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A PUBLIC ADDRESS TIME ALIGNMENT INSTRUMENT

James A S Angus and Robert M Claxton

Signal Processing: Voice and Hearing Research Group
Department of Electronics, University of York, York YO1 5DD

INTRODUCTION

In many concert venues and auditoriums electronic sound reinforcement is necessary to obtain adequate loudness and a good distribution of sound throughout the audience. Often the loudspeakers have to be placed at a number of points within the auditorium to achieve the required results. In many cases there will be a number of places within the audience that are reached by the direct sound from more than one loudspeaker.

The human auditory system will perceive two sounds whose spectrum and time envelopes are essentially the same as being separate if they arrive over 35ms apart. If they arrive within 35 ms they are not heard as separate sounds by the ear, instead they tend to reinforce the sounds heard.

If a listener can hear direct sound from two loudspeaker sources that have a path length difference between themselves and the listener of more than around 10 metres then the listener will begin to perceive echoes in the sound they hear. As the path length difference increases and hence the time delay between arrival times of signals from the speakers increases then the intelligibility of the sound will decrease. Clearly this will spoil the listening enjoyment of the audience.

Overcoming the problem firstly involves good auditorium design and well-planned loudspeaker locations. However, problems may persist and it is then that other solutions must be sought.

To overcome the problems illustrated above, the signal to a particular loudspeaker or group of loudspeakers can be delayed, thereby delaying the arrival time of the sound at the listener so that it arrives synchronously with other speaker locations. Figure 1 illustrates a situation where time delay units are necessary to achieve a good distribution of sound throughout the audience whilst maintaining the intelligibility of the sound and preventing echoes.

In this auditorium a central loudspeaker cluster is used to cover the majority of the audience. However, listeners in the under-balcony area are blocked from this central cluster by the overhang of the balcony. A second loudspeaker system is required to cover this area of the audience. There will be a zone of overlap in the audience where people will first hear sound from the under-balcony loudspeaker, and then some time later, sound from the central cluster. This will lead to the problems of echoes and intelligibility mentioned earlier. A time delay unit of some sort is needed to delay sound going to the under-balcony loudspeaker so that it arrives at approximately the same time as the sound from the central cluster.

A PUBLIC ADDRESS TIME ALIGNMENT INSTRUMENT

Setting Delay Times

Having chosen audio time delay units to eliminate echo problems, sound engineers have a number of options open to them to set the required delay time for a particular pair of loudspeakers.

1 **By ear.** The sound engineer could estimate a value for the required delay time needed to align the arrival times of sound from two loudspeakers, or loudspeaker clusters, and simply listen to the results. Further adjustments could then be made until the engineer thought that the two loudspeakers were sufficiently in line.

This is a subjective method that may be possibly justifiable because as long as it sounds right there cannot be too much wrong. However the process of checking the alignment by ear is time consuming and difficult, especially in a reverberant environment. The job is made even harder given the normal situation in an auditorium or concert venue when a band is 'setting up' for a performance; ie numerous people and crew racing around building the stage, lighting rig and generally making noise!

2 **Using floor plans and measurements of loudspeaker placements to calculate the required delay time.** Given the knowledge of the speed of sound in air and the path length difference between the loudspeakers and the point of interest within the auditorium the required time delay can be easily calculated.

Despite the apparent simplicity of this method there are a number of problems. Floor plans are rarely adhered to accurately and cannot be absolutely relied upon. Making measurements from loudspeaker clusters, which may very well be inaccessible places, to various points within the auditorium presents a difficult and time consuming problem. Even without any of these problems to contend with, the speed of sound in air is not constant. It varies with temperature, pressure and humidity. Therefore, the calculated time delay will only be accurate for a given pressure and temperature. Any large deviations from the values used, for example, in temperature, will reduce the usefulness of the calculations.

3 **Using existing purpose built time delay measuring devices**. Much of this type of equipment is computer based, for example the TEF 20 Analyser which is capable of a large range of time based measurements. These devices are quite capable of performing the job of aligning loudspeaker clusters but do have disadvantages. Costs are high not only for the package itself, but also because of the need for a dedicated computer. The actual use of the systems is relatively complicated as are the display presented to the user. In addition the measurements are not made 'live'. In other words, a measurement of delay time is made, the delay to the loudspeaker adjusted as necessary and then another measurement is made to check the adjustment.

A PUBLIC ADDRESS TIME ALIGNMENT INSTRUMENT

AN ALTERNATIVE SOLUTION

Clearly there is a need for a cheap, simple, and easy to use meter for time aligning loudspeaker clusters within an auditorium or concert venue.

Requirements
Such a meter should ideally meet the following requirements (not necessarily in order of importance)

1 It should be able to measure the relative delay in arrival times between two audio signals from loudspeakers.

2 It should display the measured delay time simply and with resolution steps of 10ms or 1ms, in order to be compatible with digital delay units.

3 It should update the display quickly so any adjustments of delay times will be quickly visible. This implies an update rate of 5Hz to 10 Hz.

4 It should operate over a range of relative time delay equivalent to at least a 30m path length difference between loudspeaker clusters and the point of interest in the auditorium. This implies a greater than 100ms relative delay measurement capability.

5 It should be able to work in an environment where some extraneous noise is present without difficulty.

6 It should have an operating sound pressure level from the loudspeakers of approximately 70dB. That is, comparable with the level of normal speech at a distance of a metre.

7 It should use standard audio inputs, outputs and casing. This would make the integration of the meter into existing professional sound systems simple.

8 It should be marketable for around £200.

METHOD OF OPERATION

The Live PA Time Alignment Meter works in the following manner (see Figure 2). Two test signals are output from the meter and are connected to the mixing desk of the sound system. Each of the signals is then patched through to the loudspeaker or loudspeaker cluster that is under test. A single microphone, which is connected to the input of the meter, is then located at the point of interest within the auditorium. This could possibly be a radio microphone. The meter generates the two test signals to be sent to the loudspeakers and received at the microphone. The meter then gives a display of the relative delay in arrival times from the two loudspeakers under test. The time delay unit on one or other of the feeds to the loudspeakers can then be adjusted

A PUBLIC ADDRESS TIME ALIGNMENT INSTRUMENT

until there is no relative delay or as much delay as the sound engineer feels is required for the best results.

TECHNIQUE OF OPERATION

In order to achieve the above operation we need two test signals which

a) can measure delay easily

b) can be distinguished from each other

c) can provide some rejection of environmental noise.

Ideal signals for this purpose are maximum length sequences similar to those used in the MLSSA system. These have the right properties for measuring delay in noisy environments. In addition Gold [1,2] has shown that it is possible to find preferred pairs of sequences that have low cross correlation and so can be easily separated.

The block diagram of the system is shown in Figure 3. It consists of two m-sequence generators that provide the two test signals. The common signal received from the test microphone is correlated by two correlators to measure the relative delay between the two signals. These are then displayed on a simple single bar display (Figure 4) in which the correlation between the two signals is represented as brightness on the LED display. Each LED in the display is a tri colour device and the two sequences control the green and red parts of the LED. Thus the absolute delay for each sequence is shown as a red or green dot. By varying the delay units delay the relative delay can be adjusted to zero. Under this condition the red and green parts of the same LED light up and so show yellow.

RESULTS

To test the system, we examined the correlator output under various conditions.

a) No interference or filtering of the test sequence.

b) The two sequences summed together but still unfiltered.

c) The sequence low pass filtered with a time constant of 10 ms (equivalent to a cut off frequency of 16Hz).

d) The sequence high pass filtered with a cut-off frequency of 16Hz.

The correlation peaks obtained are shown in Figure 5a to 5d respectively. As one can see, the presence of the second sequence [Figure 5b] and the effects of high pass filtering [Figure 5d] are minimal. Even the effect of an extreme low pass filter [Figure 5c] does not entirely remove the correlation peak, although it reduces it considerably.

As a final test the two sequences were passed through two slightly different high pass filters and then summed before correlation. The resulting output from the integrator is shown in Figure 6 Circled in the figure are some peaks in the signal. These are due to cross-correlations between the sequences. The peak height of these cross-correlations was measured as approximately 4.5v. This compares with a correlation peak height of 27.53v. The ration of these heights is $20 \log_{10} \frac{27.53}{4.5} \approx 15.7$dB. This agrees closely with the theoretical cross-correlation performance of $\approx$ 17.5dB.

CONCLUSION

We have described a technique that will allow the delays in Public Address System to be set up easily. Further work is needed to provide a product that would have general utility in setting up the delays in Public Address Systems.

REFERENCES

1 Gold, R, "Optimal Binary Sequences for Spread Spectrum Multiplexing," *IEEE Transactions on Information Theory*, **IT-13**, pp 619-621

2 Gold, R, "Maximal Recursive Sequences with 3 Valued Cross-Correlation Functions," *IEEE Transactions on Information Theory*, **IT-14**, pp 154-156

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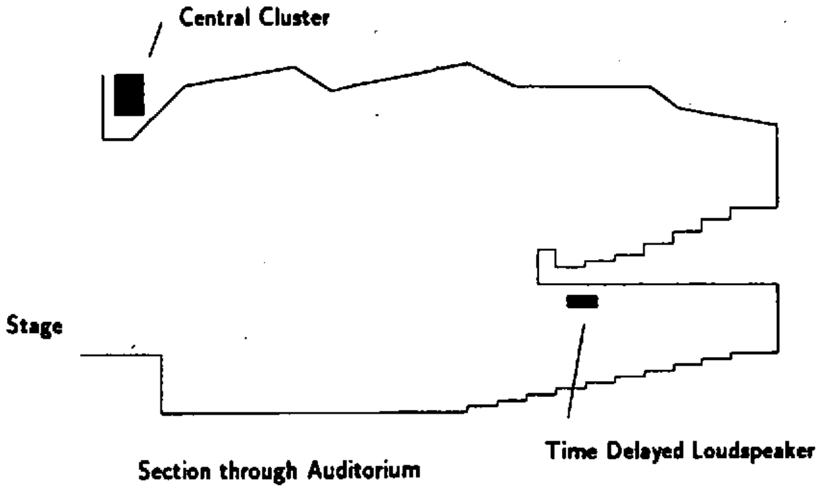

Figure 1: A Sound Reinforcement System Requiring Time Delay Units

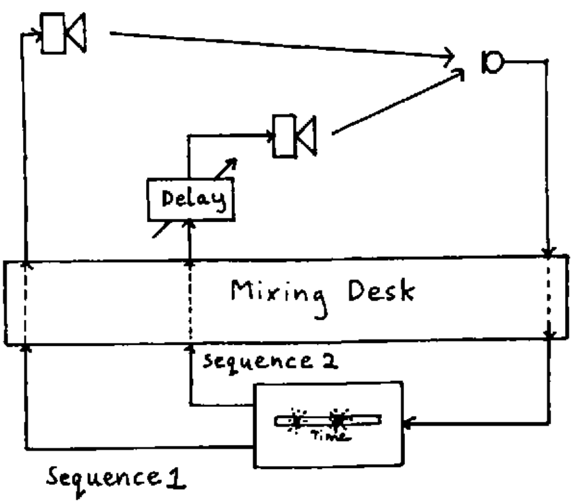

Figure 2: The PA Time Alignment Meter Within a System

A PUBLIC ADDRESS TIME ALIGNMENT INSTRUMENT

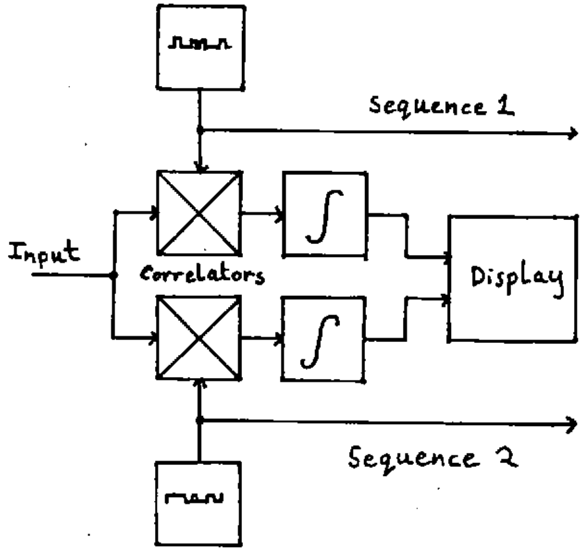

Figure 3: The PA Time Alignment Meter's Block Diagram

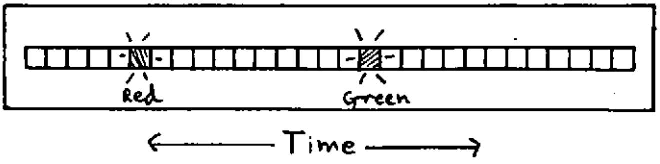

Figure 4: The PA Time Alignment Meter's Display Method

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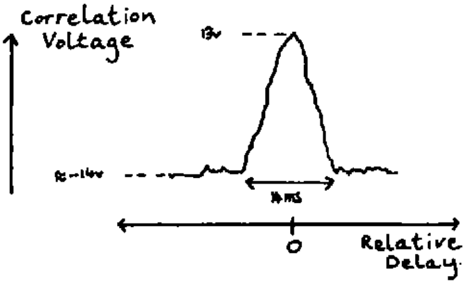

Figure 5a: Correlator Output With No Interfering Sequence

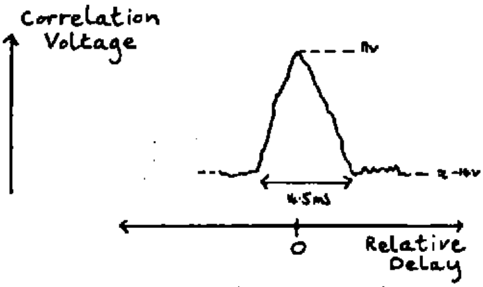

Figure 5b: Correlator Output With an Interfering Sequence

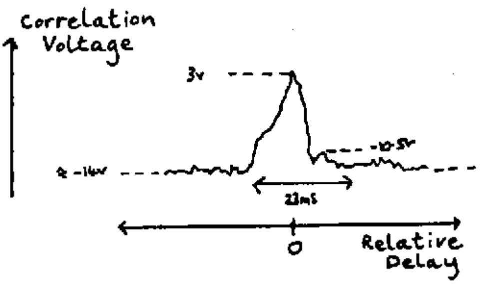

Figure 5c: Correlator Output for a Low-Pass Filtered Sequence

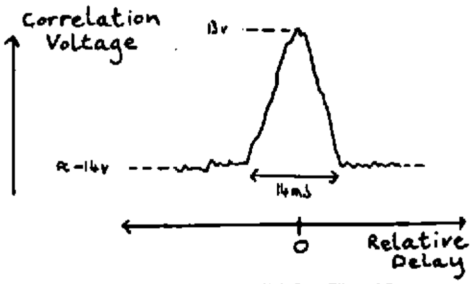

Figure 5d: Correlator Output for a High-Pass Filtered Sequence

Figure 5: The Correlator's Output Under Various Conditions

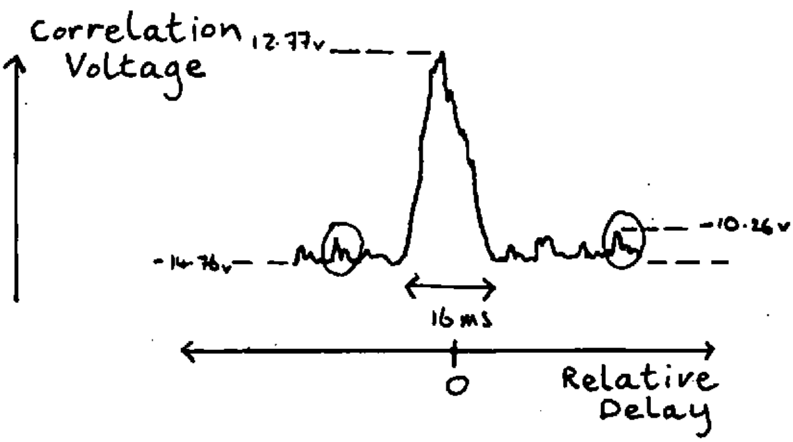

Figure 6: Integrator Output for Two Sequence Correlation