Electronic Reverberation

The first of two articles which explain the principles of electronic reverberation systems, and discuss the techniques which have recently been developed for implementing them.

by ALAN M. FOWLER, MIE Aust., MIREE*

Once considered impracticable, the fully electronic reverberation unit is now a reality. The development of multistage shift registers on a single integrated circuit chip has opened the way for the design of electronic reverberation units which will overcome many of the problems of the mechanical or acoustic systems used previously.

Reverberation occurs naturally in an enclosed or semi-enclosed space because the sound waves from a source are reflected from the walls or other surfaces, and reach the listener after the direct sound, as shown in Fig 1. The ear does not distinguish individual echoes unless they are separated by more than 40 to 60ms, and provided that the delays between successive echoes is less than this, they blend together to give the sound a pleasant fullness.

Many of the delayed sounds reaching the listener will have been reflected more than once, and at each reflection they lose energy until they are too small to make any useful contribution to the sound level. When the original sound stops suddenly the sound level at the listener will die away slowly as the delayed sounds keep arriving.

The time taken for the sound level to drop by 60dB from its original level, i.e. to one thousandth of its original value, is called the reverberation time.

In practice, acceptable reverberation times depend on the size of the room and whether it is being used for speech or music, and are generally in the range of 1 to 2 seconds. If the reverberation time is much less than 1 second, the room sounds dead, and speech or music recorded under these conditions will sound thin or lacking in body. If the reverberation time is much greater than 2 seconds the syllables in speech will run together making them unintelligible, and music will sound muddy or lacking in brilliance. It is very difficult to remove excessive reverberation from a recording, and the room must be treated to reduce the reverberation time to an acceptable value.

* Australian Post Office Research Laboratories, Melbourne

Where the reverberation time is too short, it is possible to add additional reverberation to improve the sound. There are quite a few techniques for doing this, using acoustic, mechanical, or wholly electronic methods.

In order to understand the process of generating artificial reverberation let us look at a suitable way of representing how reverberation occurs naturally. Sound travels at approximately 1130 feet per second, so that each path will introduce a delay of 885µs for each foot of length. The simple case of a direct ray and three indirect rays in Fig 1 may be thought of as four paths in parallel, each containing one or more sections introducing delay, and having an attenuator at each reflection to represent the loss as energy is absorbed. These paths are represented diagramatically in Fig 2.

Because the system is linear, i.e., the sound energy is only delayed, reflected or partially absorbed without any other changes such as the generation of harmonics or other spurious frequencies, the various paths in Fig 2 can be simplified and combined into a single delay line as shown in Fig 3. The line is tapped at points corresponding to the total delay in each path, and the respective attenuators represent the total loss in each path. A more complex case can similarily be represented by lengthening the delay line, and adding additional taps and attenuators.

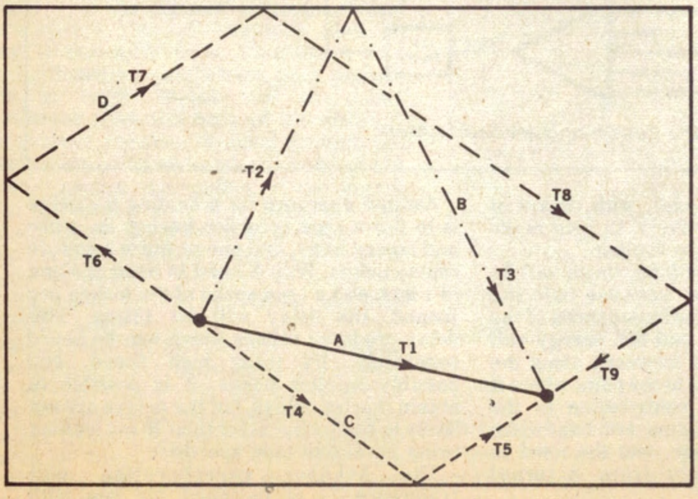

Fig 1. Reverberation occurs naturally in an enclosed space as sound waves are reflected from the walls and other surfaces to reach the ears of the listener after the direct sound.

The amount of energy lost at each reflection will vary with frequency, and will depend on the sound absorption characteristics of each reflecting surface. A complete representation of the reverberation behaviour of a room would therefore require that the frequency range be split into a number of bands, and each band represented by a unit similar to Fig 3, but having different attenuator settings.

Again, because the system is linear, only a single delay line is needed, the various taps for each frequency band are fed to separate amplifiers for each band, and the outputs of the amplifiers passed through suitable filters before combining. The case for three bands is shown in Fig 4, and this may be extended to any number of bands as required.

In practice, for many applications, it is only necessary to cover the range from 200Hz to 4500Hz using a single band.

If the time delay between each tap in Fig 3 was the same, and there was equal loss at each reflection it would be possible to simplify the design even further, by using a single delay section, and feeding part of the output back to add to the input as shown in Fig 5. This would produce a series of echoes at intervals of t seconds, each echo being smaller than the previous one by the loss around the loop. If the delay is made 10ms and a reverberation time of 1 second is required (60dB drop per second) the loop loss will be 0.6dB.

The system uses positive feedback, so that with a continuous signal at the input, each successive pass around the loop will add energy to the system, and the signal output will build up to a maximum value given by

$$V_{out} = V_{in} / (1 - r)$$

where "r" is the fraction of the voltage fed back around the loop. For a loop loss of 0.6dB, r is 0.933 and the signal output will be 15 times the input, ie, there will be a system gain of 23.5 dB.

Because positive feedback is being used, and the system is operating just below oscillation, a small change in loop gain will have a significant effect on the output level. If the loop loss changed from 0.6dB to 0.5dB, r would become 0.944 and the system gain rise to 25dB — ie the output would rise by 1.5dB for a 0.1dB variation in loop gain, and at the same time the reverberation time would rise to 1.2 seconds. This sensitivity to small changes in the loop gain of the system places very tight limits on the frequency response and overload characteristics of the loop.

While an artificial reverberation unit could be made using the arrangement in Fig 5, the results would not be entirely satisfactory. For a fixed loop delay, the phase of the signal arriving back at the input will depend on the frequency, and while the signals will be in phase for some frequencies they will be 180° out of phase at others, resulting in cancellation.

For a 10ms delay there will be a series of peaks and troughs at 100Hz spacings throughout the band. The peaks can be made closer together by increasing the delay time, but this cannot be made much longer than 40ms or separate regular echoes will be apparent. The peaks and troughs in the response can be broken up and smoothed out by taking the feedback from several taps which are not simple multiples of one delay.

So much for the theory, which is fairly simple and straightforward. It would seem easy enough to make an electronic model of Fig 3 if a suitable method of delaying the signals would be found. Until recently, it has not been easy to make a fully controllable delay line.

One of the first systems used, and which is still in use, was to place a loudspeaker and microphone a suitable distance apart in a room having the desired reverberation characteristics. While simple, it takes up space which could be more profitably used for other purposes, and hardly qualifies for portable applications. In addition, special care has to be taken to keep extraneous noises out of the room — there are many stories in broadcasting circles of rooms that had been pressed into service to provide reverberation, only to find the staff playing table tennis, or a cleaner dropping something heavy on his foot and speaking at length on the subject during an otherwise quiet passage!

The next approach was to use a long concrete pipe, or a length of water piping, preferably buried underground, with a speaker at one end, and one or more microphones along the length. This overcomes the external noise problem, but is still not portable. An extension of this approach is to use a small diameter plastic tube to carry the sound. The tubing can be coiled up conveniently, and may be mounted in acoustic insulating material to reduce the effects of external noise.

The tubing behaves like a transmission line, and the acoustic impedances of both the speaker and microphone are not matched to that of the tube, so that part of the energy will be reflected back to the speaker, where part will be absorbed, and the remainder reflected to the microphone. The combination will produce a series of echoes of decreasing amplitude, and spaced at odd multiples of the one way delay through the tube. This solution provides a compact, lightweight unit; the main problem is in obtaining sufficient bandwidth and a flat response.

Instead of using air in a tube, the sound waves may be transmitted through a liquid or metal. The Hammond reverberation unit (references 1,2) uses a long thin wire which has been coiled into a spring as the delay line. The Hammond unit actually has two delay lines in the one unit, with delays of 29ms and 37ms respectively, to help make the echoes appear more random.

As in the case of the pipe, there will be multiple reflections because of the mismatches between the impedances of the line and transducers, and the energy will bounce backwards and forwards along the line until the level drops to the point where it makes no significant contribution to the ouput. Again the problems are bandwidth and flatness of response, and the need to insulate the assembly from external vibrations.

Another approach for providing the delay is to use a tape recorder having an erase and record head, and one or more separate replay heads. With a head to head spacing of 1 inch and a tape speed of 7½ inches per second, the delay will be 133ms. This delay produces echoes which can be heard separately. By using high speed, and possibly smaller heads, it is possible to obtain shorter delays, but the delays are not likely to be much shorter than 20 ms without using excessive tape speeds.

The Schober reverberation unit (reference 3) is designed for use with

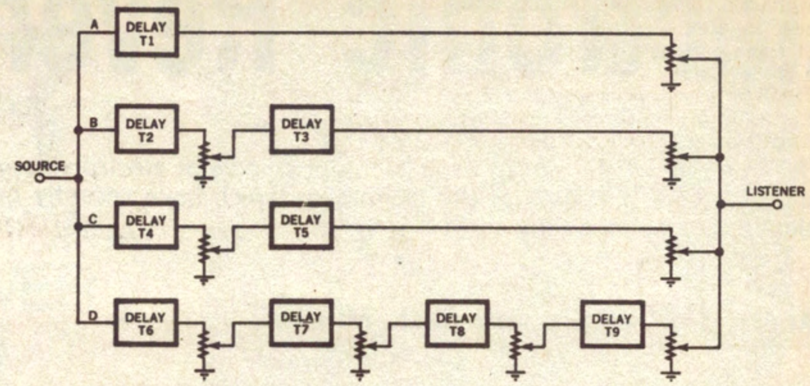

Fig 2. Showing how the various paths may be represented in a block schematic.

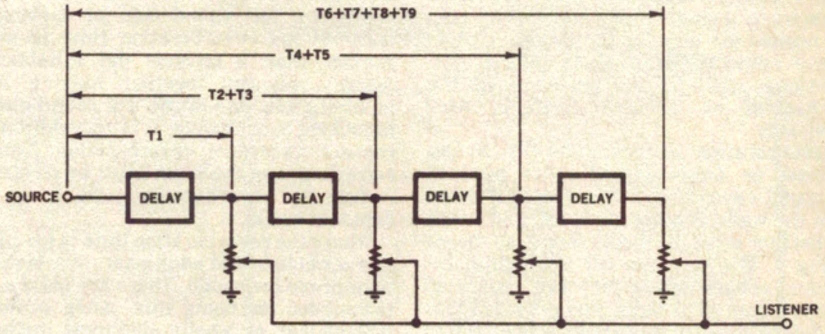

Fig 3. The various paths of Fig 2 can be simplified into a single delay line.

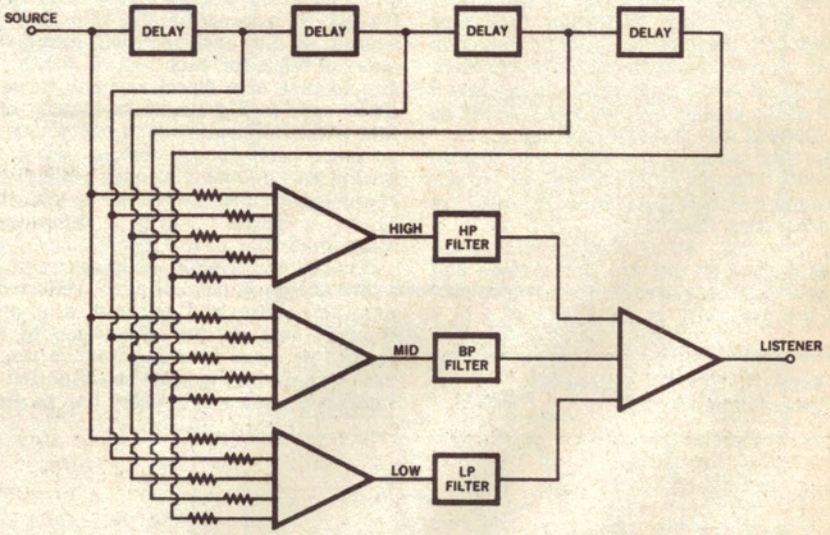

Fig 4. Schematic for a three frequency band system.

electronic organs, and has three replay heads spaced to give successive delays of approximately 110ms. The output of the third head is attenuated and added to the original signals feeding the record head. With a 330ms delay, the loop attenuation will be 20dB for a reverberation time of one second or 3.3dB for six seconds. With such big attenuations, the frequency response of the system is not affected greatly by small changes in loop loss, and eases the design considerably. Dorf and Evedon, the

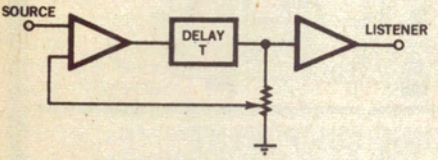

Fig 5. Showing how it is possible to simplify the design further by using a single delay section and feeding back part of the output.

designers of this unit, defend the choice of 110ms for the individual delays — a figure which will produce separately identifiable echoes — on the grounds that it produces a more natural sound at longer reverberation times.

Various workers have found that in reverberant rooms, the ear tends to select the echoes produced by the longest dimension of the room, and to judge the room size from this. They claim that systems using a short delay, and a smooth reverberation decay curve, sound unnatural at longer reverberation times, because of the absence of these longer delays. The effect of using a long delay can only be evaluated subjectively by each listener in each particular application, and any discussion is likely to result in more arguments than those about the merits of different loudspeaker systems.

The use of a tape recorder overcomes the problems of bandwidth, external noise, and portability, but introduces others. It is usual to have a continuous loop of tape and even using very special grades to minimise wear, the loops must be replaced at intervals of about 500 hours. The tape splice must be carefully made and preferably welded to prevent it pulling apart in use, and to reduce the level of the clicks as it passes over the heads. The background noise, wow, and flutter must be kept to a very low level — it is very annoying to hear flutter on the echoes when it is not present on the original sound.

All of the methods mentioned thus far use mechanical delay lines, with all the problems associated with such lines. A fully electronic delay line allows a much simpler approach, as will be shown in the next article.

The permission of the Senior Assistant Director General, Australian Post Office Research Laboratories, to publish this article is hereby acknowledged.

REFERENCES—

1. W. N. WILLIAMS, Radio, Television and Hobbies, October 1961, pp. 80-81.
2. W. N. WILLIAMS, "Using Reverberation Units", Electronics Australia, April 1966, pp. 72-73.
3. R. H. DORF and R. C. EVEDON, "Design of the Schober Reverbatape Unit," Electronics Australia, November 1968, pp. 104-113.