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A ROOM SIMULATION SYSTEM FOR ARCHITECTURAL EDUCATION

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Introduction

'Room Simulation' is the name given to the technique of using electronic delays and an array of loudspeakers to reproduce, in an anechoic space, the pattern of reflections typical of an auditorium. This technique has been used for many years in acoustics research and has yielded important information concerning the importance of early reflections and the shape of the reverberant decay curve (1-4).

Concepts established in the course of this work are now taught in courses on architectural acoustics.

It was recently suggested that room simulation might be employed as a design tool as well as a research technique (5). Developments in electronics have suggested to the authors that application of room simulation can now be extended to the teaching of acoustics in schools of architecture.

Amongst the parameters that student architects could experience, given a readily available room simulator, are: (i) the influence of echoes at various strengths and delay, relative to the direct sound (following the work of Haas (1))(ii) the effect of reverberation time on speak intelligibility (iii) the importance of the direction from which echoes arrive on auditorium quality (iv) the influence of the ratio of early to late energy on audiotrium quality etc.

This paper sets out to review the topic of room simulation and to assess the possibility of developing a low cost system suitable for use in teaching and possibly preliminary research programmes.

Delay Systems for the Generation of Discrete Echoes

A delay system for room simulation should satisfy the following requirements (i) be capable of introducing a delay of the order of tens of milliseconds (ii) have a uniform frequency response (iii) be stable (iv) introduce a negligible amount of noise (v) introduce a neglibible amount of distortion.

Two approaches can be distinguished, the first consisting of systems which work with a continuous signal and the second consisting of systems which work on sampled data.

Continuous signals can be delayed using multiple active filter sections having linear phase characteristics or tape loops having replay heads at different separations along the tape. The former can only provide delays of the order of 100 μsec and hence are not suitable for this application. Tape loop systems have been extensively used in the past but suffer from a number of drawbacks such as (i) lack of dynamic range (ii) degredation of performance in use as the tape oxide coating is destroyed or deposited on the head (iii) wow and flutter (iv) large physical size (v) require skill in setting up (vi) require skill in maintenance. The last two characteristics make them unsuitable for use by architecture students whose technical expertise is generally limited.

Delay systems working on sampled data are commercially available, being intended for use in high quality speech reinforcement systems. One such system consists of an analogue to digital converter (ADC) the output of which is fed into parallel shift registers. Each time the analogue signal is sampled, the binary words representing the analogue signal are moved along the shift register, until at the end of the register a digital to analogue converter (DAC) returns the signal to analogue form. The delay time is determined by the sampling frequency and the number of parallel shift registers employed. Although these devices are available they tend to be expensive and beyond the budget of most teaching establishments.

Similar in many respects to the shift register system are bucket brigade devices (BBD). They have the advantage, however, of not requiring ADCs or DACs. A BBD consists of an integrated circuit containing a number of capacitors arranged in series and connected through switches. On receipt of a clock pulse the analogue signal to be delayed is applied to the first capacitor which is then charged, a second pulse disconnects the analogue signal, and closes a switch which passes the charge to the second capacitor. A third pulse disconnects the two capacitors and reconnects the analogue signal to the first capacitor. The process is then repeated with the sampled analogue signal being passed down the line of capacitors as clock pulses are applied. The delay time is determined by the clocking rate and the number of capacitors on the chip. A low cost BBD chip contains typically 1024 capacitors.

The use of shift registers is essentially a digital hardware solution. A real time digital computer for simulating audio systems has been described (6) which would permit delays to be obtained using simple software. The design and construction of a dedicated computing system is, however, probably beyond most of us. The microprocessor, however, presents another possible software approach. Wilson has recently demonstrated an audio delay system using direct memory access and a microprocessor (7). It is possible to conceive of a microprocessor-based delay system in which standard processor boards containing a delay program in ROM are employed for each simulated source path. The value of the required delay for a particular source path could be set using switches on the board. If a number of such boards are constructed the cost per board could be very small.

Since both the digital systems and the BBD work with sampled data the sampling theorem holds. Thus if one wishes to cover the whole audio frequency range the sampling frequency must be at least 40 kHz. This means, for example, that a standard BBD clocked at this frequency will delay the audio signal by approximately 12.5 milliseconds. For longer delays several devices can be connected in series.

Sampling, storing and processing data at this sort of speed means working close to the limit of most microprocessors and is one reason for Wilson's recourse to DMA.

The cost of an ADC is determined by its conversion speed and resolution. Whilst the conversion rate required here is not excessive and can be achieved with low cost ADCs, their resolution is typically only 8 bits, corresponding to a dynamic range of approximately 50 dB (i.e. comparable to all but the very highest quality tape systems). Ideally one would prefer to work with 12 or 14 bit ADCs to avoid granulation effecta. Most low cost microprocessors are, however, 8 bit devices and the use of higher resolution ADCs with them greatly complicates programming. Given that the dynamic range of an 8 bit ADC will be comparable to that of the tape source likely to be employed in a teaching situation, it will probably be acceptable if dither is introduced to reduce granulation effects (8).

One advantage of digital systems, is that once the signal has been converted to digital form it can be delayed indefinitely without degradation. The analogue signal transmitted by the BBD on the other hand will be degraded as it is passed along the system.

It would seem, therefore, that whilst both BBDs and microprocessor based delay systems have certain attractions in the teaching context, their fidelity may be limited. There limitations, however, are not so great as to preclude their use in the types of room simulation experiment under consideration. Of the two methods BBDs provide the easiest method of obtaining audio delays.

A BBD system could be built and running in a day but its acoustic performance is probably slightly inferior to an 8 bit microprocessor based system.

Reverberation

A typical echogram would show early reflections as discrete echoes. As the delay time increases, however, the echo density becomes so great that individual reflections cannot be seen. At this point one is into reverberation.

The simplest way of obtaining articifical reverberation would be to feed the output of a delay device (suitably attenuated) back into the input. The impulse response of such a system would, however, be very regular and the effect on broadband sound would be to introduce unacceptable colouration. Schroeder presented a technique for modifying the feedback loop to avoid this comb filter effect and this has subsequentially been refined (9,6). A number of delay systems providing different delay times each with a feedback loop would provide reasonably natural reverberation.

Speaker Arrays

Loudspeaker appraisal is a notoriously subjective field. Perhaps the best advise that can be given is that one should employ the best that can be afforded. The speakers employed for reverberation and high order reflections need not be of as high a quality as those employed for direct sound or early reflection.

Schroeder has demonstrated that with suitable signal processing all the spatial information that can be aurally perceived can be provided by two speakers (10). This obviously warrants further investigation. It is possible that recent work on binaural sound reporduction can be married to Schroeder's method to permit room simulation using a pair of headphones in a conventional room instead of a speaker array in an anechoic room.

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Conclusions

Room simulators for teaching purposes can be built at modest cost using BBDs or microprocessor systems to provide the necessary delays. Both systems enable delay times to be simply and accurately set-up by users having a limited technical background. The low cost, availability and low maintenance requirements of these systems make them very suitable for use by students of architecture.

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