\boldsymbol{y}_b = \begin{pmatrix} y_1 \\ y_2 \\ y_3 \\ y_4 \end{pmatrix} = \begin{bmatrix} H_{p1} & H_{s1} \\ H_{p2} & H_{s2} \\ H_{p3} & H_{s3} \\ H_{p4} & H_{s4} \end{bmatrix} \begin{pmatrix} x_p \\ x_s \end{pmatrix} = \boldsymbol{H}_b\, \boldsymbol{x} \tag{2}$$

$$\boldsymbol{y}_d = \begin{pmatrix} y_5 \\ y_6 \\ y_7 \\ y_8 \end{pmatrix} = \begin{bmatrix} H_{p5} & H_{s5} \\ H_{p6} & H_{s6} \\ H_{p7} & H_{s7} \\ H_{p8} & H_{s8} \end{bmatrix} \begin{pmatrix} x_p \\ x_s \end{pmatrix} = \boldsymbol{H}_d\, \boldsymbol{x} \tag{3}$$

Rewriting Equation (*1*) using these frequency domain terms, we get:

$$C = \frac{\left\| \boldsymbol{y}_b \right\|^2}{\left\| \boldsymbol{y}_d \right\|^2} = \frac{\boldsymbol{x}^H \boldsymbol{H}_b{}^H \boldsymbol{H}_b\, \boldsymbol{x}}{\boldsymbol{x}^H \boldsymbol{H}_d{}^H \boldsymbol{H}_d\, \boldsymbol{x}} \tag{4}$$

Since the action of the secondary source is required to not affect the levels in the bright zone, an expression for the contrast maximisation can be defined as:

$$\text{Minimise } \boldsymbol{x}^H \mathbf{H}_d^H \mathbf{H}_d \boldsymbol{x}$$
$$\text{subject to the constraint } \boldsymbol{x}^H \mathbf{H}_b^H \mathbf{H}_b \boldsymbol{x} = \mathrm{k} \tag{5}$$

Through the method of Lagrange multipliers, the solution of the optimisation stated in (*5*) can be shown to be equal to the eigenvector corresponding to the largest eigenvalue of the matrix $\boldsymbol{H}_b^H\mathbf{H}_b / \boldsymbol{H}_d^H\mathbf{H}_d$, i.e.

$$\boldsymbol{x}_{opt} = eig_{\boldsymbol{max}}\left(\frac{\boldsymbol{H}_b{}^H\boldsymbol{H}_b}{\boldsymbol{H}_d{}^H\boldsymbol{H}_d}\right) \tag{6}$$

This solution describes the amplitude and phase relationship between the signals emitted by the primary and secondary sources for maximum acoustic contrast at a single frequency. Once found, the optimal vector $\boldsymbol{x}$ can be used to generate a filter $W$ to be applied to the signal sent to the secondary source by calculating this relationship for each discrete frequency in the bandwidth of interest.

$$\boldsymbol{x}_{opt} = \begin{pmatrix} x_p \\ x_s \end{pmatrix} = \begin{pmatrix} x_p \\ Wx_p \end{pmatrix} = \begin{pmatrix} 1 \\ W \end{pmatrix} x_p \tag{7}$$

The impulse response of the corresponding FIR filter can then be found through the inverse Fourier Transform of this frequency domain representation of the filter.

### 3.2. Direct Cancellation

An alternative method of calculating the filter $W$, which is the approach used in the real-time implementation, is the direct cancellation of sound at a single point. If this point is chosen to be the left ear of the person seated in the dark zone (microphone *6*), very similar results are predicted to those achieved with the contrast method, despite the absence of a constraint on the levels in the bright zone.

This method has an adaptive solution, where the level of the signal from microphone *6* is minimised by designating this to be the error signal and applying the filtered-x LMS algorithm [3]. Such a method is simple to implement, and gives more robust performance in the face of changes in the frequency responses. However, it is still reliant on the accuracy of a model of the plant response (from the secondary source to microphone *6*) which should be measured prior to operation.

In terms of the measured frequency responses, the optimal solution at a single frequency using this approach is found from:

$$W = \frac{-H_{p6}}{H_{s6}} \tag{8}$$

## 4. RESULTS

Figure 3 shows the predicted performance calculated from frequency responses measured in an anechoic chamber. Results are shown for both approaches discussed, where the first approach uses a filter that maximises the acoustic contrast, and the second a control filter designed only to cancel the pressure at microphone 6 in the dark zone. The similarity in the performance of these two approaches suggests that the cancellation method, which is simpler to implement, performs almost as well as the ideal contrast maximisation.

Figure 4 shows the measured performance of the two approaches when implemented on a real system operating in a small room (6m x 2m x 2m in size). The contrast maximisation method here uses a fixed filter calculated off-line from measured frequency responses, while the direct cancellation employs an adaptive filter. The results are obtained through use of a spectrum analyser to measure the combined frequency responses of the primary source and the filtered secondary source arriving at each microphone.

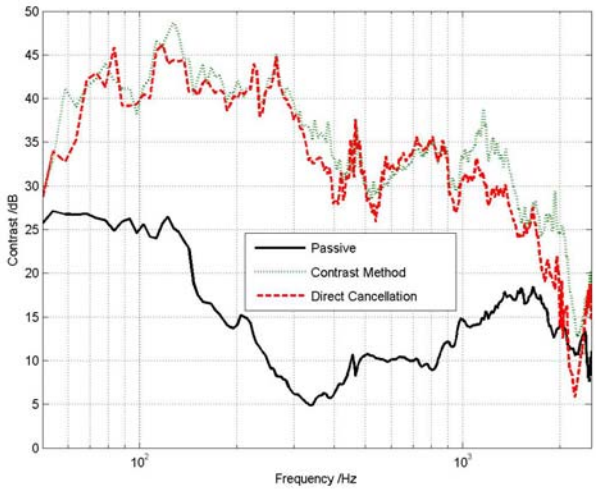


*Figure 3 – Predicted acoustic contrast using the contrast method and direct cancellation from measurements obtained in an anechoic chamber*

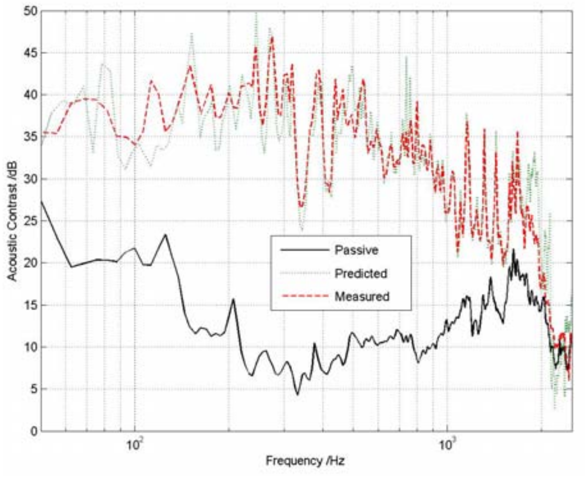


*Figure 4 – Predicted and measured acoustic contrast achieved using a real-time implementation of the (adaptive) direct cancellation method in a small room*

## 5. CONCLUSIONS

An improvement in acoustic contrast of approximately 20dB has been achieved from about 100 Hz to 1 kHz, with smaller improvements up to 3kHz. At higher frequencies the passive contrast increases due to the increased directivity of the primary source loudspeaker, and the greater absorption of the soft materials used in the construction of the headrest.

Comparing Figure 3 and Figure 4, the degree of acoustic contrast achieved is very similar. This serves to verify the results measured in the real-time implementation, and indicates that the performance is not unduly degraded by the acoustics of the room.

The subjective impression sitting in the quiet zone is that the direct sound from the adjacent seat has been effectively suppressed, and one is left with a much quieter and more diffuse sound.

## 6. FURTHER WORK

Performance may be further improved by investigating different mechanical arrangements and altering the design of the headrest/loudspeaker combination. Work is currently underway developing a BEM model of the headrest to facilitate investigation into this area.

Additional performance may also be gained by the use of a pre-filter operating on the signal sent to the primary source. For example delaying it by an integer number of samples may provide more freedom for the secondary source filter to cancel the sound, or removing particular frequencies where performance is poor may increase the perceived performance.

Future work will also involve some subjective testing in order to identify the band of frequencies to concentrate control effort to produce maximum perceived reduction. An important question will be the extent to which a subjectively acceptable solution can be obtained with a fixed filter, and whether the benefits of an adaptive arrangement outweigh the substantial increase in cost of such a system.

## 7. REFERENCES


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