Institute of Acoustics

ARTIFICIAL REVERBERATION - A DEVELOPED DIGITAL METHOD

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EARLY REVERBERATION DEVICES AND THEIR DEVELOPMENT

The most common reverberators until recently were spring or plate devices which worked on electro-mechanical principles rather than purely electronic ones. Both relied on a transducer feeding mechanical energy into a structure that could propagate and reflect sound; in the case of the spring system this was a helical spring or springs under tension and in the case of a plate, a thin sheet of metal suspended in a frame. The vibrations in the spring or plate were then converted back to an electrical signal by another transducer and subsequently mixed with the dry signal in the desired proportions. Stereo effects could be obtained by using several springs and routing their outputs to one or other side of the mix and, in the case of the plate, using two pick-up transducers mounted on different parts of the plate surface.

Though the operating principle is very similar in both cases, the plate gave generally better results due to its lower colouration and was almost universally adopted by sound studios for a number of years.

The plate sound is however not an accurate simulation of true room reverberation as it is characterised by a very rapid build up of reflections which attain a high density very early on. Also some plates tended to add more colouration to the sound than others and the effect is rather percussive. However, the achieved result was very usable and some control over the decay time could be exercised by damping the plate. What the plate could not do though was to simulate a specific acoustic environment as there was no control over the build-up or spacing of the early reflections, the reverb density or the way in which high frequencies decay relative to the low frequencies other than the changes that could be made by altering the plate tension and mechanical damping. The only variable parameters were relative level, decay time and overall equalisation. Furthermore they were extremely susceptible to mechanical vibration and were bulky.

ELECTRONIC SIMULATION

Klark-Teknik have been involved in developing a high quality reverberation simulator since 1981 and, as with any logical research exercise, they started by examining the work that had gone before, a summary of which follows.

ARTIFICIAL REVERBERATION

Schroeder

Shroeder's AES paper published in 1961 and entitled 'Colourless Artificial Reberberation' was the starting point. Here he discussed the behaviour of rooms in response to transient sounds and came up with a set of conditions which an artifical reverberator would need to satisfy before it would sound natural. These were:

1. The frequency response must be flat when measured with narrow bands of noise, with a bandwidth corresponding to that of the transients in the sound to be reverberated.

2. The normal modes of the reverberator must overlap to cover the entire audio frequency range.

3. The reverberation times of the individual modes must be essentially equal so that different frequency components within the sound will decay at equal rates.

4. A short time after the excitation, the echo density must be high enough so that individual echoes cannot be resolved by ear.

5. The echo response must be free from periodicity which would manifest itself as flutter.

6. The amplitude/frequency response must not exhibit any apparent periodicities because otherwise unnatural colourations would be generated.

Shroeder then goes on to introduce the all-pass and the comb filter which are covered again in his next paper on the subject.

In 1962, Schroeder published another AES paper entitled Natural Sounding Artificial Reverberation to follow up his previous work.

This paper started out by underlining the shortcomings of the existing systems which were mainly electro-mechanical or relied upon magnetic recording in the form of tapes or discs. These suffer from two serious defects:

1. Their amplitude responses are not flat leading to unnatural colouration.

2. The number of echoes per second in response to a single input pulse is too low compared to the manifestations in a real room. This leads to the characteristic 'fluttering' which is particularly evident on percussive sounds.

Schroeder then went on to describe the all-pass reverberator which would exhibit a flat frequency/amplitude response and so solve the problem of colouration. This was based on the all-pass filter, the block diagram for which is shown in Figure 3.

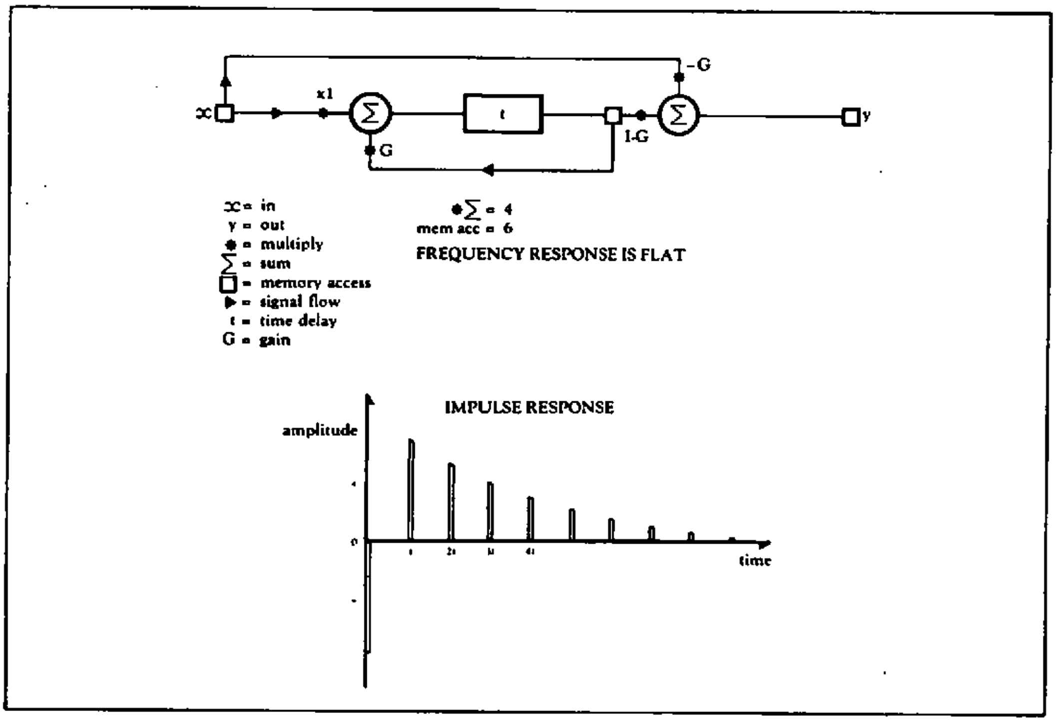

Fig 3 The All-pass filter.

The concept involved connecting several all-pass filters in series in order to create the required reverb density and, as each filter section has a flat frequency response, the overall system should preserve this characteristic. He suggested making the delay of each filter section about 1/3 of that of the previous section and with five filters in series, the calculated echo density for his values of delay was 810 per second. He originally postulated that something in the order of 1000 echoes per second was sufficient to create a natural sounding reverberation. It is now accepted that between 1000 and 3000 reflections per second are required for a truly convincing simulation.

Exploration of the possibility of connecting several comb filters in parallel to simulate the behaviour of rooms that do not have a perfectly flat amplitude/frequency response,lead to the block diagram of a comb filter as shown in figure 4. overleaf.

Those who have experimented with digital delay devices will recognise this as the same effect obtained by setting up a delay time of several milliseconds and then feeding some of the output of the delay line back to the input. If it is desired to make the reverberation time a function of frequency for greater realism, it is a simple matter to insert filter components in the feedback path. However, the bank of comb filters alone does not provide a great enough echo density and so a further section consisting of several cascaded all-pass filters must be inserted in series with the comb filter as shown in figure 5.

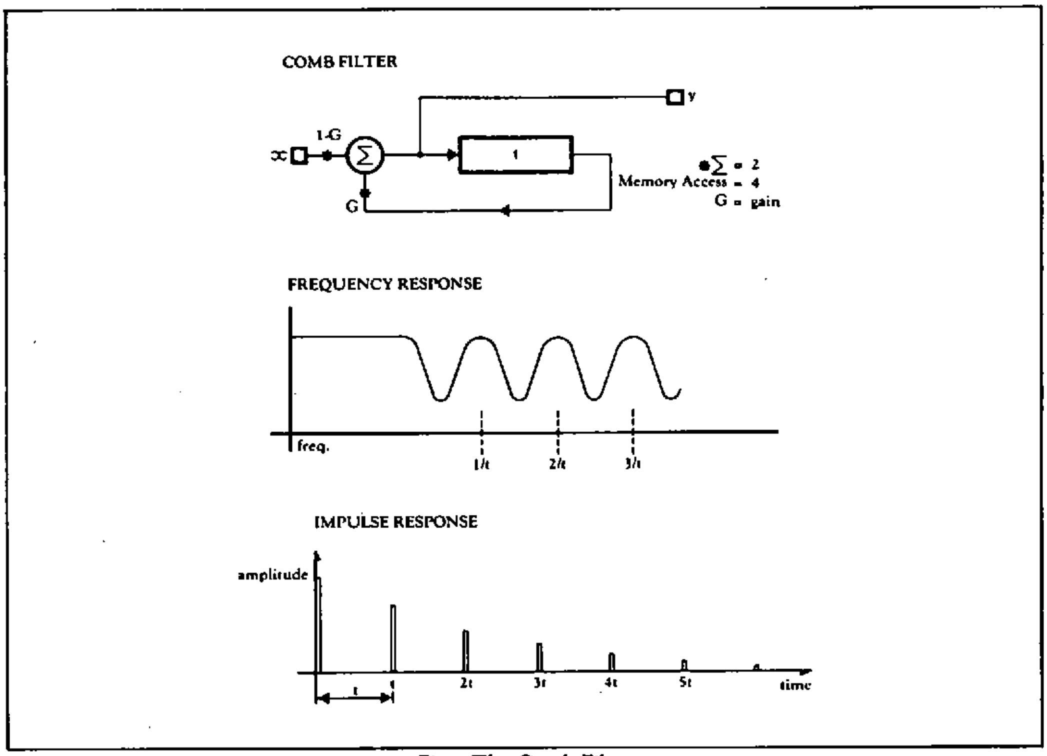

Fig 4 The Comb Filter.

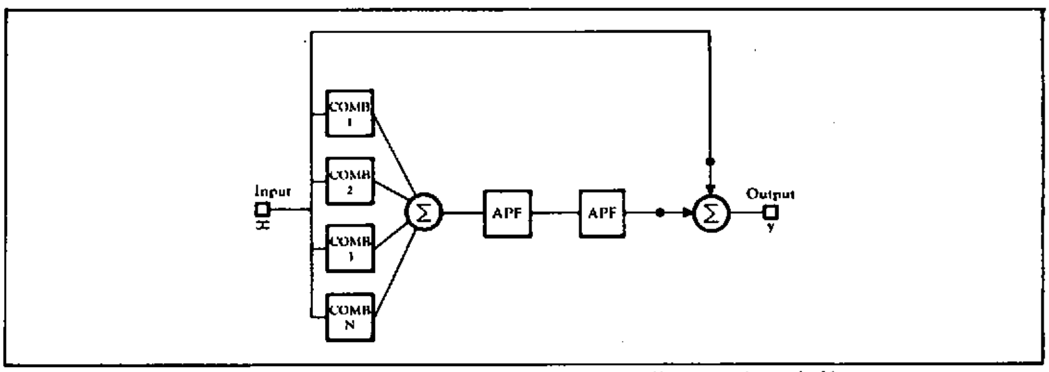

Fig 5 Block diagram of a reverberator using both all-pass and comb filters.

The paper concludes with a description of how a matrix system could be devised for simulating a multi-channel output to be fed to speakers placed around the room to give an enhanced spacial effect.

Some years later, in 1970, Shroeder suggested using a separate reverberator to simulate the first few reflections and this was followed up by a Moorer.

Moorer

In 1979, Moorer's paper, 'About this Reverberation Business' took up from where Schroeder had left off and, with the help of the more advanced computer technology that had developed in the intervening years, Moorer set out to narrow the gap between natural and synthetic reverberation. The first step was to examine the problem areas in Schroeder's original designs and these were as follows.

1. With a reverberator built up only from a number of all-pass filters in series, the decay did not start with a dense sound and die out exponentially. In fact, the higher the order, the longer the echoes took to build up to a pleasing level. This lag in reverberation could be as much as several hundreds of milliseconds.

2. Even slight changes in delay time between the successive filters could cause the smoothness of the overall decay to vary enormously.

3. The tail of the decay exhibited a metallic ringing.

These parameters defied precise mathematical analysis and the most satisfactory combination of values had to be arrived at empirically.

In his paper, Moorer examined the properties of the Boston Symphony Hall, widely held to be a particularly good example of concert hall acoustics, and published the following observations.

The reverberation time of over 1.7 seconds suggests that the sound has travelled over 600 meters before dying away and, given typical atmospheric conditions, the 4kHz signals will be attenuated some 60dB more than the 1kHz signals. This shows up the importance of simulating the correct frequency characteristics for the late decay. Furthermore, to simulate a good listening environment, the reverb must be smooth and dense with no apparent resonances. And, to accurately simulate the early reflections, these would have to include a halo of reverberation to simulate the effect of diffusion at the reflecting surfaces.

Moorer then moved on to consider the simulation of these early reflections using a finite impulse filter or FIR as illustrated in figure 6. overleaf.

This is in effect a tapped delay line where each of the taps has separate gains. The spacing of the taps must be irregular so as to avoid any obvious periodicity in the reflection pattern but the optimum delays have to be chosen empirically; attempts to calculate mathematical spacings based on prime numbers or to simulate the behaviour of actual rooms did not prove to be satisfactory. These taps may then be summed and treated with an all-pass/comb reverberator to build up the required density.

Taking all this into account, the block diagram of our reverberator might look something like figure 7.

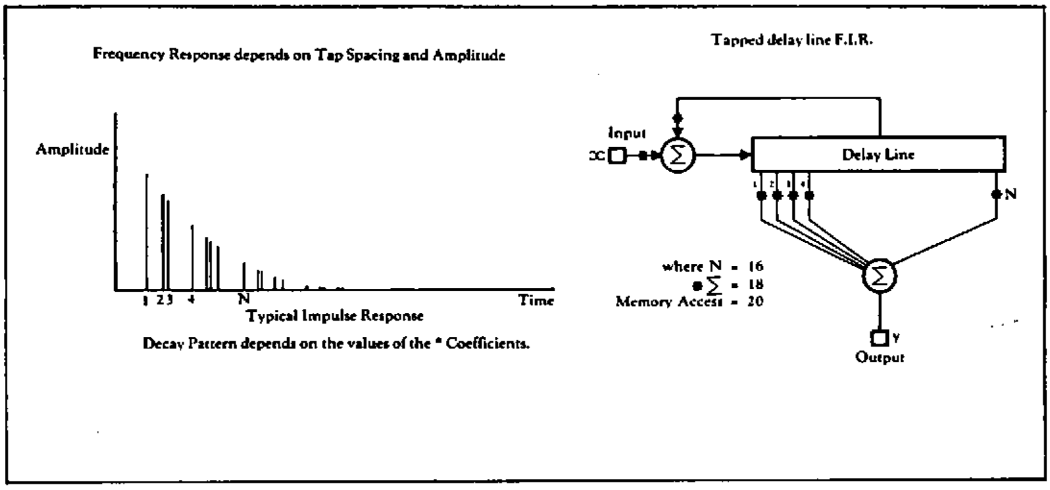

Fig 6 Finite impulse response filter.

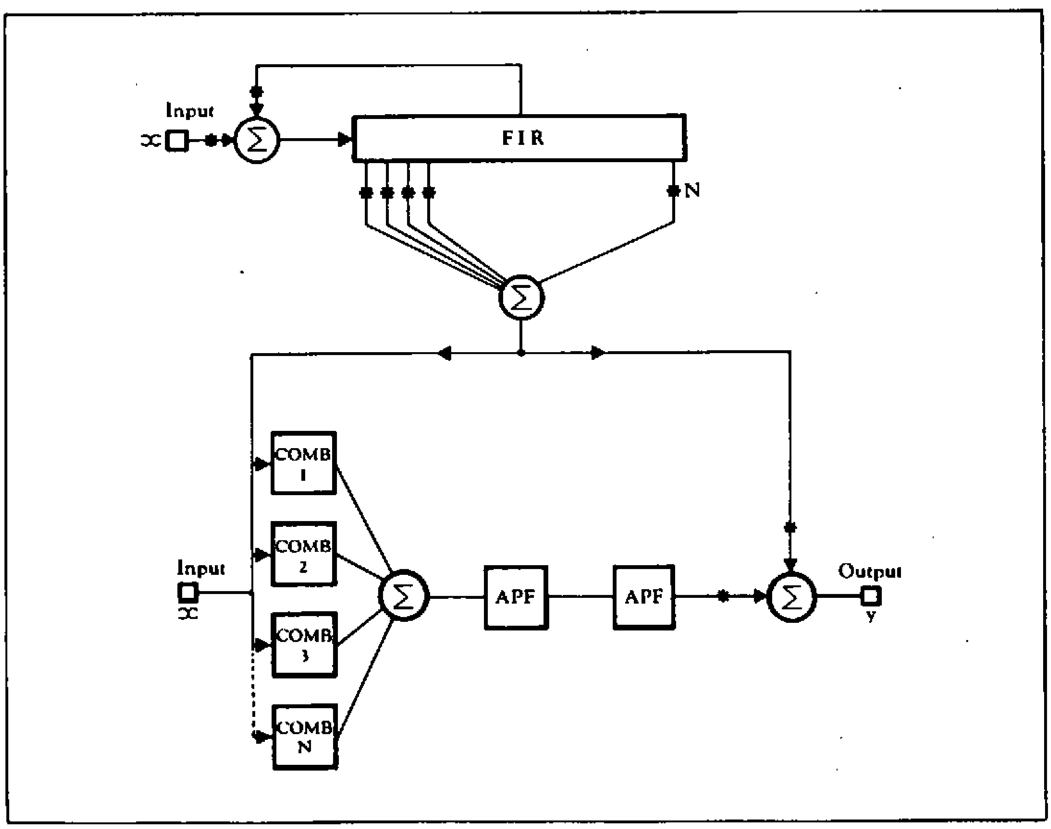

Fig 7 Reverberator with FIR plus comb and all-pass filters.

Using modern computer technology, these components can be simulated in real time but the computer needs to be capable of very fast arithmetic processing if it is to generate a convincing, dense reverberation with an acceptable bandwidth.

TONE CONTROL

In addition to the three basic but important elements, we also need to introduce frequency selective filters into to processing path. The basic low-pass filter and the full LF/HF control are shown in figure 8.

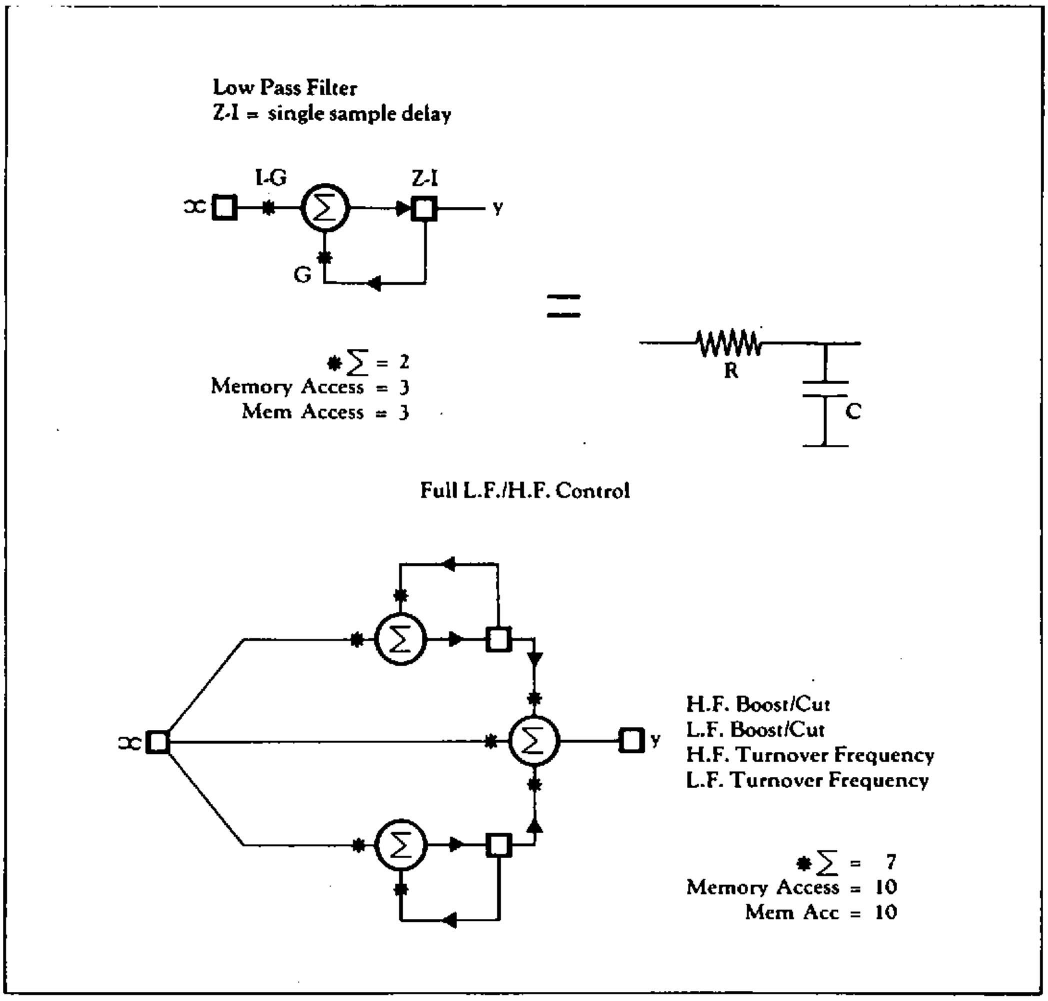

Fig 8 Low-pass and LF/HF control filters.

The full LF/HF control configuration is very flexible as it gives control over both HF and LF cut and boost and both HF and LF turnover points are variable. Again this imposes a demand on processing time which is 2 multiplier/accumulator operations and 3 memory accesses per sample period for a simple low-pass filter; 7 multiplier/accumulator operations and 10 memory accesses are needed for the full LF/HF control filter.

SUMMARY

There are three main elements or building blocks which can be used in combination to simulate reverberation and these are the all-pass filter, the comb filter and the finite impulse response filter. Additionally, some way of modifying the frequency response of feedback loops within the system must be implemented.

The characteristics of each building blocks significantly affects the subjective quality of the result and parameters cannot be optimised purely by mathematical means. Some empirical adjustment must be made.

The implementation of a high quality reverberation simulator makes heavy demands on processing speed.

COMBINING THE BLOCKS

Figure 9 illustrates a complete reverberator combining all the elements so far discussed. Notice that some taps from the FIR are fed only to the comb filter bank and that later taps are only summed into the APF section. Also notice that the feedback around the FIR is routed via an APF and a low pass filter to add diffusion and to tailor the frequency response of the decay. By duplicating the APF section and summing different taps from the FIR and outputs from different comb filter taps within the bank, two different outputs can be synthesised. These two outputs can be used in stereo applications and the results from a carefully designed system can be most effective, yielding a greatly enhanced sense of depth.

In terms of computing power, the mathematical implications of this system, using 12 combs, are what we need to perform 140 multiplier/accumulator operations and 200 memory accesses per sample for a stereo unit.

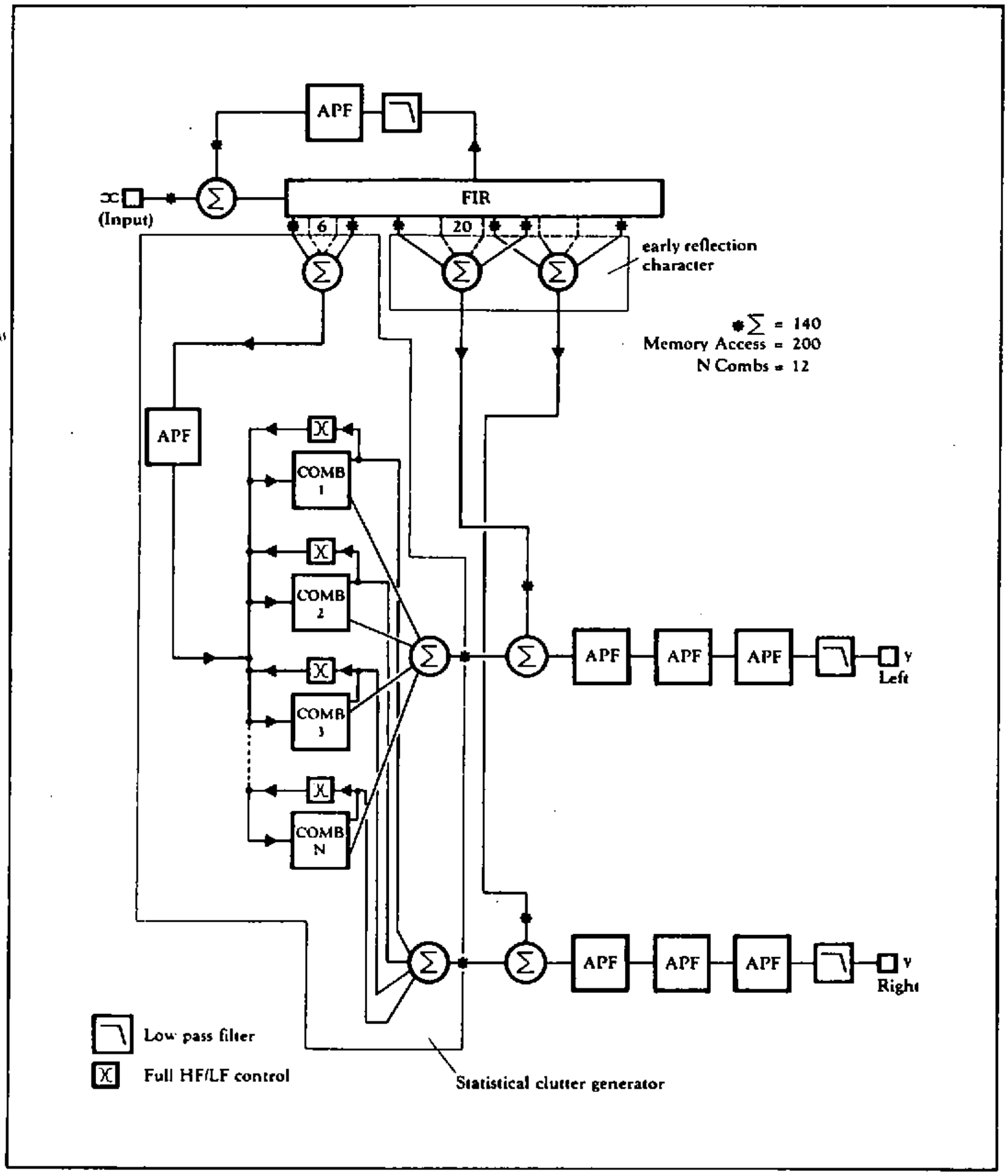

Fig 9 A complete digital reverberator.