APPROACHES TO SINGLE MICROPHONE REVERBERATION MEASUREMENT

J A S Angus University of Salford, Salford, U.K.

1 INTRODUCTION

Although reverberation time can be measured relatively easily from a known sound source in the room, such as a shot, random noise, swept sine wave, or even music, these two point methods are often inappropriate if the room is occupied. Furthermore, even when unoccupied the need to have a separate excitation source within a room can prove both inconvenient and problematical. Single microphone measurements are an ideal method for measuring Reverberation time and several methods[1,2,3] have been used to try to achieve this. This paper presents an alternative method based on adaptive filtering.

THE NATURE OF REVERBERATION

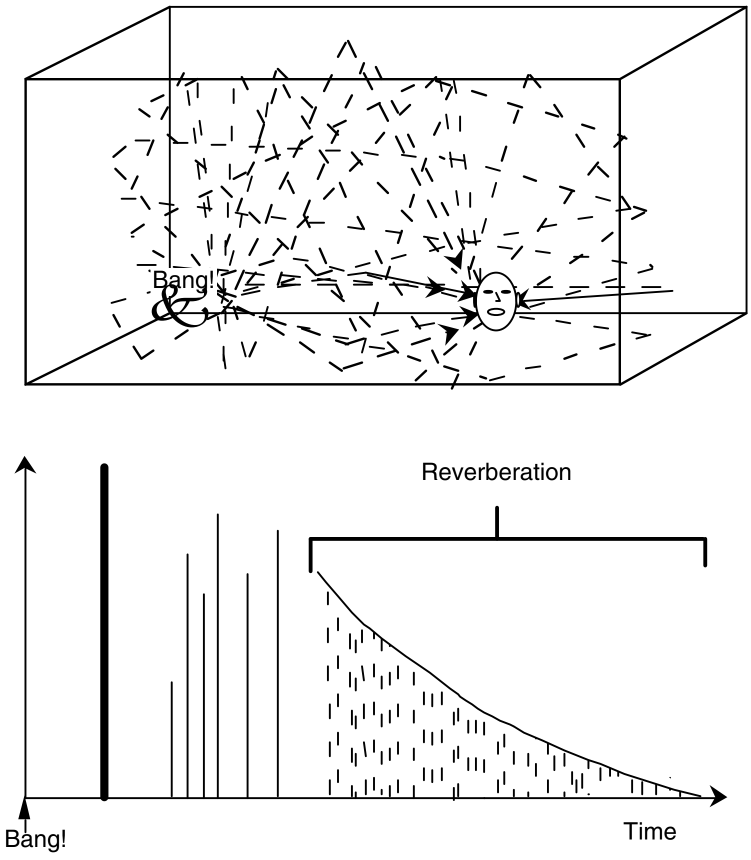

Figure 1 The reverberant sound in a room.

The reverberant sound in a room is shown in figure 1 after some time after its generation the sound has been reflected many times and is arriving at the listener from all directions, as shown in figure 1. Because there are so many possible reflection paths each individual reflection is very close in time to its neighbours and thus there is a dense set of reflections arriving at the listener. It is this part of the sound that is called reverberation and it is desirable as it adds richness to, and supports, musical sounds. Reverberation also helps integrate all the sounds from an instrument so that a listener hears a sound, which incorporates all the instrument's sounds, including the directional parts. In fact we find rooms that have very little reverberation uncomfortable and generally do not like performing music in them, it is much more fun to sing in the bathroom compared to the living room! The time taken for reverberation to occur is a function of the size of the room and will be shorter for smaller rooms, due to the shorter time between reflections. In fact the time gap between the direct sound and reverberation is an important cue to the size of the space that the music is being performed in. Because some of the sound is absorbed at each reflection it dies away eventually. The time that it takes for the sound to die away is called the reverberation time and is dependent on both the size of the space and the amount of sound absorbed at each reflection. The important thing to note is that it takes some time for the reverberation to build up. The early part of the room response is dominated by the direct sound and early reflections and it is only after 80ms or more that there is any significant reverberation component.

2 ADAPTIVE FILTERING AS A REVERBERATION TIME ESTIMATOR.

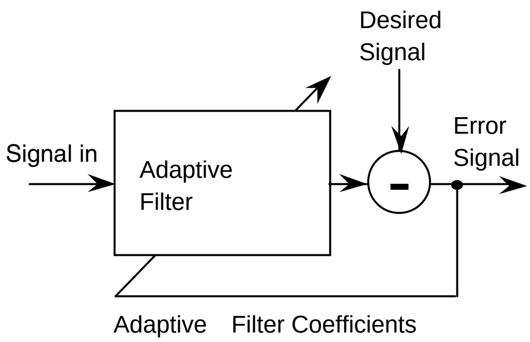

Figure 2 An Adaptive Filter.

Figure 2 shows a basic adaptive filter[4] in which the desired signal is subtracted from the filter's output and the filter's coefficients are adjusted to minimise the error. If the signal that is input to the filter is the sound source in the reverberating environment, and the desired signal is the reverberant sound field, the filter coefficients will adapt to become equal to an estimate of the impulse response of the room. This is the basic idea behind some two-point measurements, as shown in figure 3.

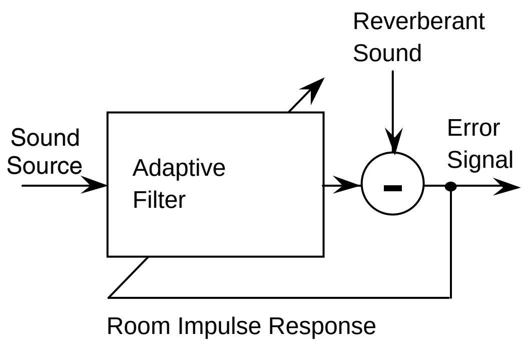

Figure 3 An Adaptive Filter reverberation measurement.

3 SINGLE POINT REVERBERATION TIME ESTIMATION.

However, if one uses this structure to do a single point measurement, as shown in figure 4, it doesn't work! This is because the both the input to the filter and the error signal are identical and therefore the filter will set itself up to be a '1' followed by lots of zeroes.

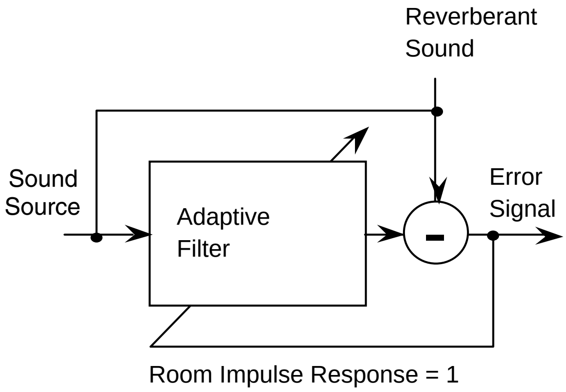

Figure 4 The wrong way to do single point estimation

However, remember that the reverberation does not really begin until over 80ms after the direct sound. Thus we can incorporate a delay between the input to the filter and the desired signal, as shown in figure 5.

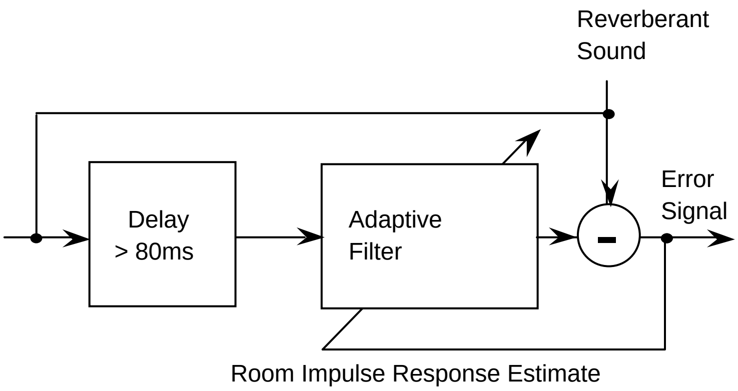

Figure 5 A possible way to do single point estimation

In this arrangement the filter 'decorrelates' the two inputs, and the filter coefficients can be free to adjust themselves according to the rooms impulse response. However this method is not without problems, long steady tones, like organ notes, will not decorrelate and the filter will misadapt.

4 POSSIBLE IMPROVEMENTS TO THE TECHNIQUE

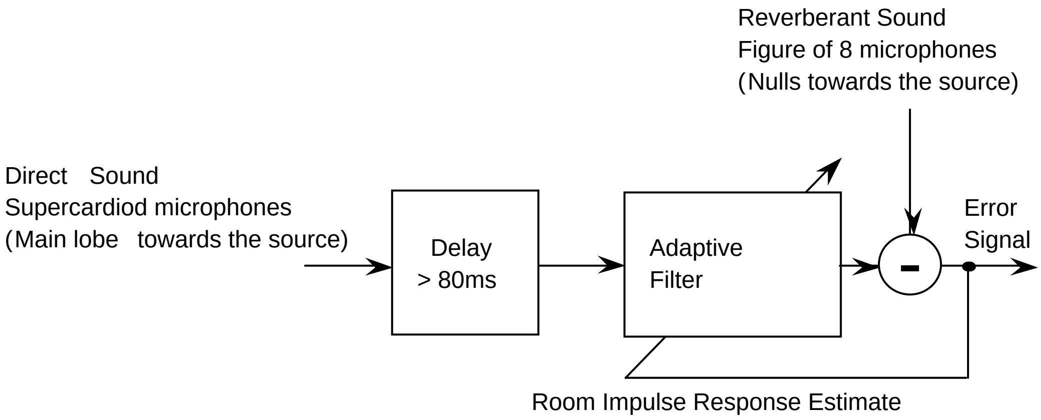

Figure 5 A possible improvement to single point estimation

One way to improve the estimation is to use co-located microphones to provide some additional separation between the source and the reverberation, as shown in figure 5. By using a hypercardioid microphone pointing at the source and figure of 8/cardioids, such that the nulls in the polar responses are pointing towards the source the difference between them is maximised. A disadvantage of this method would be the matching of the microphones and the need to correctly orientate the instrument.

5 CONCLUSION

A method for single point reverberation time measurement has been proposed, but like any such method it is critically dependent on the source material.

6 REFERENCES

1. T.J. Cox, F. Li and P. Darlington., 'Extracting room reverberation time from speech using artificial neural networks', J.Audio.Eng.Soc. 49(4) 219-230. (April 2001).
2. L. Courvreur, C. Ris, C. Couvreur: Model-based blind estimation of reverberation time: Application to robust ASR in reverberant environments. Proc. Eurospeech, 2001.
3. R. Ratnam, D. L. Jones, B. C. Wheeler, W. D. O'Brien, Jr., C. R. Lansing, A. S. Feng: Blind estimation of reverberation time. J. Acoust. Soc. Am. **114** (2003) 2877–2892.
4. B. Widrow, S. Stearns Adaptive Signal Processing, Prentice Hall, (1985), ISBN 0130040290