Natural Sounding Artificial Reverberation*

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Artificial reverberation is added to sound signals requiring additional reverberation for optimum listening enjoyment. This paper describes methods for generating, by purely electronic means, an artificial reverberation which is indistinguishable from the natural reverberation of real rooms. This artificial reverberation can be given any desired characteristics to match different types of music and personal tastes. A method for making the artificial reverberation "ambiophonic" (*i.e.*, three-dimensional) is also described.

SHORTCOMINGS OF EXISTING ELECTRONIC REVERBERATORS

PRESENTLY available electronic reverberators producing multiply delayed echoes by means of delay circuits (magnetic tape or disc, acoustic tubes, springs, etc.) suffer from two main defects:

1. Their *amplitude-frequency responses* are not flat. In fact, they deviate from a flat response so much that an unpleasant "coloration" of many sounds is heard, particularly if only little direct (unreverberated) sound is mixed with the artificially reverberated signal.

2. The *echo density* (*i.e.*, the number of echoes per second at the output of the reverberator for a single pulse at the input) is too low compared to the echo density of a real room. This leads to a "fluttering" of the reverberated sound, especially for short transients.

REQUIREMENTS FOR NATURAL SOUNDING ARTIFICIAL REVERBERATION

How can we avoid the above degradations in artificial reverberators employing delay and feedback? Obviously, the problem of *coloration* could be solved if one knew how to make an artificial reverberator with a *flat* amplitude-frequency response. Luckily, this can be done and the resulting "all-pass" (*i.e.*, passing all frequency components equally) reverberator will be described below.

Concerning the problem of *low echo density,* we have found that approximately 1,000 echoes per second are required for a flutter-free reverberation. In fact, even for extremely short transient sounds, the ear cannot distinguish an echo density of 1,000 per second from any higher value. (Higher echo densities occur in real rooms a short time after onset of the reverberation process.) Unfortunately, echo densities of 1,000 per second are not easily achieved by one-dimensional delay devices. For example, a reverberator consisting of a delay line with 40 msec delay in a feedback loop produces 25 echoes per second. Forty (!) such reverberators *in parallel* are required to give the desired echo density of 1,000 per second. Obviously, this approach is impractical.

Previous investigators have suggested multiple feedback to produce a higher echo density. However, multiple feedback has severe stability problems. Also, it leads to non-flat frequency responses and non-exponential decay characteristics. A much easier solution to the echo density problem would be at hand if one had a basic reverberating unit which one could connect *in series* any desired number of times. In this manner, each unit would effectively *multiply* the number of echoes produced by preceding units. Assuming that each pulse is "spread" into 3 of comparable size, the multiplication factor for each additional unit is about 3. Starting again with a unit which produces 25 echoes per second, only about $(1{,}000/25)^{1/3}$ or between 3 and 4 additional units are required to reach the desired echo density.

The reason why this simple remedy of the echo density problem has not been used previously can be stated quite simply: existing reverberators have highly irregular frequency responses. Connecting only 2, let alone 4 or 5, of these reverberators in tandem results in a totally unacceptable sound quality. However, if the basic reverberator unit has a *flat* frequency response, the series connection of any number of them will have a flat response, too.

Thus, it appears that reverberators with a flat frequency response (all-pass reverberators) would remove the two main obstacles (coloration and flutter) in realizing natural sounding artificial reverberation. The principle of all-pass reverberators will be described in the following section.

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ALL-PASS REVERBERATORS

The simplest echo producing arrangement consists of a delay line, a magnetic tape delay or a similar device which gives a single echo after a delay τ. Its time response to a single impulse, $\delta(\tau)$, at its input is

$$H(t) = \delta(t-\tau). \quad (1)$$

The *spectrum* of the delayed impulse is obtained by taking the Fourier transform of both sides of Eq. (1). If the applied impulse is an ideal impulse (so-called Dirac delta function), then

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

where $\omega = 2\pi f$ is the radian frequency. The absolute value of $H(\omega)$ is unity. This means that all frequencies are passed equally well, without gain or loss.

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

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

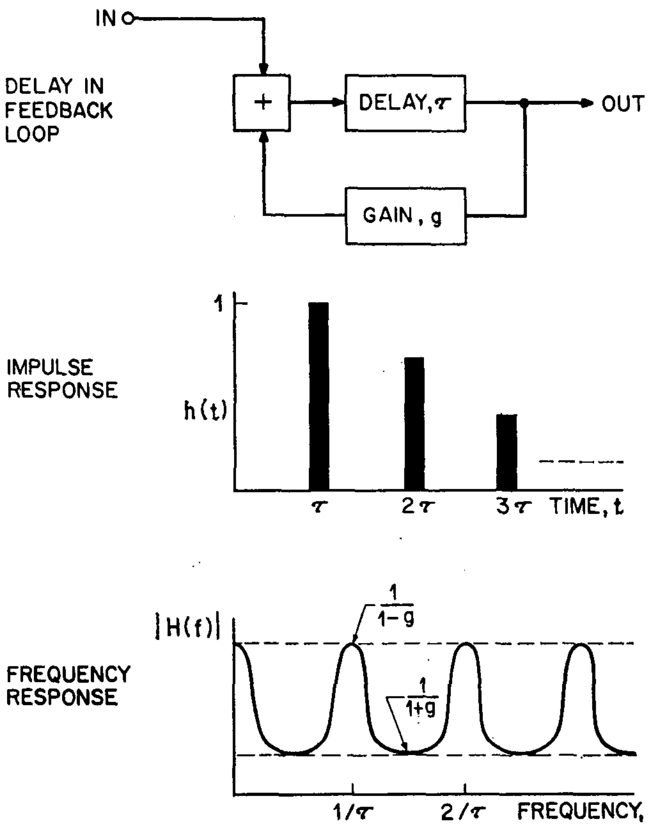

FIG. 1. Simple reverberator with exponentially decaying echo response. Frequency response resembles comb.

The corresponding spectrum is

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

or, using the formula for summing geometric series,

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

By taking the absolute value of $H(\omega)$, one obtains the amplitude-spectrum of the impulse train.

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

This is also the amplitude-frequency response of the reverberator shown in Fig. 1.

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

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

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

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

The ratio of the response maxima to minima is

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

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

The amplitude-frequency response has the appearance of a comb with periodic maxima and minima, as shown at the bottom of Fig. 1. It is these peaks and valleys which impart the undesired "colored" quality to sound reverberated by devices like that shown in Fig. 1.

In a search for better artificial reverberators, Mr. B. F. Logan and the author[1] noted that a certain mixture of the output of the multiply delayed sound and the undelayed sound would result in an equal response of the reverberator for all frequencies. The mixing proportions that accomplish this and result in unity gain for all frequencies are $(-g)$ for the undelayed sound and $(1-g^2)$ for the delayed sound. The corresponding circuit is shown in Fig. 2. Its impulse response is given by

$$h(t) = -g\delta(t) + (1-g^2) [\delta(t-\tau) + g\delta(t-2\tau) + \ldots]. \quad (10)$$

The corresponding frequency response is

$$H(\omega) = -g + (1-g^2) [e^{-i\omega\tau}/(1-ge^{-i\omega\tau})], \quad (11)$$

or

$$H(\omega) = (e^{-i\omega\tau}-g)/(1-ge^{-i\omega\tau}). \quad (12)$$

To show the all-pass character of $H(\omega)$, we rewrite Eq. (12) as follows:

$$H(\omega) = e^{-i\omega\tau} [(1-ge^{i\omega\tau})/(1-ge^{-i\omega\tau})]. \quad (13)$$

[1] M. R. Schroeder and B. F. Logan, *J. Audio Eng. Soc.* **9**, 192 (1961). The use of all-pass filters for artificial reverberators was independently proposed by the Dutch acoustician J. J. Geluk.

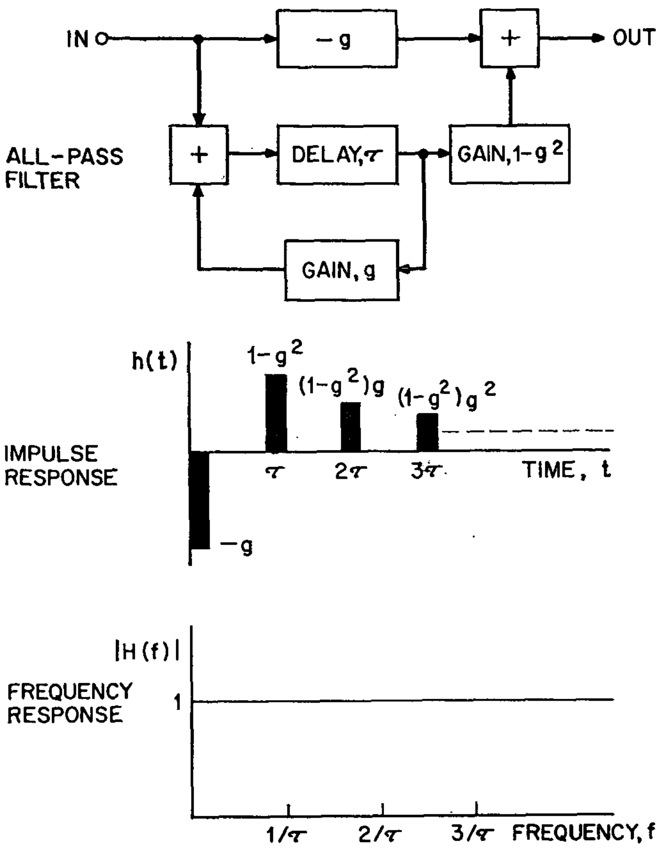

FIG. 2. Modification of simple reverberator. By adding proper amount of undelayed signal, frequency response of the reverberator becomes flat (all-pass reverberator).

The first factor on the right of Eq. (13), $(e^{-i\omega\tau})$, has, of course, absolute value one. The second factor is the quotient of two conjugate complex numbers and thus also has absolute value one. Hence,

$$|H(\omega)| = 1. \quad (14)$$

In other words, the addition of a suitably proportioned undelayed path has converted the comb filter (Fig. 1) into an all-pass filter (Fig. 2). This is not a mere academic result. The conversion of a comb filter into an all-pass filter is accompanied by a marked improvement in the quality of the reverberated sound from the hollow character of the former to the perfectly "colorless" quality of the latter.

Now we are in possession of a basic reverberating unit which passes all frequencies with equal gain and thus avoids the problem of sound coloration. Furthermore, as pointed out above, an arbitrary number of such units can be connected in series to obtain the desired high echo density. Finally, the all-pass reverberator of Fig. 2 shares with other reverberators utilizing delay and feedback the property of *exponential* decay of sound energy as exhibited by acoustically well-designed rooms.

RELATION BETWEEN LOOP GAIN, DELAY AND REVERBERATION TIME

Before discussing series connection and other modifications of the basic reverberator shown in Fig. 2, it is appropriate to establish the relationship between the loop gain g, the delay τ and the resulting reverberation time T. Since W. C. Sabine, room acousticians have defined reverberation time as the time in which the reverberating sound level decays by 60 decibels. For a feedback loop with open loop gain g, the sound level decays by $-20 \cdot \log |g|$ decibels for every trip around the feedback loop. Since every round trip takes τ seconds, the time for a 60 db decay is

$$T = [60/(-20 \cdot \log |g|)] \cdot \tau = (3/\log |1/g|) \cdot \tau. \quad (15)$$

Example: For an open loop gain of $g = 0.708$, or $\log |1/g| = 0.15$, the reverberation time T equals 20 times the delay τ. If an artificial reverberation with $T = 2$ sec is desired, the delay τ must be 100 msec. Such a delay produces 10 echoes per second or only 1/100 of the echo density required for flutter-free reverberation.

INCREASE OF ECHO DENSITY

As mentioned above, the echo density of an artificial reverberator of the all-pass type can be increased by connecting several units in series. This is illustrated in Fig. 3. The

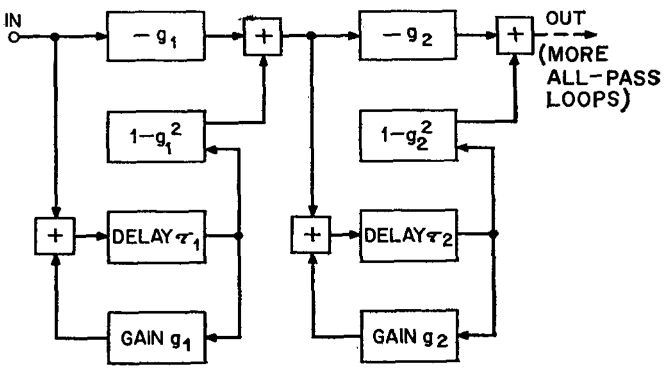

FIG. 3. Series connection of several all-pass reverberators with incommensurate delays to make echo response aperiodic and increase echo density.

delay of each section is made about ⅓ of the preceding delay. Thus, the delay of the n-th unit will be $\tau_1(1/3)^{n-1}$. The gains are most conveniently made equal to about 0.7.

The echo density at the output of the first unit is $1/\tau_1$, at the output of the second $3/\tau_1$, etc. The effective echo density of 5 loops in series will be approximately $81/\tau_1$. For $\tau_1 = 0.1$ sec, the effective echo density will be 810 per second which is sufficiently close to the required 1,000 per second. To avoid echo cancellation and superposition, it is advisable to use *incommensurate* delay ratios rather than the round number 3. Figure 4 shows the resulting echo response for 5 all-pass units connected in series.

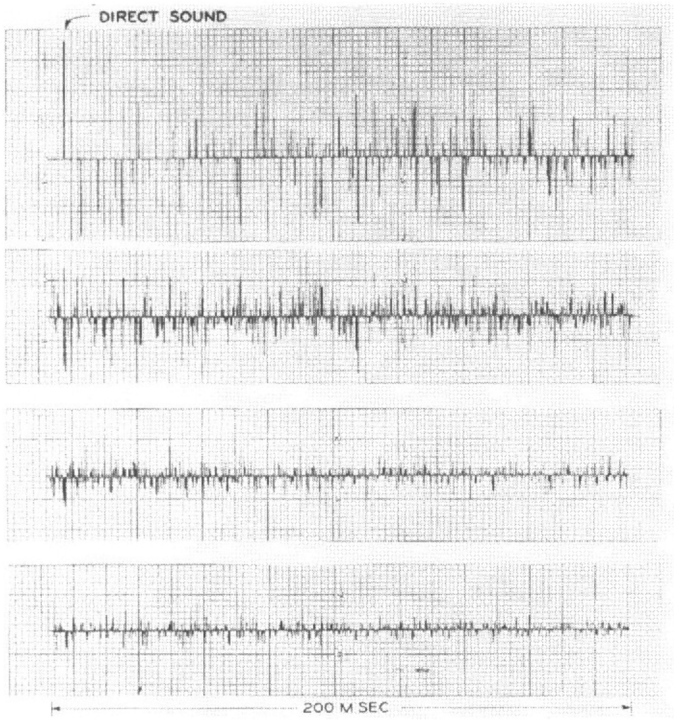

FIG. 4. Echo response of all-pass reverberator consisting of five simple reverberators connected in series as shown in Fig. 3.

FURTHER REFINEMENTS

Our listening experience with all-pass reverberators, employing several units in series, indicates that both coloration and flutter can be completely eliminated. In fact, the responses of our reverberators are *indistinguishable* on these two counts from those of real rooms. Thus, we felt encouraged to imitate some of the more subtle characteristics of natural reverberation. These are: 1. mixing of reverberated sound and direct sound, 2. the introduction of a time gap between the direct sound and the reverberation, and 3. a dependence of the reverberation time on frequency. All three modifications are important if a high degree of realism is desired. Of course, none of these modifications should introduce any audible coloration or flutter. We shall first discuss a method of mixing direct sound and reverberation without violating the all-pass principle.

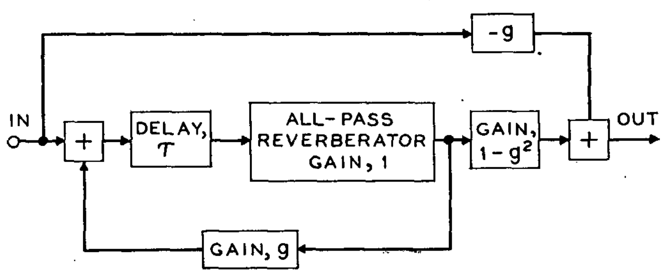

FIG. 5. All-pass reverberator with variable ratio of direct-to-reverberated sound. It produces a non-exponential decay of the reverberated sound.

MIXING OF DIRECT SOUND AND REVERBERATION

The sound emerging from a series of all-pass reverberating units contains a negligible amount of direct sound. A possible method of mixing direct sound with this reverberation without violating the all-pass principle is shown in Fig. 5, where the box labeled "all-pass reverberator" contains a series connection of all-pass units as shown in Fig. 3. The ratio of direct sound energy to reverberant sound energy is $g^2/(1-g^2)$. If it is desired to make this ratio equal to 6 db or 4:1, for example, g has to be made equal to $2/\sqrt{5} = .893$. The delay τ serves to introduce the desired time gap between the direct sound and the onset of the reverberation. In a real room, this gap, as well as the ratio of direct-to-reverberated sound, depends on the positions of sound source and listener. A typical value for τ for a large concert hall and a seat in the middle of the orchestra is 30 msec.

A peculiarity of the reverberator illustrated in Fig. 5, which is not immediately apparent, is the fact that the reverberation it produces does not decay exponentially. This may actually be pleasing for some sounds. (In real rooms non-exponential decays point to a lack of spatial "diffusion" of the sound field.)

In the following section we shall describe a reverberator which has a jagged frequency response (much like that of real rooms) but which is subjectively indistinguishable from a perfectly flat response. It allows mixing of direct and reverberated sound without producing non-exponential decays. Also, the onset of its reverberation is more like that in real rooms.

THE COMB FILTER APPROACH

One might ask oneself: "Why insist on perfectly flat frequency responses for an artificial reverberator if real rooms have highly irregular frequency responses?" In fact, for many sounds the human ear cannot distinguish between a flat response and the irregular response of a room, which fluctuates on the average about 10 db and has extreme variations of 40 db or more. Psycho-acoustic experiments now in progress at Bell Telephone Laboratories indicate that such extreme response irregularities are imperceptible when the density of peaks and valleys on the frequency scale is high enough. There are about 15 large response peaks in every 100 cps interval for a room with 1 sec reverberation time. Thus, one might hope that if an artificial reverberator has a comparable number of response peaks it might sound just as good as a real room. We have been able to confirm this expectation by subjective evaluations of the responses of reverberators consisting of several comb filters (see Fig. 1) connected in parallel. For a delay of 0.04 sec, the number of response peaks per 100 cps is 4. Thus, between 3 and 4 comb filters in parallel (and not just 2 as in some presently available reverberators), with incommensurate delays, are required to approximate the number of peaks of the frequency response of a room having a reverberation time of $T = 1$ sec. Also, the open loop gain of the comb filters should not exceed about 0.85 or –1.4 db to keep the response fluctuations from being excessive. (A single comb filter with an open loop gain of 0.85 has amplitude response fluctuations equal to $(1 + 0.85)/(1 - 0.85) = 12.3:1$ or about 22 db!)

To insure a sufficient echo density several all-pass reverberators of the kind shown in Fig. 2 have to be connected in series with the comb filters. The number of echoes per second of a comb filter having a delay of 0.04 sec is 25. Four such comb filters in parallel give 100 echoes per second, which is short by a factor of 10 from the goal of 1,000 echoes per second. Thus, two all-pass reverberator units, each multiplying the number of echoes by about 3, are required to furnish the required number of echoes per second.

A complete circuit diagram of such a reverberator is shown in Fig. 6. The values of the delays τ_1 through τ_4 are spread

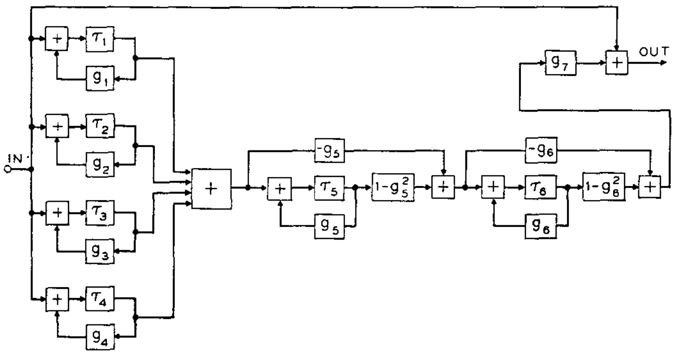

FIG. 6. Block diagram of reverberator using both comb filters (in parallel) and all-pass filters (in series). This type of reverberator allows the widest choice of specifications such as mixing ratios, delay of reverberated sound and kind of decay.

over a range of approximately 1:1.5, say between 30 and 45 msec. The shortest delay determines the time gap between direct sound and reverberation. The gains g_1 through g_4 are adjusted according to the formula $T = 3\tau_n/(-\log|g_n|)$, where T is the desired reverberation time. Good choices for the delays τ_5 and τ_6 are about 5 and 1.7 msec. The gains g_5 and g_6 are most conveniently made equal to 0.7. The gain g_7 finally determines the amount of reverberated sound mixed with the direct sound.

Extensive listening tests have shown that artificial reverberators like that shown in Fig. 6 are subjectively indistinguishable from real rooms—not only for speech and music but even for such "trying" signals as very short clicks and wideband Gaussian noise.

FREQUENCY-DEPENDENT REVERBERATION TIME

If it is desired to make the reverberation time a function of frequency, the gains g_1 through g_4 in Fig. 4 have to be made frequency dependent. For this, a simple RC-section in each feedback loop will suffice. In this manner further realism can be added to the artificial reverberation by making the reverberation time larger for the low frequencies. This trend of reverberation time with frequency is typical of many concert halls and cathedrals.

[2] R. Vermeulen, *J. Audio Eng. Soc.* **6**, 124 (1958).
[3] M. R. Schroeder, *J. Acoust. Soc. Am.* **33**, 1061 (1961).

AMBIOPHONIC REVERBERATION

One more modification of our artificial reverberator is needed to enable it to produce life-like *three-dimensional* reverberation. That is to make it "ambiophonic."[2,3] As long as one has only a single reverberated signal, even if radiated from more than one loudspeaker, one will not be able to recreate the spatially diffuse character of real reverberation. For this, one needs several differently reverberated signals fed to a number of loudspeakers distributed around the listener.

A simple way of generating several differently reverberated sound signals without additional equipment is illustrated in Fig. 7. Here the order of the comb filters (C_1

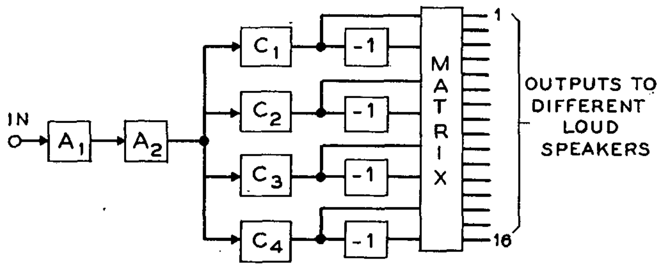

FIG. 7. Connection of all-pass filters (A) and comb filters (C) for ambiophonic installation. Matrix forms sixteen different outputs to drive loudspeakers distributed around the listening area.

through C_4) and the all-pass filters (A_1 and A_2) has been inverted compared with Fig. 6. The outputs and the inverted outputs of the 4 comb filters are connected to a resistance matrix which forms up to 16 different combinations of the comb filter outputs. Each combination uses each comb filter output or its negative exactly once. The matrix outputs are supplied to power amplifiers and loudspeakers distributed around the listening area. In this manner a highly diffuse reverberation can be achieved.

ELECTRO-ACOUSTIC CONVERSION OF EXISTING CONCERT HALLS

The artificial reverberators described here are applicable to the important problem of increasing reverberation time and sound diffusion in auditoriums, concert halls and theaters by purely electro-acoustic means without requiring architectural changes. In these applications high sound quality is of paramount importance. Ultimate acceptance of electro-acoustic techniques in concert halls and opera houses is assured only if the artificially added sound is indistinguishable from the naturally reverberated sound. The reverberators described here are an important step in this direction.

ACKNOWLEDGMENT

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Dr. Schroeder's biography appeared in the January issue of the Journal.