

July 9, 1968

P. H. PARKIN

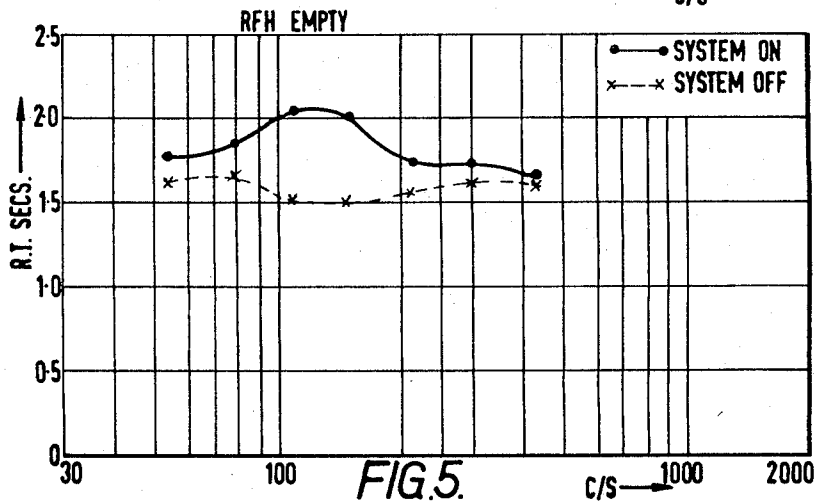
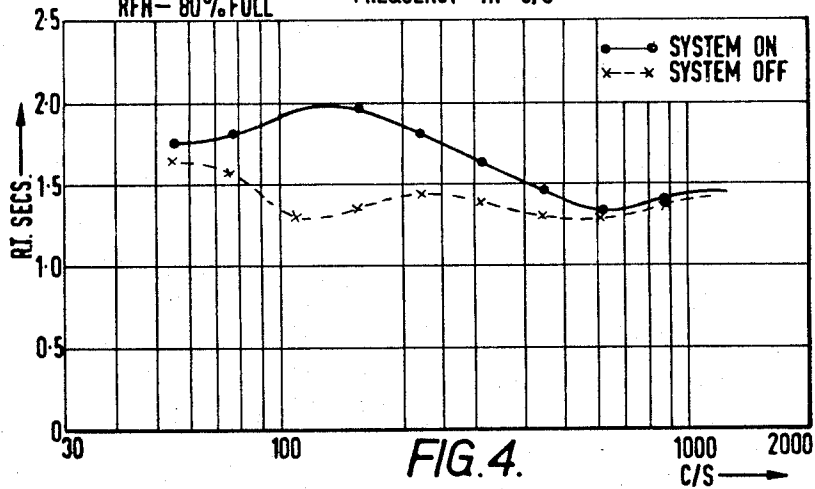
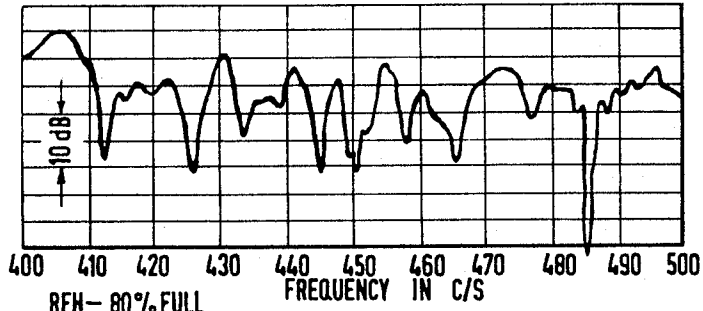
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ACOUSTIC SYSTEMS

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3 Sheets-Sheet 1

FIG. 1.



INVENTOR
PETER HUBERT PARKIN
By *Steward & Steward*
HIS ATTORNEYS.

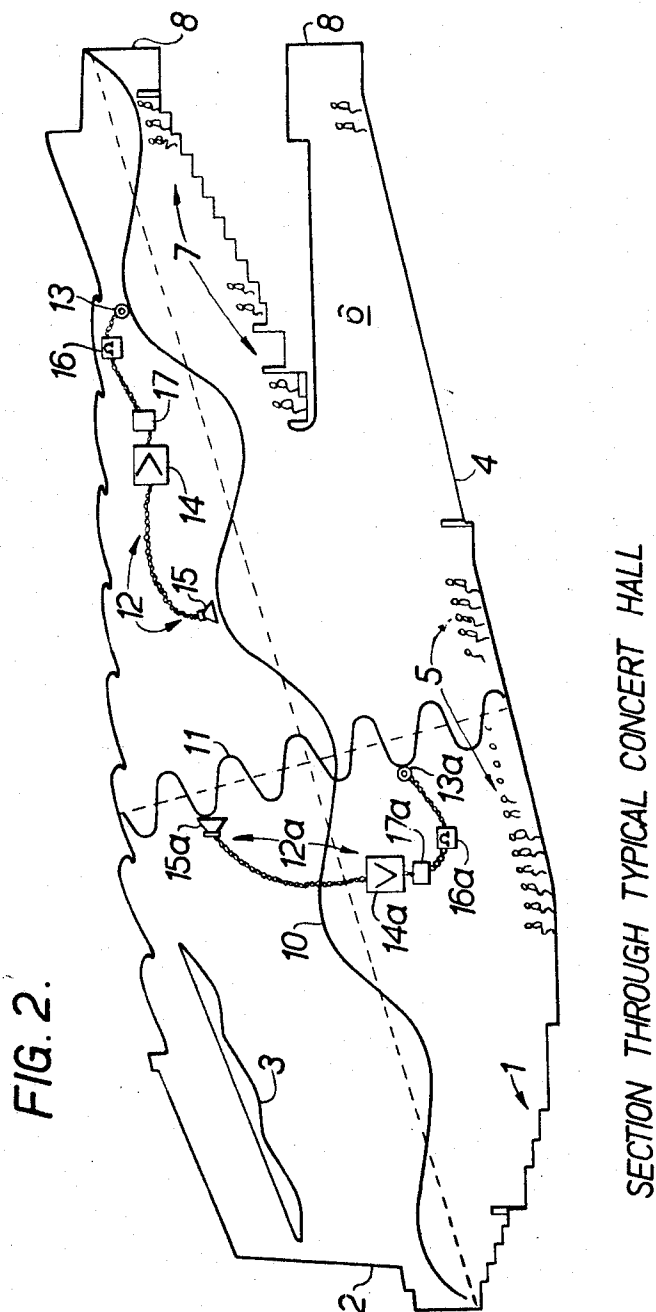
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INVENTOR
PETER HUBERT PARKIN
By *Steward + Steward*
HIS ATTORNEYS.

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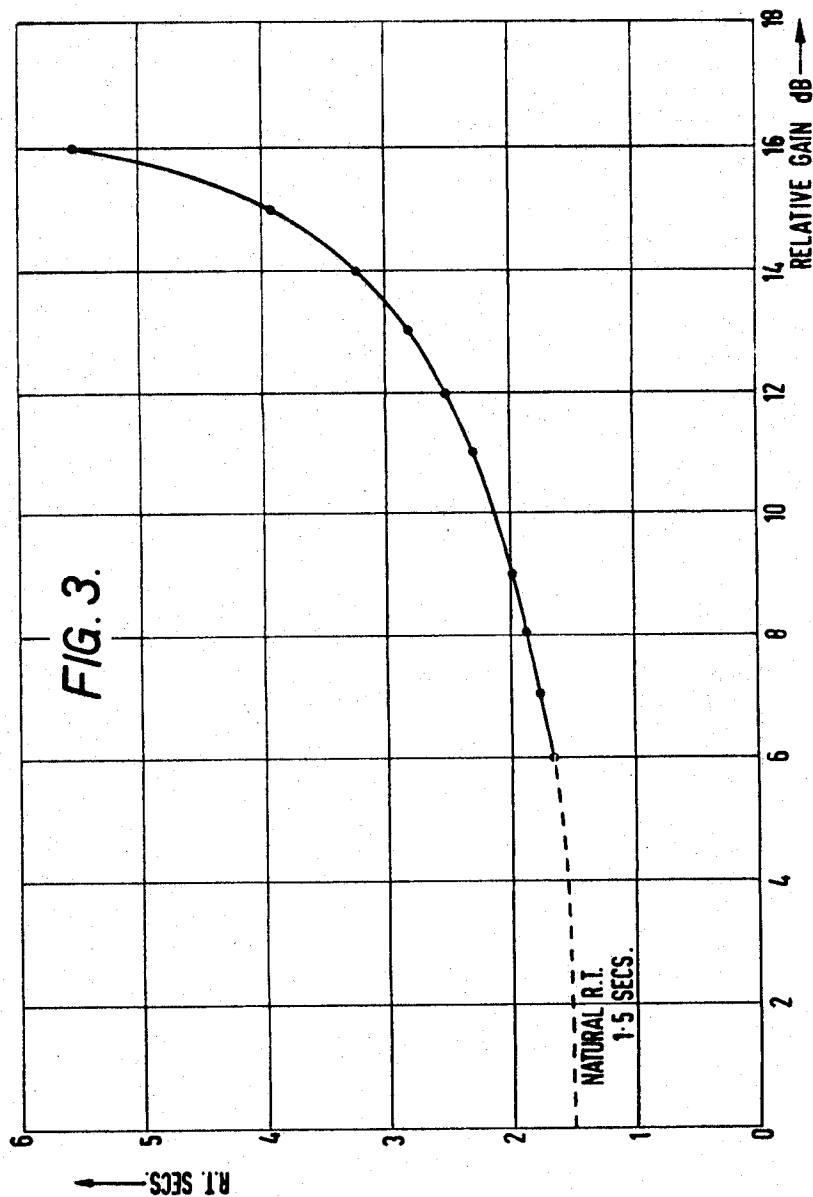
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INVENTOR

PETER HUBERT PARKIN

By *Steward + Steward*

HIS ATTORNEYS.

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ACOUSTIC SYSTEMS

Peter Hubert Parkin, Watford, England, assignor to
Council for Scientific and Industrial Research, Lon-
don, England, a corporation of the United Kingdom

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ABSTRACT OF THE DISCLOSURE

Reverberation time of an auditorium is effectively increased by sensing sound of a given frequency at a point where that frequency has a pressure peak, amplifying the signal, and reproducing the signal at a second point in the auditorium, in phase with the original sound wave.

This invention relates to acoustic systems in auditoria, and to a method of controlling their reverberation time.

In this specification, an auditorium is any air space intended to be acoustically excited by a more or less localised source of original of "live" sound (such as a speaker or singer or singers or an orchestra or musical instrument or a combination of vocal and orchestral or instrumental sound) which space is defined by surfaces that can set up a pattern of standing waves and therefore includes rooms, halls, sound- and multi-purpose recording studios, opera houses, arenas and the like.

It is a matter of common experience that for acceptable intelligibility of speech alone, auditorium acoustics require a shorter optimum reverberation time than for an orchestral or instrumental sound source. But it has also become recognised that for satisfactory quality in all cases, the nature of the music performed requires different acoustical properties of the auditorium, and that classical and some modern musical works are best heard in an auditorium in which the reverberation time is relatively short whilst romantic and some choral works are best heard when the reverberation time is longer. "Reverberation Time" is a technical term meaning the time taken for the intensity of a sound to decay by 60 db.

In his book, "Vibration and Sound" (1948, 2nd edition, McGraw-Hill, New York, p. 381), Morse makes the following observations:

"We look on the air in the room as an assemblage of resonators, standing waves that can be set into vibration by a source that will die out exponentially when the source is stopped. When the source is started there will be set up a steady-state vibration, having the frequency of the source, and a transient free vibration, having the frequencies of the normal modes, which will die out. The steady-state vibration may be considered to be made up of a large number of the standing waves (just as the forced motion of a string can be built up out of a Fourier series) whose amplitudes depend on the frequency of the source, the "impedance" of the standing wave in question, and the position of the source in the room. The transient vibration will have the form necessary to satisfy the initial conditions in the room when the source is started and will therefore also be made up of many standing waves; but each normal mode of the transient vibration will vibrate with its own natural frequency. . . .

"After the transient has died out, the steady-state vibration remaining will have only the frequency of the source. . . . When the source is turned off, these standing waves remain, only now they have their own natural frequencies, damping out exponentially according to their free vibration properties, and perhaps interfering with each other (making beat notes) as they do so. The damp-

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ing of these free vibrations . . . is called reverberation. . . ."

Reference is now made to the accompanying drawings, in which

FIG. 1 is a graph showing a typical curve of sound pressure level for a particular auditorium;

FIG. 2 is a fragmentary vertical section taken longitudinally through a typical concert hall and diagrammatically illustrating an acoustic system in accordance with the present invention;

FIG. 3 is another graph illustrating how the reverberation time of an auditorium can be controlled by employing the present invention; and

FIGS. 4 and 5 are graphs comparing the natural reverberation time of an auditorium with the reverberation time of the same auditorium after it has been modified in accordance with the present invention.

In every auditorium, for a given location of source there will be normal modes of vibration, i.e., standing wave patterns, at intervals of a few cycles, depending on the acoustic characteristics of the auditorium. The variation of sound pressure with frequency can be plotted, for any given location in the auditorium, by measuring the sound pressure level on, say, a logarithmic level recorder while the space within the auditorium is excited from a source of progressively changing frequency. A typical plot, made in the Royal Festival Hall in London, is shown in FIGURE 1 of the accompanying drawings which shows a curve of sound pressure level, in db, over the $\frac{1}{3}$ -octave frequency range from 400 c.p.s.—500 c.p.s. The irregularity of this curve, and the wide fluctuation in level (up to 40 db) are noteworthy.

A rigorous analysis of the acoustic characteristics of an auditorium by this method, whilst being theoretically possible, would in practice be too tedious and complex to carry out, and trial-and-error methods based on experience would normally be employed.

Although reverberation time is not the only criterion of the excellence of a concert hall or auditorium, it is the most important one, and as a first approximation it may be said that the longer the reverberation time—within reason—the more likely is the auditorium to have the acoustic quality which musicians variously describe as "warmth," "resonance," "body," "fullness of tone" and other names which, for present purposes, are regarded as synonymous.

The reverberation time of a hall varies directly with its volume and inversely as the amount of sound absorption present. Modern halls and auditoria have increased in size to accommodate larger audiences, but the volume has not increased at the same pace as the increase of absorption due to the larger audiences and to the wider spacing and improved upholstery of the seats. The reasons for the lower rate of increase of volume are partly cost and partly the increased risk of echoes from long path differences for the sound waves. As an example of the former, it was estimated that an increase of one foot in the roof height of the Royal Festival Hall in London would have cost about £10,000 extra to build.

Hence, if the reverberation time of any given auditorium proves to be too short, there is no economical structural way of increasing it beyond the reduction of the spacings between seats and the reduction of upholstery, and even this has strict limitations.

An object of the present invention is to provide for controllably increasing, without having recourse to structural alterations, the reverberation time of an auditorium.

According to the present invention therefore, in an auditorium a microphone and loudspeaker are located in the vicinity of pressure peaks at a resonant mode frequency, and are tunably coupled through a variable gain amplifier so as to feed amplified compensating sound

energy at that frequency alone into the auditorium at a rate less than the normal decay rate at said resonant mode frequency.

If a number of microphone-amplifier-loudspeaker units, one of each of a plurality of selected modes of vibration, is mounted in the auditorium in such a way that the decaying modal vibrations excite the units to radiate amplified vibrations which in turn reexcite the modal vibrations, it would then become possible to control the reverberation time for the structure to almost any value greater than the natural reverberation time.

The loop gain (in db) of any one of the microphone-amplifier-loudspeaker units arranged in this manner is made up of the electrical gain in the amplifier and the loss in the acoustic path between the loudspeaker and the microphone the effects of which are manifested by absorption of acoustic energy at the auditorium surfaces. If the amplifier gain is increased from zero (and assuming that the frequency response of the electrical circuit is relatively flat), the total gain round the loop will first become greater than unity at the frequency of the normal mode which is excited by the loudspeaker provided that the microphone and loudspeaker can be suitably located in relation to the pressure peaks in the mode, to minimise the above loss in the acoustic path. In these conditions, the system becomes unstable, and "howl"—which is well known in public address systems and the like—is set up. If the loop gain is only a little below unity, then the microphone-amplifier-loudspeaker unit will feed energy into the auditorium at a rate less than the loss at the frequency of the mode due to natural damping. Thus, when a transient sound external to the loop excites the chosen mode, it will decay more slowly than it would have done in the absence of the acoustic energy provided by the loudspeaker. This means that, for the frequency of the selected mode, the reverberation time can be increased.

The variation of reverberation time by the method disclosed above can be controlled without requiring unduly high accuracy or stability of the electrical components. For example, it has been found that, in a large concert hall, reverberation can be varied between, say, two or three seconds for a change in gain of 4 decibels. Consequently provided co-operation with the modal pressure peaks is observed in the manner hereinafter to be prescribed, the required order of increase of reverberation time can be achieved in any given auditorium by a system of microphone-amplifier-loudspeaker units constructed to within the normal limits of instrumental stability.

Alternatively, the coupled microphone and loudspeakers are located at points in the auditorium between which there is required, at a selected resonant mode frequency, minimum electrical gain in the amplifier to produce the desired maximum reverberation time.

Considered in yet another way, such an arrangement ensures that a given output from a loudspeaker produces the maximum effect on the reverberation.

It will be apparent that a plurality of microphone-amplifier-loudspeaker units will be required for the acoustic system of the present invention and the number thereof is determined in accordance with the following considerations.

The number of normal modes or modal channels which need to be selected in any one auditorium for a satisfactory control of the reverberation time from considerations of human aural perception will depend on a number of factors including the subjective consideration of what constitutes acceptable conditions. On the one hand, there are tens of thousands of modes in a large auditorium, but on the other hand only a comparatively few are important between any one source and any one position. The required number of units could possibly be calculated, but the calculation would certainly be very complex and in practice the required number is determined experimentally.

At frequencies in excess of about 2,000 c.p.s. air absorp-

tion becomes the dominant factor in the determination of the reverberation time, and limits the practical effectiveness of a system according to the present invention. Hence modes outside the range need not be selected for artificial excitation. This range can also be further restricted for most practical purposes, to an upper limit of 1,000 c.p.s., since it is rarely necessary to vary the reverberation time at higher frequencies. Experience with one concert hall has in fact, shown that the reverberation time thereof can be effectively increased by the use of only a few hundred microphone-amplifier-loudspeaker units, the spacing between the respective modal channels being at approximately 3 c./s. intervals. It is probable that the requirements of the average concert hall can similarly be met.

With a plurality of modal channels to be excited—not usually simultaneously—there is a risk that two or more of the above units will combine to excite another (non-selected) normal mode, most of the energy being absorbed by the non-selected mode. To guard against this it is necessary to include in each microphone-amplifier-loudspeaker unit a filter tuned to the appropriate selected frequency.

Furthermore, by deliberately shifting the phase of the amplified signal relative to the original, the electrical gain can be increased, which in turn increases the volume of sound for the same reverberation time.

The filter used can be acoustic, electrical, or mechanical, or a combination of two or all three types according to circumstances and preference. For example, the microphone of one unit could be placed inside a Helmholtz resonator which is tuned to the frequency of the selected normal mode, the pressure inside the resonator will be higher at the resonant frequency than at other frequencies and in this way the response of the unit will be practically restricted to this one frequency. Another example would be to use an electrical filter in each unit tuned to pass the appropriate frequency and to reject all other frequencies. The mechanical filter may for example be an electrically driven tuning fork or a tuned vibrating reed.

In any one loop path the minimum acoustic loss is obtained only when the sound reaching the microphone from the loudspeaker is in phase with the incident sound (from the orchestra, etc.). In theory this can be arranged by selecting the position in space of the microphone and its associated loudspeaker, but in practice it is very much easier to adjust the phase in each loop circuit by introducing an electrical adjustable phase-shift network into each circuit. It is also envisaged that a unit according to the invention may comprise two or more microphones at different positions of feeding a common amplifier and loudspeaker, tuned filters such as Helmholtz resonators being associated with each of the microphones. Between each microphone and the loudspeaker there would be one "most favoured" mode which would be excited thereby, provided of course, that the tuned frequencies of the filters are sufficiently far apart to ensure that interaction between them in the electrical circuit is quite negligible. Each microphone moreover, would have its own gain control and phase-shift control and each loudspeaker would radiate the two or more frequencies thus involved. Such an arrangement will bring about an economy in the number of amplifiers and loudspeakers required by the invention.

A reduction in the power requirements of the amplifiers can be obtained by fitting tuned acoustic devices to the loudspeakers, thus substantially improving their radiating efficiency. If a separate loudspeaker is used for each loop, then the acoustic device in the loudspeaker would be tuned to the frequency of that loop; if each loudspeaker is connected to a plurality of microphones then a multiple tube array will be necessary on each loudspeaker to deal with all the separate frequencies.

It should be noted that such a tube on a loudspeaker

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also increases the filtering effect at its appropriate frequency.

As shown in FIG. 2 the source of sound is located on the stage or platform 1 adjacent one end wall 2 of the auditorium, and a conventional sound reflector 3 is shown located above the stage 1. The auditorium consists of a floor 4 which is normally covered for most of its area by stalls at 5 and 6. Above the rear stalls 6 is one or more circles 7, cantilevered out from the rear wall 8 of the auditorium. The roof is not shown in the drawing. Two normal modes of vibration are shown in stylised form as sine waves 10 and 11, the mode 10 lying in a vertical plane between the end walls 2, 8 and the mode 11 lying in a vertical plane between the floor 4 and the roof. Associated with the mode 10 is a reverberation time controlling unit 12 consisting of a microphone 13, variable gain amplifier 14, loudspeaker 15, filter 16 and variable phase-shift network 17. The microphone 13 and loudspeaker 15 are located respectively at the two antinodes between which acoustic loss is a minimum for the mode 10. A similar reverberation time control unit 12a is correspondingly associated with the mode 11 and has a microphone 13a, variable gain amplifier 14a, loudspeaker 15a, filter 16a and phase-shift network 17a.

Although theoretically there will be only one pair of points between which acoustic loss will be a minimum, in practice there will be a selection of pairs of points between which the loss is negligibly greater than the minimum—for example, between one or two decibels—and the location of the microphone and the loudspeaker of a unit can usually be chosen to suit also the practical mechanics of the assembly, and the architectural convenience of the auditorium. Thus, in most cases, the same overall result will be achieved by siting the microphone and the loudspeaker of a unit in or near a wall or roof surface.

The drawing shows a typical arrangement in which it will be noted that the dispositions of the microphone and loudspeaker of any one unit are related solely to the pattern of the mode and not to the relevant positions of the source of sound at 1 and the audience at 5, 6, 7.

The construction of each unit is not necessarily mechanically integral—thus, the microphone, filter, amplifier, phase-shift network and loudspeaker are not necessarily mounted on or housed in a common structure. For example, the microphone and the loudspeaker of a unit may be independently suspended from some convenient part of the structure of the auditorium or from existing lighting fittings or the like. Each amplifier in a complete system of units is preferably remotely located at a central point so that the respective gains can be adjusted to suit working conditions and to ensure that all modes will have substantially the same reverberation time. Such control can be effected by any known conventional method. Practical experience generally shows that it is artistically desirable to lengthen the reverberation time at low frequencies as compared with higher frequencies.

Of the three types of filter mentioned above, the acoustic is preferred for most purposes, and in particular a Helmholtz resonator type of filter. This consists of an enclosed volume of air which is connected to the outside air through a tube or "neck." Depending on the enclosed volume and on the dimensions of the neck this system resonates strongly at one frequency so that the acoustic pressure inside the volume at this frequency is very much higher than at any other frequency. Thus, a microphone placed inside the volume will pick up this frequency much more than any other. Apart from this filtering action, this type of resonator has the advantage that because of the increased pressure at the microphone, electrical noise problems from the first stage of the amplifier are virtually eliminated. Moreover, the increased signal needs much less gain from the amplifier and can afford a saving of 2 to 3 stages of amplification. Further, there is nothing to go wrong with the resonator. Accordingly, the present invention has been developed along these lines.

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It is still necessary to select a position in space for the microphone and loudspeaker of any one channel such that a signal from the loudspeaker will produce several pressure maxima at the microphone position, one of which is at the selected frequency. It is also possible to select the microphone and loudspeaker relative positions so that the pressure received at the microphone from the loudspeaker is in phase or substantially so with the original signal arriving at the microphone from the source, i.e. so that the loudspeaker signal reinforces the original signal. A phase variation of up to $\pm 15^\circ$ on either side of synchronisation is permissible in this respect. However, in practice it is much easier to adjust the phase of the system electrically rather than to look for the correct positions, which are very critical in respect of coincidence of both pressure and phase. Thus for any given frequency, the loudspeaker of the unit is placed in a convenient position in the auditorium and the resonant mode in question is excited by a pure tone signal source (a loudspeaker) set up on the platform and radiating power at a given intensity level. A Helmholtz resonator tuned to the said frequency is then placed several wavelengths away from the loudspeaker of the unit and the microphone placed inside it. The phase-shift network of the amplifier is then adjusted so that the unit loudspeaker output arriving at the microphone is in phase with the original signal arriving there from the signal source loudspeaker.

If the gain of the amplifier be now adjusted so that—for example—the energy it supplies is equal to the energy lost to the auditorium surfaces, the reverberation time of the auditorium will be infinite. The amplifier gain is set at less than this level, with the result that energy is supplied to the auditorium at a slightly lower rate than that at which it is lost to the auditorium and the reverberation time will be lengthened at this one frequency.

FIGURE 3 of the drawings shows as an example, how the reverberation time of a pure 100 c./s. tone from a loudspeaker source on the platform in the Royal Festival Hall is easily controllable over a wide range using the system of the present invention. In this graph, reverberation time is plotted against relative gain of the amplifier. It will be seen for example, that the reverberation time is increased from 2 secs. to 3 secs. for a change of gain of about 4 db. It will also be evident that the reverberation time at this frequency can be increased to any large value.

FIGURES 4 and 5 of the drawings depict results obtained from an arrangement in the Royal Festival Hall using 89 modal channels separated at approximately 3 c./s. intervals and covering the range from about 70 to 340 c./s. All the microphones and loudspeakers for each channel were fitted to existing holes in the ceiling and objective measurements made using pistol shots both with the auditorium 80% full and empty. These measurements were of reverberation time (a) without the system switched on and (b) with the system switched on with various levels of amplification.

With the hall 80% full (FIGURE 4) it will be seen that the reverberation time measured near the platform (i.e. source of sound) at 125 c./s. was increased from 1.3 secs. to 1.8 secs., the relative amplifier gain being of course, kept below positive. This increase may be compared with an increase of not more than 0.1 sec. if instead of using the system of the invention, all the wood paneling on the sides of the hall had been removed.

With the empty hall (FIGURE 5) the increase in reverberation time at this frequency is somewhat greater, as is to be expected; the values being the average of readings taken at three different locations.

An auditorium equipped with a system of reverberation time control according to the present invention can readily be adapted for speech or the performance of musical works requiring, for their optimum presentation, different reverberation times, and adjustment to the reverberation time can readily be made between successive works in the same programme. The invention therefore enables a fixed structure of auditorium to acquire the character-

istics hitherto obtainable only by entirely different structures having different volumes for the same audience capacity and absorption characteristics. Any given auditorium thus acquires a high degree of versatility for the presentation of speech or different types of musical art. In effect, an auditorium provided with a system according to the invention can be converted at will from the equivalent of one structure to the equivalent of another.

I claim:

1. A method of improving the acoustical properties of an auditorium by controlling the reverberation time therefor, which comprises the steps of emitting signals from an original sound source at a plurality of resonant mode frequencies spaced at intervals of a few cycles per second within a given range of frequencies, locating for each such frequency the positions within the auditorium along the same resonant mode of at least two separate sound pressure peaks of such mode, picking up the signal from each such resonant mode in the vicinity of one of its sound pressure peaks, amplifying said signal, and reinjecting such amplified signal into the auditorium near another of said sound pressure peaks of the same resonant mode at an energy level such that the rate of supply of sound energy thereby produced is less than the natural rate of decay of the original signal for each such mode, thereby in effect reducing the rate of decay of each such signal and increasing its reverberation time.

2. A method as defined in claim 1, wherein the signal emitted from the original sound source for each frequency is filtered in order to eliminate other frequencies before it is re injected into the auditorium.

3. A method as defined in claim 1, which includes the further step of adjusting the phase of such amplified signal at each frequency so that it is in phase with the signal emitted from the original sound source.

4. A method as claimed in claim 3, in which the resonant mode frequency signals are spaced at intervals of about 3 c./s.

5. A method as claimed in claim 3, in which the resonant mode frequency signals are spaced at intervals of about 3 c./s. and ranging from 70 c./s. to 340 c./s.

6. An acoustic system for variably increasing the reverberation time of an auditorium with respect to the natural reverberation time therefor, comprising in combination with an auditorium containing an original sound source, a plurality of sound-reinforcing units each comprising a microphone, a variable-gain amplifier, a loudspeaker and means for tuning each of said units to a selected resonant

mode frequency, said microphone being connected to said loudspeaker through said amplifier, said microphone and loudspeaker for each of said sound-reinforcing units being located within the auditorium at positions spaced from each other along the same resonant mode frequency near separate sound pressure peaks for such frequency, whereby the signal at the frequency to which each of said sound-reinforcing units is tuned is picked up and re injected into the same resonant mode frequency at another sound pressure peak in order to partially compensate for the normal decay of the signal emitted from said original sound source at each of said selected frequencies.

7. An acoustic system as claimed in claim 6, wherein the resonant mode frequency signals from the original sound source, to which said sound-reinforcing units are tuned, are selected at intervals of about 3 c./s.

8. An acoustic system as claimed in claim 6, wherein the resonant mode frequency signals from the original sound source, to which said sound-reinforcing units are tuned are selected at intervals of about 3 c./s. within the frequency range 70 c./s. to 340 c./s.

9. An acoustic system as claimed in claim 6, wherein said tuning means for each of said sound-reinforcing units is a filter, said variable-gain amplifier for each reinforcing unit having an electrical phase-shift means for shifting the sound signal emitted by its loudspeaker into phase with the corresponding signal emitted from the original sound source.

10. An acoustic system as claimed in claim 9, wherein said filter is an acoustic Helmholtz resonator tuned to the selected resonant mode frequency, said microphone being housed within said resonator.

11. An acoustic system as claimed in claim 9, wherein said filter comprises an electrical acceptor circuit for tuning said variable-gain amplifier to the resonant mode frequency selected for the particular sound-reinforcing unit.

12. An acoustic system as claimed in claim 9, wherein said filter comprises a mechanical filter for tuning said variable-gain amplifier to the selected resonant mode frequency.

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KATHLEEN H. CLAFFY, *Primary Examiner*.

H. ZELLER, R. P. TAYLOR, *Assistant Examiners*.