An Artificial Reverberation System

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Describing a simple, practical system for producing controlled reverberation.

ARTIFICIAL reverberation has been a regular adjunct in the broadcasting business for some time, particularly to furnish special sound effects. It has been used less often for the enhancement of live or recorded music, probably because the frequency response of the more familiar systems has usually been inadequate. The following describes a system which has been in use at KFI for over a year, and which has been found suitable for use in either of the two ways mentioned above. Since the requirements for use with musical program material are the more critical, the design was carried out with this function uppermost in mind; at the same time, the system is also sufficiently flexible to produce the usual sound effects.

The pleasing results obtained when music is accompanied by the proper amounts of reverberation have long been recognized by the broadcasting, motion picture and sound recording industries. In fact, published material on the subject is so voluminous that it would be impractical to list all references here. The desired reverberation is obtained preferably from a suitably designed auditorium studio; however, this usually calls for more space than most broadcasters can provide. A next-best substitute of more moderate cost can be obtained by artificial means which, if carefully controlled and wisely used, can give a partial approach to the ideal. It is not necessarily true that the listener's pleasure goes up in proportion to the amount of reverberation present, and this is particularly the case when the reverberation is supplied through artificial means. Just as a good public address system should not intrude itself on the listening audience, artificial reverberation should likewise be so used that the listener is not aware of its presence. It can be overdone easily and extreme care is indicated in its use.

Ideal Requirements

An ultimate artificial system can be imagined which, having a sufficient number of adjustable variables, could be made to reproduce closely the reverberation characteristics of any given auditorium or studio. Such a system should operate so that each original sound is followed by a large number of trains of echoes, each corresponding to a series of reflections from one of the many surfaces in the ideal studio. It has been suggested[1] that an illusion of blending is fairly well approached if individual sounds are repeated with spacings of .05 seconds or less. Optimum reverberation times vary from 0.4 to 4 seconds approximately, depending on frequency, volume of the studio and type of program material. The trains of echoes should, therefore, have an adjustable decay time to a maximum of about 4 seconds and, in addition, the rates of decay should be made to change depending on the frequency of the original sound.

Obviously a system with this degree of complication is impractical. Severe compromises are necessary in a practical system. In the present case, only one train of echoes is provided; the spacing between echoes remains constant at about .023 seconds. The train consists of groups of four echoes repeated successively, with no breaks or spaces between groups. The decay rate of the four echoes can be controlled independently and the over-all decay rate of the entire train is also adjustable to a maximum of about 6 or 8 seconds, but both decay rates remain essentially constant with frequency and no control is provided by which this relation can be changed. Lastly, a control is available which determines the ratio of reverberation to original sound. Even with the compromises as described above and with this small number of controllable variables, the illusion is quite surprising, and with many types of music the result is satisfactory even with the ratio of reverberation to original sound advanced to between 1:2 and 1:1.

Operation

A number of media have been utilized for producing delayed echoes; they include coiled springs, magnetic tape,[1] a moving strip of phosphor coating[2], and large empty rooms.[3] The medium used in the system here described** is a long pipe with a loudspeaker at one end and a microphone at the other. Operation can be explained by reference to *Fig. 1*, a simplified diagram of the essential units in the channel, excluding equalizers. At

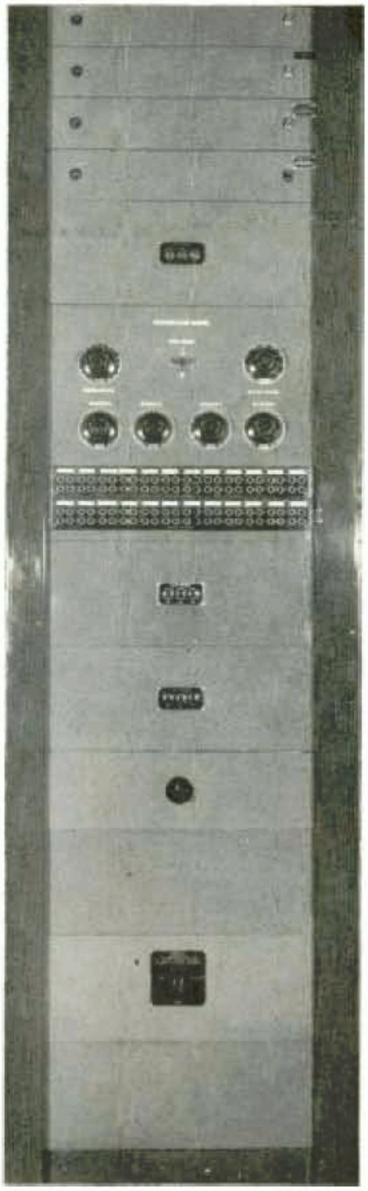

Fig. 12. Reverberation bay of system, comprising amplifiers, control panel, and power supply.

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** The general features of our channel were suggested by a conversation with Dr. H. F. Olson, of RCA Laboratories at Princeton, N. J., and a similar system has been developed by that organization. Our development, carried out independently, was done almost entirely by Wayne R. Johnson, of the KFI R & D group, to whom much credit accrues for an outstanding job.

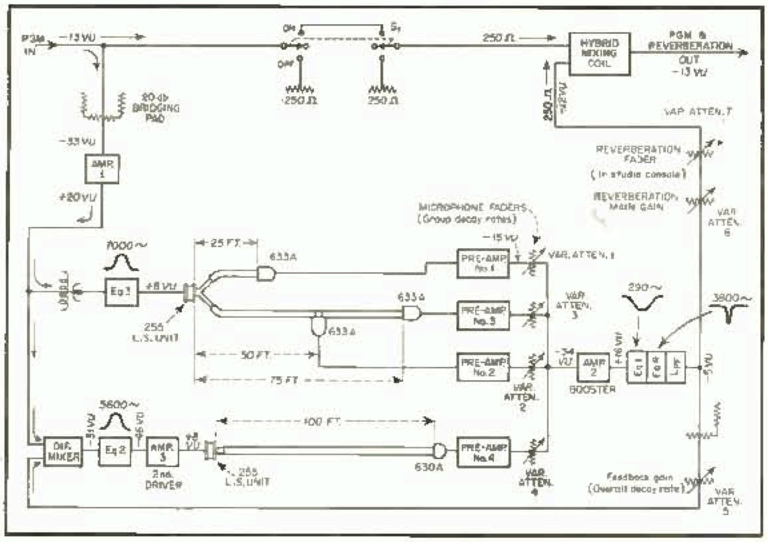


Fig. 1. Simplified diagram showing essential units of the system.

the input (*upper left*) the incoming signal splits, one portion going along a direct path through a switch S_1 and a mixing coil to the output of the channel. The switch S_1 enables use of the channel in three ways. In the "ON" position, the input terminals look like 250 ohms and a direct path is supplied within the channel itself. With the switch in the "OFF" position, the input terminals also look like 250 ohms but no direct path is provided. With the switch in the center or neutral position, the input terminals look like a high (bridging) impedance and again there is no direct path through the channel itself.

After splitting at the input terminals, the second portion of the incoming signal is fed to the bridging input of the first driver amplifier, AMP 1, which drives the loudspeaker units. The first three units are driven directly from this amplifier while the fourth unit gets its excitation through a differential mixer network and a second driver amplifier, AMP 3. The loudspeaker units are coupled to four lengths of 1" aluminum pipe 25, 50, 75 and 100 feet long with microphones at the other end to pick up the sound after it has traversed the length of pipe. The sound arriving at the first microphone is accordingly delayed with respect to the original signal by the time required to travel the length of the 25-foot section, approximately 0.023 second. The delay at the second, third and fourth microphones is, respectively, 0.046, 0.069 and 0.092 second, approximately. After passing through pre-amplifiers and microphone faders, these four delayed signals are combined and fed to a booster amplifier, AMP 2, thence through a reverberation main gain and fader, VAR ATTEN 6 and 7, and a mixing coil to the output of the channel. Mixture of the direct and delayed signals occurs in the hybrid mixing coil. Since they control the amplitude of the delayed signals, the main gain and fader serve to control the ratio of reverberation to original program. The fader is located at the studio mixing position so that the studio operator has continuous control of this ratio.

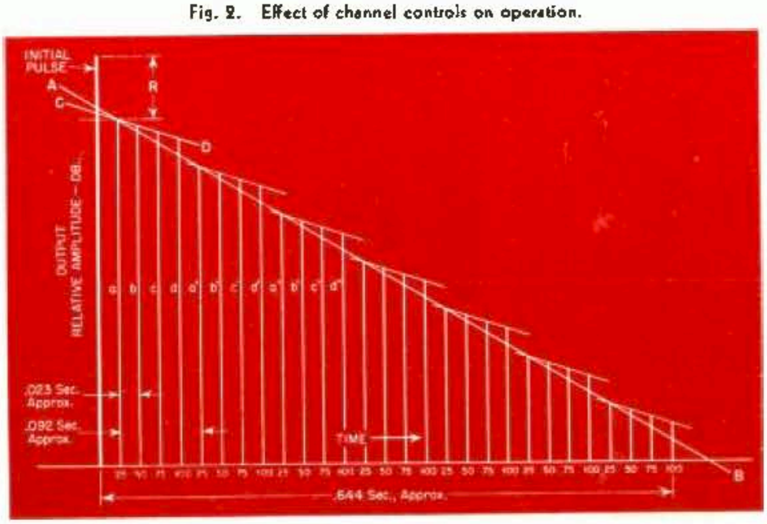
The delay afforded by the longest pipe, approximately 0.1 second, falls short of the desired 3 or 4 seconds maximum reverberation time. Additional delay could be obtained by using longer sections of pipe, but the 3,000 or 4,000 feet required would be much too large physically and the high-frequency attenuation in such a length would be prohibitive. Instead, additional delay is obtained by establishing a feedback path from the output of the booster amplifier through the gain control VAR ATTEN 5, the differential mixer, the second driving amplifier, and the 100-foot section of pipe back to the booster amplifier. Each time the first four delayed signals traverse this circuit four new echoes appear at the channel output, thus providing successive groups of four echoes with a delay between corresponding echoes of each group equal to the delay through the 100-foot section. The rate at which the groups decay will depend upon the loss in the feedback circuit and hence upon the setting of the feedback gain control. If the frequency response of the entire feedback circuit is flat, this gain control can be opened up until very long delay times are obtained; at the limit the system will, of course, break into sustained oscillation or singing. Under carefully controlled conditions, delays of from 6 to 8 seconds can be maintained; under more practical conditions the delay is usually limited to 3 or 4 seconds, while in actual use delays in excess of one second are rarely needed.

The requirements of flat frequency response through the feedback path are quite stringent. Any small peaks are increased each time a signal makes the round trip through the feedback circuit, and after ten or twenty such round trips the effect of the original small peak has been magnified to large proportions. The irregularities most usually appear in the upper audio range, and it has been our experience that when they are greater than 1 db the over-all result is very tinny in character and the system may break into oscillation before appreciable reverberation effects can be achieved.

Description

Figure 2 illustrates the action of the various channel controls. It shows the series of echoes that might result from certain settings of those controls, when one very short pulse is applied to the channel input. The initial pulse travels via the direct route and appears at the output as indicated at the left of the sketch. Approximately .023 seconds later the first

Fig. 2. Effect of channel controls on operation.



echo (a) will appear, this being the one produced by the original pulse after traveling through the 25-foot section and thence to the output through the main gain, fader and mixing coil. The ratio between the level of the initial pulse and that of the first echo will depend upon the settings of the main gain and fader, ATTEN 6 and 7, with a given setting of the microphone fader, ATTEN 1. Adjustment of the main gain and fader accordingly serves to change the ratio between the two signals indicated as R in *Fig. 2*.

In the meantime, the initial pulse has also been traversing the 50-foot section of pipe, and it will arrive at the output via the path through the ratio controls as echo (b) approximately 0.023 seconds later than echo (a). The output level of echo (b) with respect to (a), everything else being equal, will depend upon the relative settings of the two microphone faders. Similarly, echos (c) and (d) from the 75 and 100-foot lengths of pipe will arrive at the output each having an additional delay of approximately 0.023 seconds and a level depending on the settings of the associated microphone faders. The settings of these attenuators, therefore, determine the rate of decay of the group of four echoes as indicated by the slope of the line (CD) in *Fig. 2*.

While the foregoing has been occurring, echo (a) from the 25-foot pipe, having been diverted to the feedback path at the output of the booster amplifier, has been traveling the 100-foot section of pipe to arrive at the output with a total delay, referred to the original pulse, of 125 feet (echo a'). The height of signal (a') with respect to (a) is determined by the setting of the feedback gain control. Similarly, signals (b), (c) and (d) after passing through the feedback path produce signals (b'), (c') and (d') at the output. Thereafter, the process repeats itself to give the succeeding groups of echoes which comprise the whole train. In *Fig. 2*, the line AB, connecting the tops of the (a) echoes, represents the rate at which the train decays as controlled by the setting of the feedback gain control.

That a one-inch pipe is far from being an ideal transmission line can be seen by reference to the curves of *Fig. 3*, which show the relative theoretical loss versus frequency of various lengths of this size pipe[1]. In view of the large amounts of equalization that were indicated at frequencies above 7,000 cycles per second, this approximate figure was chosen as the top design limit. This decision was strengthened when the available loudspeaker drivers and microphones were brought into the picture. The response characteristics of these units, plus those of the acoustical couplings, added further to the loss at the higher frequencies.

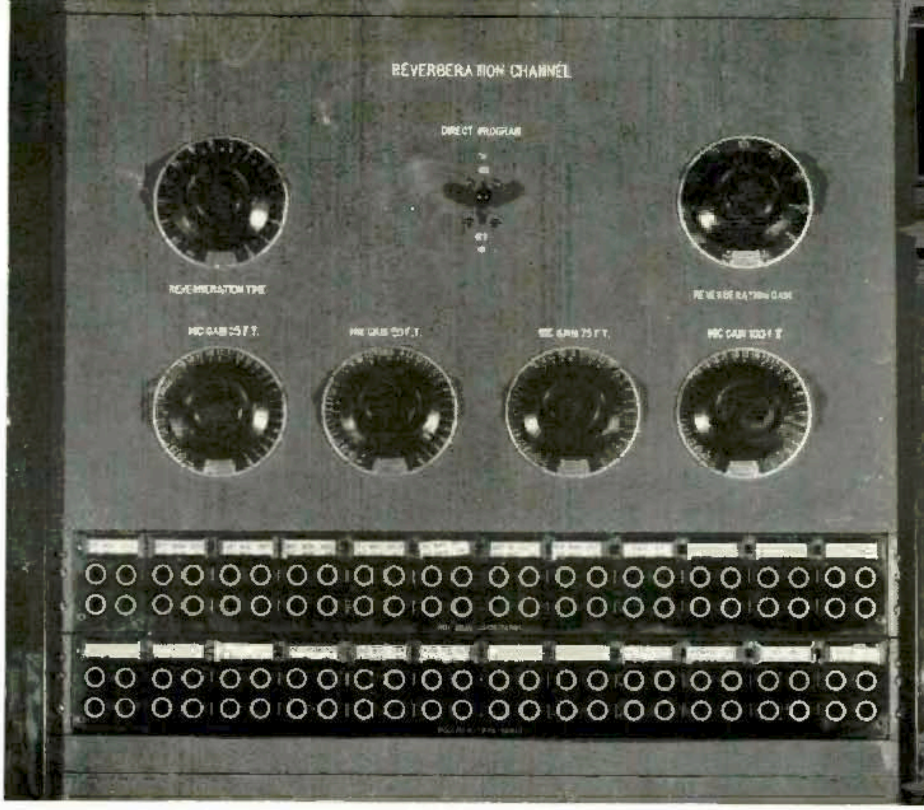


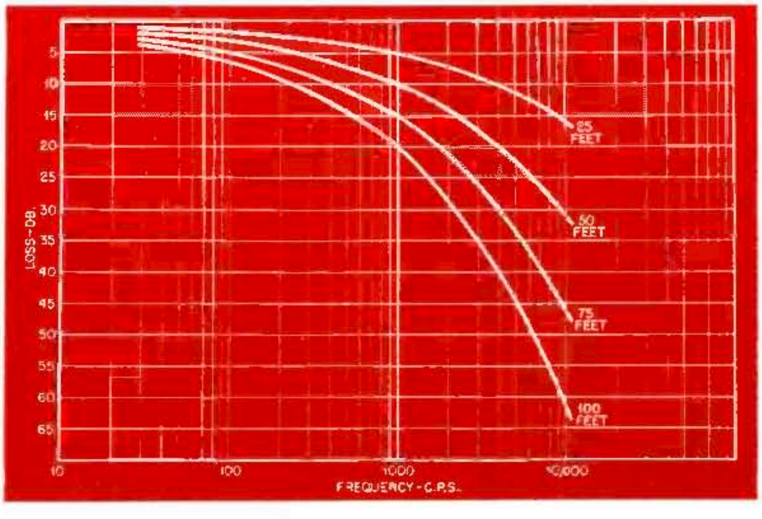
Fig. 13. Control panel of reverberation channel.

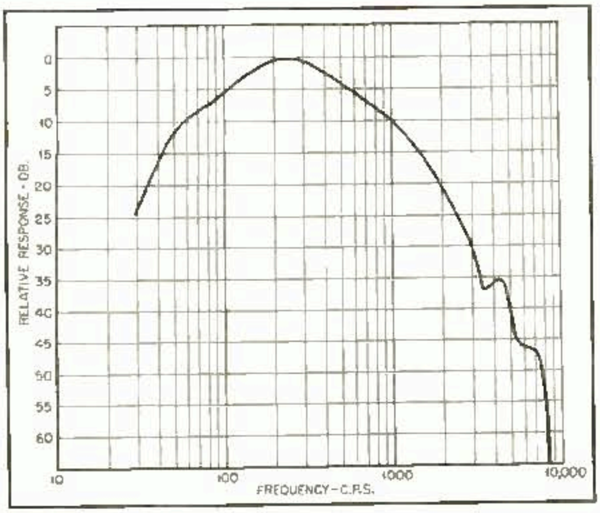
Figure 4 shows the total amount of correction necessary over the useful frequency range. This shows the overall relative response of a WE 255 driver, and 100-foot section of pipe and a WE 633A ("salt shaker") microphone without any attempt at equalization and without any especial provision for impedance matching at either the sending or receiving end. The ends of the pipe were merely butted up against the driver and microphone and taped in place. Standing waves of small amplitude (about 2 db) present at frequencies below 150 cycles have been averaged out in drawing the curve. The frequency characteristics of each amplifier used was sufficiently flat so that its effect could be neglected.

Impedance Matching

From consideration of *Fig. 4*, it was apparent that there was considerable to be done if flat transmission through the 100-foot pipe and the feedback path was to be achieved. At first, the tendency was to be rather indifferent to acoustical impedance matching at the sending and receiving ends. It seemed that the necessary equalization would be better done by electrical means with which we were more familiar. It was soon learned,

Fig. 3. Theoretical transmission loss through 1-inch pipe.





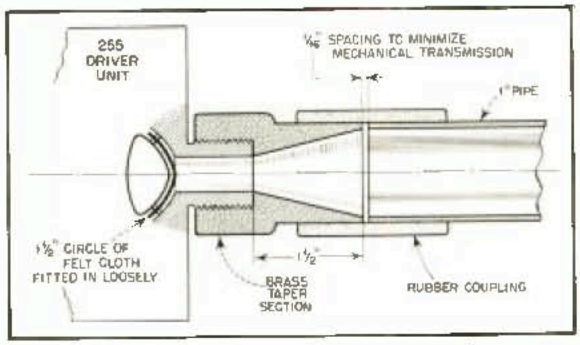


Fig. 5 (above). Method of coupling pipe to driver.

Fig. 4 (left). Response of 255 driver, 100-ft. pipe, 633A microphone, no correction.

however, that efforts spent in obtaining acoustical equalization or in trying to improve the impedance match at driver and microphone, paid worthwhile dividends. Pronounced peaks in the high-frequency region produced by small cavities in the couplings at the transducers, and standing waves which occurred at the lower frequencies, were reduced to negligible proportions when some attention was paid to the fabrication of these couplings. As for acoustical equalization, the few instances in which it was used resulted in an appreciable improvement in signal-to-noise ratio. It was easy to design electrical attenuation equalizers that would accomplish the required corrections, but their relatively high insertion losses caused one or both of two things to happen. Either the desired signal began to approach uncomfortably near to the inherent noise level in the available amplifiers, or the signal level at the receiving end of the longer pipes was so low that pickup of external noise and building rumble by the microphones became objectionable. The acoustical equalization, on the other hand, gave comparatively small insertion losses.

Acoustical Equalization

A brief description of the couplings and the means employed for acoustical correction in the final setup may be of interest. They were evolved by a combination of forethought and cut-and-try and are not necessarily the ultimate that can be accomplished. At the WE 255 loudspeaker driver unit which excited the 100-foot section of pipe, a brass taper was provided as sketched in *Fig. 5*. A thread was turned on the outside of the driver collar to accept the female thread of the taper section; the latter was turned up snug to give a tight fit at the bearing surface on the end of the collar. The flare provided a gradual transition from the small inner diameter at the output of the driver unit to the larger inner diameter of the 1-inch pipe. The pipe was attached to the brass section by means of a snug-fitting length of heavy rubber tubing about 4 inches long. A small gap, approximately 1/16 inch, was left between the end of the pipe and the end of the taper to minimize mechanical transmission of sound between the two components. The only other treatment provided at this point was to insert a small 1½-inch circle of felt cloth, fitted in loosely in front of the phasing plug inside of the driver. Small slits put in the felt to accommodate the small rods that support the phasing plug also helped to hold the felt in place. The felt was inserted to dampen a peak at approximately 4,500 cycles.

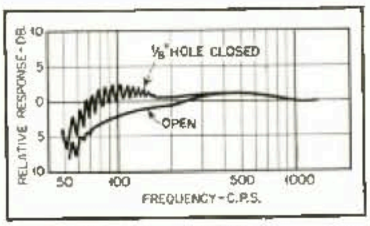


Fig. 7. 255 driver, 100-ft. pipe, 630A microphone, with electrical equalization, showing effect of 1/8-inch opening in pipe wall (see Fig. 6).

At the microphone end of the 100-foot section of pipe, a WE 630A microphone was used. Its dimensions were appropriate to fit the end of the pipe, as shown in the sketch of *Fig. 6*. Two mating tapers were put on the outside of the pipe and in the inside of the microphone baffle so as to give a force fit when joined. The pipe and microphone were then taped together for added strength and to give an air-tight joint. The importance of avoiding small air leaks and small cavities in these couplings was demonstrated again and again. Air leaks are equivalent to putting an inductance in shunt with the acoustical transmission line[4]; even small ones cause a pronounced falling off at the lower frequencies. The cavities correspond to a tuned LC circuit across the line, series-tuned if the cavity is closed and parallel-tuned if open, and gives resonant dips or peaks in the higher frequency range. One such air leak was intentionally introduced in the form of a 1/8-inch circular hole in the wall of the pipe near the microphone as shown in *Fig. 6*. It was used to help smooth out low-frequency standing waves in the long pipe; its effect is shown in the curves of *Fig. 7*. During final testing of the channel, a portion of the hole was covered with tape by cut-and-try until the desired response was obtained; the final curve used was approximately midway between the two curves. The low-frequency droop incurred was then compensated for elsewhere.

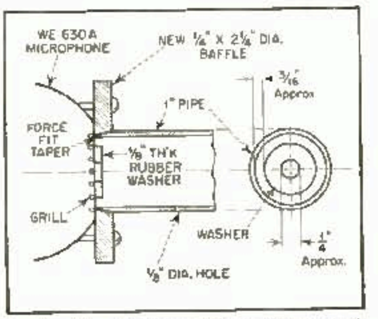


Fig. 6. Pipe coupling and acoustical equalization at 630A microphone.

High-Frequency Equalization

A very welcome amount of high frequency equalization, all out of proportion to the simplicity of the means by which it was obtained, was brought about by inserting a small rubber washer against the wire grill of the same 630A microphone as also shown in the sketch of *Fig. 6*. This microphone was arranged to face upward and the washer was held in place only by its own weight. Presumably it could have been cemented to the wire grill at a few points. The equalization effected by this simple means is depicted in *Fig. 8*. Use of this gadget enabled removal of two high-frequency electrical equalizers with a very marked improvement in signal-to-noise ratio.

The remaining equalization required in the feedback circuit was obtained from attenuation equalizers, as indicated in *Fig. 1*. The computed insertion loss of these equalizers is shown in *Fig. 9*. One so-called "dip-filter" (Eq. 4) was required to remove a resonant peak in the neighborhood of 3,800 cycles. The other two equalizers had smoother characteristics and were used to build up the less

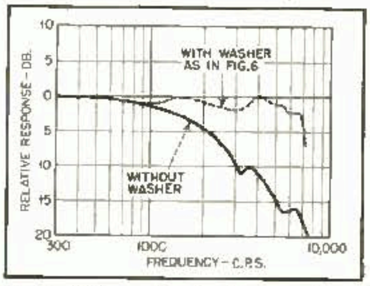


Fig. 8. Same as Fig. 7, showing effect of washer in 630A microphone.

localized deficiencies of the system. In addition, a cut-off filter, having a rather sharp roll-off above 8,000 cycles was inserted mainly to eliminate tube hiss. While all four of these elements (three equalizers and one cut-off filter) were in

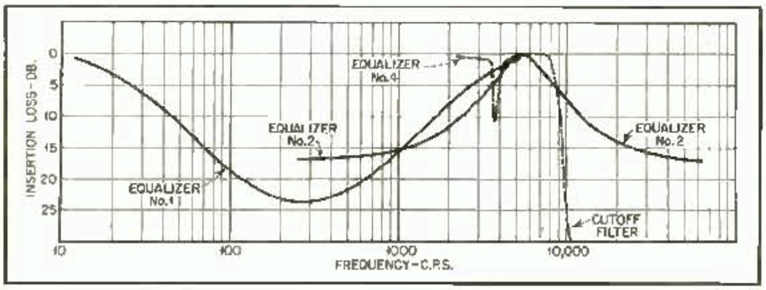


Fig. 9. Insertion loss characteristics of equalizers and filter in feedback path.

the feedback circuit, only one (Eq. 2) was identified solely with the circuit of the 100-foot pipe section. The other three were located so that their characteristics also affected the signals from the three shorter sections of pipe.

Using the acoustical and electrical components described above, the response curve of *Fig. 10* was obtained by feeding tone at constant level into the PGM IN terminals and then measuring the relative output at the PGM PLUS REVERB OUT terminals with the switch *S1* in the off position, with all microphone faders turned off except VAR ATTEN 4, and with the feedback gain control turned off. This curve is seen to be within ±1 db from approximately 80 or 90 cycles to approximately 7,000 cycles. While extending the response to somewhat lower frequencies may seem desirable, a roll-off in that region helped to keep down some irreducible building rumble. During listening tests when a program was alternately switched from the direct path through switch *S1* to the path which included the elements measured as described, it was difficult to identify one from the other. Therefore no attempt was made to extend the response further. Total noise, principally building rumble, measured about 45 db below program. When a high-pass filter with cut-off at 40 cps was included in the circuit, the remaining noise level was at approximately 53 db below normal program.

Treatment of Shorter Sections

Since the frequency response through the shorter sections was relatively unimportant compared to that through the 100-foot section, variations of plus or minus 5 db were allowed. Standing waves were also more pronounced as the sections became shorter and amplitudes of 6 db from peak-to-peak were likewise permitted.

For economy, only one WE 255 driver was used to excite the remaining lengths of pipe. A commercially-made "Y" or two-branch loudspeaker connector was attached to the driver and to 25 and 75-foot lengths of pipe. WE 633A ("salt-shaker") microphones were put at the other ends of these sections, and one was also inserted in the 75-foot section 50 feet distant from the driver. Thus the total length of pipe required and the space it would occupy were also reduced. The "Y" connector had the proper inside diameter at the driver end and no change was necessary there. At the output ends where the inner diameter was too small, there was adequate wall thickness

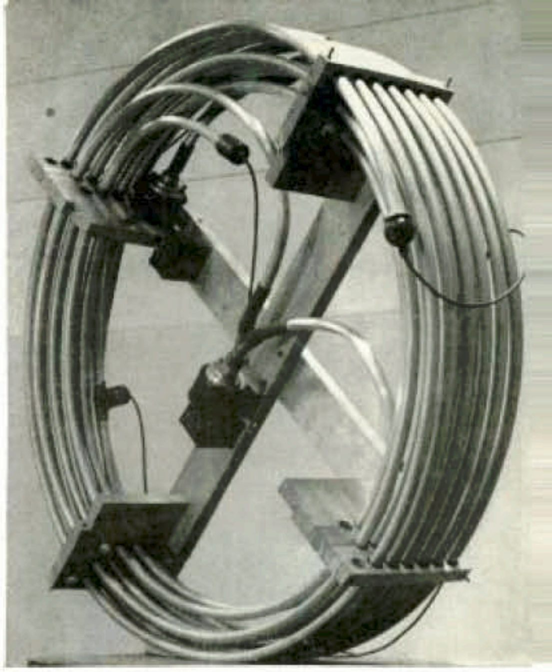


Fig. 14. Pipe unit for reverberation system.

to permit reaming out the ends of the connector to give a flare similar to that shown in *Fig. 5*. Short lengths of rubber tubing were again used to attach the pipe ends to the branches of the "Y."

In the portion of the "Y" that was connected to the 25-foot section, felt cloth was loosely bunched from the end of the pipe to the apex of the "Y." The attenuation at high frequencies thus introduced served to reduce cavity effect in the connector, and also provided the attenuation necessary to compensate for the lower high-frequency loss in the short section.

The openings in the salt-shaker microphones were of approximately the correct diameter so that they were merely butted against the ends of the pipe and taped in place after small notches had been put in the pipe ends to accommodate the three ribs on the front of the microphone. The taping had to be tight and carefully done for mechanical rigidity and to prevent forming small air cavities.

To attach the third salt-shaker microphone, which was inserted in the side wall of the pipe at the 50-foot point, a brass insert was provided which correctly fitted the pipes at each end. An opening in the side of the insert was then carefully formed to fit the contours of the microphone so that the front surface of the latter imposed a minimum of dis-

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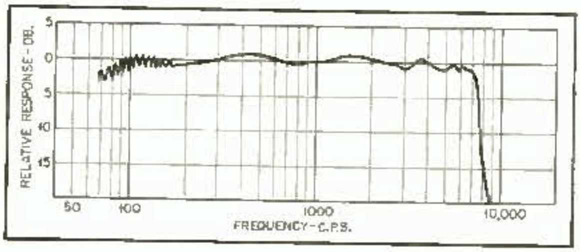
Fig. 10. Over-all response input to output through 100-ft. pipe and critical units of feedback circuit.



Reverberation System

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continuity along the air column. Tape was also used here to seal against possible leaks.

In addition to those already described, equalizer (3) was provided for this circuit branch, which was placed ahead of the loudspeaker driver. The computed insertion loss characteristic of this equalizer by itself is shown in *Fig. 11*.

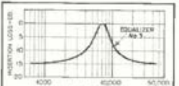

Fig. 11. Insertion loss characteristics driver equalizer (No. 3).

The various acoustical coupling and equalizing elements described were selected mainly by the cut-and-try method, observing the result of any change on the frequency response of the branch. A sweep-frequency audio oscillator is, therefore, a necessity in work of this nature; with it, the effect of any change can be noted instantly on the screen of a scope connected to the output. Tedious point-by-point measurement was resorted to only when greater accuracy was desired.

Figures 12, 13 and *14* are photographs of the units comprising the channel. The first is of the reverberation bay in the main equipment room. At the top are the four microphone preamplifiers, then an 82A amplifier, control panel, jack strip, more 82A amplifiers, metering switch and the power supply at the bottom. An enlarged view of the control panel is shown in *Fig. 13*. The four controls on the bottom row are the microphone faders; at the upper left is the feedback gain control while the one at the upper right is the reverberation main gain. Transfer switch S_1 is at upper center. Equalizer and filter elements, except Eq. 3, are mounted at the rear of this unit.

The coils of one-inch aluminum pipe shown in *Fig. 14* have a maximum diameter of approximately five feet. Taping of the microphones was done after the photograph was made. Eq. 3 was mounted in a small aluminum box below the lower driver unit. The supporting framework shown was shock mounted to one wall of a relatively unused room, and so far no trouble has been experienced from accidental pickup of voices in the room. Each turn of pipe was isolated from the wood frame by a layer of sponge rubber. Forming the coils was easy because the tubing was quite soft.

REFERENCES ¶

S. K. Wolf, Synthetic Production and Control of Acoustic Phenomena by a Magnetic Recording System, *Proceedings IRE*, p. 365, July, 1941.

P. C. Goldmark, P. S. Hendricks, Synthetic Reverberation, *Proceedings IRE*, p. 747, December, 1939.

J. K. Hilliard, Reverberation Control in Motion Picture Recordings, *Electronics*, p. 15, January, 1938.

H. F. Olson, Elements of Acoustical Engineering, 1940, D. Van Nostrand Co.