Proceedings of The Institute of Acoustics

EQUALISATION OF ROOM ACOUSTIC RESPONSES OVER SPATIALLY DISTRIBUTED REGIONS

K.D. Farnsworth, P.A. Nelson and S.J. Elliott

Institute of Sound and Vibration Research,
University of Southampton, Southampton, SO9 5NH, England.

1. INTRODUCTION

The acoustic transmission path between a source and a receiver in an enclosure depends upon the geometry of the enclosure, the absorbtivity of the enclosure's boundaries, and the directivity/distribution of the source and receiver. In many instances the natural characteristics of the environment may render the 'quality' of the transmission path unacceptable. Since modern digital signal processing techniques have made the sophisticated treatment of signals possible, systems have been developed which attempt to 'dereverberate' audio signals for recorded speech, eg [1,2].

Great advantage could be gained if this processing could be performed in real time so that a signal could be dereverberated whilst it is being generated, for example for conference-communication systems, reverberation compensating public address, etc. However, this extension calls for a suitable method of designing the dereverberation filter. Furthermore, dereverberation attempts, to date, have yielded disappointing results, which although first regarded as simply due to inaccuracy in in digital modelling, are now thought to be the result of the highly sensitive dependence of a single transmission path impulse response on small changes in the parameters (eg. geometry/directionality) which can be both spatial and temporal. A method of producing an approximate inverse which is insensitive to these detailed changes could be an improvement over existing schemes.

2. DECONVOLUTION TECHNIQUES

The dereverberation is a deconvolution of the transmission path impulse response $R(z)$, which is performed by calculating an inverse filter $R^{-1}(z)$ such that

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

and a delta function impulse response results; in this case $R^{-1}(z)$ is the unconstrained Weiner filter [3]. Most practical room impulse responses are mixed phase [4] and thus the realisation of a stable inverse filter may not always be possible. In order to produce a stable, causal approximation to the inverse of these, special processing, rather than direct inversion, is needed. It has been found [5] that the most promising technique is least squares inversion. A highly efficient version of this is the L.M.S. algorithm after Widrow [3], which has been used [6] successfully invert room-like impulse responses by "post-processing" using the scheme outlined in Figure 1(a). This algorithm is adaptive and so can take account of time dependence of the impulse sequence to be inverted, if it varies slowly compared with the convergence time of the algorithm (usually about 1-2 seconds). Thus sensitivity to changes in parameters in time can be dealt with.

The extension of the L.M.S. algorithm to the so called 'filtered-X' version [7] allows the room transmission path to become part of the feedback in the system as shown in Figure 1(b). This enables, in principle, real time inversion through "pre-processing" for the first time.

3. RESULTS OF A COMPUTER SIMULATION

The performance with various impulse sequences and the dependence on computational parameters of both L.M.S. and filtered-X algorithms have been investigated and recommendations for their optimal use in dereverberation have been made [6]. Figure 2 shows the results of the use of these techniques to approximate the inverse of a given impulse response.

In order to assess the sensitivity of the inverse to spatial displacement, the effect of the inversion of the response between a source and a point $\underline{x}$ on the impulse response of the transmission path between the source and neighbouring points $\underline{y}_i$ has been investigated. In this study, the image-expansion based computer model of a rectangular room (after Allen & Berkley [8]) was used to produce impulse sequences for source-to-$\underline{x}$ and source-to-$\underline{y}_i$ transmission paths which were convolved digitally with the inverse filter produced by the filtered-X algorithm. Preliminary time and frequency domain results are shown in figs 4 and 5.

Figure 4 shows the result of applying the inverse of the transmission path to $\underline{x}$ to points spaced at 13.6 cm along a line parallel to the x_1-axis as depicted in fig 3. These time histories clearly show how the quality of the dereverberation deteriorates with distance from the point for which the inverse was designed. Fig 5a, 5b show a comparison between the net frequency response function of the transmission path before and after pre-filtering with the inverse filter at the point $\underline{x}$ for which this filter was designed; an improvement in the 'flatness' is apparent, but narrow band "dropouts" have persisted after dereverberation. Fig 5c to f show the frequency response functions corresponding to the impulse responses at $\underline{y}_1$ to $\underline{y}_5$ shown in fig 4 and the gradual deterioration can be seen to correspond with a gradual departure from a delta-function like time history. Also, the deviation of the responses in 5c to g from that of 5b apparently increases with increasing frequency as well as the position of $\underline{y}_i$ (which increases in 13.6 cm steps from 5b to 5f). These phenomena are currently the subject of more detailed investigation.

4. CONCLUSION

A new method for dereverberating a room transmission path has been shown to be effective in computer models. Various assessments of performance have been carried out and the viability of in-situ real time pre-processing has been demonstrated in principle. The quality of the inversion at one point when applied to a neighbouring point has received preliminary study.

REFERENCES

EQUALISATION OF ROOM ACOUSTIC RESPONSES OVER SPATIALLY DISTRIBUTED REGIONS

[1] J. Mourjopoulos, 'The removal of room reverberation from signals', PhD Thesis, I.S.V.R., Southampton University (1985).

[2] J.B. Allen, D.A. Berkley, J. Blauert, 'Multimicrophone signal processing technique to remove reverberation from speech signals', J.A.S.A. 62(4), (1977).

[3] B. Widrow et al, 'Adaptive noise cancelling: principles and applications', Proc. IEEE, 63, 1692, (1975).

[4] S.T. Neely, J.B. Allen, 'Invertibility of a room impulse response', J.A.S.A. 66(1), 165, (1979).

[5] J. Mourjopoulos, P.M. Clarkson, J.K. Hammond, 'A comparative study of least-squares and homomorphic techniques for the inversion of mixed-phase signals', IEEE Int. Conf. on Ac. Sp. & Sig. Proc., 1858, (1982).

[6] K. Farnsworth, 'Equalisation of room acoustic responses over spatially distributed regions', MSc Thesis, I.S.V.R., Southampton University, (1986).

[7] B. Widrow, S.D. Stearns, 'Adaptive signal processing', Prentice-Hall, (1985).

[8] J.B. Allen, D.A. Berkley, 'Image method for efficiently simulating small-room acoustics', J.A.S.A. 65, 943, (1979).

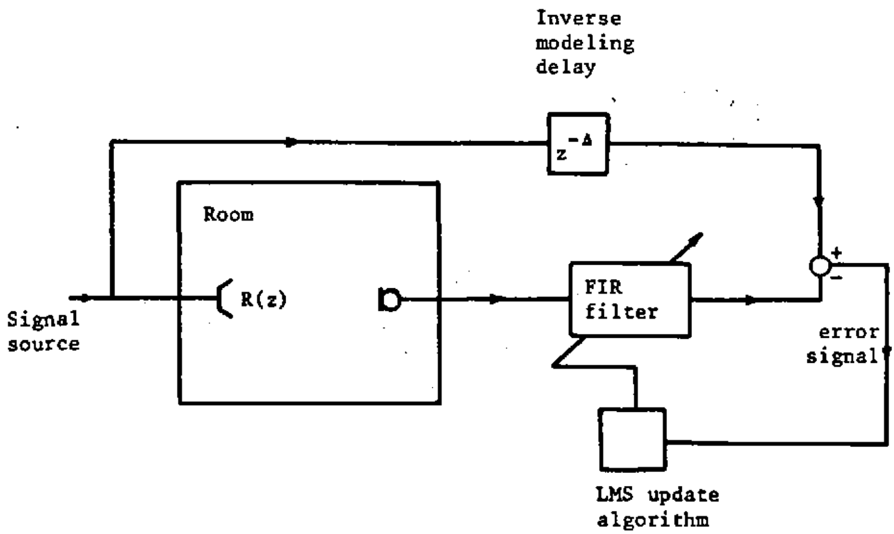


FIGURE 1(a) Post processing arrangement for the dereverberation of the source-receiver transmission path R(z). Note that the FIR filter is adaptively adjusted in order to produce a net transmission path which best approximates a pure delay. The delay Δ samples is chosen to be that which minimises the energy of the error signal [7].

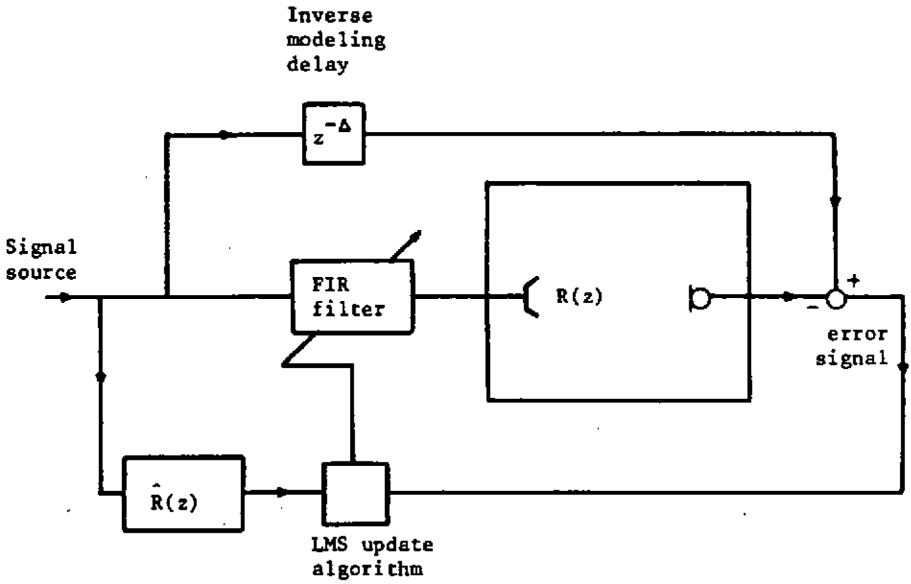


FIGURE 1(b) Pre-processing arrangement for the dereverberation of the source-receiver transmission path R(z). In this case, the "Filtered-X" LMS update algorithm requires the input signal to be passed through a filter which is an estimate $\hat{R}(z)$ of the source-receiver transmission path [7].

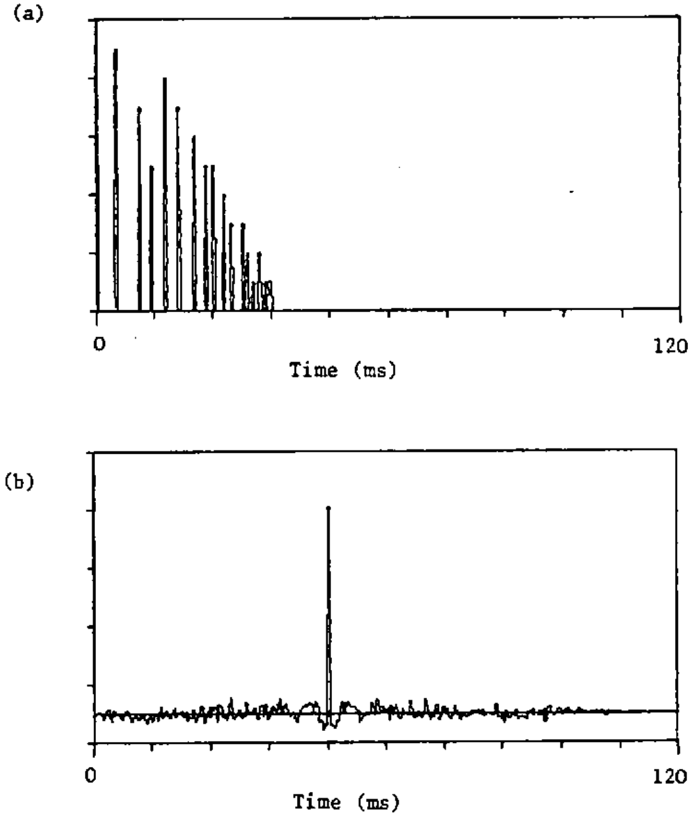


FIGURE 2 Showing (a) an arbitrary computer simulated impulse response of the source-receiver transmission path and (b) the impulse response of the net transmission path produced using the post processing arrangement of Figure 1(b).

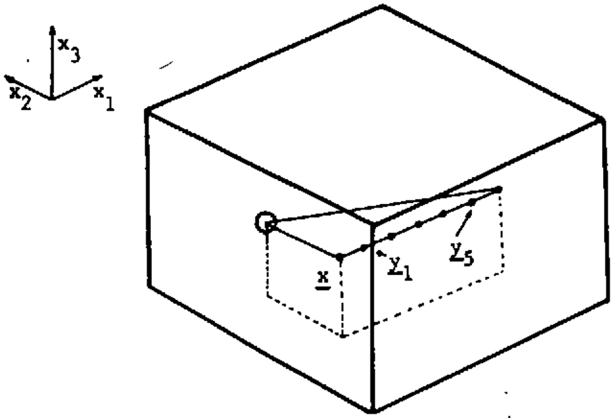


FIGURE 3
The room synthesised by the computer model for the production of the transmission path responses from the source to receivers at $\underline{x}$ and $\underline{y}_1$ to $\underline{y}_5$. The sequence of receiver positions are spaced 13.6 cm apart. The room volume is approximately $50m^3$ and all the wall surfaces have an absorption coefficient of 0.4.

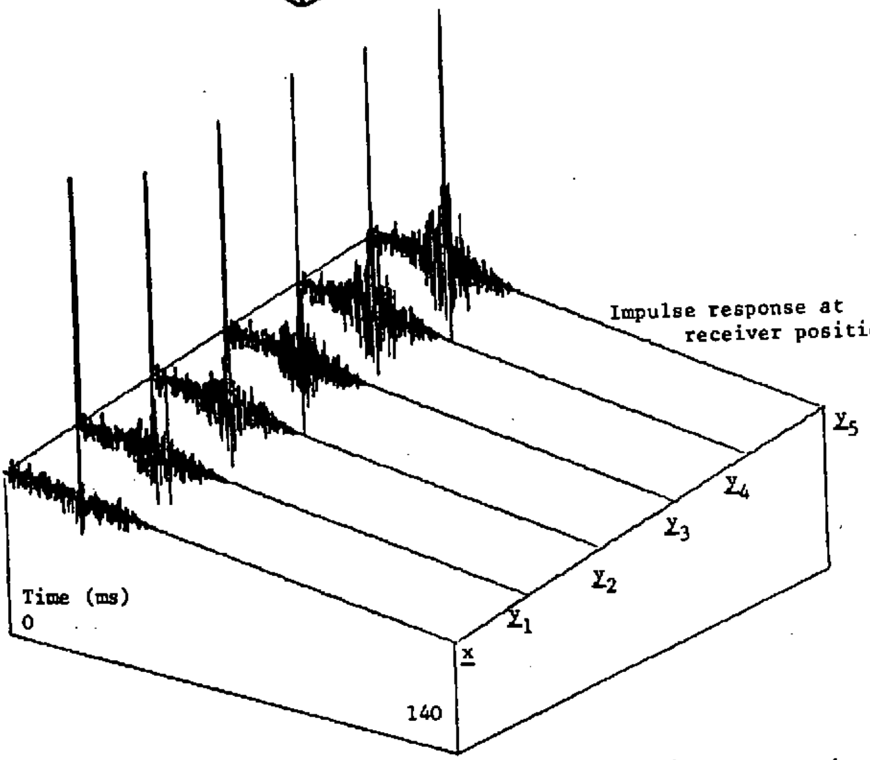
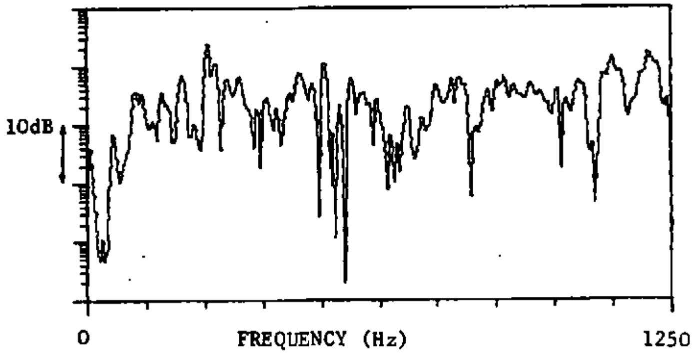


FIGURE 4 Impulse response of net transmission path to receivers at $\underline{x}$ and $\underline{y}_1$ to $\underline{y}_5$ when a "pre-processing" filter is used to dereverberate the signal at $\underline{x}$.

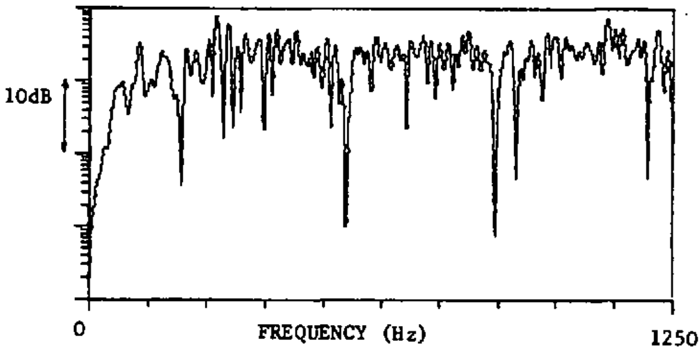
EQUALISATION OF ROOM ACOUSTIC RESPONSES OVER SPATIALLY DISTRIBUTED REGIONS

FIGURE 5

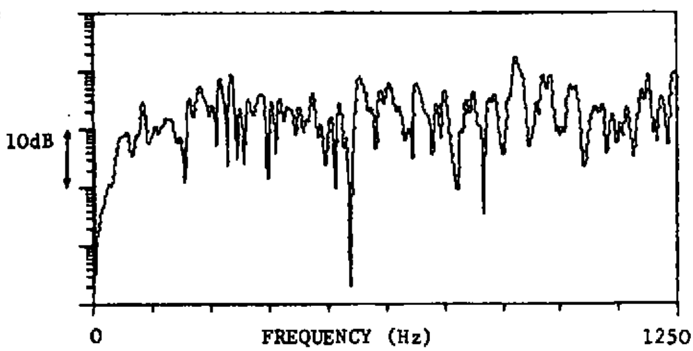
(a) Frequency response of transmission path to $\underline{x}$ before dereverberation.



(b) Frequency response of transmission path to $\underline{x}$ after dereverberation

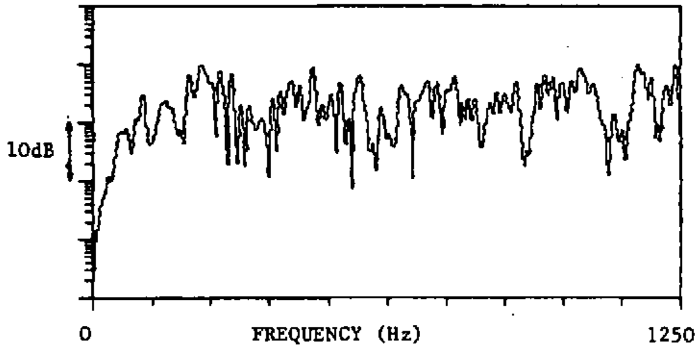


(c) Frequency response of transmission path to $\underline{y}_1$ after dereverberation at $\underline{x}$.

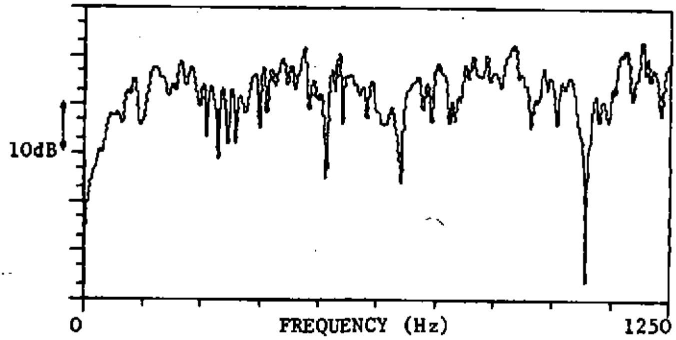


EQUALISATION OF ROOM ACOUSTIC RESPONSES OVER SPATIALLY DISTRIBUTED REGIONS

(d) Frequency response of transmission path to $\underline{y}_2$ after dereverberation at $\underline{x}$.



(e) Frequency response of transmission path to $\underline{y}_4$ after dereverberation at $\underline{x}$.



(f) Frequency response of transmission path to $\underline{y}_5$ after dereverberation at $\underline{x}$.