

*Symposium
on
Auditory
Perspective*

Symposium on Auditory Perspective

In 1964, Paul Klipsch reprinted this paper. Here is his introduction – I cannot say it better:

The following paper is a reprint of one of the most important papers in the field of audio. Fundamentals do not change. The laws of physics endure. In reprinting the Symposium, the fundamentals are restated.

One is tempted to editorialize on a paper that is thirty years old [now seventy years! – jm], but to do so would inject what the editor thinks the authors meant. Rather, in this case, the reader may at least read what the author said. But to yield just a little to the temptation one may suggest judging any “major breakthrough” in the light of these fundamentals.

To preserve references, page numbers from the original printing have been preserved.

It is intended to reprint other papers, and readers are invited to submit suggestions for reprinting of papers which, like this one, are truly milestones in the art.

Paul W. Klipsch 30 April, 1964

Our thanks to Mike Durff for loaning us the Klipsch reprint, which I have scanned and present here as a “searchable image”.

John G. (Jay) McKnight, Chair
AES Historical Committee
2002 Dec 23

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A Symposium on

Wire Transmission of Symphonic Music^{and its} Reproduction in Auditory Perspective

On April 27, 1933, another milestone in the development of the communication art was passed when the music of the Philadelphia Symphony Orchestra was picked up in the Academy of Science Hall in Philadelphia and reproduced in Constitution Hall in Washington, D. C., with a fidelity, depth, and spatial effect that effectively created the illusion of the orchestra's presence behind the stage curtain. To achieve this effect it was necessary that the frequency, intensity, and phase relations of the sound in each ear of each listener be reproduced so accurately as to convey not only the sounds of the various instruments, but also their spatial relations with respect to each other. In this experiment a close approximation to complete

facsimile reproduction of symphonic music was obtained by using a 3-channel system, each channel involving its own microphone, amplifier control, transmission, and reproducing equipment. With such a system the auditory illusion was substantially complete and the effect upon the listening audience in the distant hall was essentially the same as though the orchestra had been behind the stage curtains there instead of miles away in another city. Details of the various principles and apparatus involved in the auditory perspective system used in the Philadelphia-Washington experiment are treated in the 6 papers of this symposium appearing on this and the following 23 pages, and on p. 214-19, inclusive, of this issue.

Auditory Perspective

—Basic Requirements

The fundamental requirements involved in a system capable of picking up orchestral music, transmitting it a long distance, and reproducing it in a large hall, are discussed in this first paper of the symposium.

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IN THIS ELECTRICAL ERA one is not surprised to hear that orchestral music can be picked up in one city, transmitted a long distance, and reproduced in another. Indeed, most people think such things are commonplace. They are heard every night on the radio. However, anyone who appreciates good music would not admit that listening even to the best radio gives the emotional thrill experienced in the concert hall. Nor is it evident that a listener in a small room ever will be able to

get the same effect as that experienced in a large hall, although it must be admitted that such a question is debatable. The proper answer will involve more than a consideration of only the physical factors.

This symposium describes principles and apparatus involved in the reproduction of music in large halls, the reproduction being of a character that may give even greater emotional thrills to music lovers than those experienced from the original music. This statement is based upon the testimony of those who have heard some of the few concerts reproduced by the apparatus which will be described in the papers of this symposium.

It is well known that when an orchestra plays, vibrations which are continually changing in form are produced in the air of the concert hall where the orchestra is located. An ideal transmission and reproducing system may be considered as one that produces a similar set of vibrations in a distant concert hall in which is executed the same time-sequence of changes that takes place in the original hall. Since such changes are different at different positions in the hall, the use of such an ideal system implies that at corresponding positions in the two halls this time-sequence should be the same. Obviously, this never can be true at every position unless the halls are the same size and shape; corresponding positions would not otherwise exist. Let us consider the case where the two halls are the same size and shape and also have the same acoustical properties. Let us designate the first hall in which the music originates by O , and the second one in which the music is reproduced by R . What requirements are necessary to obtain perfect reproduction from O into R such that any listener in any part of R will receive the same sound effects as if he were in the corresponding position in O ?

Suppose there were interposed between the orches-

Full text of a paper recommended for publication by the A.I.E.E. committee on communication, and scheduled for discussion at the A.I.E.E. winter convention, New York, N. Y., Jan. 23-26, 1934. Manuscript submitted Oct. 31, 1933; released for publication Dec. 4, 1933. Not published in pamphlet form.

tra and the audience a flexible curtain of such a nature that it did not interfere with a free passage of the sound, and which at the same time had scattered uniformly over it microphones which would pick up the sound waves and produce a faithful electrical copy of them. Assume each microphone to be connected with a perfect transmission line which terminates in a projector occupying a corresponding position on a similar curtain in hall *R*. By a perfect transmission line is meant one that delivers to the projector electrical energy equal both in form and magnitude to that which it receives from the microphone. If these sound projectors faithfully transform the electrical vibrations into sound vibrations, the audience in hall *R* should obtain the same effect as those listening to the original music in hall *O*.

Theoretically, there should be an infinite number of such ideal sets of microphones and sound projectors, and each one should be infinitesimally small. Practically, however, when the audience is at a considerable distance from the orchestra, as usually is the case, only a few of these sets are needed to give good auditory perspective; that is, to give depth and a sense of extensiveness to the source of the music. The arrangement of some of these simple systems together with their effect upon listeners in various parts of the hall is described in the paper by Steinberg and Snow. (Page 12)

In any practical system it is important to know how near these ideal requirements one must approach before the listener will be aware that there has been any degradation from the ideal. For example, it is well known that whenever a sound is suddenly stopped or started, the frequency band required to transmit the change faithfully is infinitely wide. Theoretically, then, in order to fulfill these ideal requirements for transmitting such sounds, all 3 elements in the transmission system should transmit all possible frequencies without change. Practically, because of the limitations of hearing, this is not necessary. If the intensities of some of the component frequencies required to represent such a change are below the threshold of audibility it is obvious that their elimination will not be detected by the average normal ear. Consequently, for high-grade reproduction of sounds it is obvious that, except in very special cases, the range of frequencies that the system must transmit is determined by the range of hearing rather than by the kind of sound that is being reproduced.

Tests have indicated that, for those having normal hearing, pure tones ranging in frequency from 20 to 20,000 cycles per second can be heard. In order to sense the sounds at either of these extreme limits, they must have very high intensity. In music these frequencies usually are at such low intensities that the elimination of frequencies below 40 cps and those above 15,000 cps produces no detectable difference in the reproduction of symphonic music. These same tests also indicated that the further elimination of frequencies beyond either of these limits did begin to produce noticeable effects, particularly on a certain class of sounds produced in the orchestra. For example, the elimination of all frequencies above 13,000 cps produced a detectable change in the repro-

duced sound of the snare drum, cymbals, and castanets. Also, the elimination of frequencies below 40 cps produced detectable differences in reproduced music of the base viol, the bass tuba, and particularly of the organ.

Within this range of frequencies the system (the combination of the microphone, transmission line, and loud speaker) should reproduce the various frequencies with the same efficiency. Such a general statement sounds correct, but a careful analysis of it would reveal that when any one tried to build such a system or tried to meet such a requirement he would have great difficulty in understanding what it meant.

For example, for reproducing all the frequencies within this band, a certain system may be said to have a uniform efficiency when it operates between two rooms under the condition that the pressure variation at a certain distance away from the sound projector is the same as the pressure variation at a certain position in front of the microphone. It is obvious, however, that in other positions in the 2 rooms this relation would not in general hold. Also, if the system were transferred into another pair of rooms the situation would be entirely changed. These difficulties and the way they were met are discussed in the papers of this symposium that deal with loud speakers and microphones (p. 17) and with methods of applying the reproducing system to the concert hall (p. 216). It will be obvious from these papers that the criterion for determining the ideal frequency characteristics to be given to the system is arbitrary within certain limits. However, solving the problem according to criteria adopted produced a system that gave very satisfactory results.

Besides the requirement on frequency response just discussed, the system also must be capable of handling sound powers that vary through a very wide range. If this discussion were limited to the type of symphonic music that now is produced by the large orchestras, this range would be about 10,000,000 to 1, or 70 decibels. To reproduce such music then, the system should be capable of handling the smallest amount of power without introducing extraneous noises approaching it in intensity, and also reproduce the most intense sounds without overloading any part of the transmission system. However, this range is determined by the capacities of the musical instruments now available and the man power that conveniently can be grouped together under one conductor. As soon as a system was built that was capable of handling a much wider range, the musicians immediately took advantage of it to produce certain effects that they previously had tried to obtain with the orchestra alone, but without success because of the limited power of the instruments themselves. For these reasons it seems clear that the desirable requirements for intensity range, as well as those for frequency range, are determined largely by the ear rather than by the physical characteristic of any sound. An ideal transmission should, without introducing an extraneous audible sound, be capable of reproducing a sound as faintly as the ear can hear and as loudly as the ear can tolerate. Such

a range has been determined with the average normal ear when using pure tones. The results of recent tests are shown in Fig. 1.

The ordinates are given in decibels above the reference intensity which is 10^{-16} watts per square centimeter. The values are for field intensities existing in an air space free from reflecting walls. The most intense peaks in music come in the range between 200 and 1000 cps. Taking an average for this range it may be seen that there is approximately a 100-db range in intensity for the music, provided about 10 db is allowed for the masking of sound in the concert hall even when the audience is quietest.

The music from the largest orchestra utilizes only

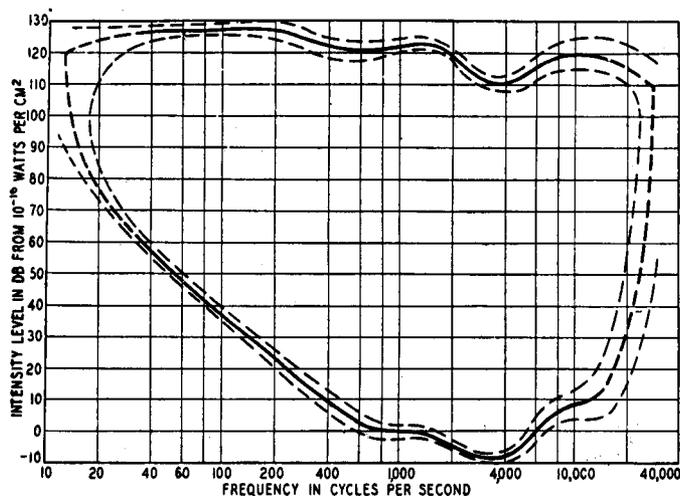


Fig. 1. Limits of audible sound as determined by recent tests

70 db of this range when it plays in a concert hall of usual size. To utilize the full capabilities of the hearing range the ideal transmission system should add about 10 db on the *pp* side and 20 db on the *ff* side of the range. The capacity of the sound projectors necessary to reach the maximum allowable sound that the ear can tolerate varies with the size of the room. A good estimate can be obtained by the following consideration.

If T is the time of reverberation of the hall in seconds, E the power of the sound source in watts, I the maximum energy density per cubic centimeter in joules, and V the volume of the hall in cubic centimeters, then it is well known that

$$I = \frac{1}{6 \log_e 10} \cdot \frac{ET}{V} \quad (1)$$

Measurements have shown that when the sound intensity in a free field reaches about 10^{-4} watts per square centimeter, the average person begins to feel the sound. This maximum value is approximately the same for all frequencies in the important audible range. Any higher intensities, and for some persons somewhat lower intensities, become painful and may injure the hearing mechanism. This intensity corresponds to an energy density I of 3×10^{-9} joules. Using this figure as the upper limit to be tolerated

by the human ear, then, the maximum power of the sound source must be given by

$$E = 4.1 \times 10^{-3} \frac{V}{T} \quad (2)$$

For halls like the Academy of Music in Philadelphia and Carnegie Hall in New York City, in which the volume V is approximately 2×10^{10} cubic centimeters and the reverberation time about 2 sec, E , the power of the sound source, is approximately 400 watts. For other halls it may be seen that the power required for this source is proportional to the volume of the hall and inversely proportional to the reverberation time. A person would experience the sense of feeling when closer than about 10 meters to such a source of 400 watts power, even in free open space. Hence it would be unwise to have seats closer than 10 or 15 meters from the stage when such powers are to be used.

These, then, are the general fundamental requirements for an ideal transmission system. How near they can be realized with apparatus that we now know how to build will be discussed in the papers included in this symposium.

A system approximately fulfilling these requirements was constructed and used to reproduce the music played by the Philadelphia Orchestra. The first public demonstration was given in Constitution Hall, Washington, D. C., on the evening of April 27, 1933, under the auspices of the National Academy of Sciences. At that time, Dr. Stokowski, Director of the Philadelphia Orchestra, manipulated the electric controls from a box in the rear of Constitution Hall while the orchestra, led by Associate Conductor Smallens, played in the Academy of Music in Philadelphia.

Three microphones of the type described in the paper by Wente and Thuras (p. 17) were placed before the orchestra in Philadelphia, one on each side and one in the center at about 20 ft in front of and 10 ft above the first row of instruments in the orchestra. The electrical vibrations generated in each of these microphones were amplified by voltage amplifiers and then fed into a transmission line which was extended to Washington by means of telephone cable. The construction of these lines, the equipment used with them, and their electrical properties, are described in the paper by Affel, Chesnut, and Mills (p. 28). In Constitution Hall at Washington, D. C., these transmission lines were connected to power amplifiers. The type of power amplifiers and voltage amplifiers used are described in the paper by Scriven (p. 25). The output of these amplifiers fed 3 sets of loud speakers like those described in the paper by Wente and Thuras. They were placed on the stage in Constitution Hall in positions corresponding to the microphones in the Academy of Music, Philadelphia.

Judging from the expression of those who heard this concert, the development of this system has opened many new possibilities for the reproduction and transmission of music that will create even a greater emotional appeal than that obtained when listening to the music coming directly from the orchestra through the air.

Auditory Perspective —Physical Factors

In considering the physical factors affecting it, auditory perspective is defined in this paper as being reproduction which preserves the spatial relationships of the original sounds. Ideally, this would require an infinite number of separate microphone-to-speaker channels; practically, it is shown that good auditory perspective can be obtained with only 2 or 3 channels. This is the second paper in the symposium.

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ABILITY to localize the direction, and to form some judgment of the distance from a sound source under ordinary conditions of listening, are matters of common experience. Because of this faculty an audience, when listening directly to an orchestral production, senses the spatial relations of the instruments of the orchestra. This spatial character of the sounds gives to the music a sense of depth and of extensiveness, and for perfect reproduction should be preserved. In other words, the sounds should be reproduced in true *auditory perspective*.

In the ordinary methods of reproduction, where only a single loud speaking system is used, the spatial character of the original sound is imperfectly preserved. Some of the depth properties of the original sound may be conveyed by such a system,¹ but the directional properties are lost because the audience tends to localize the sound as coming from the direction of a single source, the loud speaker. Ideally, there are 2 ways of reproducing sounds in true auditory perspective. One is binaural reproduction which aims to reproduce in a distant listener's ears, by means of head receivers, exact copies of the sound vibrations that would exist in his ears if he were listening directly. The other method, which was described in the first paper of this series, uses loud speakers and aims to reproduce in a distant hall an exact copy of the pattern of sound vibration that exists in the original hall. In the limit, an

infinite number of microphones and loud speakers of infinitesimal dimensions would be needed.

Far less ideal arrangements, consisting of as few as 2 microphone-loudspeaker sets, have been found to give good auditory perspective. Hence, it is not necessary to reproduce in the distant hall an exact copy of the vibrations existing in the original hall. What physical properties of the waves must be preserved then, and how are these properties preserved by various arrangements of 2- and 3-channel loudspeaker reproducing systems? To answer these questions, some very simple localization tests have been made with such systems. Perhaps attention can be focused more easily on their important properties by considering briefly the results of these tests.

LOCALIZATION AFFORDED BY MULTICHANNEL SYSTEMS

In Fig. 1 is shown a diagram of the experimental set-up that was used. The microphones, designated as *LM* (left), *CM* (center), and *RM* (right), were set on a "pick-up" stage that was marked out on the floor of an acoustically treated room. The loud speakers, designated as *LS*, *CS*, and *RS*, were placed in the front end of the auditorium at the Bell Telephone Laboratories and were concealed from view by a curtain of theatrical gauze. The average position of a group of 12 observers is indicated by the cross in the rear center part of the auditorium.

The object of the tests was to determine how a caller's position on the pick-up stage compared with his apparent position as judged by the group of observers in the auditorium listening to the reproduced speech. Words were uttered from some 15 positions on the pick-up stage in random order. The 9 positions shown in Fig. 1 were always included in the 15, the remaining positions being introduced to minimize memory effects. The reproducing system was switched off while the caller moved from one position to the other.

In the first series of tests, the majority of the observers had no previous experience with the set-up. They simply were given a sheet of coordinate paper with a single line ruled on it to indicate the line of the gauze curtain and asked to locate the apparent position of the caller with respect to this line. Following these tests, the observers were permitted to listen to speech from various announced positions on the pick-up stage. This gave them some notion of the approximate outline of what might be called the "virtual" stage. These tests then were repeated. As there was no significant difference in results, the data from both tests have been averaged and are shown in Fig. 1.

The small diagram at the top of Fig. 1 shows the caller's positions with respect to the microphone positions on the pick-up stage. The corresponding average apparent positions when reproduced are shown with respect to the curtain line and the loudspeaker positions. The type of reproduction is indicated symbolically to the right of the apparent position diagrams.

With 3-channel reproduction there is a reasonably

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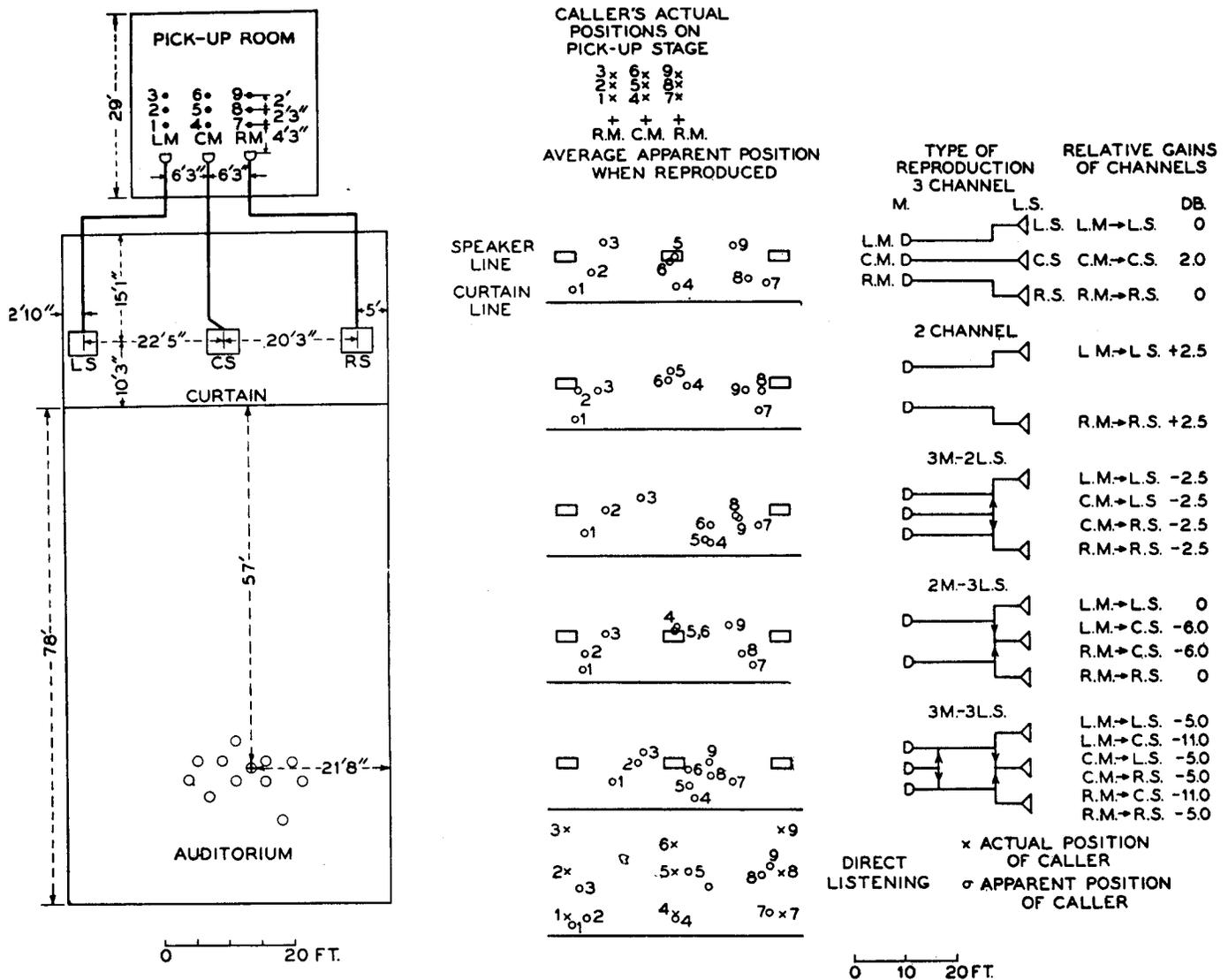


Fig. 1. Diagram of arrangement (left) for sound localization tests and (right) the results obtained

good correspondence between the caller's actual position on the pick-up stage and his apparent position on the virtual stage. Apparent positions to the right or left correspond with actual positions to the right or left, and apparent front and rear positions correspond with actual front and rear positions. Thus the system afforded lateral or "angular" localization as well as fore and aft or "depth" localization. For comparison, there is shown in the last diagram the localization afforded by direct listening. The crosses indicate a caller's position in back of the gauze curtain and the circles indicate his apparent position as judged by the observers listening to his speech directly. In both cases, as the caller moved back in a straight line on the left or right side of the stage, he appeared to follow a curved path pulling in toward the rear center; e. g., compare the caller positions 1, 2, 3, with the apparent positions 1, 2, 3. This distortion was somewhat greater for 3-channel reproduction than for direct listening.

The results obtained with the 2-channel system show 2 marked differences from those obtained with 3-channel reproduction. Positions on the center line of the pick-up stage (i. e., 4, 5, 6) all appear in

the rear center of the virtual stage, and the virtual stage depth for all positions is reduced. The virtual stage width, however, is somewhat greater than that obtained with 3-channel reproduction.

Bridging a third microphone across the 2-channel system had the effect of pulling the center line positions 4, 5, 6, forward, but the virtual stage depth remained substantially that afforded by 2-channel reproduction, while the virtual stage width was decreased somewhat. In this and the other bridged arrangements the bridging circuits employed amplifiers, as represented by the arrows in Fig. 1, in such a way that there was a path for speech current only in the indicated direction.

Bridging a third loud speaker across the 2-channel system had the effect of increasing the virtual stage depth and decreasing the virtual stage width, but positions on the center line of the pick-up stage appeared in the rear center of the virtual stage as in 2-channel reproduction.

Bridging both a third microphone and a third loud speaker across the 2-channel system had the effect of reducing greatly the virtual stage width. The width could be restored by reducing the bridging

gains, but fading the bridged microphone out caused the front line of the virtual stage to recede at the center, whereas fading the bridged loud speaker out reduced the virtual stage depth. No fixed set of bridging gains was found that would enable the arrangement to create the virtual stage created by 3 independent channels. The gains used in obtaining the data shown in Fig. 1 are indicated at the right of the symbolic circuit diagrams.

FACTORS AFFECTING DEPTH LOCALIZATION

Before attempting to explain the results that have been given in the foregoing, it may be of interest to consider certain additional observations that bear more specifically upon the factors that enter into the "depth" and "angular" localization of sounds. The microphones on the pick-up stage receive both direct and reverberant sound, the latter being sound waves that have been reflected about the room in which the pick-up stage is located. Similarly, the observer receives the reproduced sounds directly and also as reverberant sound caused by reflections about the room in which he listens. To determine the effects of these factors, the following 3 tests were made:

1. Caller remained stationary on the pick-up stage and close to microphone, but the loudness of the sound received by the observer was reduced by gain control. This was loudness change without a change in ratio of direct to reverberant sound intensity.
2. Caller moved back from microphone, but gain was increased to keep constant the loudness of the sound received by the observer. This was a change in the ratio of direct to reverberant sound intensity without a loudness change.
3. Caller moved back from microphone, but no changes were made in the gain of the reproducing system. This changed both the ratio and the loudness.

All of the observers agreed that the caller appeared definitely to recede in all 3 cases. That is, either a reduction in loudness or a decrease in ratio of direct to reverberant sound intensity, or both, caused the sound to appear to move away from the observer. Position tests using variable reverberation with a given pick-up stage outline showed that increasing the reverberation moved the front line of the virtual stage toward the rear, but had slight effect upon the rear line. When the microphones were placed outdoors to eliminate reverberation, reducing the loudness either by changing circuit gains or by increasing the distance between caller and microphone moved the whole virtual stage farther away. It is because of these effects that all center line positions on the pick-up stage appeared at the rear of the virtual stage for 2-channel reproduction.

It has not been found possible to put these relationships on a quantitative basis. Probably a given loudness change, or a given change in ratio of direct to reverberant sound intensity, causes different sensations of depth depending upon the character of the reproduced sound and upon the observer's familiarity with the acoustic conditions surrounding the reproduction. Since the depth localization is inaccurate even when listening directly, it is difficult to obtain sufficiently accurate data to be of much use in a quantitative way. Because of this inaccuracy,

good auditory perspective may be obtained with reproduced sounds even though the properties controlling depth localization depart materially from those of the original sound.

ANGULAR LOCALIZATION

Fortunately, the properties entering into lateral or angular localization permit more quantitative treatment. In dealing with angular localization, it has been found convenient to neglect entirely the effects of reverberant sound and to deal only with the properties of the sound waves reaching the ob-

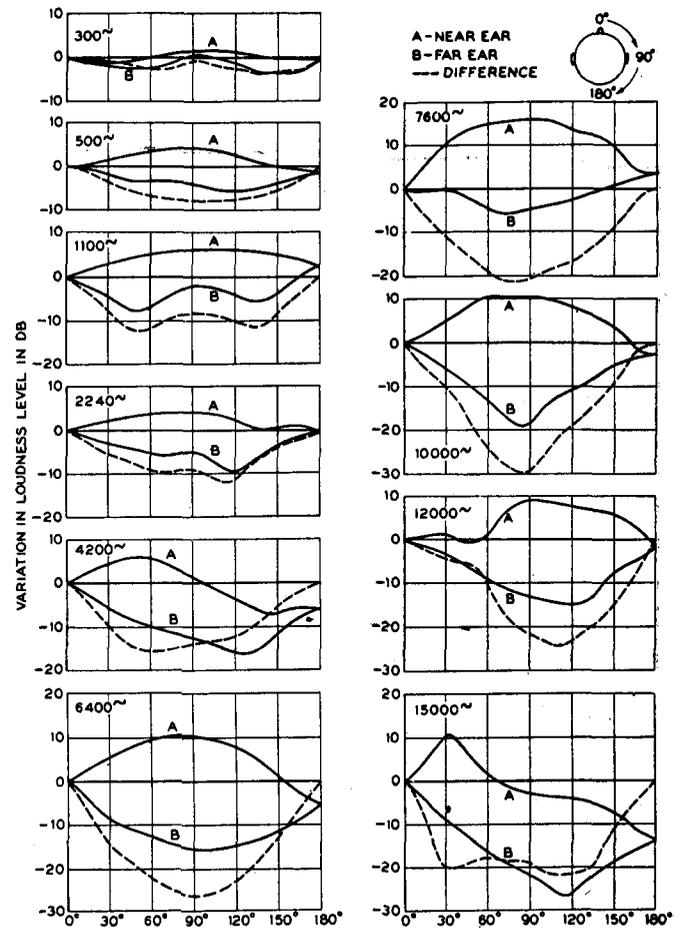


Fig. 2. Variation in loudness level as a sound source is rotated in a horizontal plane around the head

server's ears without reflections. The reflected waves or reverberant sounds do appear to have a small effect on angular localization, but it has not been found possible to deal with such sound in a quantitative way. One of the difficulties is that, because of differences in the build-up times of the direct and reflected sound waves, the amount of direct sound relative to reverberant sound reaching the observer's ears for impulsive sounds such as speech and music is much greater than would be expected from steady state methods of dealing with reverberant sound.

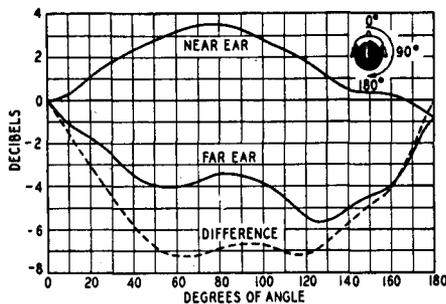
For the case of a plane progressive wave from a single sound source, and where the observer's head

is held in a fixed position, there are apparently only 3 factors that can assist in angular localization: namely, phase difference, loudness difference, and quality difference between the sounds received by the 2 ears.

In applying these factors to the localization of sounds from more than one source, as in the present case, the effects of phase differences have been neglected. It is difficult to see how phase differences in this case can assist in localization in the ordinary way. The 2 remaining factors, loudness and quality differences, both arise from the directivity of hearing. This directivity probably is due in part to the shadow and diffraction effects of the head and to the differences in the angle subtended by the ear openings. Measurements of the directivity with a source of pure tone located in various positions around the head in a horizontal plane have been reported by Sivian and White.² From these measurements, the loudness level differences between near and far ears have been determined for various frequencies. These differences are shown in Fig. 2 from which, using the pure tone data given, similar loudness level differences for complex tones may be calculated. Such calculated differences for speech are shown in Fig. 3.

As may be inferred from the varying shapes of the curves of Fig. 2, the directive effects of hearing introduce a frequency distortion more or less characteristic of the direction from which the sound comes. Thus the character or quality of complex sounds varies with the angle of the source. There are quality differences at each ear for various angles of source, and quality differences between the two

Fig. 3. Variation in loudness as a speech source is rotated in a horizontal plane around the head



ears for a given angle of source. In Fig. 4 is shown the frequency distortion at the right ear when a source of sound is moved from a position on the right to one on the left of an observer. It is a graph of the "difference" values of Fig. 2 for an angle of 90 deg. Frequencies above 4,000 cycles per second are reduced by as much as 15 to 30 decibels. This amount of distortion is sufficient to affect materially the quality of speech, particularly as regards the loudness of the sibilant sounds.

Reference to the difference curve of Fig. 3 shows that if, for example, a source of speech is 20 deg to the right of the median plane the speech heard by the right ear is 3 db louder than that heard by the left ear. A similar difference exists when the angle is 167 deg. Presumably, when the right ear hears

speech 3 db louder than the left, the observer localizes the sound as coming from a position 20 deg or 167 deg to the right, depending upon the quality of the speech. If this be assumed to be true, even though the difference is caused by the combination of sounds of similar quality from several sources, it should be possible to calculate the apparent angle.

LOUDNESS THEORY OF LOCALIZATION

Upon this assumption the apparent angle of the source as a function of the difference in decibels between the speech levels emitted by the loud speakers of the 2- and 3-channel systems has been calculated. Each loud speaker contributes an amount of direct sound loudness to each ear, depending upon its distance from, and its angular position with respect to, the observer. These contributions were combined on a power basis to give a resultant loudness of direct sound at each ear, from which the difference in loudness between the 2 ears was determined. The calculated results for the 2- and 3-channel systems are shown by the solid lines in Fig. 5. The y axis shows the apparent angle, positive angle being measured in a clockwise direction. The x axis shows the difference in decibels between the speech levels from the right and left loud speakers. The points are observed values taken from Fig. 1. The observed apparent angles were obtained directly from the average observer's location and the average apparent positions shown in Fig. 1. The speech levels from each of the loud speakers were calculated for each position on the pick-up stage. This was done by assuming that the waves arriving at the microphone had relative levels

Fig. 4. Loudness difference produced in the right ear when a source of pure tone is moved from a position on the right to the left of an observer

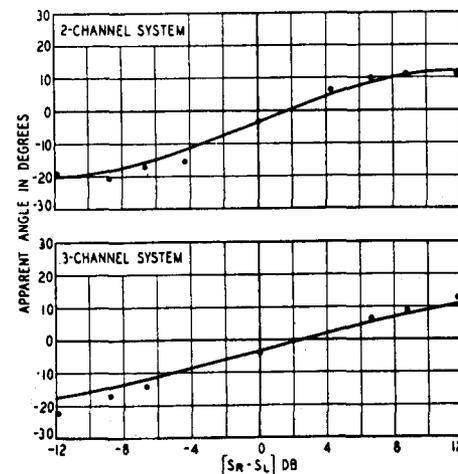
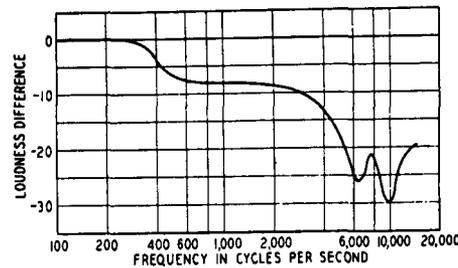


Fig. 5. Calculated and observed apparent angles for 2- and 3-channel reproduction

inversely proportional to the squares of the distances traversed. By correcting for the angle of incidence and for the known relative gains of the systems, the speech levels from the loud speakers were obtained.

A comparison of the observed and calculated results seems to indicate that the loudness difference at the 2 ears accounts for the greater part of the apparent angle of the reproduced sounds. If this is true, the angular location of each position on the virtual stage results from a particular loudness difference at the 2 ears produced by the speech coming from the loud speakers. When 3 channels are used a definite set of loud speaker speech levels exists for each position on the pick-up stage. To create these same sets of loud speaker speech levels with the bridging arrangement of 3-microphones 3-loud speakers already discussed, it would be necessary to change the bridging gains for each position on the pick-up stage. Hence it could not be expected that the arrangement as used (i. e., with fixed gains) would create a virtual stage identical with that created by 3-channel reproduction. However, with proper technique, bridging arrangements on a given number of channels can be made to give better reproduction than would be obtained with the channels alone.

EXPERIMENTAL VERIFICATION OF THEORY

Considerations of loudness difference indicate that all caller positions on the pick-up stage giving the same relative loud speaker outputs should be localized at the same virtual angle. The solid lines of Fig. 6 show a stage layout used to test this hypothesis with the 2-channel system. All points on each line have a constant ratio of distances to the microphones. The resulting direct sound differences in pressure ex-

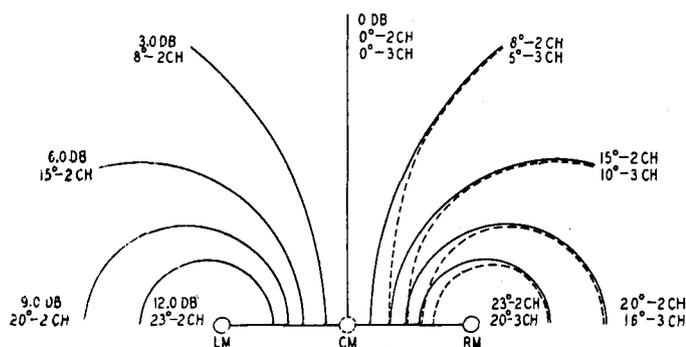


Fig. 6. Pick-up stage contour lines of constant apparent angle

pressed in decibels and the corresponding calculated apparent angles are indicated beside the curves. The apparent angles were calculated for an observing position on a line midway between the 2 loud speakers but at a distance from them equal to the separation between them. The microphones were turned face up at the height of the talker's lips to eliminate quality changes caused by changing incidence angle. It was found that a caller walking

along one of these lines maintained a fairly constant virtual angle. For caller positions far from the microphones the observed angles were somewhat greater than those computed. For highly reverberant conditions, the tendency was toward greater calculated than observed angles. Reverberation also decreased the accuracy of localization.

A change of relative channel gain caused a change in virtual angle as would be expected from loudness difference considerations. For instance, if the caller actually walked the left 3-db line, he seemed to be on the 6-db line when the left channel gain was raised 3 db. Many of the effects of moving about the pick-up stage could be duplicated by volume control manipulation as the caller walked forward and backward on the center path. With a bridged center microphone substituted for the 2 side microphones similar effects were possible and, in addition, the caller by speaking close to the microphone could be brought to the front of the virtual stage.

For observing positions near the center of the auditorium the observed angles agreed reasonably well with calculations based only upon loudness differences. As the observer moved to one side, however, the virtual source shifted more rapidly toward the nearer loud speaker than was predicted by the computations. This was true of reproduction in the auditorium, both empty and with damping simulating an audience, and outdoors on the roof. Computations and experiment also show a change in apparent angle as the observer moves from front to rear, but its magnitude is smaller than the error of an individual localization observation. Consequently, observers in different parts of the auditorium localize given points on the pick-up stage at different virtual angles.

Because the levels at the 3 microphones are not independent, and because the desired contours depend upon the effects at the ears, a 3-channel stage is not as simple to lay out as a 2-channel stage. For a given observing position, however, a set of contour lines can be calculated. The dashed lines at the right of Fig. 6 show 4 contours thus calculated for the circuit condition of Fig. 1 and the observing position previously mentioned. The addition of the center channel reduces the virtual angle for any given position on the pick-up stage by reducing the resultant loudness difference at the ears. Although the 3-channel contours approach the 2-channel contours in shape at the back of the stage, a given contour results in a greater virtual angle for 2- than for 3-channel reproduction.

Similar effects were obtained experimentally. As in 2-channel reproduction, movements of the caller could be simulated by manipulation of the channel gains. From an observing standpoint the 3-channel system was found to have an important advantage over the 2-channel system in that the shift of the virtual position for side observing positions was smaller.

EFFECTS OF QUALITY

If the quality from the various loud speakers differs, the quality of sound is important to localiza-

tion. When the 2-channel microphones were so arranged that one picked up direct sound and reverberation while the other picked up mostly reverberation, the virtual source was localized exactly in the "direct" loud speaker until the power from the "reverberant" loud speaker was from 8 to 10 db greater. In general, localization tends toward the channel giving most natural or "close-up" reproduction, and this effect can be used to aid the loudness differences in producing angular localization.

PRINCIPAL CONCLUSIONS

The principal conclusions that have been drawn from these investigations may be summarized as follows:

1. Of the factors influencing angular localization, loudness difference of direct sound seems to play the most important part; for certain observing positions the effects can be predicted reasonably well from computations. When large quality differences exist between the loudspeaker outputs, the localization tends toward the more natural source. Reverberation appears to be of minor importance unless excessive.
2. Depth localization was found to vary with changes in loudness, the ratio of direct to reverberant sound, or both, and in a manner not found subject to computational treatment. The actual ratio of direct to reverberant sound, and the change in the ratio, both appeared to play a part in an observer's judgment of stage depth.
3. Observers in various parts of the auditorium localize a given source at different virtual positions, as is predicted by loudness computations. The virtual source shifts to the side of the stage as the observer moves toward the side of the auditorium. Although quantitative data have not been obtained, qualitative data on these effects indicate that the observed shift is considerably greater than that computed. Moving backward and forward in the auditorium appears to have only a small effect on the virtual position.
4. Because of these physical factors controlling auditory perspective, point-for-point correlation between pick-up stage and virtual stage positions is not obtained for 2- and 3-channel systems. However, with stage shapes based upon the ideas of Fig. 7, and with suitable use of quality and reverberation, good auditory perspective can be produced. Manipulation of circuit conditions probably can be used advantageously to heighten the illusions or to produce novel effects.
5. The 3-channel system proved definitely superior to the 2-channel by eliminating the recession of the center-stage positions and in reducing the differences in localization for various observing positions. For musical reproduction, the center channel can be used for independent control of soloist renditions. Although the bridged systems did not duplicate the performance of the physical third channel, it is believed that with suitably developed technique their use will improve 2-channel reproduction in many cases.
6. The application of acoustic perspective to orchestral reproduction in large auditoriums gives more satisfactory performance than probably would be suggested by the foregoing discussions. The instruments near the front are localized by every one near their correct positions. In the ordinary orchestral arrangement, the rear instruments will be displaced in the reproduction depending upon the listener's position, but the important aspect is that every auditor hears differing sounds from differing places on the stage and is not particularly critical of the exact apparent positions of the sounds so long as he receives a spatial impression. Consequently 2-channel reproduction of orchestral music gives good satisfaction, and the difference between it and 3-channel reproduction for music probably is less than for speech reproduction or the reproduction of sounds from moving sources.

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Auditory Perspective —Loud Speakers and Microphones

In ordinary radio broadcast of symphony music, the effort is to create the effect of taking the listener to the scene of the program, whereas in reproducing such music in a large hall before a large gathering the effect required is that of transporting the distant orchestra to the listeners. Lacking the visual diversion of watching the orchestra play, such an audience centers its interest more acutely in the music itself, thus requiring a high degree of perfection in the reproducing apparatus both as to quality and as to the illusion of localization of the various instruments. Principles of design of the loud speakers and microphones used in the Philadelphia-Washington experiment are treated at length in this, the third paper of this symposium.

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AS EARLY AS 1881 a large scale musical performance was reproduced by telephone instruments at the Paris Electrical Exhibition. Microphones were placed on the stage of the Grand Opera and connected by wires to head receivers at the exposition. It is interesting to note that separate channels were provided for each ear so as to give to the music perceived by the listener the "character of relief and localization." With head receivers it is necessary to generate enough sound of audible intensity to fill only a volume of space enclosed between the head receiver and the ear. As no amplifiers were available, the production of enough sound to fill a large auditorium would have been entirely outside the range of possibilities. With the advent

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1. Bell Telephone Laboratories, Inc.

of telephone amplifiers, microphone efficiency could be sacrificed to the interest of good quality where, as in the reproduction of music, this was of primary interest. When amplifiers of greater output power capacity were developed, loud speakers were introduced to convert a large part of the electrical power into sound so that it could be heard by an audience in a large auditorium. Improvements have been made in both microphones and loud speakers, resulting in very acceptable quality of reproduction of speech and music; as is found, for instance, in the better class of motion picture theaters.

In the reproduction, in a large hall, of the music of a symphony orchestra the approach to perfection that is needed to satisfy the habitual concert audience undoubtedly is closer than that demanded for any other type of musical performance. The interest of the listener here lies solely in the music. The reproduction therefore should be such as to give to a lover of symphonic music esthetic satisfaction at least as great as that which would be given by the orchestra itself playing in the same hall. This is more than a problem of instrument design, but this paper will be restricted to a discussion of the requirements that must be met by the loud speakers and microphones, and to a description of the principles of design of the instruments used in the transmission of the music of the Philadelphia Orchestra from Philadelphia to Constitution Hall in Washington. Some of the requirements are found in the results of measurements that have been made on the volume and frequency ranges of the music produced by the orchestra.

GENERAL CONSIDERATIONS

The acoustic powers delivered by the several instruments of a symphony orchestra, as well as by the orchestra as a whole, have been investigated by Sivian, Dunn, and White. Figure 1 was drawn on the basis of the values published by them.¹ The ordinates of the horizontal lines give the values of the peak powers within the octaves indicated by the positions of the lines. For a more exact interpretation of these values the reader is referred to the original paper, but the chart here given will serve to indicate the power that a loud speaker must be capable of delivering in the various frequency regions; if the reproduced music is to be as loud as that given by the orchestra itself. However, it was the plan in the Philadelphia-Washington experiment to re-

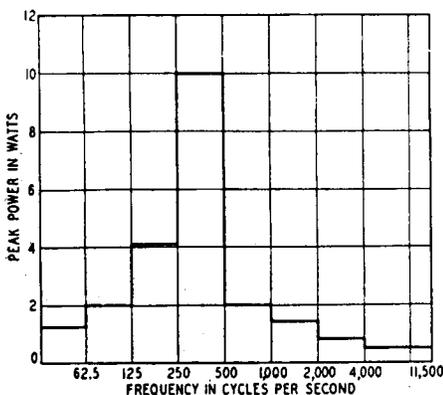


Fig. 1. Peak powers delivered by an orchestra within various frequency regions

produce the orchestra, when desired, at a level 8 or 10 db higher, so that with 3 channels each loud speaking system had to be able to deliver 2 or 3 times the powers indicated in Fig. 1. Sivian, Dunn, and White also found that for the whole frequency band the peak powers in some cases reached values as high as 65 watts. In order to go 8 db above this value, each channel would have to be capable of delivering in the neighborhood of 135 watts.

The chart (Fig. 1) shows that the orchestra delivers sound of comparable intensity throughout practically the whole audible range. Although it is conceivable that the ear would not be capable of detecting a change in quality if some of the higher or lower frequencies were suppressed, measurements published by W. B. Snow² show that for any change in quality in any of the instruments to be undetectable the frequency band should extend from about 40 to about 13,000 cps. The necessary frequency ranges that must be transmitted to obviate noticeable change in quality for the different orchestral instruments are indicated in the chart of Fig. 2, which is taken from the paper by Snow.

Thus far only the sound generated by the orchestra itself has been considered. However, it is well known that the esthetic value of orchestral music in a concert hall is dependent to a very great extent upon the acoustic properties of the hall. At first thought one might be inclined to leave this out of account in considering the reproduction by a loud speaking system, as one should normally choose a hall known to have satisfactory acoustics for an actual orchestra. There would be no further problem in this if the orchestral instruments and the loud speaker radiated the sound uniformly in all directions, but some of the important instruments are quite directive; i. e., they radiate much the greater portion of their sound through a relatively small angle. As an example, a polar diagram giving the relative intensities of the sound radiated in various directions by the violin is given in Fig. 3, which is taken from a paper published by Backhaus.³ The directional characteristics of some of the instruments is one of the chief reasons why the music from an orchestra does not sound the same in all parts of a concert hall. The music which we hear comes to us in part directly and in part indirectly; i. e., after one or more reflections from the walls. Both contribute to the esthetic value of the music. The ratio of the direct to the indirect sound, which has been designated by Hughes⁴ as the *acoustic ratio*, is to a first approximation inversely proportional to the product of the reverberation time and the angle through which the sound is radiated.⁵ For a steady tone by far the greater part of the intensity at a given point in a hall remote from the source is attributable to the indirect sound. However, inasmuch as many of the tones of a musical selection are of short duration, the direct sound is of great importance; it is this sound alone which enables us to localize the source. So far as this ratio is concerned, a decrease in the radiating angle of a loud speaker is equivalent to a reduction in the reverberation time of the hall. The effect on the music, however, is not entirely equivalent, for the rate of decay of sound in the room

is unaltered by a change in directivity of the source, as this depends only on the reverberation time.

As already pointed out, some of the instruments of the orchestra are quite directive and others are nondirectional. In general, it may be said that the instruments of lower register are less directive than those of higher register. To have each instrument as reproduced by the loud speaker sound just as the instrument itself would sound in the same hall, the loud speaker would have to reproduce the music from each instrument with a directivity corresponding to that of the instrument itself. This manifestly is impossible. The best that can be hoped for is a compromise. Let the loud speaking system be designed so that it is nondirective for the lower frequencies, and at the higher frequencies it will radiate the sound through a larger angle than the most directive of the instruments and through a smaller angle than the least directive. Although this compromise means that the individual instruments will not sound exactly like the originals, it carries with it one advantage: At all the seats in the hall included in the radiating angle and at a given distance from the loud speaker the music may be heard to equal advantage, whereas with the orchestra itself the most desirable seats comprise only a certain portion of the hall. The optimum radiating angle is largely a matter of judgment; if it is too small the music will lack the spatial quality experienced at indoor concerts; if it is too large there will be a loss in definition.

There is another respect in which the directivity of the source can greatly affect the tone quality. Most loud speakers radiate tones of low frequency

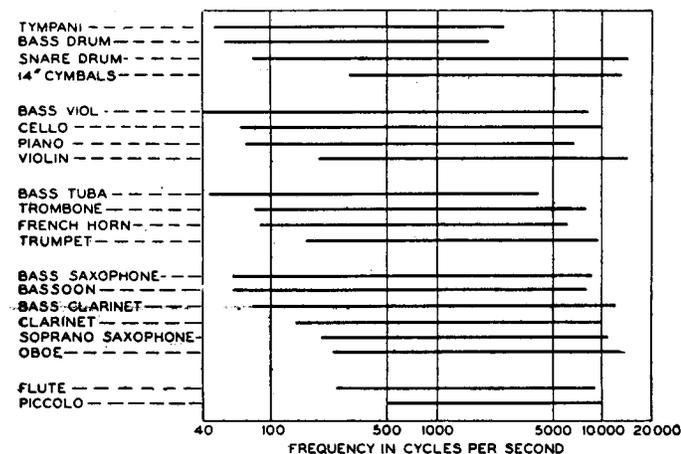


Fig. 2. Frequency transmission range required to produce no noticeable distortion for orchestral instruments

through a relatively large angle, but as the frequency is increased this angle becomes smaller and smaller. Under this condition the relation between the intensities of the high and low frequency tones as received directly will be different for almost all parts of the hall. Hence, even with equalization by electrical networks, the reproduction at best can be good only at a few places in the hall. Therefore,

the sound radiated not only should be contained within a certain solid angle, but the radiation throughout this angle should be uniform at all frequencies.

THE LOUD SPEAKER

At present 2 kinds of loud speakers are in wide commercial use, the direct radiating and the horn types. Each has its merits, but the latter was used in the Philadelphia-Washington experiment because it appears to have definite advantages where such large amounts of power are to be radiated. The horn type can be given the desired directive properties more readily, and higher values of efficiency throughout a wide frequency range are more easily realized. In consideration of the large power requirements, high efficiency is of special importance because it will keep to the lowest possible value the power capacity requirements of the amplifiers and because, with the heating proportional to one minus the efficiency, the danger of burning out the receiving units is reduced.

For efficiently radiating frequencies as low as 40 cps, a horn of large dimensions is required. In order that the apparatus may not become too unwieldy the folded type of horn is preferable, but a large folded horn transmits high frequency

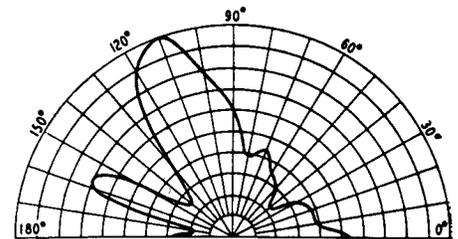


Fig. 3. Variation of intensity with direction of the sound radiated by a violin (660 cps)

tones very inefficiently. As actually used, therefore, the loud speaker was constructed in 2 units: one for the lower and the other for the higher frequencies, an electrical network being used to divide the current into 2 frequency bands, the point of division being about 300 cps.

THE LOW FREQUENCY HORN

When moderate amounts of power are transmitted through a horn the sound waves will suffer very little distortion, but when the power per unit area becomes large, second-order effects, usually neglected in considering waves of small amplitude, must be taken into account. The transmission of waves of large amplitude through an exponential horn has been investigated theoretically by M. Y. Rocard.⁶ His investigation shows that if W watts are transmitted through the throat of an exponential horn a second harmonic of intensity RW will be generated, where R is given by the relation

$$R = \frac{(\gamma + 1)^2 f^2 \times 10^7 W}{2\rho c^2 f_0^2 A} \quad (1)$$

in which f is the frequency of the fundamental, f_0

the cut-off frequency of the horn, c the velocity of sound, ρ the density of air, and A the area of the throat of the horn, all expressed in cgs units. It may be noted that the intensity of the harmonic increases with the ratio of the frequency to the cut-off frequency of the horn; this is another argument against attempting to cover too wide a range of frequencies with a single horn. In Fig. 1 it is shown that in the region of 200 cps the orchestra gives peak powers of about 10 watts. If, therefore, 30 watts be set as the limit of power that the horn is to deliver at 200 cps, 32 cps as the cut-off frequency of the horn, and 30 db below the fundamental be assumed as the limit of tolerance of a second harmonic, from eq 1 a throat diameter of about 8 in. is determined.

If the radiation resistance at the throat of a horn is not to vary appreciably with frequency, the mouth opening must be a substantial fraction of a wave length. This condition calls for an unusually large horn if frequencies down to 40 cps and below are to be transmitted. However, the effect of variations in radiation resistance on sound output can be kept down to a relatively small value if the receiving unit is properly designed. This will be explained in the next section. The low frequency horn used in these reproductions has a mouth opening of about 25 sq ft. As computed from well-known formulas⁷ for the exponential horn the impedance of this horn with a throat diameter of 8 in. is shown in Fig. 4. These curves were computed under the assumption that the mouth of the horn is surrounded by a plane baffle of infinite extent, a condition closely approximated if the horn rests on a stage floor.

LOW FREQUENCY RECEIVING UNIT

When a moving coil receiving unit, coupled to a horn, is connected to an amplifier having an output resistance equal to $n-1$ times the damped resistance R of the driving coil, it can easily be shown that the sound power output is

$$P = \frac{\left(\frac{EBLT}{nR}\right)^2 r \times 10^{-9}}{\left[T^2 r + \frac{B^2 L^2 \times 10^{-9}}{nR}\right]^2 + [x_d + T^2 x]^2} \text{ watts} \quad (2)$$

where E is the open circuit voltage of the amplifier, L the length of wire in the receiver coil, T the ratio of the area of the diaphragm to the throat area of the horn, $r + jx$ the throat impedance of the horn, and x_d the mechanical reactance of the diaphragm and coil, the mechanical reactance of which is assumed to be negligibly small. From Fig. 4 it may be seen that the mean value of x increases as the frequency decreases to a value below 40 cps, and that x is smaller than r except at the very lowest frequencies. If, therefore, the stiffness of the diaphragm be adjusted so that x_d is equal to T^2 times the mean value of x at 40 cps, the second term in the denominator may be neglected without much error because it will have but little effect upon the sound output except at the higher frequencies, where the mass reactance of the coil and diaphragm may have to be taken into account.

If minimum variations in sound output are desired for variations in r ,

$$\frac{B^2 L^2 \times 10^{-9}}{nR T^2} = r_0 \quad (3)$$

where r_0 is equal to the geometric mean value of r , which is approximately equal to $A\rho c$.

If α is the ratio of the resistance at any frequency to the mean value, and if the second term in the denominator is neglected, eq 2 becomes

$$P = \frac{E^2}{nR} \frac{\alpha}{(1 + \alpha)^2} \quad (4)$$

In Fig. 4 it is shown that above 35 cps α has extreme values of 2.75 and 0.36, at which points there will

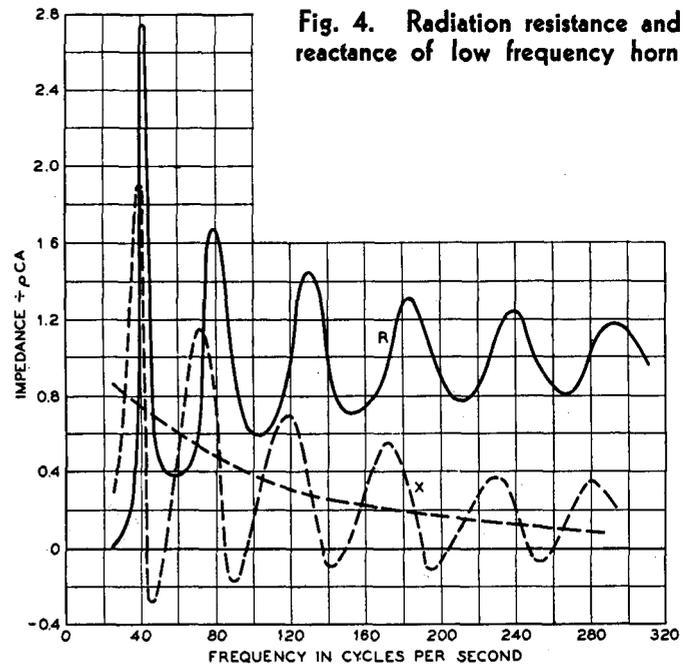


Fig. 4. Radiation resistance and reactance of low frequency horn

be minimum values in P , but these minimum values will not lie more than 1 db below the maximum values. Hence, if the receiver satisfies the condition of eq 3, the extreme variations in the sound output will not exceed 1 db, although the horn resistance varies by a factor of 7.5. Also it may be stated here that when the condition of eq 3 is satisfied the horn is terminated at the throat end by a resistance equal to the surge resistance of the horn. Thus eq 3 establishes a condition of minimum values in the transient oscillations of the horn.

The mean motional impedance of the loud speaker is $\frac{B^2 L^2 \times 10^{-9}}{T^2 r_0}$ which, from eq 3, is equal to nR . The condition of eq 3 therefore specifies that the efficiency of the loud speaker shall be $\frac{n}{n+1}$. The maximum power that an amplifier can deliver without introducing harmonics exceeding a specified value is a function of the impedance into which it operates. Therefore, to obtain the maximum acoustic power for a specified harmonic content, the load impedance should have the value for which the prod-

uct of the loud speaker efficiency and the power capacity of the amplifier has a maximum value. This optimum value of load impedance for the amplifier and loud speaker used in the Philadelphia-Washington experiments was found to be about 2.25 times the output impedance of the amplifier; the corresponding value of n then is 2.6 and the required efficiency 72 per cent. For best operating condition a definite value of receiver efficiency thus is specified.

The receiver may be made to satisfy the foregoing conditions regardless of the value of T , the ratio of diaphragm area to throat area. The area of the diaphragm has, however, a definite relation to the

construct the loud speaker to be able to deliver 25 watts in this region.

As the coil moves out of its normal position in the air gap, the force factor varies. Harmonics thus will be generated, the intensities of which increase with increasing amplitude. A limit to the maximum value of the amplitude ξ thus is set by the harmonic distortion that one is willing to tolerate. In this receiver the maximum value of ξ was taken equal to 0.060 in. Figure 4 shows that $\alpha\omega^2$ has a minimum value at about 50 cycles, where α is equal to about 0.4. These values give a ratio of 4.5 for T .

Inasmuch as $R = \frac{\sigma L^2}{v}$, where σ is the resistivity of the wire used for the coil and v the volume of the coil, from eq 3 is obtained

$$B^2v = n\sigma T^2r_010^9 \quad (5)$$

The first member gives the total magnetic energy that must be set up in the region occupied by the driving coil. This value is fixed by the fact that all factors in the second member are specified. The same performance is obtained with a small coil and high flux density as with a large coil and low flux density, provided B^2v is held fixed, but the coil in any case should not be made so small that it will be incapable of radiating the heat generated within it without danger of overheating, nor so large that the mass reactance of the coil will reduce the efficiency at the higher frequencies.

This receiver unit, when constructed according to the above principles and when connected to an amplifier and a horn in the specified manner, should be capable of delivering power 3 or 4 times that delivered by the orchestra in the frequency region lying between 35 and 400 cps, with an efficiency of about 70 per cent, and with a variation in sound output for a given input power to the amplifier of not more than 1 db throughout this range.

THE HIGH FREQUENCY HORN

It is well known that a tapered horn of the ordinary type has a directivity which varies with frequency. Sound of low frequency is projected through a relatively large angle. As the frequency is increased this angle decreases progressively until, at frequencies for which the wave length is small compared with the diameter of the mouth opening, the sound beam is confined to a very narrow angle about the axis of the horn.

If we had a spherical source of sound (i. e., a source consisting of a sphere, the surface of which has a radial vibratory motion equal in phase and amplitude at every point of the surface), sound would be radiated uniformly outward in all directions; or, if we had only a portion of a spherical surface over which the motion is radial and uniform, uniform sound radiation still would prevail throughout the solid angle subtended at the center of curvature by this portion of the sphere, provided its dimensions were large compared with the wave length. Throughout this region the sound would appear to originate at the center of curvature. Hence, for the ideal distribution of a spherical source within

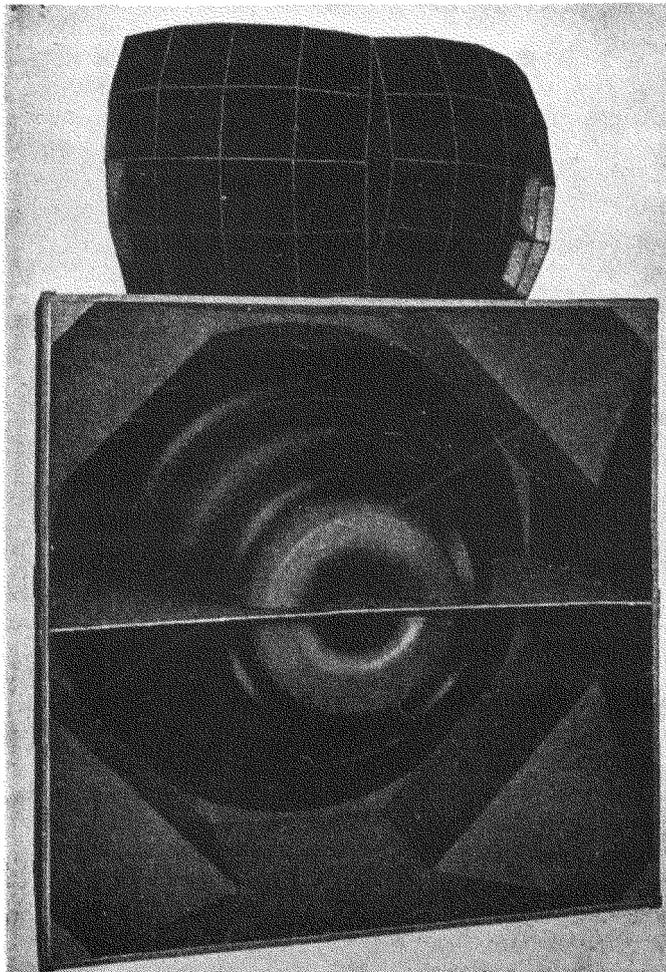


Fig. 5. Special loud speaker developed for auditory perspective experiment

maximum power that the receiver can deliver at the low frequencies. The peak power delivered by the receiver is equal to $T^2\alpha r_0\xi^2\omega^3 \times 10^{-7}$ peak watts where ξ is the maximum amplitude of motion of the diaphragm. Figure 1 shows that in the region lying between 40 and 60 cps, peak powers reach a value of from 1 to 2 watts. However, the low frequency tones of an orchestra are undesirably weak and may advantageously be reproduced at a relatively higher level. Therefore it was decided to

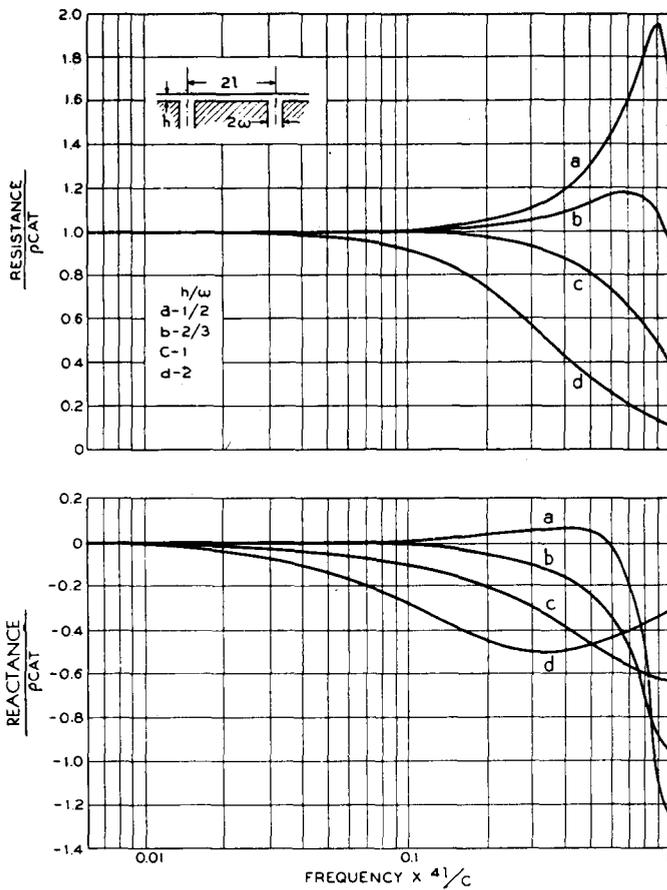


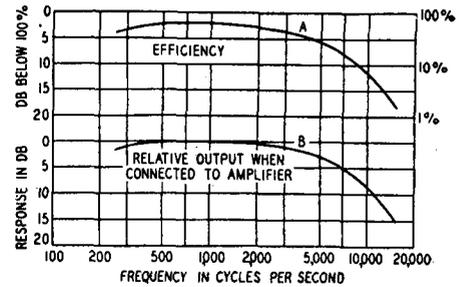
Fig. 6. Load impedance of speaker diaphragm

a region to be defined by a certain solid angle, it is necessary and sufficient that the radial motion be the same in amplitude and phase over the part of a spherical surface intercepted by the angle and having its center of curvature at the vertex and located at a sufficient distance from the vertex to make its dimensions large compared with the wave length. If, further, these conditions are satisfied for this surface at all frequencies, all points lying within the solid angle will receive sound of the same wave form. A horn was designed to meet these requirements for the high frequency band.

The horn, shown in the upper part of Fig. 5, comprises several separate channels, each of which has substantially an exponential taper. Toward the narrow ends these channels are brought together with their axes parallel, and are terminated into a single tapered tube which at its other end connects to the receiver unit. Sound from the latter is transmitted along the single tube as a plane wave and is divided equally among the several channels. If the channels have the same taper, the speed of propagation of sound in them is the same. The large ends are so proportioned and placed that the particle motion of the air will be in phase and equal over the mouth of the horn. This design gives a true spherical wave front at the mouth of the horn at all frequencies for which the transverse dimensions of the mouth opening are a large fraction of a wave length.

As the frequency is increased, the ratio of wave length to transverse width of the channels becomes less, and the sound will be confined more and more to the immediate neighborhood of the axis of each channel. The sound then will not be distributed uniformly over the mouth opening of the

Fig. 7. Relative computed sound output of high frequency receiver



horn, but each channel will act as an independent horn. To have a true spherical wave front up to the highest frequencies, the horn would have to be divided into a sufficient number of channels to make the transverse dimension of each channel small compared with the wave length up to the highest frequencies. If it is desired to transmit up to 15,000 cps, it is not very practical to subdivide the horn to that extent. Both the cost of construction and the losses in the horn would be high if designed to transmit also frequencies as low as 200 cps, as is the case under consideration. However, it is not important that at very high frequencies a spherical wave front be established over the whole mouth of the horn. For this frequency region it is perfectly satisfactory to have each channel act as an independent horn, provided that the construction of the horn is such that the direction of the sound waves coming from the channels is normal to the spherical wave front.

The angle through which sound is projected by this horn is about 60 deg, both in the vertical and in the horizontal direction. For reproducing the orchestra 2 of these horns, each with a receiving unit, were used. They were arranged so that a horizontal angle of 120 deg and a vertical angle of 60 deg were covered. These angular extensions were sufficient to cover most of the seats in the hall with the loud speaker on the stage. The vertical angle determines to a large extent the ratio of the direct to the indirect sound transmitted to the audience. The vertical angle of 60 deg was chosen purely on the basis of judgment as to what this ratio should be for the most pleasing results.

THE HIGH FREQUENCY RECEIVING UNIT

In the design of the low frequency receiver one of the main objectives was to reduce to a minimum the variations in sound transmission resulting from variations in the throat impedance of the horn. However, the high frequency horn readily can be made of a size such that the throat resistance has relatively small variations within the transmitting region. On the other hand, whereas the diameter of the diaphragm of the low frequency unit is only

a small fraction of the wave length, that of the high frequency unit must be several wave lengths at the higher frequencies in order to be capable of generating the desired amount of sound. Unless special provisions are made there will be a loss in efficiency because of differences in phase of the sound passing to the horn from various parts of the dia-

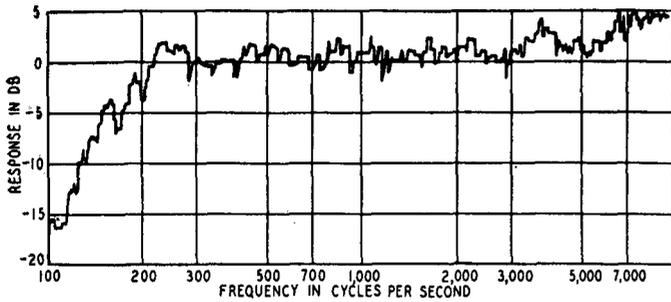


Fig. 8. Output-frequency characteristic of high frequency receiver as measured in a small room

phragm. The high frequency receiver therefore was constructed so that the sound generated by the diaphragm passes through several annular channels. There are enough of these channels to make the distance from any part of the diaphragm to the nearest channel a small fraction of a wave length. These channels are so proportioned that the sound waves coming through them have an amplitude and phase relation such that a substantially plane wave is formed at the throat of the horn.

In the appendix it is shown that, for the higher frequencies where the impedance of the horn may be taken as equal to ρc times the throat area and for the type of structure adopted, the radiation resistance is equal to

$$\rho c a T^2 \left[\frac{1}{k^2 h^2 T^2 + k^2 l^2 \cot^2 kl} \right] \quad (6)$$

and the reactance

$$-j \frac{\rho c a}{k h} T \left[1 - \frac{1}{kl \cot kl + \left(\frac{h T}{l} \right)^2 kl \tan kl} \right] \quad (7)$$

where a is the area of the throat of the horn, T the ratio of the area of the diaphragm to the throat area, $k = \frac{\omega}{c}$, and the other designations are those indicated in Fig. 11. At the lower frequencies the resistance is $T^2 r$ and the reactance $T^2 x$ where r and x are, respectively, the resistance and reactance of the throat of the horn.

Equation 6 shows that at a given frequency, other conditions remaining the same, the radiation resistance will have a maximum value when l is approximately equal to $\frac{\pi}{2k} = \frac{c}{4f}$. In Fig. 6 the resistances as computed from eq 6 are plotted as a function of frequency for several values of $\frac{h}{w}$. It is seen from these curves that the resistance at the higher frequencies is determined very largely by the relation

of $\frac{h}{w}$ but is independent of it at the lower frequencies,

where it is equal to $\rho c a T^2$. At the lower frequencies where the mechanical impedance of the diaphragm is negligible, the efficiency, as was the case for the low frequency receiver, depends upon the value of $B^2 v$ where v is the volume of the coil, but at the higher frequencies the efficiency decreases with increasing mass of the coil. It is advantageous, therefore, to keep v small and to make B as large as is practically possible. Values were selected to give the receiver an efficiency of 55 per cent at the lower frequencies. For these conditions the relative sound power output was computed by eq 2 on the assumption that the receiver was connected to an amplifier having an output impedance equal to 0.45 times that of the receiver at the lower frequencies. Figure 7 shows the values so obtained. Corresponding values obtained experimentally when the receiver was connected to the horn previously described are shown in Figs. 8 and 9, where the sizes of the rooms in which the values were obtained were, respectively, 5,000 and 100,000 cu ft. Both of these curves differ considerably from the computed curve, particularly as regards loss at high frequencies. The curve of Fig. 8 shows less, and that of Fig. 9 more, loss at high frequencies. The computed curve, however, refers to the total sound

Fig. 9. Output-frequency characteristic of combined low and high frequency receivers as measured in a large room

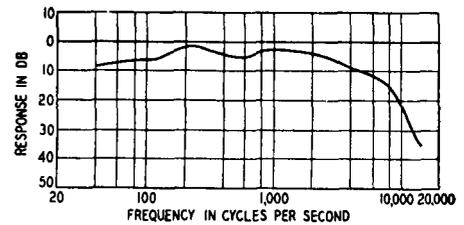
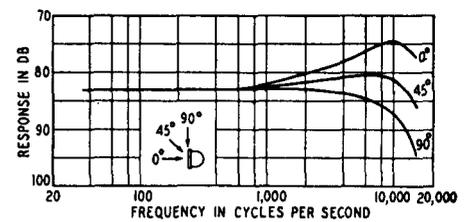


Fig. 10. Output-frequency characteristic of moving coil microphone



output, whereas the measured curves give average values of sound intensity in a certain part of the room, values dependent upon the acoustic characteristics of the room.

The number of high frequency receivers that must be used for each transmitting channel is governed largely by the amount of power that the system is to deliver before harmonics of an objectionable intensity are introduced. The generation of harmonics in a horn when transmitting waves of large amplitude already has been discussed. Let it suffice here to say that, for a given percentage harmonic distortion, the power that can be transmitted through the horn is proportional to the area of the throat and inversely proportional to the square of the ratio of the frequency to the cut-off frequency.

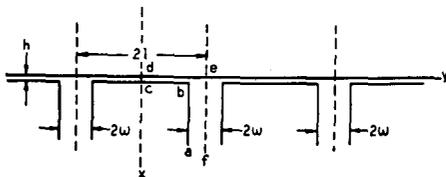
Inasmuch as the moving coil microphones used for the transmission of music in acoustic perspective have been described previously⁸ they will not be discussed here at length. Their frequency response characteristic as measured in an open sound field for several different angles of incidence of the sound wave on the diaphragm are shown in Fig. 10 where it is seen that the response at the higher frequencies becomes less as the angle of incidence is increased. In general, this is not a desirable property, but with the instruments as used in this experiment the sound observed as coming from each loud speaker is mainly that which is picked up directly in front of each microphone; sound waves incident at a large angle do not contribute much.

At certain times the sound delivered by the orchestra is of very low intensity. Therefore it is important that the microphones have a sensitivity as great as possible, so that the resistance and amplifier noises may readily be kept down to a relatively low value. At 1,000 cps these microphones, without an amplifier, will deliver to a transmission line 0.05 microwatt when actuated by a sound wave having an intensity of 1 microwatt per sq cm. This sensitivity is believed to be greater than that of microphones of other types having comparable frequency response characteristics, with the possible exception of the carbon microphone.

Appendix—Load Impedance of a Diaphragm Near a Parallel Wall With Slot Openings

First assume a diaphragm and a parallel wall of infinite extent separated by a distance h , and that the wall is slotted by a series of equally spaced openings as shown in Fig. 11. From symmetry it

Fig. 11. Schematic diagram of diaphragm and parallel slotted wall of infinite length



is known that when the diaphragm vibrates there will be no flow perpendicular to the plane of the paper or across the planes indicated by the dotted lines. Therefore only one portion of unit width, such as $abcdef$ need be considered. Let the x and y reference axes be located as shown. If the general field equation

$$\frac{\partial^2 \varphi}{\partial x^2} + \frac{\partial^2 \varphi}{\partial y^2} + k^2 \varphi = \xi \quad (8)$$

is applied when the diaphragm has a normal velocity equal $\xi e^{i\omega t}$ the following boundary conditions are obtained:

$$\begin{aligned} \text{When } x = 0, \quad \frac{\partial \varphi}{\partial x} &= -\xi \\ x = h, \quad \frac{\partial \varphi}{\partial x} &= 0 \\ y = 0, \quad \frac{\partial \varphi}{\partial y} &= 0 \end{aligned}$$

and when $y = l$, the pressure is equal to the product of acoustic impedance and volume velocity or

$$\frac{\rho}{h} \int_0^h \left(\frac{\partial \varphi}{\partial t} \right)_{y=l} dx = \frac{c\rho}{w} \int_0^h \left(\frac{\partial \varphi}{\partial y} \right)_{y=l} dx$$

where φ is the velocity potential, $k = \frac{\omega}{c}$, and c is the velocity of sound.

The appropriate solution of eq 8 then is

$$\varphi = \frac{\xi}{k} \left[\frac{\cos ky}{kh \left(\cos kl + i \frac{h}{w} \sin kl \right)} - \frac{\cos k(x-h)}{\sin kh} \right]$$

The average reacting force per unit area of the diaphragm is

$$\frac{ik\rho c}{l} \int_0^l (\varphi)_{x=0} dy$$

Thus, for the impedance per unit area, which is equal to the force divided by the velocity, is obtained

$$\frac{\rho cl}{w} \left\{ \frac{\frac{\sin^2 kl}{k^2 l^2} \frac{1}{\cos^2 kl + \left(\frac{h}{w}\right)^2 \sin^2 kl}}{\frac{kh \cos kh}{\sin kh} kl - \frac{\sin kl \cos kl}{\cos^2 kl + \left(\frac{h}{w}\right)^2 \sin^2 kl}} \right\} \equiv r' + jx'$$

In all practical types of loud speakers $\frac{kh \cos kh}{\sin kh}$ would be very nearly equal to 1; then

$$\begin{aligned} r' &= \frac{\rho cl}{w} \left[\frac{1}{k^2 l^2 \left(\left(\frac{h}{w}\right)^2 + \cot^2 kl \right)} \right] \\ x' &= -\frac{\rho cl}{h} \left[\frac{kl - \frac{1}{\cot kl + \left(\frac{h}{w}\right)^2 \tan kl}}{k^2 l^2} \right] \end{aligned}$$

If the total area of the diaphragm is A and that of the corresponding channels a , then $\frac{A}{a} = \frac{l}{w}$, approximately, and the total impedance becomes

$$\begin{aligned} &= \frac{\rho c A^2}{a} \cdot \frac{1}{\left(\frac{kh}{a}\right)^2 A^2 + k^2 l^2 \cot^2 kl} \\ x &= -j \frac{\rho c A}{kh} \left[1 - \frac{1}{kl \cot kl + \left(\frac{hA}{l}\right)^2 kl \tan kl} \right] \end{aligned}$$

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Auditory Perspective

—Amplifiers

Appreciable care is required in the design of a system which must amplify with great fidelity practically the whole range of audible frequencies and be capable of delivering a high level while at the same time providing a wide volume range. Some of the problems involved are discussed, particularly as applying to the equipment used in the reproduction in Washington, D. C., of the Philadelphia Symphony Orchestra playing in Philadelphia. This is the fourth paper in this symposium.

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VACUUM TUBE AMPLIFIERS have been closely identified with the extension of the channels of communication since, with completion of the initial transcontinental telephone line 20 years ago, they first enabled New York to converse with San Francisco. There are now thousands of audio frequency amplifiers in telephone circuits and in sound picture theaters, public address systems, and other similar services as well as in the millions of radio receiving sets.

Along with the extension of the field of usefulness of audio amplifiers there has been continuing progress toward more faithful reproduction, better transmitters, better receivers, and better amplifiers. Those first telephone repeaters, although quite adequate for their immediate purpose, transmitted a frequency band only a few octaves wide. Very few radio sets even now cover a range above 3,000 cps without distortion, and the most up-to-date sound picture installation rarely can be depended upon for accurate reproduction of frequencies above 7,000 or 8,000 cps. The requirements as to frequency range and freedom from distortion for any particular service are, in the last analysis, determined by public demand.

However, when one undertakes to reproduce an orchestra like the Philadelphia Symphony and to reproduce it in such a manner as to satisfy the critical ear of the director, or that of the devotee of sym-

phonic concerts, one has to provide something out of the ordinary in audio amplifiers.

In his paper, which forms a part of this symposium, Dr. Fletcher has pointed out that "only the elimination of those frequencies below 40 cps and those above 15,000 cps produces no detectable difference in the reproduction of symphonic music." This, then, is the frequency spectrum that the amplifier must be designed to handle. Also, it is important that there shall be uniform amplification of all parts of the frequency range and that no extraneous frequencies shall be introduced.

Of importance commensurate with the distortionless amplification of the complete frequency range of the orchestra is the provision of an equivalent volume of sound. The amplifier must be capable of supplying to the loud speakers without distortion an amount of energy that will produce a sound volume at least equivalent to that produced by the orchestra (the Philadelphia-Washington installation was designed to produce about 10 times this amount). And equally important, the amplifier must be so free from internal disturbances and from self-induced electrical fluctuations that the softest music, the weakest input to the microphone, can be reproduced without appreciable background noise. According to Fletcher the ratio of the heaviest playing of a large orchestra such as the Philadelphia Symphony Orchestra to the softest music such as that of a violin is about 10,000,000 to 1, or 70 db. Thus it is required that any noise be at least 75 db below the loudest tones; that is, there must be at least a 75-db volume range.

The sources of noise may be divided into 2 groups. In the first group are included the 60-cycle alternating current power supply, vibration or jar of mechanically unstable vacuum tubes, contact and thermoelectric potentials, and similar disturbances, which may be reduced to practically any degree depending upon the lengths to which one is willing to go to reduce them. In the second group are those electronic irregularities intimately associated with the operation of the vacuum tube and which depend somewhat upon the design, manufacture, and method of operation of the vacuum tube; and which, when sufficiently amplified and fed into a loud speaker, may be heard as noise. In general, the maximum volume range of an amplifier is reached when all other disturbances are reduced to the level of this tube noise.

It is evident, then, that under ordinary circumstances the limiting volume range of an amplifier is a function of the amount of amplification following the first tube. In other words, the magnitude of the signal voltage with respect to the noise voltage in the plate circuit of the first tube in a multistage amplifier determines the limiting volume range obtainable with that amplifier.

It will appear that in a sound reproduction system a highly efficient microphone simplifies the amplifier volume range requirements, and that loud speakers of high efficiency reduce the volume required from the amplifier.

Perhaps it is in order to inquire as to what makes an amplifier free from frequency distortion over a

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wide range. The answer might well be: attention to impedance relations. A compact, efficient amplifier requires several pieces of reactive apparatus such as transformers, retardation coils, and capacitors. One must remember that an inductance of one henry is equivalent to an impedance of 250 ohms at 40 cps but that it is nearly 100,000 ohms at 15,000 cps; that the grid circuit of the vacuum tube is not actually an open circuit even though the grid is maintained negative with respect to the cathode, but has a reactance which becomes important at high frequencies or with large ratio input transformers. Many years of development in this field have advanced the art to the point where transformers transmitting extremely wide bands now can be designed. The commercial production of such designs requires rigid inspection including shop transmission measurements under the actual conditions of use. The transformer must be designed for the particular type of vacuum tube with which it is to be used. First, however, the tube must be designed to permit its use under the proposed conditions and then it must be manufactured to close limits, every tube of a type like every other tube of that type.

This is, then, the general requirement for a wide frequency range amplifier: (1) attention to impedance relations; (2) meticulous design of each component for the particular job it has to do, and rigid inspection to insure that it does that job.

One might suppose that when the tube designer and the coil designer each had done his part the job was done. Such is not the case. The various pieces of apparatus have to be gathered together into a unit (often a current supply set for supplying anode, cathode, and grid potential is assembled with the

amplifier) and out of this electrical and physical association is apt to arise "feed-back" and "noise."

When there is coupling between 2 parts of the amplification circuit which are at different potential or different phase there is feed-back. Feed-back sometimes is employed designedly to modify an amplifier characteristic, but, feed-back which may arise as a result of a particular arrangement of apparatus or wiring ordinarily will cause more or less severe frequency distortion. It may be induced due to stray electromagnetic or electrostatic fields, which must be eliminated by rearrangement of apparatus or by shielding; or it may be caused by common circuit impedance, requiring circuit modifications. In general, a low gain amplifier or one with limited frequency range presents no feed-back problems, but a study of a high-gain wide-range equipment usually is necessary in order to determine the best arrangement. Often modifications of tentative circuit or apparatus must be made to obtain satisfactory operation.

The provision of a volume range of some 75 db on an energy basis became largely a matter of the suppression of a-c hum. The low inherent electronic noise effect of the Western Electric No. 262A vacuum tube and the relatively high level from the microphones kept electronic tube noise well in the background. Careful and in some cases rather elaborate shielding of audio transformers and leads and the segregation of the 60-cycle power equipment coupled with the use of vacuum tubes having indirectly heated cathodes and specially designed to have small stray fields prevented a-c hum trouble in the early stages. However, the Western Electric No. 242A vacuum tubes used in the push-pull final

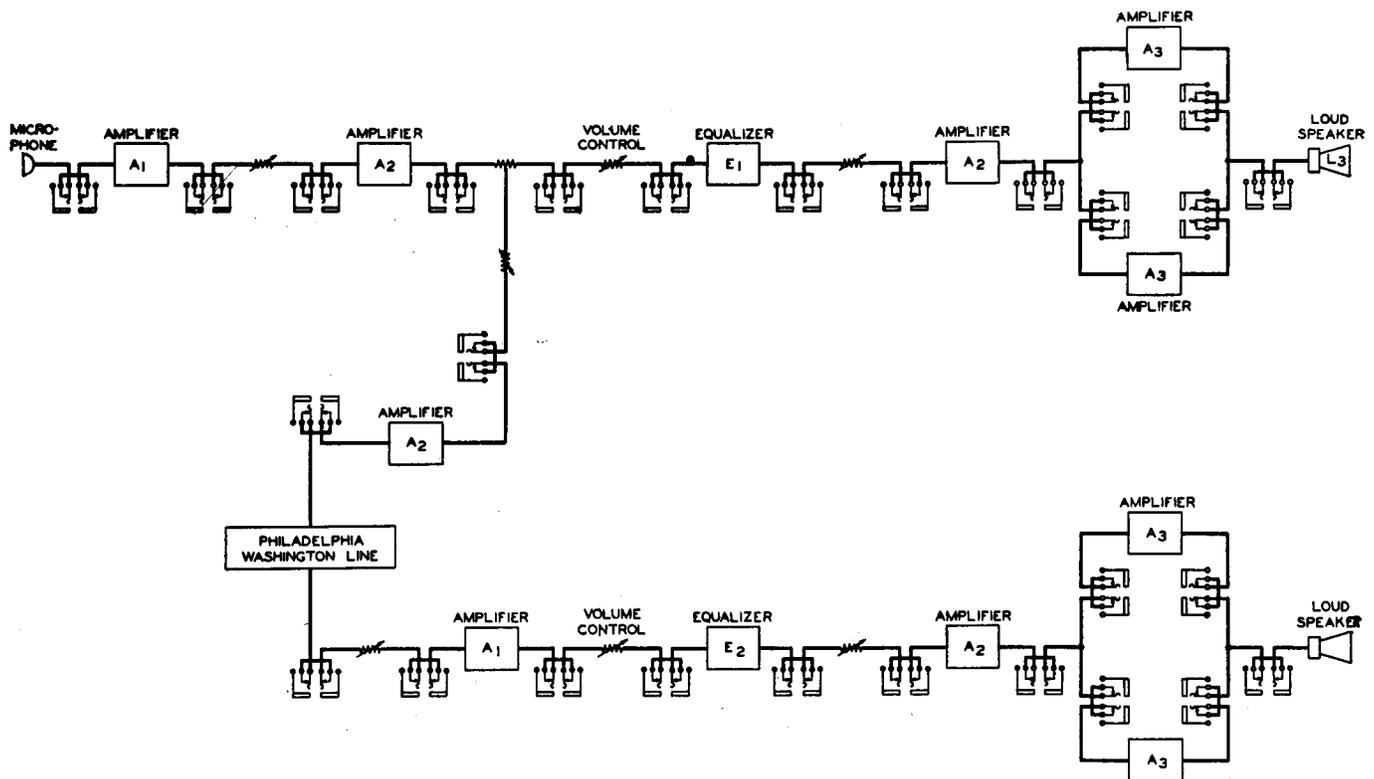


Fig. 1. Schematic diagram of the amplification system used in conveying Philadelphia symphonic music to Constitution Hall in Washington, D. C., and there reproducing it in auditory perspective

stage have filamentary cathodes, and when such tubes have raw a-c filament supply, a very appreciable 120-cycle component appears in the space current. Although theoretically in a perfectly balanced push-pull amplifier this component would be eliminated, in practice an exact balance cannot be obtained. As a final step in noise elimination, advantage was taken of the fact that each channel employed 2 amplifiers in parallel. Under such conditions and with proper phasing of the power supply to the 2 amplifiers the net a-c noise output of the 2 amplifiers in parallel will be less than that of either one alone.

Having reduced feed-back and noise to tolerable values, it remains to determine the operating conditions for maximum output. The vacuum tube is not strictly a linear device, but, when properly used, the total harmonic content can be held to a low figure. For a high quality system the total harmonics produced in the system should not exceed one per cent of the fundamental. This requires that impedance and potential relations in the vacuum tubes should be adjusted to give approximately linear operation; and also that the design of audio transformers, particularly those carrying considerable levels, must be scrutinized carefully to insure that they operate over an essentially linear portion of the magnetization curve of the core material.

An instrument really essential to the design of high quality amplifiers is a high sensitivity harmonic analyzer that is capable of quickly and accurately resolving a complex wave into its simple components. By this means the effect of variations in circuit relations can be evaluated and the optimum condition for maximum distortionless power output determined.

It may be desirable at this point to examine the make-up of the audio amplification system used in the Philadelphia-Washington experiments. It should be noted that the arrangement of equipment provided for simultaneous reproduction at both Philadelphia and Washington. There were 3 complete and essentially equivalent channels of equipment actually in use and a fourth complete channel held in reserve as a spare.

Several stages of so-called voltage amplification were required preliminary to the final or power stage. There is, of course, no essential difference between a voltage amplifier and a power amplifier, the term "voltage amplifier" being applied to those preliminary stages of an amplification system the function of which is so to amplify the output of the pick-up device as to supply adequate driving voltage to the grids of the power stage. Theoretically, inasmuch as no energy is absorbed in the ideal grid circuit, this voltage increase might be supplied entirely by a high ratio input transformer. However, there are practical difficulties to the design of such a single stage amplifier and therefore multistage vacuum tube amplification is employed.

As a matter of convenience the voltage amplifica-

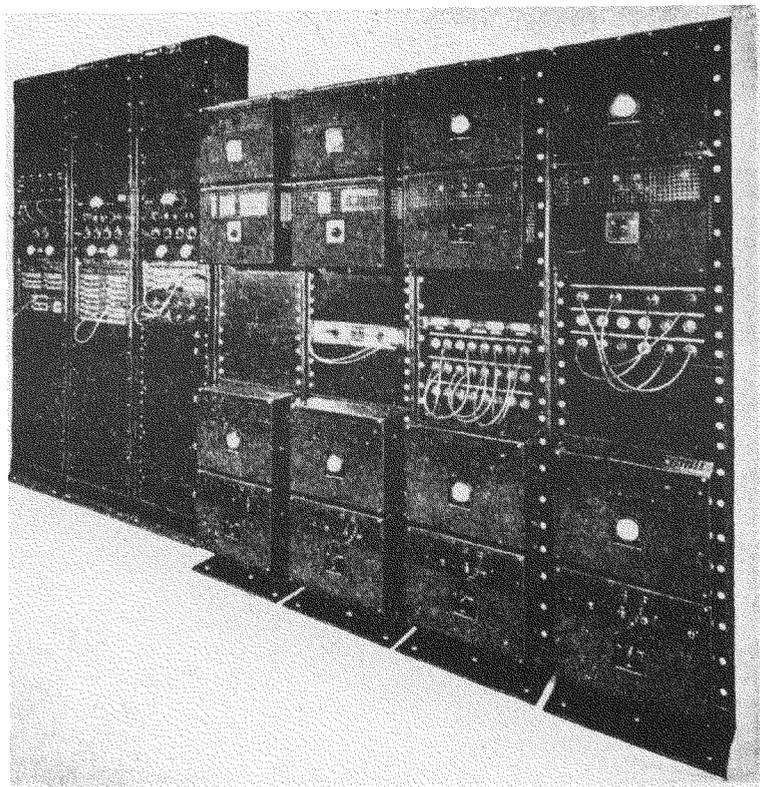


Fig. 2. Amplifying equipment used at Philadelphia. The taller racks are 8 ft high and contain A_1 and A_2 amplifiers, volume indicators, and various controls

tion for this system was obtained through the use of several separate amplifier units in tandem. This arrangement not only enabled the ready replacement of any unit of the system in case of failure, but it also facilitated the insertion of a pad, control potentiometer, or other network at any desired point. Several of these devices were required, and of course each introduced a loss. Thus the gross amplification of the system used for reproduction at Philadelphia was approximately 160 db and for Washington 240 db, although the actual difference in level between microphone output and loud speaker input was but from 80 to 90 db.

The general scheme of the amplification system is shown in Fig. 1. A_1 is a single-stage, single-tube Western Electric No. 80A amplifier slightly modified to meet the particular conditions of use; it has a gain of 30 db, and employs a Western Electric No. 262A vacuum tube. This tube has an equipotential cathode, the heater being operated on 10-volt 60-cycle alternating current and the anode being supplied from rectified alternating current. A_2 is a 2-stage amplifier having a single Western Electric No. 262A vacuum tube in the first stage and push-pull Western Electric No. 272A tubes in the second stage. It has a gain of 50 db. The cathodes of the tubes are energized with low-voltage 60-cycle alternating current and the anodes with rectified alternating current. A_3 , the final or power amplifier, is a single stage amplifier employing 2 Western Electric No. 242A vacuum tubes in parallel on each side of a push-pull circuit, thus having 4 tubes per amplifier.

Two of the A_3 amplifiers were used in parallel on each channel, and were capable of supplying 60 watts each, or a total of 120 watts, to the loud speakers. These are rms values. The instantaneous peaks of power of course could equal twice this value, or 720 watts, for the 3 channels. E_1 and E_2 are equalizers to compensate for any amplitude distortion that would cause a listener to obtain a different tone effect from the loud speakers than he would from the actual orchestra. These equalizers are loss networks and principally equalize for the acoustic characteristic of the loud speakers in the particular

hall, but they are placed in a low energy part of the amplification circuit so as not to waste the energy of the final power stage.

In view of the inclusion of the equalizers in the amplification system, and particularly because of the fact that the amplification of the A_3 amplifier deliberately was made to increase with frequency in order to compensate in part for acoustic losses in the overall system, the actual amplification-frequency curves of the amplifiers are of little importance. The equalizers of the system are discussed in a paper by Bedell and Kerney. (See page 216.)

Auditory Perspective —Transmission Lines

Describing methods whereby high quality sound reproduction in auditory perspective can be accomplished over long distances, this discussion centers largely upon a description of the exact technique employed in providing communication transmission circuits for the Philadelphia-Washington demonstration. Problems that might be involved in carrying out such transmission on a more widespread scale also are touched upon. This is the fifth paper in this symposium.

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MICROPHONES have been described that will pick up without noticeable distortion all the sounds given forth by a symphony orchestra. Loud speakers and amplifiers also have been described that will accurately reproduce this highest quality music in its full range of tone quality and

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volume. Therefore, the situation obviously requires connecting transmission paths so perfect in their characteristics that reproduction 100 or 200 miles away may not suffer in comparison with reproduction which may be only 100 or 200 ft from the source of music.

There are several respects in which a long line circuit possibly may distort the speech or music passed over it, unless considerable effort is expended to overcome these tendencies. For example, there may be frequency-amplitude distortion; i. e., all the notes and overtones may not be transmitted with the proper relative volumes. Similarly there may be phase or delay distortion, the different frequencies may not arrive at the receiving end of the line circuit in the same time relationships in which they originated. A line circuit is subject also to possible inductive disturbances from other communication circuits ("crosstalk"), or from power or miscellaneous circuits which cause "noise" at the receiving terminal. If the circuit contains amplifiers, transformers, and inductances having magnetic cores, it is subject to possible nonlinearity effects; i. e., the current at the receiving end of the line may not follow exactly the amplitude variations of the current applied to the transmitting end or, what is more important, spurious intermodulation frequencies may be generated within the transmission circuit and mar the purity of the musical tones. The problem of reproduction in auditory perspective, using 2 or 3 paralleling channels, also adds the requirement that these channels must be substantially identical in their transmission characteristics.

With the exception of the last, all these aspects of the problem are, of course, not peculiar to symphony music transmission. They exist as part of the problem of satisfactorily transmitting any telephone message. However, the requirements of this new high quality transmission have set a new high standard of refinement, even as compared with that required for ordinary radio chain broadcasting. For example, ordinary telephone message transmission commonly is carried out by circuits having a frequency range not exceeding 200 to 3,000 cycles per second. Much present-day radio broadcasting involves a transmission band only from about 100 to 5,000 cps. This new high quality transmission,

however, requires a range from approximately 40 to 15,000 cps. Further, with reference to the required freedom from interference, ordinary radio broadcasting seldom exceeds a volume range greater than 30 decibels. The new high-quality system, however, requires a volume range of at least 65 db, which is more than 3,000,000 to 1 expressed as a power ratio.

In considering the specific problem of transmitting from Philadelphia to Washington for the demonstration given on April 27, 1933, several alternative methods of providing the required transmission paths presented themselves. The arrangement chosen consisted in bridging the distance between the 2 cities by means of carrier channels over cable conductors. From the telephone toll office in Philadelphia to the toll office in Washington, 3 carrier transmission paths were provided in which the music frequencies were stepped up from their normal position in the audible range to considerably higher frequencies. The frequency range from 40 to 15,000 cps picked up by the microphones was transmitted over line circuits in a range from 25,000 to 40,000 cps. After being thus stepped up in frequency, the high frequency currents were applied to 3 nonloaded pairs in an all-underground cable which was equipped with repeaters at approximately 25-mile intervals. At Washington, step-down or demodulation apparatus restored the frequencies to their normal position in the spectrum.

For transmission between the auditorium in Philadelphia and the toll office there, a distance of approximately 3 miles, and for transmission in Washington between the telephone toll office and the auditorium, about half this distance, normal frequency transmission over small-gauge pairs in ordinary exchange cables was employed.

The use of the carrier method for the long distance transmission has several advantages. In general, it permits multiplex operation; i. e., more than one message or program on the same pair of wires. As a matter of expediency in this particular case this feature of operation was not used, and 3 separate pairs were employed, one for each channel. In the future the same technique undoubtedly would permit 2 or possibly more of these extra-broad-band transmission paths to be obtained on the same pair of conductors. The most important reason for choosing the carrier method rather than transmission in the natural audio-frequency range in this particular case was that, because all other transmission circuits

in the same cable were at a considerably lower frequency and because the lead sheath of the cable acts efficiently at the high frequencies to shield the pairs from induced disturbances from the outside, it offered a special freedom from crosstalk and noise.

With these arrangements, which will be described in somewhat greater detail in what follows, requirements of transmission were met very satisfactorily and the reproduction of the symphony music in Washington with the orchestra playing in Philadelphia suffered not the least in comparison with the reproduction of the same program in an auditorium in Philadelphia located but a few feet from the hall in which the orchestra played. It is believed that, if necessary, by the use of the same principles, line circuits may be set up and comparable quality reproduction given throughout the country. However, as will be evident from part of the discussion which follows, in some respects the problem of meeting the requirements in transmission between Philadelphia and Washington was not as difficult as might be encountered in other localities. Hence even more complex arrangements might be necessary if it were desired to establish such transmission circuits to other points, and particularly for greater distances.

LINE CIRCUITS

There are several all-underground cables between Philadelphia and Washington. As described in a paper¹ by Clark and Green given before the A.I.E.E. in 1930, recently laid cables contain several 16-gauge conductors distributed throughout the cross section of the cable for possible use as program circuits in chain broadcasting. These pairs, however, ordinarily are loaded and equipped with repeaters at approximately 50-mile intervals so that they transmit a frequency range up to about 8,000 cps.

In one of the cables several pairs of this type had not yet been loaded, and these pairs were used for this newer transmission. Because of the higher frequencies employed and the greater attenuation encountered, it was necessary to install repeaters at more frequent intervals. As may be noted in Fig. 1, the normal cable layout between Philadelphia and Washington includes 2 intermediate repeater stations, one at Elkton and one at Baltimore. Additional repeater stations were established accordingly at in-between points—Holly Oak, Abingdon, and Laurel. One of these repeater points, Holly Oak,

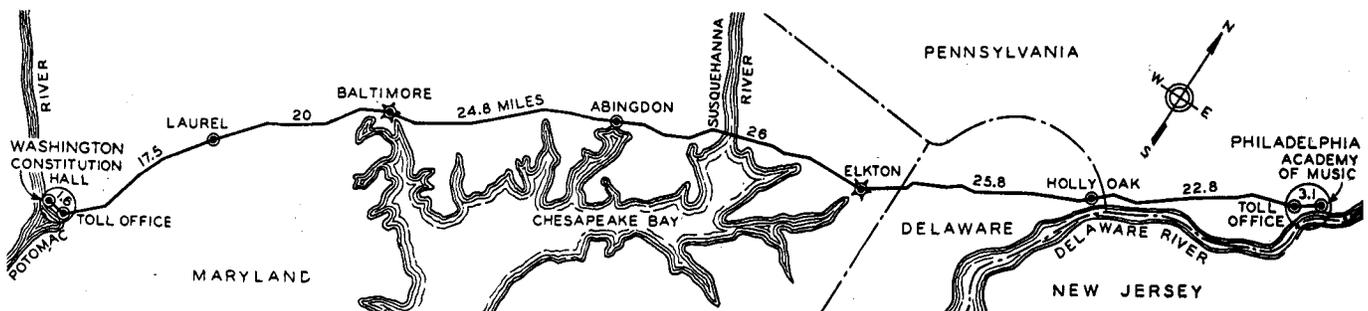
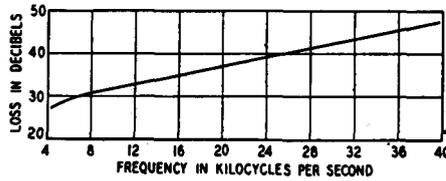


Fig. 1. Geographical layout of 3-circuit communication line used to carry Philadelphia symphony music to Washington, D. C., for reproduction in auditory perspective

was established in a local telephone office. No such convenient housing existed at the other 2 points, and it was necessary to establish new repeater stations. These were small metal structures large enough to house only the repeaters, their power supply, and

Fig. 2. Line attenuation characteristic of typical repeater section



testing equipment. This apparatus was arranged to be normally nonattended, various switching actions being remotely controlled from the nearest regular repeater station.

The line attenuation between repeater points is shown in Fig. 2. It may be noted that the attenuation is approximately 50 db for the highest carrier frequency involved. A diagram showing the variations in power level as the carrier waves traverse the complete circuit is shown in Fig. 3. Because of the variation in attenuation over the frequency range employed it was necessary, of course, to use equalizers at the input of each repeater; i. e., networks having an attenuation variation with frequency approximately the inverse of that of the line circuit.

NOISE

In setting up these circuits various tests, including measurements of noise currents picked up by the conductors to be employed, were made prior to the actual installation of the apparatus. It was discovered that on the cable circuits north of Baltimore these pairs were picking up sufficient noise even at the higher frequencies to constitute a possible limitation in the volume range that might be delivered. This noise was generated chiefly as a by-product of relay and other similar operations within the Baltimore office and was propagated over the longitudinal circuits of various pairs in the cable from which, by induction, it entered the special selected pairs. As a remedial measure, longitudinally acting choke coils applied to all but the specially selected pairs in the cable greatly reduced the noise. Shielding and physical separation were employed in the Baltimore office to prevent induction between the repeaters and the connection to the main cable. If it is desired to use existing cables for carrier transmission, particularly for such high grade transmission circuits, it seems likely that filtering arrangements of this kind, or other precautions, generally will be required.

CARRIER APPARATUS

The carrier system employed may be characterized briefly as single-sideband carrier-suppressed, with perfectly synchronized carrier frequencies of 40,000 cps. Most present-day commercial telephone carrier systems are of the single-sideband carrier-suppressed type. Suppressing one sideband saves frequency

space and suppressing the carrier reduces the load on the line amplifiers or repeaters. Ordinarily the exact synchronization of the carrier frequencies at the sending and receiving ends is not required for message telephone service.

Obtaining a single sideband after modulation commonly is carried out by providing band filters which transmit the desired sideband and suppress the unwanted sideband. For the requirements of message telephone transmission this does not impose severe requirements in the design of filters because audio frequencies less than about 100-200 cps ordinarily are not transmitted, in which case, if the filter in suppressing the unwanted sideband tends to cut off the lower frequencies of the desired sideband, it is not important.

For the requirements of this new high quality system, however, where it was desired to transmit

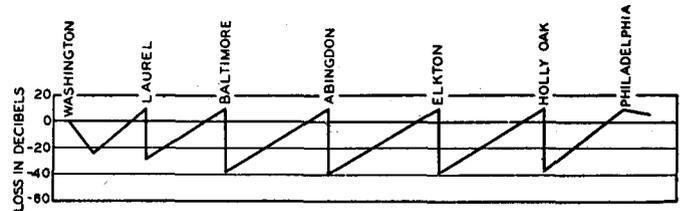


Fig. 3. Transmission level diagram

all frequencies to at least as low as 40 cps, the problem was considerably more difficult. Two alternatives presented themselves in the design of the required filters. The first consisted in attempting to provide the required selectivity in the filters themselves, perhaps supplementing the actions of inductance coils and condensers (which normally make up such a filter structure) by quartz crystals to provide the sharp selectivity required on the sides of the band. The other alternative consisted in providing a filter of moderate selectivity so that in the neighborhood of the carrier frequency the unwanted sideband is not completely suppressed, and in arranging that the resultant reproduced music at the receiving terminal is obtained by the proper coordination of the desired and the vestige of the unwanted sideband. The "vestigial" sideband method was decided upon. Although this does not require filters having particularly sharp selectivity on the sides of the band, it does, however, impose more severe requirements upon the control of the phase characteristics of the filters in the neighborhood of the carrier frequency. It makes it necessary also to have the carrier frequencies at the sending and receiving ends not only synchronized, but phase controlled as described later.

For the modulating elements in the system at both the sending and receiving terminals, copper oxide rectifying disks were chosen. These elements can be made very simple. In stability, with respect to transmission loss and the ability to suppress the unwanted carrier frequency by balanced circuits, this arrangement is superior to the usual vacuum tube circuits.

In Fig. 4 is shown schematically the arrangements

of the carrier circuit at the transmitting and receiving ends. At the transmitting terminal the circuit from the microphones is led first through low- and high-pass filters to limit the bands to the desired width; i. e., 40 cps to 15,000 cps. The 40-cycle limiting filter was included because tests had demonstrated that lower frequencies are not required for the satisfactory transmission of music of symphony character, and because it was feared that occasional high energy pulses of subaudible frequency might cause overloading. When these 40-cycle filters are omitted, as was done in tests, the carrier channels are capable of transmitting frequencies down to and including zero frequency, a characteristic which could not possibly be obtained in a single sideband system by other than such a vestigial sideband technique.

As may be noted further in Fig. 4, carrier current is supplied to the rectifying disks of the modulator along with the incoming music frequencies. The balancing connection of the 4 rectifying disks making up the modulator is arranged to suppress the carrier frequency, the final degree of suppression being adjusted by means of the variable condenser and resistance shown, which were included to make up for slight dissimilarities in the characteristics of the individual copper disks. A very high degree of carrier suppression can be achieved by this means. No difficulty was experienced in maintaining a ratio of at least 60 db between the carrier voltage applied to the unit and the residual carrier current not completely balanced out. Over short periods an even higher degree of balance can be readily obtained.

There is a certain amount of electrical noise generated in the rectifying disks over and above that caused by thermal agitation^{2,3} effects. The amount of this noise compared with the maximum permissible modulation output determines the volume range possibilities of a modulator of this type. Measurements indicated that this range was approximately 90 db, which obviously was more than sufficient to meet the requirements desired.

The circuit includes a relay and a meter through both of which flows the d-c component produced by the rectification of the carrier frequency. These supplementary units give a check on the magnitude of the carrier supply and afford an alarm in case of failure. From the modulator unit the circuit is connected to the band filter which transmits only the lower sideband lying between approximately 25,000 and 40,000 cps and the vestige of the upper sideband. From the band filter the currents are led to an amplifier and thence to the line circuit leading to the farther terminal.

It may be noted that at the transmitting terminal the 40,000-cycle carrier current is derived from a 20,000-cycle oscillator by passing its output through a series of copper oxide rectifiers connected to form a frequency doubler. Part of the originally generated 20,000 cycles also is connected to the input of the transmitting amplifier and sent over the line to be used in producing the 40,000-cycle carrier supply for demodulation.

At the receiving terminal a similar modulation or demodulation process occurs through the use of copper oxide disk circuits. A relay and meter also are included in the circuit to check the carrier supply,

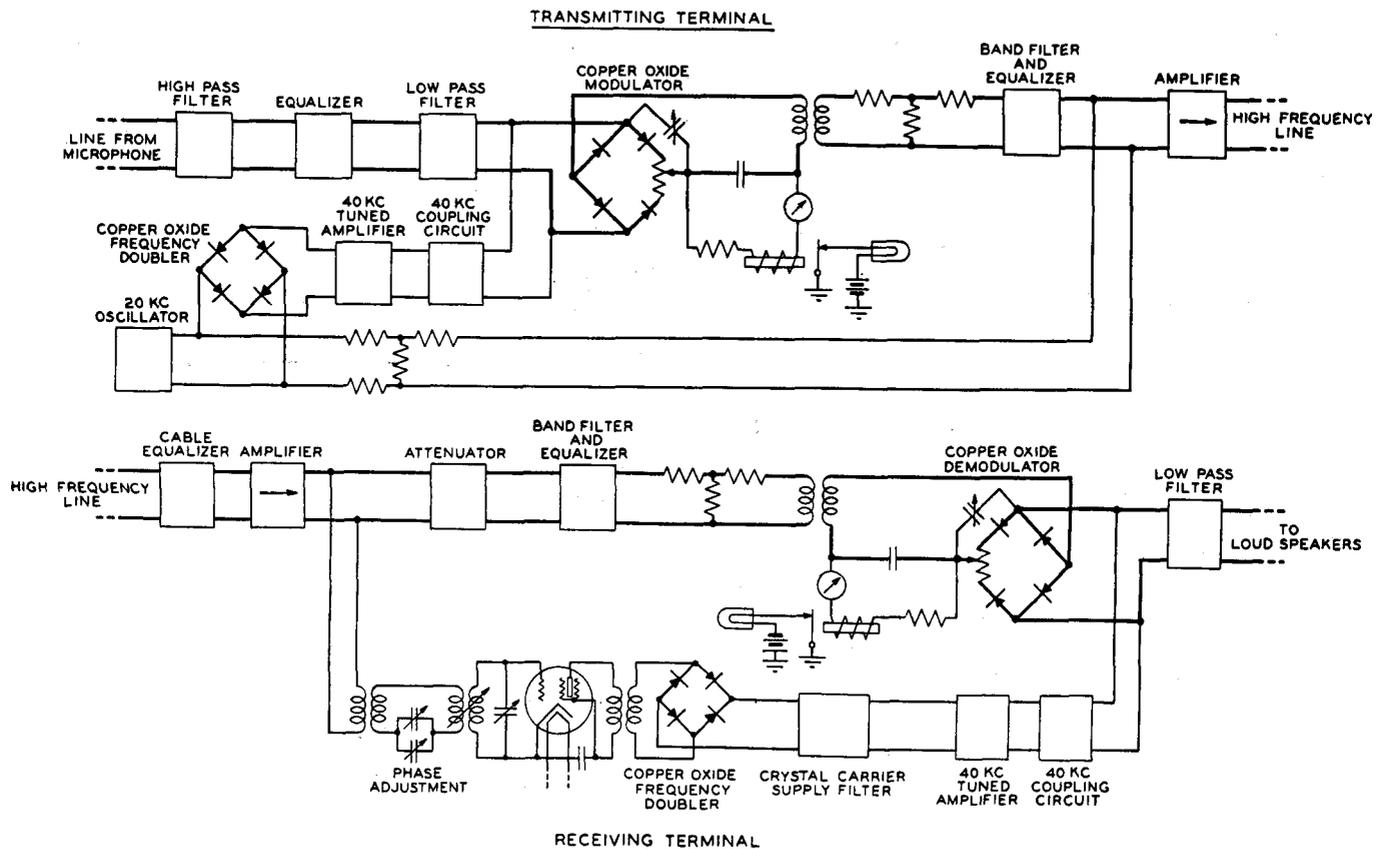


Fig. 4. Schematic diagram of carrier terminal circuits

in this case providing also a check or pilot of the transmission over the long line circuit. The 20,000-cycle synchronizing current is selected at the receiving terminal, amplified and applied to a frequency doubler, and thence applied to the demodulator circuit. The input of this carrier supply circuit includes also a phase adjusting variable condenser arrangement so that the phase of the carrier supplied to the demodulator may be adjusted properly in relation to that of the carrier supplied to the modulator at the sending end. An interesting feature of the receiving terminal carrier supply is the quartz crystal filter employed to select the 40,000-

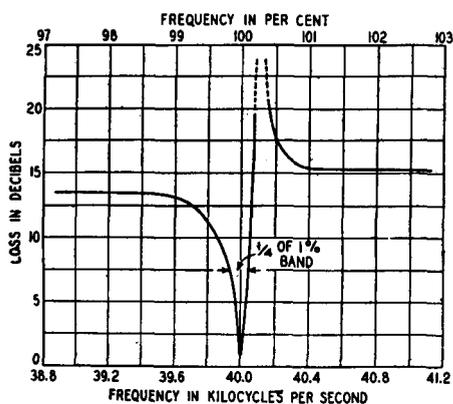


Fig. 5. Transmission characteristic of carrier supply crystal filter

cycle carrier after frequency doubling. The transmission characteristic of this extremely selective filter is shown in Fig. 5.

FILTERS

The transmission characteristics of the carrier channels are determined largely by the filters and associated equalizers. The filters principally affect-

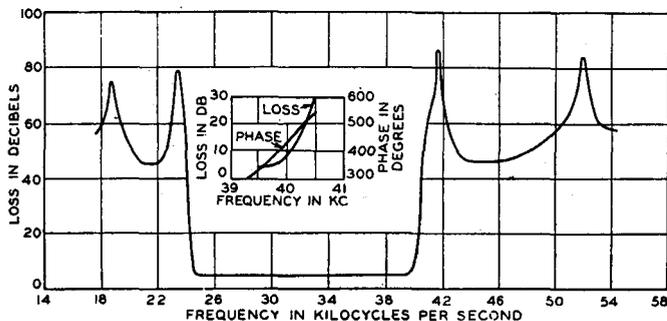


Fig. 6. Transmission and phase characteristics of band filter

ing transmission are the band filters. Identical units are employed at the sending and receiving ends. The transmission and phase shift characteristics of one of these units are shown in Fig. 6. These band filters are equalized to produce the desired squared band characteristic.

The characteristics in the frequency region near the carrier (i. e., at 40,000 cycles) are shown on a large scale. This region is of particular interest

because it is here that the degree of success in the application of the vestigial sideband method, for the purpose of insuring the satisfactory transmission of the low music frequencies, is determined. If for a given frequency interval above the carrier the phase change is arranged to be equal and opposite to that of the same frequency interval below the carrier, then in the action of demodulation the demodulated current produced by the action of one sideband adds

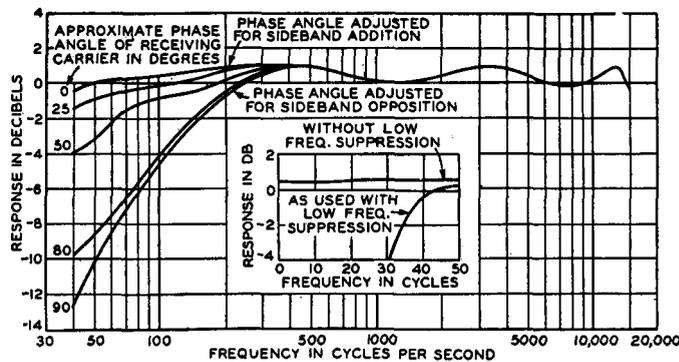


Fig. 7. Over-all transmission characteristics as a function of the phase relation of the receiving carrier

itself arithmetically to that produced by the other sideband. It will be noted that this desirable phase characteristic has been achieved closely in the characteristics shown. If, in addition, the attenuation loss in the filter is adjusted so that the sum of the regular and vestigial sideband amplitudes corresponding to the low music frequencies is substantially constant and equal to the amplitude of the frequencies at midband, the desired flat transmission characteristic is assured.

As was noted previously, this action can be carried out only if the phase angle of the receiving carrier is properly related to that of the sideband frequencies and, in turn, to the carrier applied to the modulator at the transmitting terminal. The curves shown in Fig. 7 illustrate the influence that the phase adjustment of the carrier frequency has on the transmission of the lower frequencies in a system of this kind.

The upper curve shows the transmission frequency characteristic of one of the carrier channels measured from terminal to terminal between distortionless lines, when the phase angle of the receiving carrier is adjusted for its optimum value. Under these conditions the vestigial sideband and normal sideband supplement each other in their effects to produce substantially flat transmission. (The insert indicates the sustained transmission toward zero frequency when the 40-cycle highpass filter is omitted from the circuit.) It may be noted also that with this proper phase adjustment the full band transmission characteristic provided is sub-

This paper continued on p. 214. The sixth and final paper in this symposium entitled "Auditory Perspective—System Adaptation" follows the remainder of this paper, beginning on p. 216.

Auditory Perspective —Transmission Lines

This paper continued from p. 32. The sixth and final paper in this symposium entitled "Auditory Perspective—System Adaptation" follows the remainder of this paper, beginning on p. 216.

stantially flat within a fraction of a decibel from 40 cps to 15,000 cps. The lower curves indicate successively what happens if the phase angle of the receiving carrier is adjusted different amounts from the optimum adjustment. It may be noted that for a 90-deg departure the transmission of a 40-cycle tone over the carrier channel would suffer more than 12 db in comparison with a 1,000-cycle tone.

REPEATERS

As noted previously, the line circuit between Philadelphia and Washington included 5 intermediate repeater points. A schematic drawing of

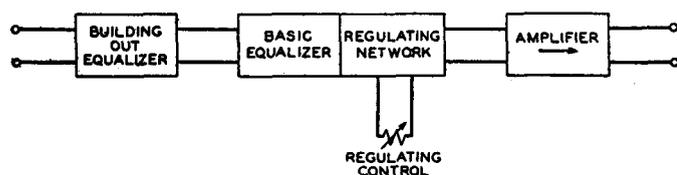


Fig. 8. Schematic diagram of repeater station apparatus

the apparatus installed at each point is shown in Fig. 8. The amplifiers at these points, as well as those used at the transmitting and receiving terminal, consisted of a new form of amplifier employing the principle of negative feed-back. The principal virtues of amplifiers of this type are their remarkable stability with battery and tube variations and great freedom from nonlinearity or modulation effects. Each amplifier is supplemented at its input by an equalizer designed to have its attenuation approximately complementary in loss to that of the line circuit in a single section. The amplifiers actually employed for the purpose were taken from a trial of a cable carrier system described in a recent A.I.E.E. paper by A. B. Clark and B. W. Kendall.⁴

The losses in the cable circuits do not, of course, remain absolutely constant with time, and slow variations due to change of temperature are compensated for by occasional adjustments of the variable equalizer arrangements provided. These adjustments were required only infrequently; approximately at weekly intervals because in an underground cable the temperature experiences only slow, seasonal variations.

As noted, new repeater stations were established at 2 points. The housing arrangements for one of these points, Abingdon, is shown in Fig. 9. The equipment at this repeater point also included relays remotely controlled from the nearest attended repeater station to permit the repeaters to be turned on and off at will and the power supply, which consisted of storage batteries, to be switched from the

regular to the reserve battery or either battery put on charge if required.

OVER-ALL PERFORMANCE

While the system was set up specifically to provide transmission for the demonstration into Washington on April 27, 1933, it was operated over a period of several weeks and complete tests and measurements were carried out for the purpose of gathering information on cable carrier systems. The complete layout of apparatus and lines provided between Philadelphia and Washington is shown in Fig. 10.

The over-all frequency transmission characteristics of the 3 channels that were set up are shown in Fig. 11. These curves differ from those shown in Fig. 7, and include the complete high frequency line circuit with its 150 miles of cable, repeaters, equalizers, and other equipment. It may be seen that between the desired frequency limits the circuit is substantially flat in transmission performance to within ± 1 db. Various noise measurements made on the over-all circuit indicated that the circuits fully met the requirements that had been set up, and that the line and apparatus noise was inaudible in the auditorium at Washington even during the weakest music passages. The circuit also was found to be free from nonlinear distortion to a satisfactory degree. Harmonic components generated when single-frequency tones were applied to the channels at high volumes were found with one unimportant exception to be more than 40 db below the fundamental.

As a means of obtaining a further increase in volume range, which was not actually required for this demonstration, tests were made with a so-called predistortion-restoring technique. In this the higher frequency components of the music were transmitted over the carrier channels at a volume much higher than normal in relation to the volume of the lower frequencies. By this means any noise entering the carrier channels at frequencies equivalent to the higher music frequencies is greatly minimized in effect.

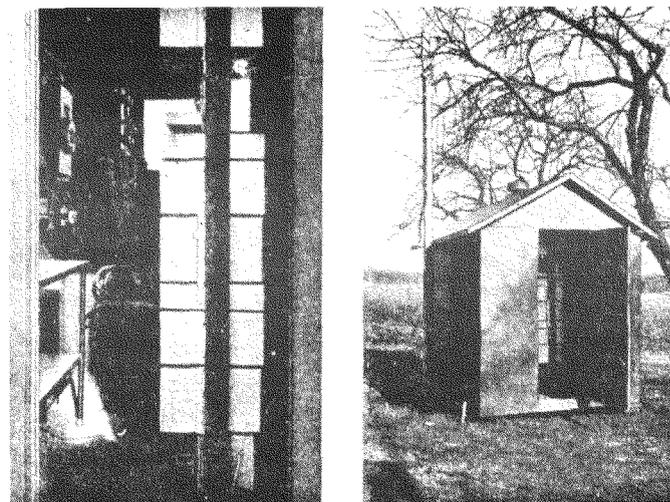


Fig. 9. Interior and exterior of special intermediate repeater station at Abingdon, Md.

Fig. 11 (right). Frequency characteristics over carrier channels used between Philadelphia and Washington, D. C.

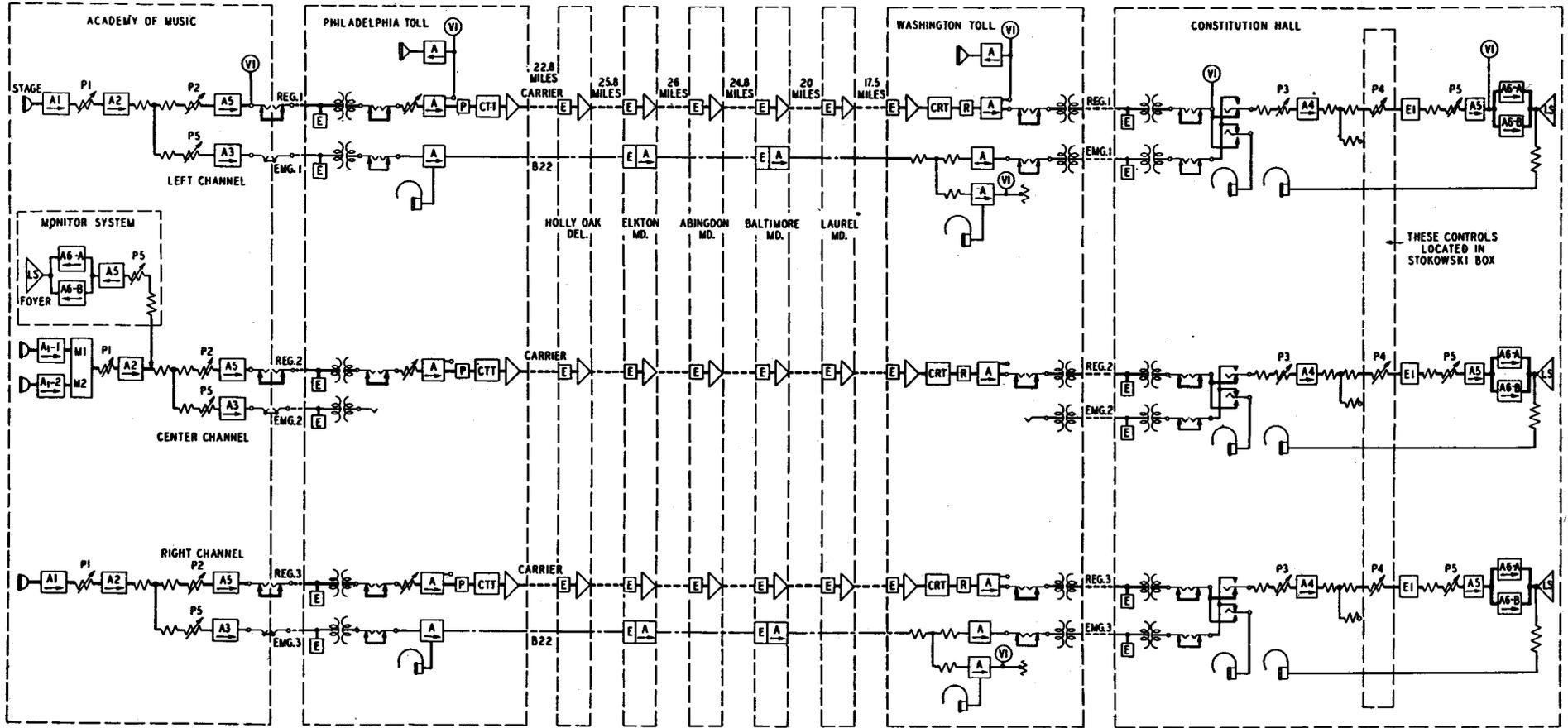
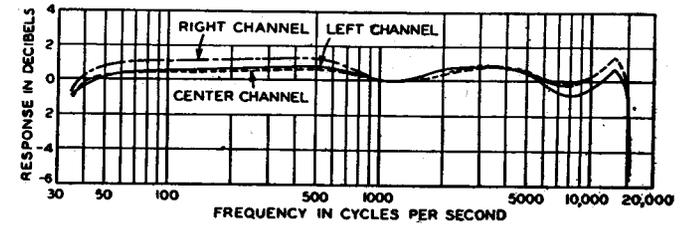


Fig. 10. Schematic diagram of circuit layout for 15-kc channel used for symphonic program demonstration

- SYMBOLS —
- 40-15,000 CYCLE AMPLIFIER
 - CARRIER AMPLIFIER
 - VOLUME INDICATOR
 - MICROPHONE
 - CARRIER TRANSMITTING TERMINAL
 - PRESTORING AND RESTORING NETWORKS
 - VARIABLE ATTENUATORS OR POTENTIOMETERS
 - TRANSFORMER
 - MONITORING HEAD RECEIVER
 - MIXING PANEL
 - CARRIER RECEIVING TERMINAL

This predistortion is accomplished by including in the circuit at the input to the modulator a network having relatively high loss for the lower frequencies and tapering to low loss for the higher frequencies. Its maximum loss is compensated for by adding in the circuit an equivalent amount of additional amplification. The characteristics of such a network are illustrated in Fig. 12. To restore the normal volume relationships between the different tones and over-tones a restoring network having complementary transmission frequency characteristics is, of course, included at the output of the receiving circuit. It was found with this predistortion-restoring technique that a volume range increase of something like 10 db could be obtained over the circuits described.

There is available also another method which might have been employed for obtaining a further increase in volume range. This method, the so-called volume compression-expansion system, very likely will be necessary if in the future it is desired to obtain such high quality circuits on long routes where the carrier frequency range is being used also for regular telephone message transmission or for other purposes, and where the problem of freedom from noise and crosstalk no doubt will be more serious than experienced in the Philadelphia-Washington demonstration. Such a volume compression-expansion system requires additional apparatus at the sending and receiving terminals of the line circuit. At the sending end this apparatus is used to raise in volume the weak passages of the music or other

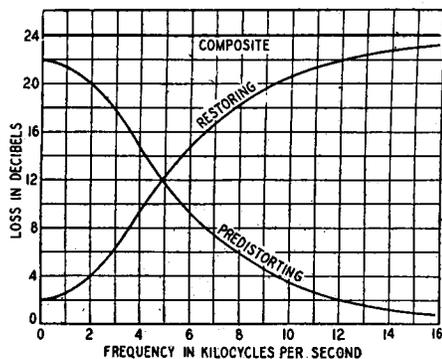


Fig. 12. Attenuation characteristics of "predistorting and restoring" networks

program for transmission over the line circuits in order that the proper ratio between the desired program and unwanted noises may be retained. At the receiving terminal coördinating apparatus reexpands the compressed volume range to the volume range originally applied to the transmitting terminal.

In the demonstration, to provide supplementary control features required by Dr. Stokowski at Washington for communicating with the orchestra at Philadelphia, additional wire circuits were established between these points. Order wire circuits also were provided for communication between the terminals and repeater points to make possible the location troubles if any should arise. Rather elaborate switching means were included at the

terminals to permit switching the carrier channels to different microphones and to different amplifier equipment at the loud speaker end. To take care of the contingency of a cable pair failure, spare pairs of wires were made available to be switched in at short notice. Fortunately, none of the reserve facilities actually were required for the demonstration.

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Auditory Perspective —System Adaptation

A communication system for the pick-up and reproduction in auditory perspective of symphonic music must be designed properly with respect to the acoustics of the pick-up auditorium and the concert hall involved. The reverberation times and sound distribution in the two auditoriums, the location of the microphones and loud speakers, and the response-frequency calibration of the system and its equalization are considered. These and other important factors entering into the problem are treated in this, the sixth and final paper of the symposium.

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WHEN THE EFFECT of music or the intelligibility of speech is spoiled by bad acoustics in an auditorium, the audience is well aware that acoustics do play a most important part

in the appreciation of the program. One may not be conscious of this fact when the acoustical conditions are good, but a simple illustration will show that the effect still is present. Thus, of the sound energy reaching a member of the audience as much as 90 per cent may have been reflected one or more times from the various surfaces of the room, and only 10 per cent received directly from the source of the sound.

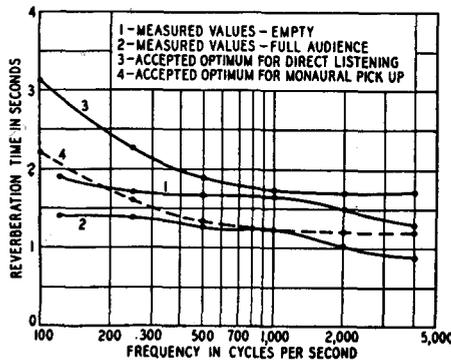


Fig. 1. Reverberation characteristics of Academy of Music, Philadelphia, Pa.

In listening to reproduced sound in an auditorium or concert hall, the effect of the room acoustics is perhaps even more important, for in this case the audience does not see any one on the stage and must rely entirely upon the auditory effect to create the illusion of the presence there of an individual or a group. Imperfections in the reproduced sound that are caused by defects in the acoustics of the auditorium may destroy the illusion and be ascribed improperly to the reproducing system itself.

In some types of reproduced sound, radio broadcast for example, where the reproduction normally takes place in a small room, the attempt is made to create the illusion that the listener is present at the source.^{1,2} In the case considered here, however, where symphonic music is reproduced in a large auditorium, the ideal is to create the illusion that the orchestra is present in the auditorium with the audience. Since the orchestra is playing in one large room and the music is heard in another, the acoustical conditions prevailing in both must be considered.

PICK-UP CONDITIONS

The source room is the auditorium of the American Academy of Music in Philadelphia. This room has a volume of approximately 700,000 cu ft, and a seating capacity of 3,000. Measured reverberation time curves for this auditorium, and preferred values^{3,4} for a room of this volume, are given in Fig. 1. It may be seen that with a full audience this room might be considered somewhat dead, but would be considered generally satisfactory for pick-up either with or without an audience. A floor plan of the Academy

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auditorium and stage, showing the location of the 3 microphones used, is given in Fig. 2. The microphone positions were selected after judgment tests using several locations and are much nearer the orchestra than they would be for single channel pick-up.² The use of the microphones near the orchestra results in picking up a high ratio of direct to reverberant sound and thus reduces the effect of reverberation in the source room upon the reproduced music. A high ratio of direct sound is desirable in the present case also because of the use of 3 channels. The perspective effect obtained with 3 channels depends to a considerable extent upon the relative loudness at the 3 microphones, and since the change in loudness with increasing distance from the source is marked for the direct sound only, and not for the reverberant, there would be a definite loss in perspective effect if the microphones were placed at a greater distance from the orchestra. This effect is discussed more fully in another paper of this symposium.

With the microphones located close to the orchestra their response-frequency characteristics will be essentially those given by the normal field calibration, since relatively little energy is received from the sides and back. For a distant microphone position it would be necessary to use the random incidence

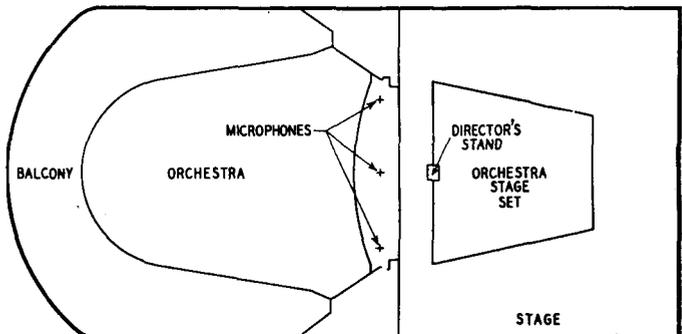


Fig. 2. Floor plan of Academy of Music, showing location of microphones

response characteristic, which differs from the normal because of the variation in directional selectivity of the microphones as the frequency varies. This difference in response characteristic depends upon the size of the microphone and may amount to as much as 10 db at 10,000 cps. It may be pointed out here that this difference in response is one factor frequently overlooked in the placement of microphones.

In addition to the 3 microphones regularly used, a fourth was provided to pick up the voice when a soloist accompanied the orchestra. In this case only the 2 side channels were used for the orchestra, the voice being transmitted and reproduced over the center channel. The solo microphone was so shielded by a directional baffle that it responded mainly to energy received from a rather small, solid angle. This arrangement permitted independent volume and quality control for the vocal and orchestral music.

The music was reproduced before the audience in Constitution Hall in Washington, D. C. This hall has a volume of nearly 1,000,000 cu ft, and a seating capacity of about 4,000. A floor plan of the auditorium showing the location of the loud speakers and of the control equipment is given in Fig. 3. The loud speakers are placed so that each of the 3 sets radiates into a solid angle including as nearly as possible all the seats of the auditorium. Figure 4 shows the reverberation-frequency characteristics of Constitution Hall. The values given by the curve for the empty hall were measured through the use of the 3 regular loud speakers and several microphone positions in the room. The values for the hall with

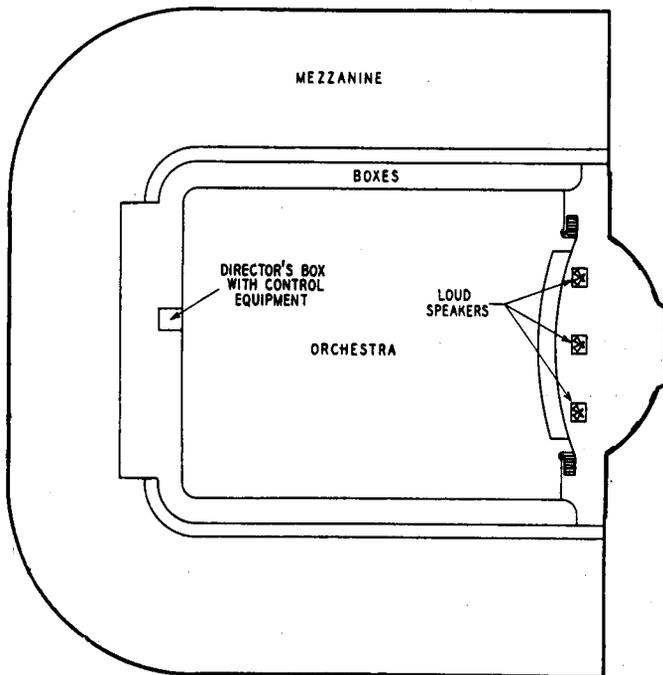


Fig. 3. Floor plan of Constitution Hall, Washington, D. C., showing locations of loud speakers

an audience present were calculated from known absorption data for an audience, and the optimum values are taken from accepted data for an auditorium of the volume of this one.³ The reverberation times were considered satisfactory and no attempt was made to change them for this demonstration. The reverberation time measurements for both Constitution Hall and the Academy of Music were made with the high speed level recorder.⁵ This instrument measures and plots on a moving paper chart a curve the ordinate of which is proportional to the logarithm of the electrical input furnished to it. When used in connection with a microphone for reverberation time measurements, curves are obtained showing the intensity of sound at the microphone during the period of sound decay. The rates of decay, and hence the reverberation times, are obtainable immediately from the slopes of these recorded curves and the speed of the paper chart.

In calibrating the system, a heterodyne oscillator connected to the loud speakers through the amplifiers was used. The oscillator was equipped with a motor drive to change the frequency, and as the frequency was varied through the range from 35 to 15,000 cps the sound was picked up with a microphone connected to the level recorder. Continuous curves of microphone response as a function of frequency thus were obtained for several positions in the auditorium, and for each channel independently. These response curves provided a check on a uniform coverage of the audience by each loud speaker, and also provided data for the design of the equalizing networks required to give an over-all flat response-frequency characteristic. If the system, including the air path from the loud speakers to one position in the auditorium, is made flat, it will not, in general, be flat for other positions or for other paths in the room. This variation in characteristic is due partly to the variation in the ratio of direct to reverberant sound, and partly to the fact that the sounds of higher frequency are absorbed more rapidly by the air during transmission.^{6,7} This latter effect is of considerable importance; it depends upon the humidity and temperature of the air, and may cause a loss of more than 10 db in the high frequencies at the more distant positions in a large auditorium. Some compromise in the amount of equalization employed therefore is necessary. Probably the most straightforward procedure would be to design the networks according to the response curves obtained with the microphones near the loud speakers. This would insure that for both the response measurements and the pick-up the microphone characteristics would be the same, and any deviation from a uniform response in the microphones would be corrected for in this way, along with variations in the loud speaker output. This procedure was modified somewhat for the case under discussion,

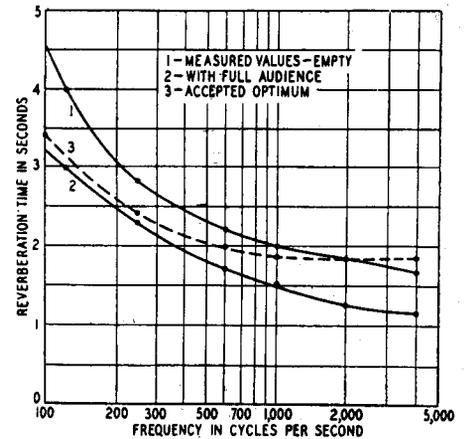


Fig. 4. Reverberation characteristics of Constitution Hall

however, because by far the greater portion of the audience was at a distance from the stage such that they received a relatively large ratio of reverberant sound, and it was believed that a better effect would

be achieved by equalizing the system characteristic in accordance with response measurements taken at some distance from the loud speakers.

CONTROL EQUIPMENT

In addition to the equalizing circuits used to obtain a uniform response characteristic, 2 sets of quality control networks which could be switched in or out of the 3 channels simultaneously were employed. One set modified the low frequencies as shown at A, B, and C of Fig. 5, while the other gave high frequency characteristics as shown at D, E, F, and G. These latter networks permitted the director to take advantage of the fact that the electrical transmission and reproduction of music permits the introduction of control of volume and quality which can be superimposed on the orchestral variations. Quality of sound can be divorced from loudness to a greater degree than is possible in the actual playing of instruments, and the quality can be varied while the loudness range is increased or decreased. Electrical transmission therefore not only enlarges the audience of the orchestra, but also enlarges the *capacity* of the orchestra for creating musical effects.

The quality control networks and their associated switches were mounted in a cabinet (Fig. 6) at the right side of the director's position. Continuously variable volume controls for the 3 channels were mounted on a common shaft and housed in the center cabinet of Fig. 6. A separate control for the center channel was provided when that was used for

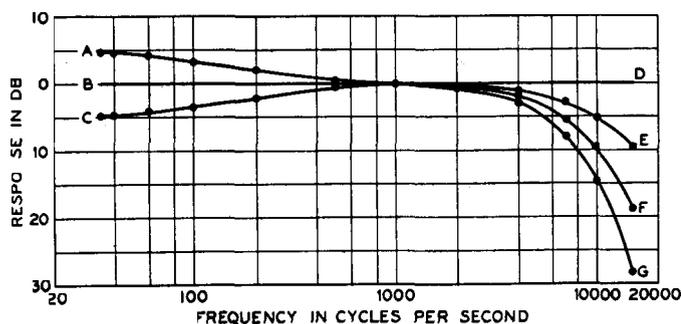


Fig. 5. Transmission characteristics of quality control networks used in the Philadelphia-Washington experiment

the soloist. In addition to the high quality channels certain auxiliary circuits were supplied to aid the smoothness of performance. Supplementing the order wire connecting all technical operators, a monitor circuit was provided in the reverse direction. The microphone was located on the cabinet before the director, and loud speakers were connected in the control rooms and on the stage with the orchestra, enabling the control operator to hear what went on in the auditorium and allowing the director to speak to the orchestra. Two useful signal circuits were employed; one giving the orchestra a "play" or "listen" signal, and at the same time connecting

either the auditorium or the orchestra's loud speakers, respectively; the other being a "tempo" signal to the assistant director leading the orchestra that could be operated during the rendition of the music. The switches for the auxiliary circuits and the order wire subset are shown at the control operator's position at the left in Fig. 6.

That a reproducing system may have quite different characteristics in different auditoriums is well illustrated in the case of the 2 halls considered here. From Fig. 3 it may be seen that in Constitution Hall the stage is built into the auditorium itself, and that

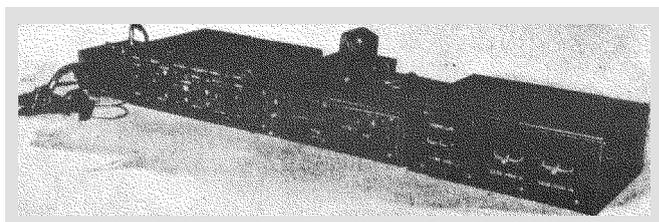


Fig. 6. Cabinets housing quality control networks and providing communication facilities for operation

there is no back stage space. The Academy of Music, however, has a large volume back stage. When the orchestra plays in the Academy the reflecting shell shown in Fig. 2 is used to concentrate the radiated sound energy toward the audience. When the reproducing system was set up in the Academy the shell could not be used because of the stage and lighting effects desired, and a large part of the energy radiated by the loud speakers at the low frequencies was lost back stage. The loss of low frequency energy is attributable partly to the fact that the loud speakers cannot well be made as directional for the very low frequencies as for the higher. The loss amounts to about 10 db at 35 cps, and becomes inappreciable at 300 cps or more, as measured in comparable locations in the 2 auditoriums. This difference in characteristics emphasizes the fact that for perfect reproduction the acoustics of the auditorium must be considered as a part of the system, and that in general the equalizing networks must have different characteristics for different auditoriums.

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