"COLORLESS" ARTIFICIAL REVERBERATION

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Abstract

1. Introduction

Electronic devices are widely used today to add reverberation to sound. Ideally, such artificial reverberators should act on sound signals exactly like real, three-dimensional rooms. This is not simple to achieve, unless one uses a reverberation chamber or the electrical equivalent of a three-dimensional space. Reverberation chambers (and plates[1]) are preferred by broadcast stations and record manufacturers because of their high quality and lack of undesirable side effect, but they are not truly artificial reverberators.

In this paper, we shall focus our attention on electronic reverberators consisting of delay-lines, disc or tape-delay, and amplifiers. Electronic reverberators are both cheaper than real rooms and have wider applicability, notably in the home (unless one wants to convert the basement into a reverberation chamber). They can also be employed to increase the reverberation time of auditoriums, thereby adapting them to concert hall use, without changing the architecture.

Before attempting the difficult task of reproducing, to the ear's satisfaction, room characteristics by delay-lines, it is wise to recall some of the important properties of large rooms.

2. The frequency response of large rooms

A room can be characterized by its normal modes of vibration. It has been shown[2] that the density of modes is nearly independent of room shape and is proportional to the square of the frequency:

$$\text{Number of modes per cps} = \frac{4\pi V}{c^3} f^2.$$

Here V is the volume, c = the velocity of sound, and f the frequency.

Above a certain critical frequency,[3]

$$f_c = 2000\sqrt{T/V} \text{ (reverberation time T in seconds, V in m}^3\text{)},$$

the density of modes becomes so high that many modes overlap. In this frequency range, which is of prime interest for large rooms, the concept of individual normal modes looses its practical (though not its theoretical) significance. The behavior of the room is governed by the collective action of many simultaneously excited and interfering modes, resulting in a very irregular amplitude-frequency response.[4,5] However, the fluctuations are so rapid (on the frequency scale) that the ear, in listening to a non-steady sound, does not perceive these irregularities.[6] (The response fluctuations can be heard only by exciting the rooms with a sinewave of slowly varying frequency and listening with one ear.) When the room response is measured using instead of a sinewave a psychoacoustically more appropriate test signal, such as narrow bands of noise, the response would indeed be much smoother.

It is this apparent smoothness of a room's frequency response which people have found particularly difficult to imitate with artificial reverberators. In this paper we shall describe electronic reverberators which have perfectly flat amplitude-frequency responses. Thus, they not only overcome this long-standing difficulty but are actually superior to rooms <u>in this one respect</u>.

However, a flat frequency response is not the only requirement for a high-quality reverberator. Before we can hope to successfully design one, we must also know something about the <u>transient</u> behavior of rooms.

3. <u>The transient behavior of rooms</u>

How does a room respond to excitation with a short impulse? If we record the sound pressure at some location in the room as a function of time, we first observe an impulse corresponding to the direct sound which has traveled from the sound source to the pick-up point without reflection at the walls. After that we see a number of discrete low-order echos which correspond to one or a few reflections at the walls and the ceiling. Gradually, the echo density increases to a statistical "clutter." In fact, it can be shown[7] that the echo density is proportional to the square of the elapsed time:

$$\text{Number of echos per second} = \frac{4\pi c^3}{V}\, t^2.$$

The time after which the echo response becomes a statistical clutter depends on the width of the exciting impulse. For a pulse of width Δt, the critical time after which individual echos start overlapping is about

$$t_c = 5 \cdot 10^{-5} \sqrt{V/\Delta t} \text{ (V in m}^3\text{)}.$$

Thus, for transients of 1 msec. duration and a volume of 10,000 m^3 (350,000 ft^3), the response is statistical for times greater than 150 msec. In this region, the concept of the individual echo looses its practical significance. The echo response is determined by the collective behavior and interference of many overlapping echos.[8]

Another important characteristic of large "diffuse" rooms is that all modes have the same or nearly the same reverberation time and thus decay at equal rates as evidenced by a straight-line decay when plotting the sound level in decibels versus elapsed time.

Still another property of acoustically good rooms is the absence of "flutter" echos, i.e., periodic echos resulting from sound waves bouncing back and forth between parallel hard walls. Such periodicities in the echo response are closely associated with one-dimensional modes of sound propagation which can be avoided by splayed walls and the placement of "diffusors" into the sound path.

4. The conditions to be met by artificial reverberators

After this brief review of room behavior, we are in a position to formulate conditions to be met by artificial reverberators.

(1) The frequency response must be flat when measured with narrow bands of noise, the bandwidth corresponding to that of the transients in the sound to be reverberated. This condition is, of course, fulfilled by reverberators which have a flat response even for sinusoidal excitation.

(2) The normal modes of the reverberator must overlap and cover the entire audio frequency range.

(3) The reverberation times of the individual modes must be equal or nearly equal so that different frequency components of the sound decay with equal rates.

(4) A short interval after shock excitation, the echo density must be high enough to give a coherent reverberation for even the shortest audible transients.

(5) The echo response must be free from periodicities (flutter echos).

In addition to these five conditions, a sixth one must be met which is not apparent from the above review of room behavior but easily violated by electronic reverberators:

(6) The amplitude-frequency response must not exhibit any apparent periodicities. Periodic or comblike frequency responses produce an unpleasant hollow, reedy, or metallic sound quality and give the impression that the sound is transmitted through a hollow tube or barrel.

This condition is a particularly important one because long reverberation times are achieved by circulating the sound by means of delay in feedback loops. The responses of such loops, which are the equivalent of one-dimensional sound transmission, are inherently periodic and special precautions have to be taken to make these periodicities inaudible.

In the following, a basic reverberator is described which fulfills conditions (1), (3) and (6) ideally. By connecting several of these reverberating elements in series, conditions (2), (4) and (5) can also be satisfied without violating the others.

5. All-pass reverberators

The simplest reverberator consists of a delay-line, disc or tape-delay which gives a single echo after a delay time τ. Its impulse response is

$$h(t) = \delta(t-\tau), \tag{1}$$

where $\delta(t)$ is the Dirac delta-function (an ideal impulse). The spectrum of the delayed impulse is

$$H(\omega) = e^{-i\omega\tau}, \tag{2}$$

where ω is the radian frequency. The absolute value of $H(\omega)$ is one. This means that all frequencies are passed equally well and without gain or loss.

In order to produce multiple echos without using more (expensive) delay, one inserts the delay line into a feedback loop, as shown in Figure 1, with gain g less than one (so that the loop will be stable). The impulse response, illustrated in the center of Figure 1, is now an exponentially decaying repeated echo:

(3) $$h(t) = \delta(t-\tau) + g\delta(t-2\tau) + g^2\delta(t-3\tau) + \dots$$

The corresponding complex frequency response is

(4) $$H(\omega) = e^{-i\omega\tau} + ge^{-2i\omega\tau} + g^2e^{-3i\omega\tau} + \dots$$

or, using the formula for summing geometric series,

(5) $$H(\omega) = \frac{e^{-i\omega\tau}}{1-ge^{-i\omega\tau}}$$

By taking the absolute square of $H(\omega)$, one obtains the squared amplitude-frequency response:

(6) $$|H(\omega)|^2 = \frac{1}{1+g^2-2g\cos\omega\tau}.$$

As can be seen, $|H(\omega)|$ is no longer independent of frequency. In fact, for $\omega = 2n\pi/\tau$ ($n = 0, 1, 2, 3, \dots$), the response has maxima

(7) $$H_{max} = \frac{1}{1-g},$$

and, for $\omega = (2n+1)\pi/\tau$, minima

(8) $$H_{min} = \frac{1}{1+g}.$$

The ratio of the response maxima to minima is

(9) $$H_{max}/H_{min} = \frac{1+g}{1-g}.$$

For a loop gain of $g = 0.7$ (-3db), this ratio is $1.7/.3 = 5.7$ or 15db!

The amplitude-frequency response of a delay in a feedback loop has the appearance of a comb with periodic maxima and minima, as shown at the bottom of Figure 1. Each response maxima corresponds to one normal mode. The natural frequencies are thus spaced $1/\tau$ cps apart.

The 3 db-bandwidth of each peak is approximately

$$\Delta f = \frac{-\ln g}{\pi\tau}, \tag{10}$$

where "$\ln$" denotes the logarithm to the base $e = 2.718\ldots$ Converting to logarithms to the base 10 ($\log$), one obtains

$$\Delta f = \frac{1}{20\pi \log e}\frac{-\gamma}{\tau} = .0367\frac{-\gamma}{\tau}, \tag{11}$$

where γ is the loop gain in decibels: $\gamma = 20 \log g$. For $\gamma = -3$ db, the bandwidth is about $.11/\tau$ or only one-ninth of the spacing of the natural frequencies. The subjective effect of this resonant response is the hollow or reedy sound quality mentioned above.

In our search for better reverberators we discovered that a certain mixture of the output of the multiply delayed sound and the undelayed sound resulted in an equal response of the reverberator for all frequencies. The mixing ratio that accomplishes this is and results in unity gain for all frequencies is $(-g)$ for the undelayed sound and $(1-g^2)$ for the multiply delayed sound. The corresponding circuit is shown in Figure 2. Its impulse response is given by

$$h(t) = -g\delta(t) + (1-g^2)\,[\delta(t-\tau) + g\delta(t-2\tau) + \ldots]. \tag{12}$$

The corresponding frequency response is

$$H(\omega) = -g + (1-g^2)\,\frac{e^{-i\omega\tau}}{1-ge^{-i\omega\tau}}, \tag{13}$$

or

$$H(\omega) = \frac{e^{-i\omega\tau} - g}{1-ge^{-i\omega\tau}}, \tag{14}$$

or

(15) $$H(\omega) = e^{-i\omega\tau}\,\frac{1-ge^{i\omega\tau}}{1-ge^{-i\omega\tau}}.$$

What is the absolute value of this $H(\omega)$? The first factor on the right has, of course, absolute value one. The second factor is the quotient of two conjugate complex "vectors," i.e., its absolute value is also one. Thus,

(16) $$|H(\omega)| = 1.$$

In other words, the addition of a suitably proportioned undelayed path has converted the comb filter (equation (6)) into an all-pass filter (equation (16)). This is not a mere academic result. The conversion of a comb filter into an all-pass filter is accompanied by a marked improvement of the sound quality from the hollow sound of the former to the perfectly "colorless" quality of the latter.

Now we are in possession of a basic reverberating element which passes all frequencies with equal gain and thus fulfills conditions (1) and (6) above. The spacings and decay rates of the normal modes (though no longer "visible" as resonant peaks of the amplitude-frequency response) are the same as those for the previously discussed comb filter. Thus, condition (3), requiring equal decay rates for the normal modes, is also fulfilled.

Whether the normal modes overlap (condition (2)) or not can no longer be judged on the basis of the amplitude-frequency response because it is constant. However, the phase-frequency response still reflects the distribution of normal modes and thus must conform to condition (2). The phase-lag of $H(\omega)$ as a function of frequency is with equation (15)

(17) $$\varphi(\omega) = \omega\tau + 2\cdot \arctan \frac{g \sin \omega\tau}{1-g \cos \omega\tau}.$$

A more convenient quantity to consider is the rate of change of phase-lag with respect to radian frequency:

(18) $$\frac{d\varphi}{d\omega} = \frac{1 - g^2}{1 + g^2 - 2g \cos \omega\tau}\,\tau,$$

which has exactly the same dependence on ω as the square amplitude-frequency response $|H(\omega)|^2$ of the corresponding comb filter (see equation (6)). The physical significance of $d\varphi/d\omega$ is that of the envelope or "group" delay of a narrow band of frequencies around ω. According to (18), for a loop gain of $g = .7$ this envelope delay fluctuates as much as 32:1 for different frequency bands, the long delays occurring, of course, for frequencies near the natural frequencies, $2\pi n/\tau$ ($n = 0, 1, 2, \ldots$), of the filter. The half-width of the envelope delay peaks is the same as that for squared amplitude (see equation (10)). Thus, for a loop gain of -3 db, only one-ninth of all frequency components suffer a large envelope delay while the remaining frequencies are much less delayed. This constitutes a very unequal treatment of different frequency components and violates condition (2).

The remaining two conditions, (4) and (5), are also violated as we shall see immediately. The relationship between reverberation time T (defined by a 60 db decay) and the two parameters of the reverberator, the delay τ and the loop gain γ in decibels, is as follows. For every trip around the feedback loop the sound is attenuated γ db. Thus, the 60 db decay time is

$$T = \frac{60}{-\gamma}\,\tau. \tag{12}$$

For $\gamma = -3$ db,* we have $T = 20 \cdot \tau$. Thus, in order to achieve, for example, 2 seconds of artificial reverberation, the loop delay must be 0.1 sec. With this loop delay the basic reverberating element shown in Figure 2 produces one echo every one-tenth of a second. This constitutes a most undesirable periodic flutter echo. Also, the echo density (ten echos per second) is much too low to give a continuous reverberation. Thus, conditions (4) and (5) are violated.

How can one obtain a less periodic time response and a greater echo density without giving up the all-pass characteristic?** If several all-pass feedback loops with incommensurate loop delays are connected in <u>series</u>, as illustrated in Figure 3, the combined frequency response remains flat, while the echo response becomes aperiodic and the echo density increases.

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* We do not consider open loop gains greater than .7 (-3 db) because in practice it is difficult to maintain the desired closed loop characteristics with gains too close to one.

** Nico Franssen[9] of Philips Research Laboratories (Eindhoven, Netherlands), has suggested the use of multiple feedback, proportioning the gains to give a flat open-loop frequency response. This allows larger loop gains without incurring instability. However, a feedback loop around an all-pass filter results in a non-flat frequency response, unless supplemented by a direct path of suitable gain as described above.

In addition, a better coverage of the frequency axis with normal modes is achieved. In fact, the envelope delay response of the series connection is a sum of terms like (18) with different τ's. Since each of these terms "covers" only one-ninth of the frequency axis, at least five and possibly ten all-pass feedback loops in series are required. On the other hand, one can also show that too many all-pass feedback loops in tandem are bad because they lead to a very unnatural, non-exponential reverberation which builds up to its maximum intensity rather slowly before it starts decaying. I shall spare the reader the mathematical details of this peculiar reverberation because he has suffered already too much, I am afraid.

Figure 4 shows the impulse response of five all-pass filters connected in series with loop delays of 100, 68, 60, 19.7, 5.85 msec. The loop gains are +.7, -.7, +.7, +.7, +.7. This combination of delays and gains was arrived at after considerable experimentation observing the response to a variety of sounds, both on the oscilloscope and by listening, and using a smooth envelope of the decay as a criterion. The appearance of the echo response is quite random and not unlike that of real rooms.[10,11] One also notices an increase in pulse density with increasing time.

The impulse response shown in Figure 4 was obtained from a large digital computer (IBM 7090) in conjunction with special digital-to-analog conversion and plotting apparatus. In addition to using the computer as a "draftsman," many of the actual reverberation experiments were performed with the help of the digital computer. In this research method (sometimes called "digital simulation"[12]) ordinary ("analog") tape recordings of the sound to be reverberated are prepared and converted into digital tapes by means of special conversion equipment. The digital tape is then read by the computer and acted upon exactly like any desired real equipment would act on the sound signal. To facilitate the programming of the computer, our engineers make use of a special translation program, developed at Bell Telephone Laboratories, which translates their block diagrams into the computer language. Thus, the computer first "compiles" its own program on the basis of the block diagram information and then acts on the sound as the block diagram would do. It then prepares one (or several) digital output tapes which are converted back into analog tape recordings and evaluated by listening. Needless to say that this is a very powerful research tool, especially when complex equipment is to be evaluated! In this manner we have studied the subjective quality of a great variety of reverberators with both flat and non-flat frequency responses.

Our listening experience with all-pass reverberators indicates that the problem of unequal response to different frequencies has been solved and sound "coloration" completely eliminated. Audible flutter echos have been avoided by the use of several all-pass feedback loops with incommensurate delays. The only audible degradation observed in the reverberator utilizing five all-pass feedback loops is a certain roughness in the early part of the echo response. This is probably due to the low echo density in that portion of the impulse response (see Figure 4). If this conjecture is true, one or two additional feedback loops with very short loop delays (high echo density) should give the desired improvement. Work to test this point is in progress. Further study is required to optimize performance, in terms of the six donditions stated above, for a given complexity of instrumentation.

6. Application to quasi-stereophony

The late Holger Lauridsen[13] of the Danish State Radio has discovered a method of splitting a single audio signal into two "quasi-stereophonic" signals which give the listener all the "ambience" of multi-channel stereophony but permits, of course, no correct localization of individual sound sources. In order to achieve this, Lauridsen has used delay networks connected to form a pair of interleaved comb filters.[14] However, these comb filters give rise to unpleasant sound qualities - not quite as pronounced as in artificial reverberators, but nevertheless easily perceptible. We have overcome this disadvantage by using a pair of all-pass filters, like the one shown in Figure 2, to split the single-channel audio signal. This idea is described in greater detail in a forthcoming publication.[15]

7. Summary

Several reverberators of the all-pass type were successfully simulated on a digital computer (IBM 7090). Others were instrumented with delay-lines and tape-delay. No coloration of the reverberation sound was detected in any of these electronic reverberators. Further work remains to be done to optimize other important characteristics for a given complexity of the electronic circuitry.

Finally, the application of all-pass reverberators to the problem of increasing the reverberation time of auditoriums and concert halls by purely electro-acoustic means,[16] will have to be studied in detail. Here the flat frequency response is particularly important for two reasons: (1) It minimizes acoustic feedback problems. The "ringing" and instability due to the unavoidably irregular frequency response of the room can be reduced by shifting all frequency components of the reverberated sound by a small constant amount.[17] (2) A flat response of the reverberator contributes to the high sound quality required in concert hall applications. Ultimate acceptance of electro-acoustic techniques in concert halls and opera houses is assured only if the artificial effects are not recognized as such by the music-loving public.

8. Acknowledgments

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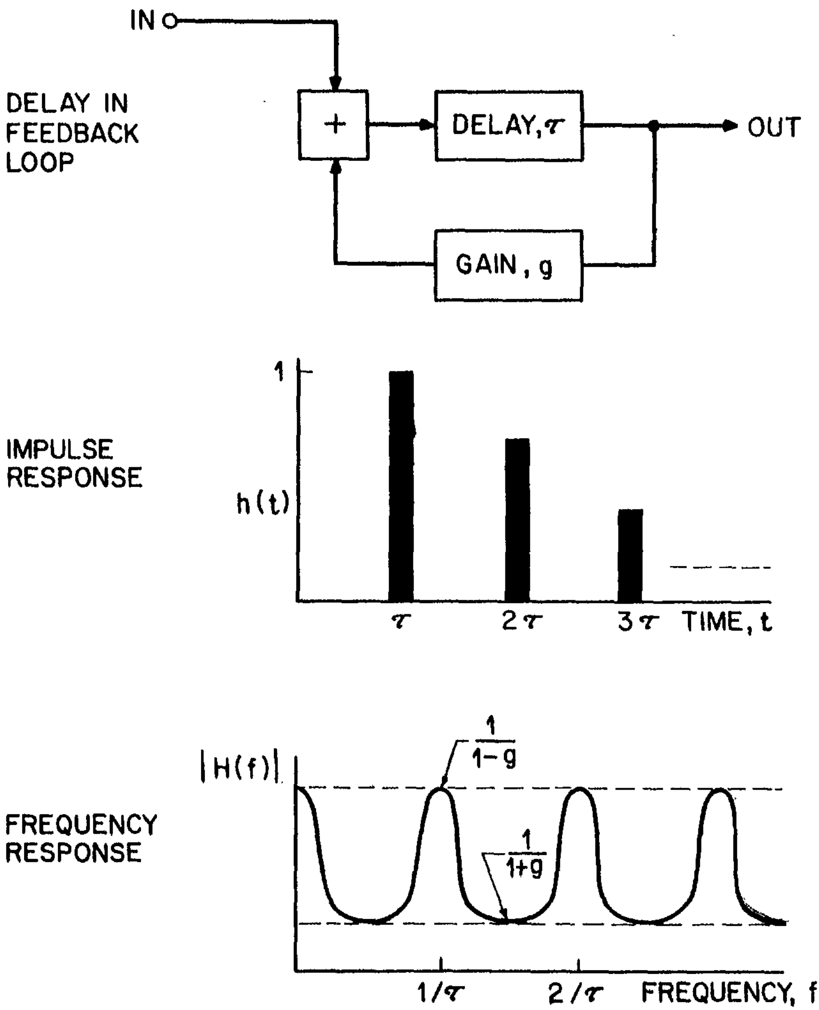


Fig. 1 Simple reverberator with exponentially decaying echo response. Frequency response resembles comb.

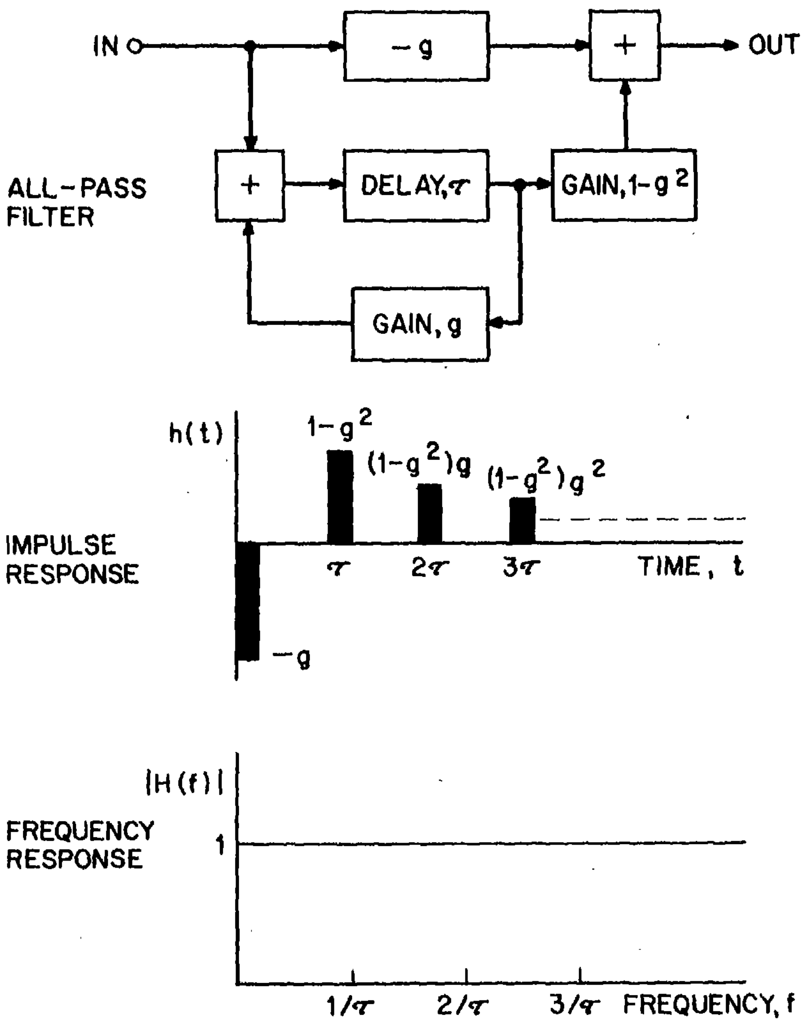


Fig. 2 Modification of simple reverberator. By adding proper amount of undelayed signal, frequency response of reverberator becomes flat (all-pass reverberator).

SERIES CONNECTION OF ALL-PASS LOOPS

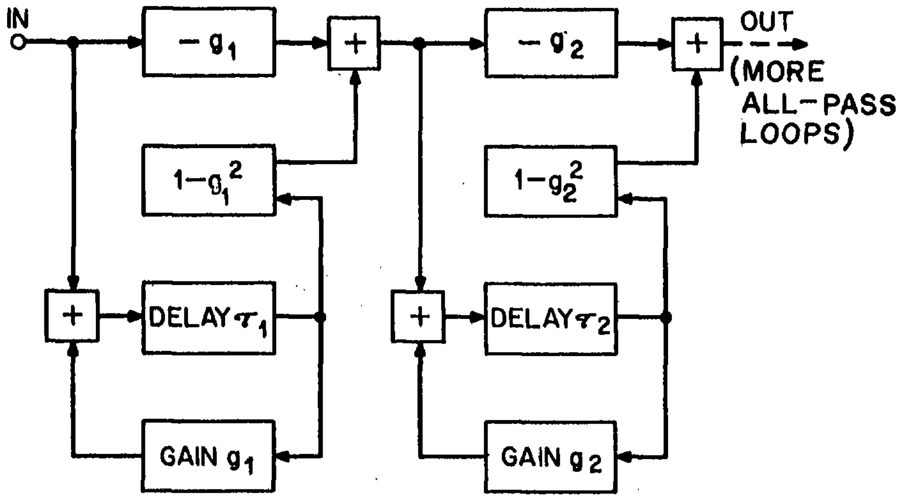
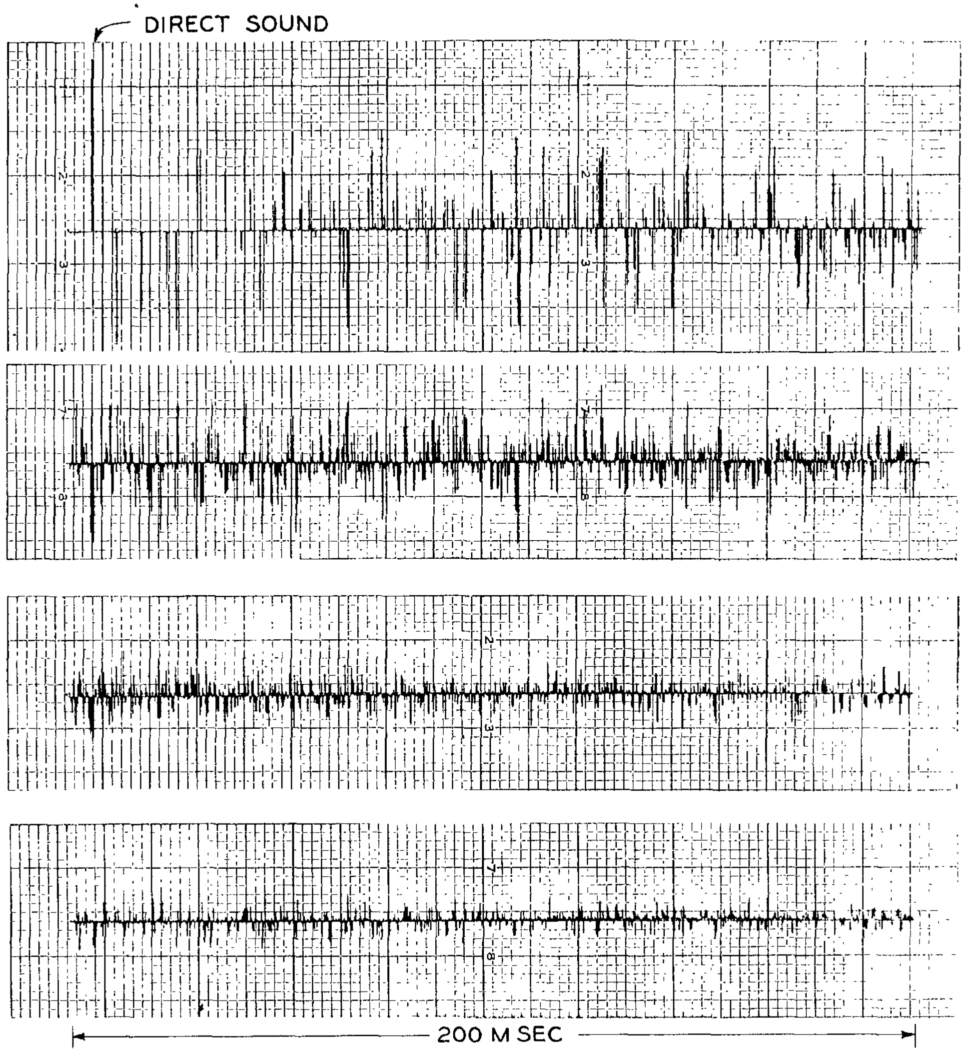


Fig. 3 Series connection of several all-pass reverberators with incommensurate delays to make echo response aperiodic and increase echo density.



IMPULSE RESPONSE OF ALL-PASS REVERBERATOR

Fig. 4 Echo response of all-pass reverberator consisting of five simple reverberators connected in series, as shown in Fig. 3, with delays 100, 68, 60, 19.7, and 5.85 msec. Effective reverberation time: 3 sec.