

Nov. 12, 1963

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3,110,771

ARTIFICIAL REVERBERATION NETWORK

Filed Sept. 29, 1960

3 Sheets-Sheet 1

FIG. 1

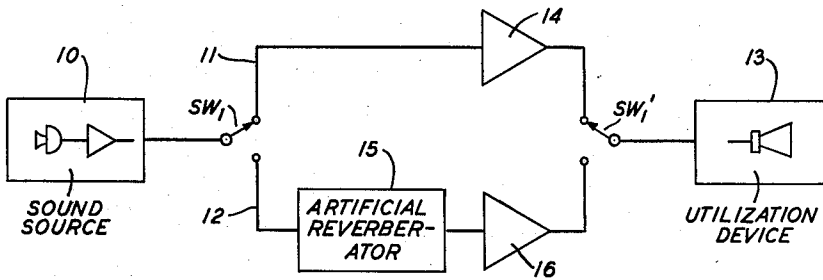


FIG. 4

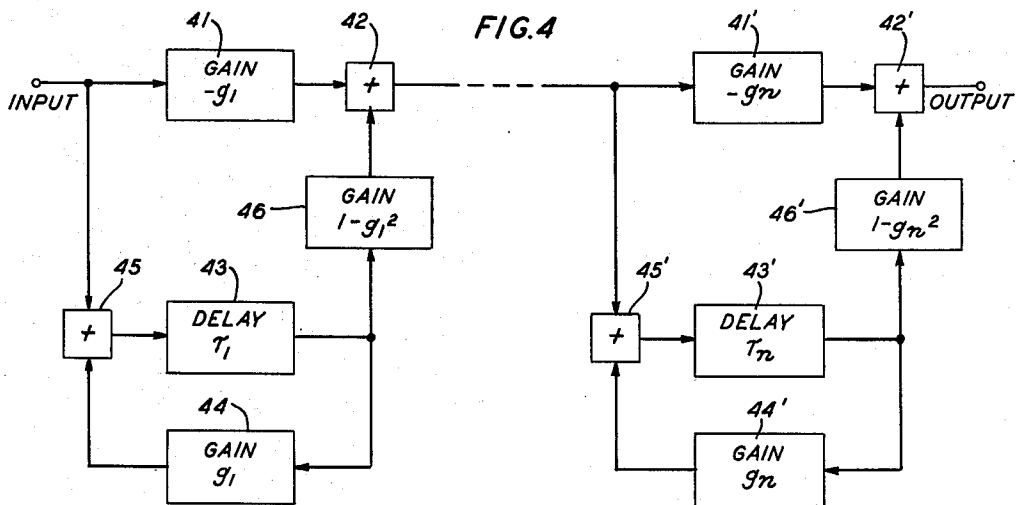
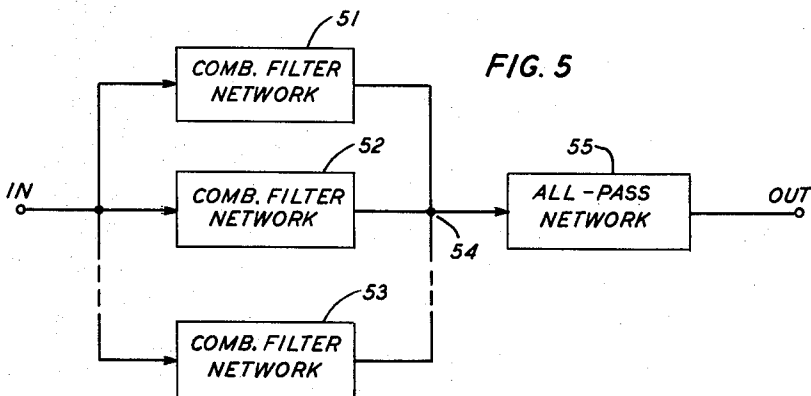


FIG. 5



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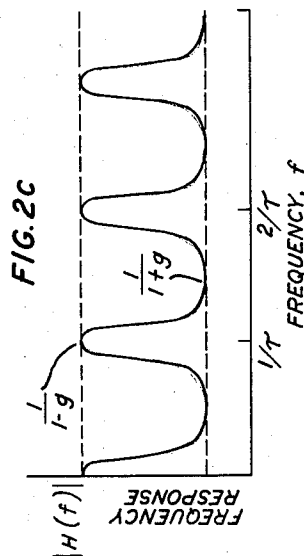
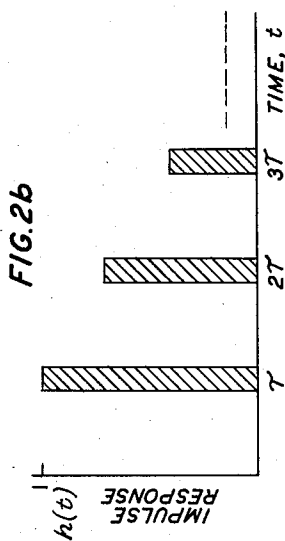
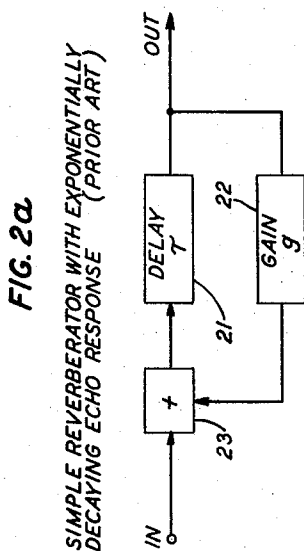
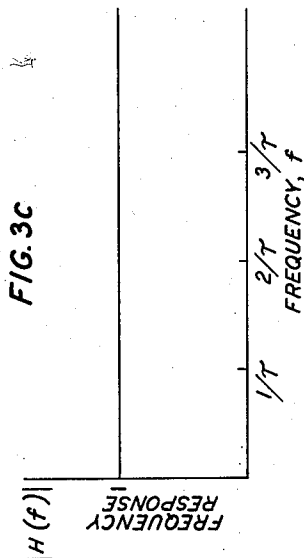
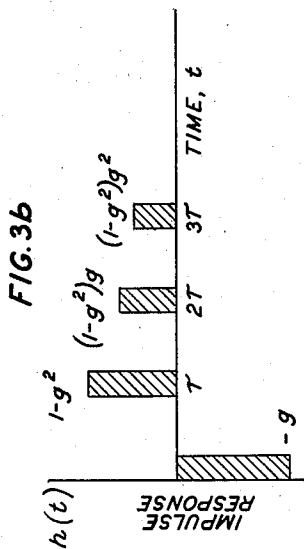
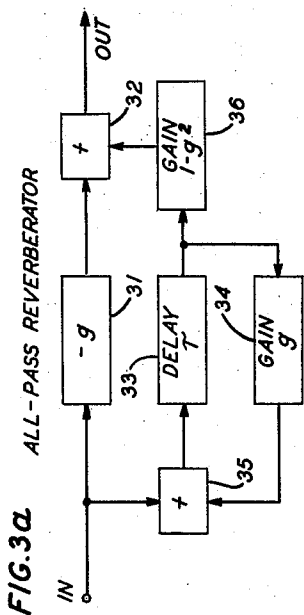
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ARTIFICIAL REVERBERATION NETWORK

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3 Sheets-Sheet 2



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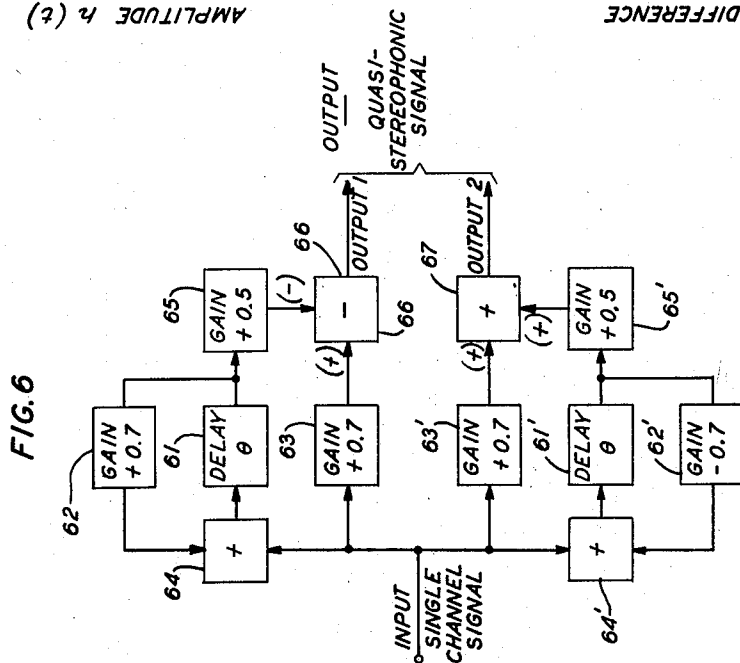
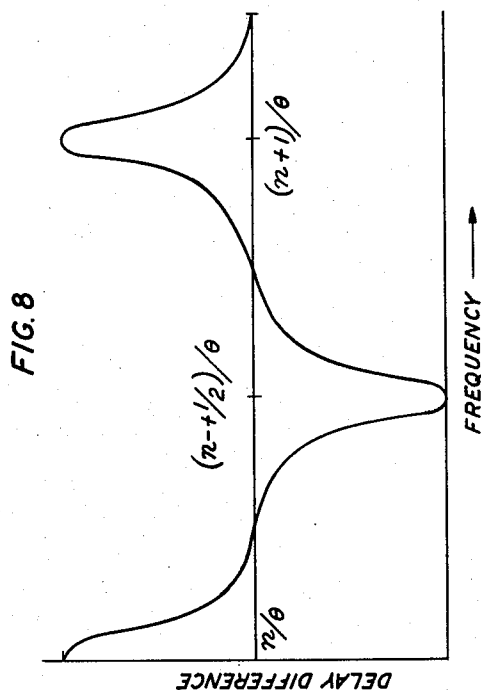
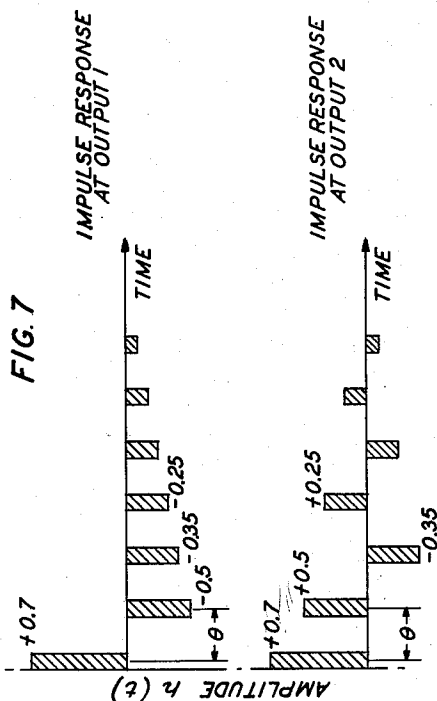
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ARTIFICIAL REVERBERATION NETWORK

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3 Sheets-Sheet 3



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ARTIFICIAL REVERBERATION NETWORK

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8 Claims. (Cl. 179—1)

This invention relates to the processing of speech and music signals, and more particularly to apparatus and methods for introducing artificial or synthetic reverberation in such signals to enhance their subjective quality.

In an auditorium, sound proceeds from its source to each of the boundaries of the auditorium where it is repeatedly reflected from one surface to another. At each impact some of the energy of the sound is absorbed by the surface so that the reflected sound is weaker than the original. As a result a finite time is required after the sound excitation has stopped for the sound energy to become inaudible. This effect is called reverberation and is evidenced by the persistence of sound for a finite time after the emission of sound by its source has ceased.

In the study of the acoustics of auditorium it has been found that a considerable time of reverberation is desired particularly for music since the blending of tones produces a pleasing effect. Consequently, electronic devices are widely used to enhance the subjective quality of sound by adding reverberation. For example, record manufacturers and broadcasters employ artificial reverberation to enhance a performance recorded in the open air or in other locations that lack the preferred degree of natural reverberation. Room acousticians make use of artificial reverberation to increase the reverberation in auditoriums designed for speaking in order to convert them for use as concert halls. In the home, artificial reverberators are used to suit the particular taste of the individual listener. Ideally, artificial reverberators should act on sound signals exactly as real three-dimensional rooms. This is not simple to achieve, however, unless a reverberation chamber or the electrical equivalent of a three-dimensional space is employed. As a result large and expensive reverberation chambers have been widely used by broadcasting stations and record manufacturers because of their high quality and lack of undesirable side effects. Of course, these chambers are not truly artificial reverberators.

Electronic reverberators are much to be preferred, however, since they are both cheaper than real rooms and may be used in nonprofessional installations, for example, in home music systems, as well as in professional applications. They can also be employed to increase the reverberation time of auditoriums, thereby adapting them to concert hall use without changing the architecture. Unfortunately, electronic reverberators consisting of delay-lines, disc or tape-delay, amplifiers and the like produce rather unnatural reverberation due to detrimental fluctuations in their amplitude-frequency response characteristic; fluctuations that give them comb filter-like frequency responses.

It is the principal object of the present invention to enhance a sound with reverberation by means of an electronic artificial reverberator free from the defects outlined above.

Before entering upon a discussion of the theoretical aspects of the invention and of the apparatus by which the theoretical principles are turned to account it is believed helpful to consider some of the important properties of large auditoriums.

A room can be characterized by its normal modes of vibration. The density of its modes is nearly independent

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of room shape and is proportional to the square of the frequency:

$$\text{Number of modes per c.p.s.} = \frac{4\pi V}{c^3} \times f^2 \quad (1)$$

where V is the volume of the room in cubic meters, c is the velocity of sound in meters per second, and f is the frequency in cycles per second. Above a certain critical frequency,

$$f_c = 2000 \sqrt{T/V} \quad (2)$$

where T is the reverberation time in seconds, the density of modes becomes so high that many modes overlap. In this frequency range, which is of prime interest for large rooms, 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. However, the fluctuations are so rapid on the frequency scale that the ear, in listening to a non-steady sound, does not perceive the irregularities. Actually the response fluctuations can be heard only by exciting the room with a sine wave of slowly varying frequency and listening with one ear. When the room response is measured, using instead of a sine wave a psycho-acoustically more appropriate test signal such as a narrow band of noise, the response is much smoother. It is this apparent smoothness of a room's frequency response which has been found to be particularly difficult to imitate with artificial reverberators.

A flat frequency response is not the only requirement for a high quality reverberator. The transient behavior of a room is also an important factor in natural sounding reverberation; specifically the response of a room to excitation with a short impulse is of great significance. If the sound pressure at some location in a room is recorded as a function of time, an impulse corresponding to the direct sound that has traveled from the sound source to the pickup point without reflection at the walls is first observed. After that, a number of discrete low order echoes are produced which correspond to one or a few reflections at the walls and ceiling of the room. Gradually the echo density increases to a statistical "clutter." In fact, the echo density is proportional to the square of the elapsed time:

$$\text{Number of echoes per second} = \frac{4\pi c^3}{V} \times t^2 \quad (3)$$

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 echoes start overlapping is about

$$t_c = 5.10 \cdot 10^{-5} \sqrt{V/\Delta t} \quad (V \text{ in m.}^3) \quad (4)$$

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. and is determined by the collective behavior and interference of many overlapping echoes.

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" echoes, i.e., periodic echoes 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 "diffusers" in the sound path.

From this brief consideration of room behavior, the

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conditions that must be met by artificial reverberators may be tabulated as follows:

(1) The frequency response must be flat when measured with narrow bands of noise and of a bandwidth commensurate with 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 echoes).

In addition to these five conditions, a sixth one must be met which is not apparent from the above review of room behavior. It is violated by most 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 an extremely important one, particularly in systems in which 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 transmissions, are inherently periodic and special precautions have to be taken to make these periodicities inaudible. It is because most prior art reverberation apparatus fails in this condition that professional recording studios and the like heretofore have found it necessary to resort to large reverberation chambers or other analog devices to provide the desired effect.

As distinguished from prior art reverberators using simple feedback loops that include delay-line, disc or tape-delay elements for producing a sequence of echoes (typically with a comb filter-like response) the present invention employs a passive network that introduces phase shift or delay without however introducing appreciable attenuation at any frequency. In particular, a feedback network with an all-pass characteristic that combines at its output portions of a multiply delayed sound and a portion of the undelayed sound has the required flat amplitude-frequency response. To achieve a sufficiently high echo density and to avoid audible periodicities, a number of all-pass feedback loops with incommensurate loop delays are connected in tandem. The over-all network retains an all-pass characteristic, i.e., it has a flat frequency response, but also has a high echo density and an aperiodic echo response. Thus the impulse response is not repetitive at an audible date. Although any number of individual stages may be connected in series, it has been found that the most natural sounding reverberation effect results with two to five stages.

The artificial reverberator, in accordance with the present invention, thus fulfills, in its simplest form, conditions (1), (3), and (6) ideally, and when iterated satisfies conditions (2), (4), and (5) without violating the others.

The all-pass characteristic of the reverberation network is further turned to account in the present invention effectively to separate a single channel source of sound into a dual channel signal that, in many respects, behaves as a dual channel stereophonic signal. That is to say, the apparatus may be employed in a parallel network configuration whereby a single channel signal, passed in parallel through individual reverberation networks selected with different phase characteristics, is effectively split into two quasi-stereophonic signals that give a listener all of the

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fullness of multichannel stereophony but which, of course, do not permit correct localization of individual sound sources.

The invention will be further apprehended from the following detailed description of illustrative embodiments thereof taken in connection with the appended drawings, in which:

FIG. 1 is a simplified schematic block diagram of a sound system adapted to add artificial reverberation to sound signals;

FIGS. 2a, 2b, and 2c are, respectively, a block diagram of the basic unit of prior art reverberation apparatus and the impulse response and frequency response relations which hold for the reverberation apparatus;

FIGS. 3a, 3b, and 3c are, respectively, a block diagram of an all-pass reverberation network in accordance with the present invention and its impulse response and frequency response characteristics;

FIG. 4 is a block schematic diagram showing in somewhat greater detail an iterated network for producing artificial reverberation in accordance with the present invention;

FIG. 5 is a block schematic diagram showing another form of artificial reverberator in accordance with the invention;

FIG. 6 is a block schematic diagram of an artificial reverberation network adapted to produce a quasi-stereophonic response from a single channel audio signal;

FIG. 7 illustrates the impulse response at the two outputs of the apparatus of FIG. 6; and

FIG. 8 is a graph illustrating the envelope delay difference between output signals 1 and 2 of the apparatus of FIG. 6 as a function of frequency.

The function and manner of using the apparatus to be described are shown in the block schematic diagram of FIG. 1 wherein an audio signal originating, for example, in sound source 10 that includes a microphone, tape transducer, or the like, is applied by one of two parallel paths 11 or 12 to a utilization device 13, for example, a loudspeaker. In normal operation switch SW1 transfers signals from source 10 to utilization device 13 via path 11 wherein nonreverberant amplification takes place in conventional amplifier 14. For the enhancement of sound signals by synthetic reverberation SW1 transfers signals from source 10 to the utilization device via path 12. With this connection the signals are passed through artificial reverberator 15 wherein a controller degree of reverberation is introduced into the signal to transform it to one with an optimum degree of reverberation, i.e., a degree of reverberation commensurate with the auditorium in which the sound eventually is reproduced. In addition, the modified signals may, if desired, be passed through conventional amplifier 16 to raise the signal to any desired level. It will be appreciated by those skilled in the art, of course, that the switching arrangement of FIG. 1 merely indicates the environment in which the artificial reverberator 15 finds use. In most installations, however, the artificial reverberator will be an integral part of the amplification apparatus so that the switching arrangement of FIG. 1 may be dispensed with. By suitable control, the degree of reverberation may be adjusted so that reverberation may be effectively removed from the circuit entirely by adjusting the degree of reverberation to zero. It is in this latter form that it finds particular use in home music systems; i.e., in so-called "high-fidelity" apparatus.

FIG. 2A shows by way of introduction a simple prior art reverberator that yields multiple echoes that decay exponentially. It comprises a delay line, disc, or tape-delay element 21 that gives an echo after a delay time τ . By itself, the delay element passes all frequencies equally well without gain or loss. In order to produce multiple echoes without using additional physical delay apparatus, the delay line is placed in a feedback loop 22 with gain g less than one so that the loop will be stable. The im-

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pulse response of the network, illustrated at FIG. 2b, is an exponentially decaying repeated echo whose impulse response $h(t)$ is:

$$h(t) = \delta(t-\tau) + g\delta(t-2\tau) + g^2\delta(t-3\tau) + \dots \quad (5)$$

where $\delta(t)$ is the Dirac delta-function (an ideal impulse), and g is the feedback loop gain. The corresponding complex frequency response $H(\omega)$ is

$$H(\omega) = e^{-i\omega\tau} + ge^{-2i\omega\tau} + g^2e^{-3i\omega\tau} + \dots \quad (6)$$

where ω is the radian frequency. Using the formula for summing geometric series, the response becomes

$$H(\omega) = \frac{e^{-i\omega\tau}}{1 - ge^{-i\omega\tau}} \quad (7)$$

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

$$|H(\omega)|^2 = \frac{1}{1 + g^2 - 2g \cos \omega\tau} \quad (8)$$

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

$$H_{\max.} = \frac{1}{1-g} \quad (9)$$

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

$$H_{\min.} = \frac{1}{1+g} \quad (10)$$

The ratio of the response maxima to minima is

$$\frac{H_{\max.}}{H_{\min.}} = \frac{1+g}{1-g} \quad (11)$$

For a loop gain of $g=0.7$ (-3 db), this ratio is $1.7/0.3=5.7$ or 15 db.

The amplitude-frequency response of a delay in a feedback loop thus has the appearance of a comb with periodic maxima and minima. This response is illustrated in FIG. 2c. Each response maxima corresponds to one normal mode. The natural frequencies are thus spaced $1/\tau$ c.p.s. apart. The 3 db-bandwidth of each peak is approximately

$$\Delta f = \frac{-\ln g}{\pi\tau} \quad (12)$$

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

$$\Delta f = \frac{1}{20\pi \log e} \frac{-\gamma}{\tau} = .0367 \frac{-\gamma}{\tau} \quad (13)$$

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 accordance with the present invention, an equal response of the reverberator for all frequencies is obtained by adding a selected amount of undelayed sound to the multiply delayed sound of the feedback network. A suitable network is shown in FIG. 3a. It includes both a direct path for applied signals via amplifier 31 and a path that includes a feedback loop. Signals from both of these are combined in adder 32. Delay element 33 provides the required delay τ for input signals, and an amplifier 34 feeds back the delayed signals via adder 35 to the input of the delay element 33. To establish the mixing ratio (of multiply delayed and undelayed signals) that results in an equal response of the network for all frequencies and, in addition, maintains unity gain for all frequencies, amplifier 31 in the undelayed path is provided with a gain ($-g$) and the multiply delayed path is provided with a gain of $(1-g^2)$. With the arrangement shown, the feedback network itself has unity gain so that

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it is convenient to pass the multiply delayed signals through amplifier 36 with a gain $(1-g^2)$ before combining them with undelayed signals in adder 32. The impulse response of the over-all network is given by

$$h(t) = -g\delta(t) + (1-g^2)[\delta(t-\tau) + g\delta(t-2\tau) + \dots] \quad (14)$$

and is illustrated in FIG. 3b. The corresponding frequency response is

$$H(\omega) = -g + (1-g^2) \frac{e^{-i\omega\tau}}{1 - ge^{-i\omega\tau}} = \frac{e^{-i\omega\tau} - g}{1 - ge^{-i\omega\tau}} \quad (15)$$

or expressed in another algebraic form

$$H(\omega) = e^{-i\omega\tau} \frac{1 - ge^{i\omega\tau}}{1 - ge^{-i\omega\tau}} \quad (16)$$

Since the first factor on the right-hand side of the equation has an absolute value of one, and the second factor is the quotient of two conjugate complex "vectors," the absolute value of $H(\omega)$ is also one. Thus

$$|H(\omega)| = 1 \quad (17)$$

In other words, the addition of a suitably proportioned undelayed path has converted the comb filter (Equation 8) into an all-pass network (Equation 17), with a frequency response of the sort shown in FIG. 3c. The conversion 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. Hence, the reverberating element 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 from Equation 16

$$\phi(\omega) = \omega\tau + 2 \arctan \frac{g \sin \omega\tau}{1 - g \cos \omega\tau} \quad (18)$$

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

$$\frac{d\phi}{d\omega} = \frac{1 - g^2}{1 + g^2 - 2g \cos \omega\tau} \tau \quad (19)$$

which has exactly the same dependence on ω as the square amplitude-frequency response $|H(\omega)|^2$ of the corresponding comb filter (see Equation 8). The physical significance of $d\phi/d\omega$ is that of the envelope or "group delay" of a narrow band of frequencies around ω . According to (19), 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, $2\pi n/\tau$ ($n=0, 1, 2 \dots$), of the filter. The half-width of the envelope delay peaks is the same as that for squared amplitude (see Equation 12). 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 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 feed-

back loop the sound is attenuated γ db. Thus, the 60 db decay time is

$$T = \frac{60}{-\gamma} \tau \quad (20)$$

For $\gamma = -3$ db, $T = 20\tau$. Open loop gains greater than .7 (-3 db) are not considered because in practice it is difficult to maintain the desired closed loop characteristics with gains too close to one. Thus, in order to achieve 2 seconds of artificial reverberation, for example, the loop delay must be 0.1 sec. With this loop delay the basic reverberating element shown in FIG. 3a produces one echo every one-tenth of a second. This constitutes a most undesirable periodic flutter echo. Also, the echo density (ten echoes per second) is much too low to give a continuous reverberation. Thus, conditions (4) and (5) are also violated.

In accordance with the present invention a less periodic time response and a greater echo density is obtained without sacrificing the all-pass characteristic by connecting several all-pass feedback loops with incommensurate loop delays in series as illustrated in FIG. 4. Conditions (2), (4), and (5) are met with this arrangement. By incommensurate delays it is meant that the delays are selected with values having no common divisor, i.e., ratios of the delays are not rational. With a tandem arrangement of all-pass networks 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 (as in Equation 19) with different τ 's. Since each of these terms appreciably delays only approximately one-ninth of all frequency components, it appears that at least five or possibly ten all-pass feedback loops in series are required. Actually, for many applications, especially those in which musical sounds are to be enhanced, it has been found that two or three serially connected networks suffice.

In the artificial reverberator shown in FIG. 4 only two all-pass networks are shown. The element designation numerals correspond to the element of the network of FIG. 3a. As suggested above, additional networks may in practice be connected in tandem with those shown (as indicated by the dashed line interconnecting the networks) if the particular application benefits from the additional stages. The structure of each of the networks is identical but the parameters of each are selected for progressively diminishing loop delays τ . A geometric staggering of delays is preferred, however, with the longest delay selected to be essentially equal to the over-all reverberation time desired for the reproduced sound. It must be noted that the over-all reverberation time is not precisely equal to the sum of the individual reverberation times; it is essentially a function of the longest delay element used in any one of the networks. The loop gains g are selected to give nearly equal pulse amplitudes in the output wave. Preferably, loop gains of successive networks are equal. From a cursory observation of a single all-pass network, e.g., that of FIG. 3a, it can immediately be observed that for loop gains nearly equal to zero, only one impulse appears in the response; that is, the network is reduced to a simple delay that yields a single echo after delay τ . Similarly, when the loop gain g is nearly one, only the undelayed signal reaches the output. It has been found that a loop gain in the range of 0.5 to 0.8 provides the best over-all distribution of echo amplitudes in the output signal. Optimally, a value of 0.707 results in successive pulse amplitudes (in the impulse response) of 0.5, 0.35, 0.25, .175 . . . in an exact geometric or exponential progression. This, of course, is a particular advantage since the decay pattern for natural reverberation is also exponential.

For completeness, the parameters for a typical reverberator for $T=1$ sec. with five tandemly connected stages are given below. These values have been found, both in listening tests and by analyses of measured data, to

eliminate completely "sound coloration." The problem of unequal response to different frequencies is solved with these values and audible flutter echoes are avoided. The addition of one or two supplemental feedback loops with longer delays allows longer reverberation time.

Stage	Delay τ (milli-sec.)	Loop gain g
1	40	+0.7
2	27	-0.7
3	24	+0.7
4	7.8	+0.7
5	2.8	+0.7

Since the all-pass reverberator networks in accordance with the invention have essentially flat frequency responses they may be used to improve the echo density of any form of electronic reverberator, e.g., reverberators that comprise a parallel network of comb filters. Thus, for example, the number of parallel network comb filter branches in such a reverberator may be reduced by the addition of one or more tandemly connected all-pass networks to achieve the same echo density. Alternatively, the addition of one or more all-pass reverberators to an existing comb filter network (without reducing the number of parallel branches) further improves the reverberation characteristic by filling in the impulse response with additional pulses to yield a considerably higher echo density than possible with the comb filter network alone. It is thus apparent that the use of one or more of the relatively simple all-pass reverberators, as an accompaniment to electronic reverberators with imperfect impulse characteristics, can result in a substantial reduction in the complexity of the reverberator network with the same or improved performance. Economic considerations ordinarily dictate the specific combination of reverberation networks that should be used to fulfill a specific need. In any event, the addition of an all-pass reverberator network to a reverberator that already has an acceptable response characteristic results in an increased echo density with no deterioration of the response characteristics. Similarly, for a reverberator with a poor response characteristic, the echo density is substantially improved and the over-all characteristic also is generally improved.

FIG. 5 illustrates, by way of example, one possible application of an all-pass reverberator to an electronic comb filter reverberator to improve its characteristic. Comb filters 51, 52 . . . 53 receive non-reverberant signals from an input terminal in parallel and supply at terminal 54 reverberant signals that in general have a comb filter-like response, e.g., that of FIG. 2c, with relatively large gaps in the frequency spectrum. Each of the comb filters typically comprises a network of the sort illustrated in FIG. 2a. Improvement in the response characteristic and echo density is obtained by passing the processed signals through all-pass network 55 to an output terminal. All-pass network 55 may conveniently comprise the all-pass reverberator illustrated in FIG. 4. In general, the more tandemly connected all-pass reverberators employed the better will be the improvement.

While the invention finds its primary use in increasing the reverberation time of auditoriums and concert halls by purely electro-acoustic means, and in the enhancement of single-channel sound signals, it also may be used to advantage for creating, from a single-channel sound, a spatial acoustic impression when the sound signal is supplied to two or more loudspeakers or earphones.

In creating a quasi-stereophonic effect, the single-channel signal has heretofore been split into two signals by means of two networks, e.g., delay networks having different impulse responses connected in the sound channel to form a pair of interleaved comb filters. The resulting spatial impression represents only one, though important, aspect of truly stereophonic sound, namely, its am-

bience. The individual sound source cannot be located correctly and in most cases cannot be separated by the listener. The effect, nevertheless, has been found subjectively pleasing and in many cases a worthwhile improvement over single-channel reproduction. However, networks using comb filters give rise to unpleasant sound qualities which, although not quite as pronounced as in reverberation apparatus using comb filters, are nevertheless easily perceptible.

This disadvantage is overcome in accordance with the present invention by using a pair of all-pass reverberation networks, e.g., a pair of networks of the sort shown in FIG. 3a, to split the single-channel audio signal into two individual portions. In particular, because the intensity versus frequency response of the all-pass networks is flat, the networks do not introduce the hollow or reedy sound quality associated with comb filters.

Apparatus for creating a spatial acoustic impression is shown in FIG. 6. A pair of all-pass reverberator networks connected with a common input and individual outputs is employed. Each network comprises a delay line 61 in a feedback loop 62. For one network, e.g., 62, a loop gain of $+1/\sqrt{2}=+.707$ is employed and for the other, e.g., 62', a loop gain of $-.707$ is used. A straight, undelayed, path 63 with $+.707$ gain is provided in each network. Adders 64 are used to complete both feedback loops, and amplifiers 65 with a gain $1-g^2=+0.5$ are used to insure that the over-all gain of each network is unity. The multiply delayed and undelayed components produced in the networks are algebraically combined to provide composite signals that are supplied to the individual output terminals. For the network with positive loop gain, the algebraic difference is obtained in subtractor 66, and for the network with negative loop gain, the algebraic sum of the two components is obtained in adder 67.

The optimal magnitude of the delay θ associated with delay lines 61 varies with the kind of program material. It has been found that delays as short as $\theta=5$ milliseconds give very pronounced spatial effects. For delays in excess of 50 milliseconds, considerable reverberation is added to the input signal. For example, a delay of θ milliseconds corresponds to $T=20\theta$ milliseconds of artificial reverberation because for each θ millisecond delay the signal is attenuated by 3 db and reverberation T is defined by a 60 db sound decay. This is, of course, often a very desirable condition, e.g., in a home music system it adds a degree of "realism" to the quasi-stereophonic signal. Delay lines 61 typically are selected with delay intervals of from 5 to 150 milliseconds. It has been found that a delay of 10 milliseconds yields an eminently suitable separation.

FIG. 7 illustrates the impulse response of each of the individual networks of the apparatus of FIG. 6.

The phase lag difference between outputs 1 and 2 of the networks of FIG. 6, as a function of frequency, is

$$\varphi_1 - \varphi_2 = 2 \arctan (2\sqrt{2} \sin \omega\theta) + \pi \quad (21)$$

where θ is the delay of the delay lines. By differentiation with respect to the radian frequency ω , the envelope (group) delay difference $\Delta\tau$ between the two outputs 1 and 2 becomes

$$\Delta\tau = \frac{4\sqrt{2} \cos \omega\theta}{1 + 8 \sin^2 \omega\theta} \cdot \theta \quad (22)$$

The delay difference $\Delta\tau$ as a function of frequency, f , is shown in FIG. 8. It oscillates with a "period" of $1/\theta$ about zero delay difference. Maxima of $\Delta\tau$ occur for $f=n/\theta$, where $n=0, 1, 2, \dots$. For these frequencies $\Delta\tau$ equals $4\sqrt{2}\theta$ or about 28 msec. if θ is chosen 5 msec. This delay is sufficient to make all frequency components near $f=n/\theta$ appear to be coming from the

earphone or loudspeaker connected to output 1. Conversely, for frequency bands around

$$f = \left(n + \frac{1}{2}\right) / \theta$$

the sound seems to emerge from output 2. For frequency bands around

$$f = \left(n + \frac{1}{4}\right) / \theta$$

or

$$f = \left(n + \frac{3}{4}\right) / \theta$$

the delay difference is zero and the sound image is central. The proportion of sound that appears to come from or near the center depends on the peakedness of the $\Delta\tau$ -response which in turn is controlled by the loop gain in the feedback loop around the delay lines. The higher the loop gain, the more peaked the $\Delta\tau$ -response. For a given maximum value of $\Delta\tau$, this means more center image because $\Delta\tau$ is relatively small for most frequencies. By adjusting both the delay θ and the loop gain, a fairly uniform spatial distribution of sound images can be achieved.

Experience has shown that the artificial reverberation apparatus described above is a distinct improvement over known electronic reverberators because in the process of introducing reverberation it does not alter the sounds, tones, color or timbre. In other words, it reproduces all sounds with equally high quality whether music or vocalization or both and regardless of pitch just as does a real concert hall. Neither does the reverberator add any unpleasant distortions of its own to the original sound and therefore does not have the usual hollow reedy quality or a false echo response.

The above-described arrangements are, of course, merely illustrative of the application of the principles of the invention. Numerous other arrangements may be devised by those skilled in the art without departing from the spirit and scope of the invention.

What is claimed is:

1. Apparatus comprising an input terminal supplied with audio frequency signals, an output terminal, and a plurality of interconnected all-pass networks coupling said input and said output terminals, each of said all-pass networks comprising means for delaying signals applied thereto, means for altering the gain of the delayed signals, means for mixing a portion of said altered delayed signals with signals applied to said network to produce input signals for said delay means, means for adjusting the gain of said applied signals, means for mixing said gain adjusted signals with said delayed signals to produce network output signals.

2. Apparatus for producing artificial reverberation that comprises an input terminal supplied with audio frequency signals, an output terminal, and a plurality of all-pass networks connected in series coupling said input and said output terminals, each of said all-pass networks comprising means for delaying signals applied thereto, means for altering the gain of said delayed signals, means for algebraically combining a portion of said altered delayed signals with signals applied to said network to produce input signals for said delay means and, means for algebraically combining a selected portion of the signals applied to said network with said delayed signals to produce signals with a prescribed phase characteristic but with no appreciable attenuation over a wide band of frequencies.

3. Apparatus for altering the reverberation character of a signal wave by the addition to the wave of a high density of echoes of the wave, said apparatus comprising an input terminal supplied with signal waves, an output terminal, and a plurality of networks characterized by prescribed delay and attenuation responses connected in series between said input and said output terminals, each

of said networks comprising means for delaying signals for an interval τ , means for adjusting the gain of delayed signals to a level g , means for combining said gain adjusted, delayed, signals with said signal waves applied to said network to produce modified signal waves, means for supplying said modified signal waves to the input of said delay means, means for adjusting the gain of said modified delayed signals to a level $1-g^2$, means for adjusting the gain of said applied signal waves to a level $-g$ where g is related to the delay interval τ and to the reverberation time T desired for said apparatus by the relation

$$T = \frac{3\tau}{-\log g}$$

and means for algebraically combining said gain adjusted applied signal waves with said gain adjusted, modified, delayed signals to produce an output signal.

4. Apparatus as defined in claim 3 wherein said means for delaying said signals in each of said networks is selected such that the delay interval of said means in one of said plurality of networks is essentially equal to the overall reverberation time desired for signals supplied to said output terminal, and the delay intervals of said delay means in the others of said plurality of networks are geometrically staggered in a progressively diminishing manner.

5. Artificial reverberation apparatus that comprises, in combination, comb filter network means having a comb filter-like response for adding artificial reverberation to an applied signal, and network means supplied with signals from said comb filter network means for substantially increasing the echo density of said signals, said network means comprising at least one all-pass network which includes means for multiply delaying said supplied signals and means for algebraically combining selected portions of said multiply delayed signals with said signals from said comb filter network means.

6. Apparatus for splitting single channel audio signals into two quasi-stereophonic signals that give a listener the spatial acoustic impression of multichannel stereophonic signals that comprises an input terminal; a pair of output terminals; a first all-pass network connected between said input terminal and a first one of said pair of output terminals and a second all-pass network connected between said input terminal and the second one of said

pair of output terminals, said first all-pass network comprising a feedback loop, means included in said feedback loop for adjusting the gain of signals passed therethrough to $+g$, and for delaying signals passed therethrough by a delay interval θ , means for supplying single channel audio signals applied to said input terminal to said loop, means for adjusting the gain of multiply delayed signals derived from said loop to a value of $1-g^2$, means for adjusting the gain of said audio signals to $+g$, means for subtractively combining said multiply delayed signals and said gain adjusted audio signals, and means for supplying said subtractively combined signals to said first output terminal; said second all-pass network comprising a feedback loop, means included in said feedback loop for adjusting the gain of signals passed therethrough to $-g$, and for delaying signals passed therethrough by a delay interval θ , means for supplying single channel audio signals applied to said input terminal to said loop, means for adjusting the gain of multiply delayed signals derived from said loop to a value of $1-g^2$, means for adjusting the gain of said audio signals to $+g$, and means for additively combining said multiply delayed signals and said gain adjusted audio signals; and means for supplying said additively combined signals to said second output terminal.

7. Apparatus as defined in claim 6 wherein the parameter g is $+0.707$, and the parameter θ is selected from the range of 5 to 150 milliseconds.

8. Apparatus as defined in claim 6 wherein the parameter g is $+0.707$ and the parameter θ is 10 milliseconds.

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