LIVENESS IN BROADCASTING

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When your friend tells you the news, do you prefer to have him sit comfortably in your home with you or to talk at you from a box? When you attend a concert, do you prefer to be in the audience and hear the sweep of the music through the hall, or to have the sound shot at you from a cabinet? Your radio can now bring your favorite newscaster, in living reality, to your home, or can transport you to the best seats in the concert hall. This article tells you how it is done.

Under normal conditions, you listen to an orchestra, a singer, or perhaps to someone telling you the latest news, with two ears. The binaural sense which results from the use of the two ears enables you to pay attention to the sound arriving from any desired direction and to partially exclude the sound from other directions. Similarly, you easily separate nearby sounds from the more distant ones. You have, therefore, two means of accentuating, at will, certain parts of the sound.

If, however, the sound has been picked up by one or more microphones, and reproduced through a single loudspeaker, your binaural ability to pay attention to the sound from any desired direction is completely lost. This results in an apparent increase in the "liveness" or reverberation present and also in the intensity of the incidental noises. However, your ability to distinguish between nearby and distant sounds is in no way impaired, but is frequently enhanced.[1,2]

Therefore, the situation may be summarized as follows:

(1) You have lost all ability to accentuate at will certain parts of the sound such as solo artist, by the help of the direction from which that particular sound comes.

(2) You still maintain your ability to accentuate by the distinction between nearby and distant sounds.

(3) The liveness, i.e., the apparent amount of reverberation has been automatically accentuated.

Any studio technique which is to reproduce life-like and realistic programs must (1) provide the studio engineer with a means of supplying the necessary accentuation lost by the failure of the binaural sense, (2) provide the engineer with means of making full use of the distinction between nearby and distant sounds, (3) eliminate the undesired accentuation of the apparent liveness.

This is particularly true, as the sense of realism experienced by the listener is as much dependent on the microphone placement and the studio acoustics as it is on his home conditions.

GREATER COVERAGE

The purpose of this article is to describe a technique of studio and auditorium sound pick-up which fulfills the above requirements and which places control of the desired accentuations on the dials of the studio mixer panel. Fortunately, the correction for the increased apparent reverberation can be accomplished by the initial placement of the microphones used.

One of the important advantages of this live type of pick-up is as much as 6 db gain in coverage at no extra expense to the sponsor or the broadcasting company.[3]

This unexpected gain is a result of the manner in which the ears of the listener perform. For a given power supplied to the loudspeaker, the loudness of a program picked up with this new technique can be 6 to 8 db greater than the loudness of programs from "dead" pick-ups. Since this gain in loudness permits the listener to operate his receiving set with a correspondingly lower electrical gain, static and other noises are reduced by this amount. Thus, this effect is a real gain in coverage.

In view of this apparently complicated situation, a search was made for some simple acoustic constant which would clarify the studio problems. Such a constant has been derived mathematically and checked by practical application to studio practice.

This constant is called *liveness* and represents the acoustic properties of an enclosed space, such as a studio or auditorium, including the effect of the distances from the artists to the pick-up microphones.[4] The properties of this constant are such that the formula can be readily applied to the use of one general or "over-all"

[1,2] Superior figures refer to Bibliography

microphone in combination with the necessary additional microphones for accentuation purposes.

The liveness formula is:

$$L = \frac{1000T^2D^2}{G_p V} \qquad (1)$$

where L = Liveness
T = Reverberation Time in Seconds
D = Distance from Sound to Microphone in Feet
V = Volume of Studio in Cubic Feet
G_p = Directivity of Pick-up Microphone from source to microphone.

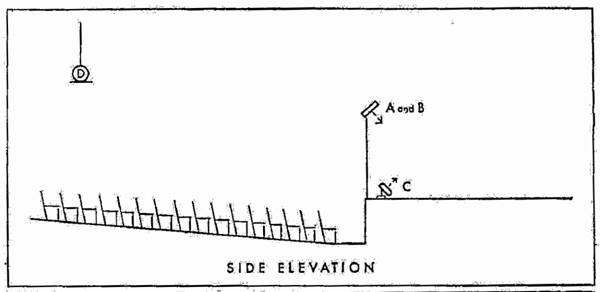
The value of T used for the practical application of this technique to broadcast pick-up is an average of the values over the frequency range from 500 cps, to 2000 cps. Where this average is unknown, the value at 1000 cps or even 500 cps may be used as a guide.

The range of the limits of liveness for satisfactory binaural listening is very great but is quite narrow for monaural or single channel reproduction. However, experience has shown that with single mike pick-up the limits of the value of liveness selected for best monaural pick-up always lie within those acceptable for direct listening. This means that in the concert hall, for example, the microphone position is always farther from the sound source than the front row of acceptable seats but nearer to the sound source than the rear row. The center of the monaural range is always closer to the source than that position generally rated as best for direct listening. This increased closeness of the monaural microphone automatically removes the accentuation of the apparent excess reverberation present.

The full useful range of liveness for monaural pick-up varies materially from one type of sound to another as, for instance, from symphony orchestra to solo singing to speech. Table 1 shows the values of the monaural liveness range for several different types of sound when picked up for reproduction in average living rooms. It may be of interest to know that where the reproducing space is abnormally live, both limits of the useful range are moved upward, not downward. Where the listening space is abnormally dead, the values must be decreased accordingly.

TABLE 1

Type of Sound	Liveness Range
Piano	4-16
Symphony Orchestra	5-20
Small Orchestra	3-12
Solo violin, cello, etc.	1- 4
Solo singing	1/2- 3
Speech	1/6-2/3



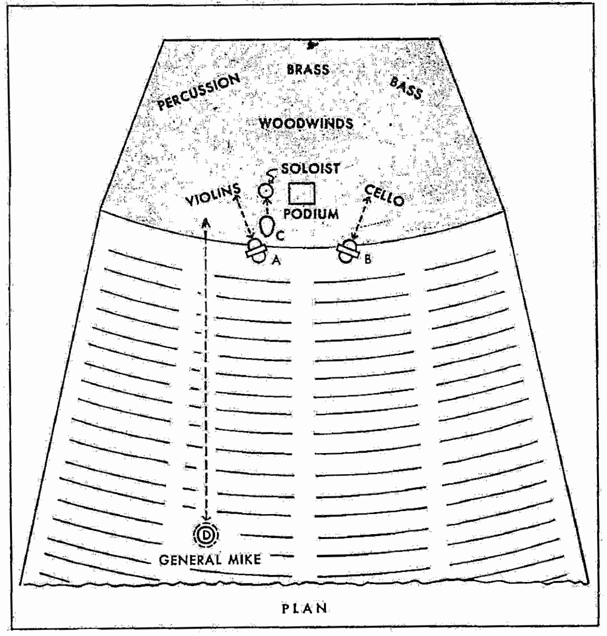


Figure 1--Large auditorium with a symphony orchestra. A and B, accentuation microphones; C, solo microphone; D, general microphone. Dotted lines show distances which can be computed as described elsewhere.

INCREASED SENSE OF REALITY

If sound is reproduced from a pick-up in which the liveness is controlled within the useful range as shown in Table 1, the subjective effect might be described as the acoustic recreation of the pick-up space around and behind the loudspeaker position. This effect adds greatly to a sense of reality and renders music or speech both natural and "easy to listen to." Under these circumstances, it is difficult to locate the position of the loudspeaker laterally, the sound appearing to flood in from behind it through an opening completely across the room. In other words, the effect is that of adding the studio space behind the plane of the loudspeaker without any intervening wall.

When the liveness is near the lower limit of the useful range, you get the impression that the sound is situated in the near end of this added space. In the case of a person speaking, there is the illusion of a real person speaking from the position of the loudspeaker.

When, however, the liveness is near the upper limit of the useful range, the source of sound appears to be considerably behind the plane of the loudspeaker as if it were coming to the hearer from a position in the remote end of the added space. In the case of broadcasting large symphony orchestras, this control of liveness enables one to so broadcast a concert that the listener in his home may seem to occupy any seat from the front to the back row of the auditorium. Since most auditoria have seats which music critics consider to be best, it is desirable to control the liveness of the broadcast so that the listeners are placed acoustically in that portion of the auditorium.

When pick-ups are made with a liveness value well below the useful range, this effect of added space disappears and one is aware of the sound being projected from the box containing the loudspeaker. Under these conditions, it has an artificial quality which could never be mistaken for the presence of a real person or a real orchestra. This effect might be called "absence" as opposed to the much desired "presence" of good broadcast pick-up.

Under these dead conditions, the lateral position of the loudspeaker can be accurately located by ear and the interpretation of quality is quite sensitive to one's position with respect to the high frequency beam of the loudspeaker and to the volume at which the sound is being reproduced.

On the other hand, when the liveness value is well above the upper limit of the useful range, one can again locate, with ease, the position of the loudspeaker. However, instead of feeling that the sound is being projected from a point source, the hearer experiences the effect of the sound reaching him through an open window from a room which is much more reverberant than the one in which he is listening.

Considerable evidence has been obtained that the public much prefers recordings made well within the useful range and in the case of orchestral music near its upper limit.

The advantages of this type of pick-up may be summarized as follows:

(1) The 6 db gain in coverage previously mentioned.

(2) When operating within the useful liveness range, the amount of manual volume control normally necessary with dead pick-up is markedly decreased without either overloading the equipment or causing the sound to sink into background noise, and therefore permits a higher average per cent modulation.

(3) For a given volume range as indicated by the vu meter, reproductions from monaural sound pick-ups made within the useful range have an apparent volume range nearly twice that of similar reproductions from dead pick-ups.

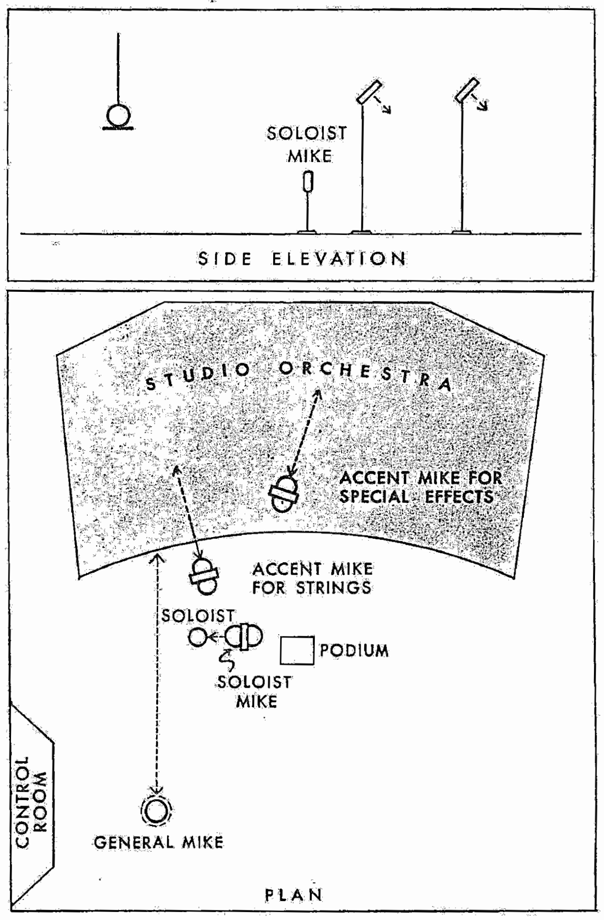


Figure 2--Plan and side elevation of normal studio, set up for orchestra with vocals. It should be noted that all of the Accentuation Microphones in this arrangement are not necessarily used at the same time.

(4) The change in quality of the monaurally reproduced sound as a function of the loudness of reproduction is materially reduced. This characteristic may be best illustrated by two contrasting cases.

Case 1--Assume that the sound from an orchestra, for instance, has been picked up under conditions of liveness well below the useful range and that this sound has been balanced for reproduction at an average intensity level of 75 db at the ear.

If this sound is now reproduced at an average ear level substantially lower than 75 db, marked distortion of the balance takes place. The lower notes and the high harmonics appear to be greatly attenuated. A similar effect in the reverse direction occurs if the sound is reproduced at a level substantially higher than that for which it was balanced. An equalizer introduced into the reproducing circuit will correct this unbalance if its characteristics correspond with the differences in the loudness contours of the Fletcher-Munson curves.[5]

Case 2--Assume that the sound referred to in Case 1 has been picked up under conditions of liveness well within the useful range and as before, has been balanced for reproduction at an average ear level of 75 db.

If this sound is reproduced at either an average ear level substantially lower or substantially higher than 75 db, very little if any apparent change of quality is noticeable. This advantage is of great value to the listening audience as it enables the listener to reproduce a sound in his living room at any desired level without a corresponding loss of quality.

TWO OR MORE MICROPHONES

The pick-up technique being described consists basically of the use of (1) a microphone situated at some distance from the performers to pick up the general blend of sound and (2) one or more accentuation microphones for accenting desired portions of the orchestra, soloists, etc.[6] This accentuation is obtained by controlling the liveness instead of the loudness.

The general microphone preferably has nondirectional characteristics as typified by the Western Electric 640 AA or the 633 Type. The accentuation microphones are usually of the bidirectional or of the cardioid type typified by the Western Electric 639 Type. Any high quality microphone, having the proper characteristics, will operate in an entirely satisfactory manner.

Figure 1 shows a qualitative arrangement for a symphonic broadcast with soloist, and some orchestral accentuation, while Figure 2 shows a typical studio set-up for orchestra with vocals. It should be realized, of course, that all of the accentuation microphones are not necessarily used simultaneously.

Arrangements such as these insure the fulfillment of the following requirements:

(1) Over-all liveness control is available to the sound engineer at all times during the broadcast.

(2) Accentuation control is similarly available at all times.

(3) The loss, by failure, of any one microphone does not render the pick-up unsuitable for broadcast.

(4) The arrangement is versatile and capable of rapid adjustment during rehearsal.

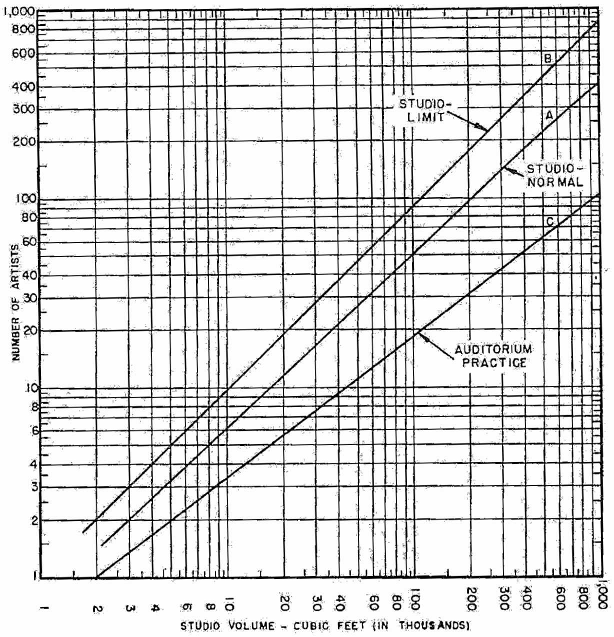


Figure 3--Relation between number of artists and studio size. A shows good studio practice; B maximum crowding without loss of realism; C, for comparison, shows auditorium conditions for symphonic music.

The employment of this technique requires studios with acoustic properties of a "pleasing" nature, i.e., studios of good acoustic properties.[7] It also requires that studios shall not be overcrowded.

Figure 3 shows the relation between the number of artists and the studio size.[8] Curve A represents good studio practice while Curve B represents the maximum crowding possible without loss of realism. Curve C is given for comparison only and represents auditorium conditions for symphonic music.

Figure 4 shows the optimum reverberation time with artists in place, as a function of studio or auditorium size.[4] Any values within 30 per cent of those shown can be compensated for by a proper choice of microphone positions.

PLACEMENT AND CONTROL OF MICROPHONES

A. Positioning the General or Over-all Microphone.

1. Choose from Table 1 the maximum value of liveness necessary for any part of the program. For instance, for a studio pick-up of a dance orchestra with vocals the maximum value of L is 12 for small orchestra.

2. Choose a value 1.5 times this (L = 18) as suitable for the overall microphone. The

increase of one and one-half is to allow you margin for leaving some accentuation microphones in circuit at all times without reducing the general liveness too much.

3. Determine the distance D from equation (2) below. D represents the distance of the microphones from the front of the orchestra.

Equation (1) may be solved for D and we get

$$D = \frac{\sqrt{L \times V \times G_p}}{31.6\ T} \qquad (2)$$

Where studios are in active use, a set of curves as shown in Figure 5 may be prepared. To aid you in preparing such a chart the following typical case is worked out in detail.

Assume a studio whose volume V is 30,000 cu. ft. and whose reverberation time T, with musicians in place, is 1.2 seconds. For nondirectional microphones $G_p = 1$ and for bidirectional or cardioid type $G_p = 3$ for sound sources on their beams. Assume a range of L from 0.3 to 30.

From equation (2) for a nondirectional microphone we get

$$D = \frac{\sqrt{30 \times 30{,}000 \times 1.0}}{31.6 \times 1.2}$$
$$= 24.6 \text{ ft. for } L = 30.$$

Similarly D = 2.46 ft. for L = 0.3

Plot these two points (A and B of Figure 5) and connect them with a straight line. From this chart the distance D corresponding to any desired value of liveness may be obtained for a nondirectional microphone.

To obtain the plot for bidirectional or cardioid microphones proceed in a similar manner letting $G_p = 3$. Then we obtain the points M and N, Figure 5. Connect these with a straight line.

This completed chart is now available for use in positioning the general microphone and, as described in the next section, for positioning the accentuation microphones also.

B. Positioning the Accentuation Microphones.

1. Choose from Table 1 the minimum liveness for the portion of the orchestra, the soloist or other source to be accentuated. For instance, for solo parts in the string section choose L = 1.0 (Solo Violin, etc.) or for a vocalist choose L = 1/2 (Solo Singing).

2. Choose a value which is two-thirds of that obtained from Table 1 as the practical operating liveness for the accentuation microphone. This decrease to two-thirds is to allow

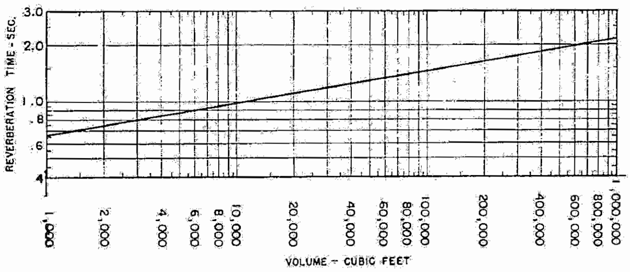


Figure 4--Curve showing optimum reverberation time with artists in place, as a function of studio or auditorium size. Any values within 30 per cent of above can be compensated for by microphone placement.

you margin for the increase in liveness due to the over-all microphone which is always in circuit.

3. Choose a suitable type of microphone, cardioid, bidirection or nondirectional. One of the directional types is usually preferred as the accentuation microphone, since it can be "beamed," i.e., partially limited to the sound sources on or adjacent to the high sensitivity axis.

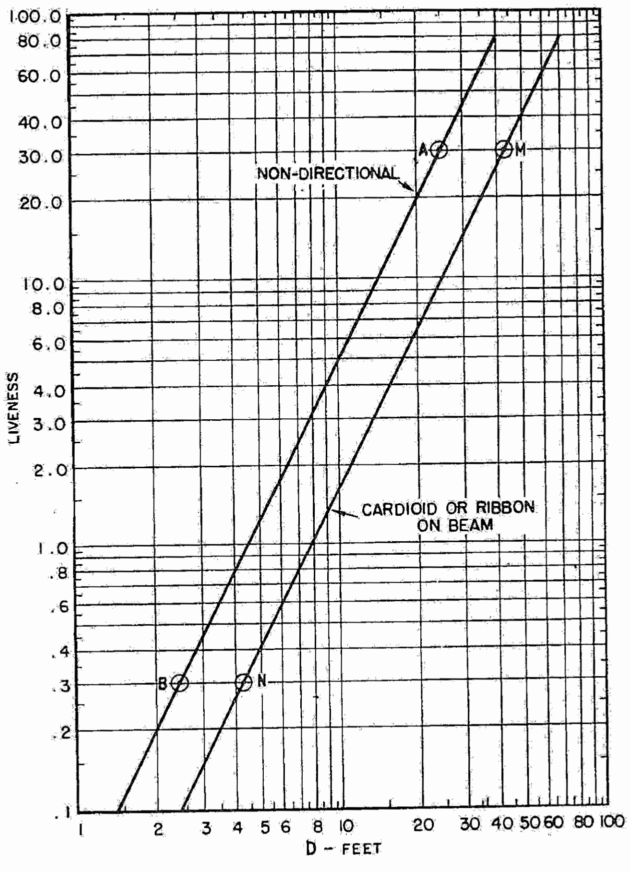


Figure 5--Typical chart used for positioning General and Accentuation Microphones in active studios.

4. Determine the distance D from Equation (2) or from the studio chart typified by Figure 5. If you are using this microphone for accentuation of the string section or any other group of artists, D represents the distance from the microphone to the nearest artist in that group.

5. Proceed in a similar manner for any other accentuation microphones which may be necessary.

C. Determination of Approximate Mixer Dial Settings.

No hard and fast rules can be given for control of the amount of accentuation necessary. This amount depends upon the type of program and upon the nature of the esthetic or dramatic illusion you are trying to create for the listener.

However, there are some general considerations which will help you acquire experience more rapidly than is possible with mere "cut and try" methods. In the first place, make it a rule to start your rehearsal with the general microphone only--all others being out of circuit. Then slowly fade in the accentuation microphones until the desired result has been obtained. When in doubt, use less accentuation than appears desirable over your monitor. This is due to the fact that most monitoring rooms are both smaller and acoustically more dead than the average living room.

The details of this mixing technique are described in Appendix 1. General adherence to the methods outlined there will result in the production of acceptable programs with very little rehearsal.

IMPORTANT APPLICATIONS

There are three broad classes of programs to which this technique has been successfully applied: namely, (1) Large concert hall pick-up, such as symphony orchestra, opera, choral singing, etc., (2) Studio music programs with or without vocals, (3) Speech only, such as news, lectures, announcements, etc.

1. *The Large Concert Hall Type of Program.* The most pleasing broadcast of a symphonic or operatic program is the one which creates for the listener the illusion that he is actually present in the auditorium. That effect is obtained when the liveness of the orchestra, including accentuation of any section such as strings, woodwinds, etc., lies between 8 and 20. A good average value for heavy music is 16 while for light delicate music a value of 10 is often preferable.

If the orchestra is accompanying a soloist, for instance, a violinist, a singer or a pianist, the liveness value for the soloist should never be less than one-quarter of the orchestral liveness and should preferably be between one-half and one-third. When the one-half of orchestral liveness is used the soloist is well out in front of the orchestra. As the solo liveness is increased the voice or solo instrument seems to move back and finally becomes merely an accentuated part of the orchestra itself.

The method of determining the dial settings to obtain these effects will be described in detail in Appendix 1. If you will use the quantitative method for setting the dials on your first few rehearsals with this new technique you will soon find that you easily recognize the desired effects by ear and no longer require the computed values, except for an approximate check.

Figure 1 shows the approximate arrangement of the microphones for the broadcast of the New York Philharmonic Symphony concerts by CBS on Sunday afternoons. This arrangement was arrived at in cooperation with Howard A. Chinn, CBS's Chief Audio Engineer after experimentation, with the valuable assistance of engineers and production personnel of the Columbia Broadcasting System.

Table 2 shows the liveness value for each of the three microphones and the power used, expressed in db above (+) or below (-) the power supplied by the over-all microphone D, Figure 1.

TABLE 2

Mike	L	db
D (general)	21	0
A (violin section)	0.6	-18 to-12
B (cello section)	0.6	-18 to-12
C (soloist)	0.5	not used

The effective liveness of this arrangement is about 13, which value is well within the useful liveness range shown in Table 1.

2. *Studio Music Programs with or without vocals.* As before, the most pleasing result is obtained when listener feels that he is present in the studio. A liveness between 6 and 12 for the orchestra yields this effect. Solo voices should have a value 1/2 to 1/3 of the orchestra value, while crooners may operate as low as 1/6 of it. These values are easily obtainable in a good studio which is not overcrowded much beyond curve B in Figure 3.

Figure 2 shows a typical studio set-up. The method of setting the distances and choosing the types of microphone has already been described. However, a sample computation may be helpful.

Assume a 20-piece orchestra and a crooner in a studio crowded to the curve marked B, Figure 3. Therefore, the studio volume is about 22,000 cu. ft. and it should have a reverberation time of about 1.1 seconds (see Figure 4) with orchestra.

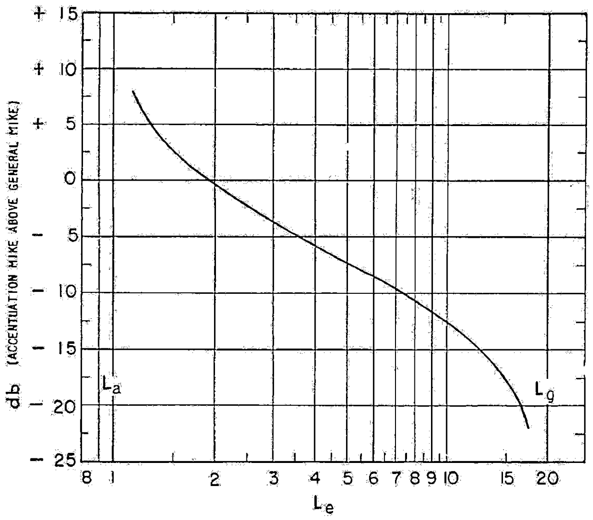


Figure 6--Chart showing effective liveness of two microphone combinations as function of relative level.

Assume

(1) the desired liveness, L, for the orchestra is 10.
(2) the desired liveness, L, for the crooner is about 1.6.
(3) the general microphone has non-directional characteristics.
(4) the microphone for the crooner and any accentuation microphones for parts of the orchestra have bi-directional characteristics.

Proceed as follows:

a. Set liveness of general microphone at a value 1.5 x 10 = 15.
b. Set liveness of crooner microphone at a liveness not greater than 1.6 x 2/3 = 1.0 approximately. Use 0.5 if in doubt, as slightly more flexibility is assured.
c. Set orchestral accentuation microphone at a liveness not greater than 2.0 x 2/3 = 1.3. Use 1.0.

From equation (2) you get the values in Table 3.

TABLE 3

Microphone	Type	G_p	L	Distance
General	Non-direct	1.0	15	16'--17'
Crooner	Bidirect	3.0	1/2	4'-- 6'
Accentuation	Bidirect	3.0	1.0	7'-- 9'

If the distance of 4'--6' for the crooner worries you cut it down to any value not less than 2' and mix accordingly (see Appendix 1).

3. *Speech such as announcers, newscasters, lecturers, etc.* These programs usually originate in small rooms of 1000 to 2000 cu. ft. with reverberation times of the order of 1/2 second. They get the full benefit of the extra coverage and naturalness due to liveness, for values of L greater than 1/6 to 1/4. From equation (2) the distance for a non-directional microphone for a 2000 cu. ft. studio, having a reverberation time of 1/2 second, would be 1-1/4 ft. and for a ribbon or cardioid microphone on beam would be 2 ft.

APPENDIX I

Method of Setting Relative Gain for General and Accentuation Microphones

Let L_a = liveness for the accentuation mike, and
L_g = liveness for the general mike, and
L_e = effective liveness of the combination
P_r = equal the ratio of the power contributed by the accentuation microphone to the power contributed by the general microphone; then db = 10 log P_r.

It can be shown that

$$P_r = \frac{1 - L_e/L_g}{L_e/L_a - 1} = \frac{L_a}{L_g}\,\frac{L_g - L_e}{L_e - L_a} \qquad (3)$$

and

$$L_e = \frac{L_a\,L_g\,(1 + P_r)}{L_a + P_r L_g} \qquad (4)$$

For example if L_a = 1.0 and L_g = 20.0 we may use equation (3) and compute the data shown in Table 4.

TABLE 4

L_e	P_r	db	Approx. db
1.25	3.75	+ 5.8	+6
1.50	1.85	+ 2.7	+3
2.0	0.90	- 0.5	0 or -1
3.0	0.43	- 3.7	-4
4.0	0.27	- 5.7	-6
6.0	0.14	- 8.5	-8 or 9
8.0	0.086	- 10.7	-10 or 11
12.0	0.036	- 14.4	-14
16.0	0.013	- 18.8	-19

Where a studio or auditorium is in regular use, it is worth the time to plot this data as a curve. Figure 6 shows such a plot.

In Table 4 and Figure 6, negative values of db mean that the level of the accentuation mike is below that of the general mike by the number of db shown.

Since the sensitivities of the various microphones are not equal, the following procedure must be used to determine this relative gain value.

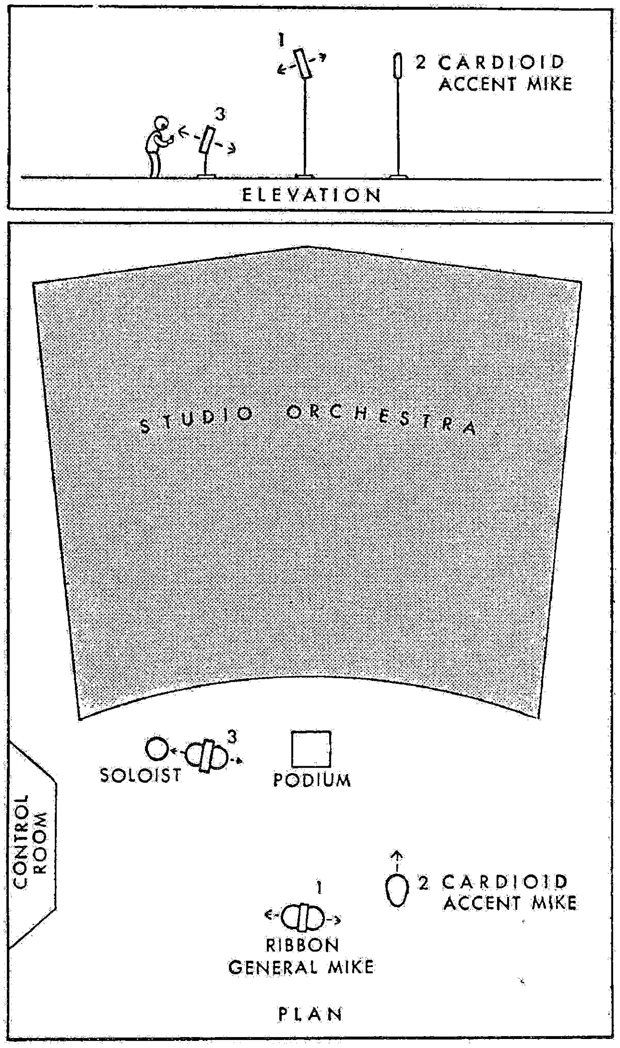


Figure 7--Plan and elevation of overcrowded studio.

During rehearsal, set the general microphone attenuator control and master gain control as if the broadcast were to be made on this microphone alone. Read attenuator dial setting for maximum peaks on vu meter.

Then turn off general microphone and turn up accentuation microphone, until maximum peaks have the same vu reading. Read attenuator dial for the accentuation mike. This becomes the zero of the db scale illustrated in Table 4 or in Figure 6.

For example, assume the general microphone had an attenuator dial setting 10 db and the accentuation microphone had a dial setting of 14 db for the same maximum peaks as read by the vu meter. This means that with a setting of 14 db the accentuation microphone is contributing the same power as the general microphone.

Then if the desired over-all liveness is 9 we find from Figure 6 that the accentuation microphone should be operated at -12 db. Therefore, the setting would be 26 db on the attenuator dial of the accentuation mike.

It sometimes happens that this low setting of the accentuation microphone causes a blend of the sound which does not seem to have given the accentuation to the desired instruments. This usually indicates that the general mike has been placed in a poor spot and that the accentuation mike is being used to mask this trouble.

Therefore, seek a new location for the general mike as a first step of correction.

APPENDIX 2

Music Programs in Overcrowded Studio

Under these conditions it is usually impossible to place the general microphone at a sufficient distance from the front row of the orchestra. Therefore, a trick must be resorted to.

(1) Place a bidirectional microphone, such as "1," Figure 7, with its insensitive direction pointing toward the orchestra. In practice this microphone will act as if G_B (see Equations 1 and 2) had a value of 1/4 to 1/3.

(2) Place the necessary accentuation microphones in the standard manner except that bidirectional microphones must not be placed too close to the studio wall. Where the crowding is extreme the use of cardioid microphones for accentuation purposes is preferred.

(3) Use the minimum contribution from the accentuation microphones, necessary to obtain the desired effect. Too much accentuation does more damage under overcrowded than under normal conditions, and can easily push the pick-up into the region below the useful liveness range. You then obtain "absence" instead of "presence".

This article has attempted to describe a semi-quantitative sound pick-up technique. If followed as a general guide, experience has shown that these methods can almost insure programs with a new realism - programs that can give your station as much as 6 db gain in coverage and your listeners the sense of being in the presence of the living artists.

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HOW TO IMPROVE PROGRAM PICKUPS

STUDIO PRACTICES REQUIRED FOR FM IMPROVE AM, TOO. HERE ARE IDEAS THAT CAN BE EMPLOYED WITH FACILITIES AVAILABLE AT ANY STATION--By HAROLD E. ENNES*

(Reprinted through courtesy of FM and Television)

Make it good. Then make it better. This is the creed of radio engineers. Over a period of twenty-eight years, broadcasting has been made good. With the advent of FM, it can be made very much better. Broadcast equipment has reached the state of design where the noise level is negligible, and the complete spectrum of audible frequencies is available for transmission and reproduction.

Yet it is apparent to all concerned that the techniques of co-ordinating and operating this equipment are as confused and almost as varied as the number of stations and the number of operating personnel at each station.

What has not been so apparent is the fact that this situation can spell the success or failure of a well-designed, well-maintained layout. There is extensive literature covering the theory, design, construction, and repair of the equipment, but the importance of using such equipment properly is just beginning to gain recognition of its true worth.

Microphone Facts & Fancies:

"The microphone is a mechanical extension of the human ear." How often have you heard that one? The fault in this definition lies not so much in its literal meaning as in the implications involved. If it were true, even though you had only one ear, you could walk into a studio and place the microphone at the spot where your one good ear could hear the orchestra, the soloist, the chorus, and the announcer. Only it doesn't work that way in practice. Yet this conception is probably the basic factor in the reasoning of the operator who just sticks up the mike in a studio and proceeds to broadcast the show by riding gain. And there is the more ambitious type who spots a microphone for each section of the orchestra, then one for the soloist and announcer and several for the chorus, and then becomes very indignant when the conductor reports that his musician friends listened to the show and thought someone else had been on the podium.

To understand why a microphone cannot be considered an extension of the human ear, it is necessary to review the fundamental theory. It is an old story, but absolutely essential to understanding the mike from an operational point of view.

Unlike the human hearing system, which is binaural (two-eared), the microphone is monaural (one-eared). It should be emphasized that this is true regardless of the number of microphones used, since the sound is collected into one channel, and reproduced by one loudspeaker, while each of our two ears has a separate channel to the brain. Physically, the difference is that in a monaural system the sense of direction is lost, while reverberation is somewhat more noticeable, making the apparent distance to the sound source seem greater than when listening binaurally. Any operator who has set up a microphone in a particularly live hall has experienced this phenomenon.

The conversation between two people anywhere in the hall can be heard quite clearly when listening with two ears, but when listening with headphones connected to an amplifier and microphone, the sounds seem much more distant, while the extraneous noise is high. This brings up an all-important psychological factor closely linked to the physical effect just explained.

This is the focusing power exerted unconsciously when listening binaurally, or even with one ear plugged up as much as is possible. The association of the ear with the nervous system and the brain tends to exclude recognition of the extraneous noise, and to focus attention on particular sounds. The microphone, as a mechanical device not associated with any means of concentration, exercises no discrimination between wanted and unwanted sound sources.

A practical example of this difference is to be found in any restaurant where there is dinner music. Despite the high level of ambient noise, you can carry on a conversation or listen to the music, and enjoy doing either one, but not both at the same time. But imagine the result if a microphone were placed on your table for a broadcast! Very little of the music would be heard, and the few strains coming through would sound far, far away. The radio listener would hear a hopeless hodge-podge of voices, noise, and confusion.

First, then, let's do away with the idea that the microphone is an extension of the human ear. This applies equally to microphone techniques for AM and FM. In practice, every effort must be made to utilize the possibilities and lim-

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*Engineer, Station WIRE Indianapolis Broadcasting, Inc., Indianapolis, Ind., author of "Broadcast Operator's Handbook" published by John F. Rider, N. Y. C.

itations of the mike in order to deliver an exact replica of the original program content to the ears of the radio listeners.

Frequently we hear it said that microphone techniques are different for AM and FM. As to this, there are ample grounds for disagreement. Discussions with control-room operators and production men seem to indicate that they do not know how to set up for optimum on FM. Therefore, they adopt the attitude that FM and AM techniques are different and, since the AM audience is larger, AM practices should be favored.

Actually, experience shows that the rule should be: "Precise or sloppy techniques show up more on FM than on AM." In other words, the difference between careful and careless handling of a program is disclosed on AM to only a limited degree, but FM listening shows up the difference very clearly, even with a set of no more than average audio capabilities.

In this connection, there is another point that calls for revision of control room practice at many stations. It is not unusual to find three speakers in one control room, for the operator may be called upon to keep up with the program on the air, another under rehearsal, and a third being aired by a local competitor. Under such conditions, critical listening by the operator is impossible!

All this leads up to the fact that FM is setting new and very high standards of studio practice. Haphazard program setups that get by on AM may prove to be poison for FM. This is no exaggeration. On the other hand, the employment of high-fidelity methods for FM results in improved quality on AM. This will be explained in detail.

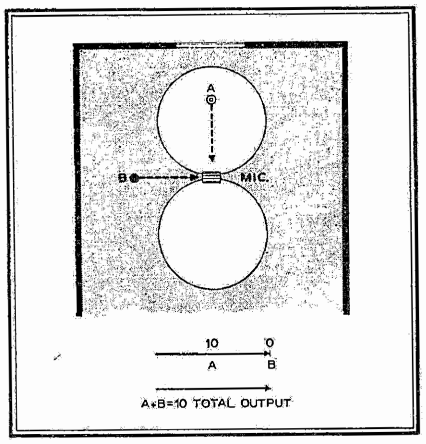

Fig. 1. Bi-directional pattern in dead room

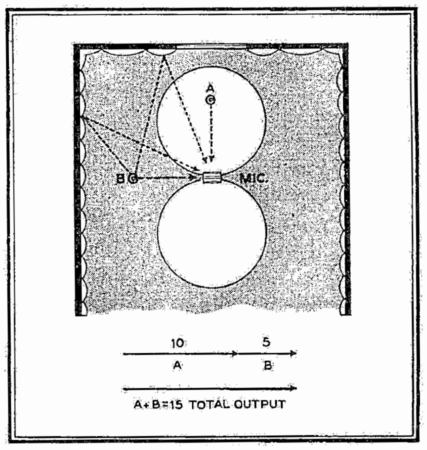

Fig. 2. Results are modified in live studio

Understanding Response Patterns:

Insofar as program setups are concerned, the foremost characteristic of a broadcast microphone is its response pattern. There are several important factors which must be considered when using a microphone pattern to obtain a desired result.

The basic point to keep in mind is that a response pattern as illustrated for a given mike is plotted in a perfectly dead room, to avoid all reflections of sound waves. But when a microphone is set up in a studio room, although the theoretical response pattern does not change, the mike is acted upon by reflections which vary in direction and magnitude according to the shape of the walls and the ceiling, and their ability to absorb or reflect sound waves. This will be made clear by the hypothetical case of a setup involving two sound sources and a bi-directional microphone.

Fig. 1 illustrates the ideal response curve of a bi-directional mike in a completely dead room. Sound sources A and B are of the same intensity and at the same distance from the mike, but B is directly on the zero-response axis. Let us say that each source represents 10 units of sound. Since sound source B will excite both sides of the ribbon equally, no movement of the ribbon will result from this source, and no output voltage will be created in the microphone. Therefore, the total output voltage will be 10 units, and consist only of the impulses received from A. Thus the pattern holds true.

Fig. 2 shows the same setup in a live room. It is still true that B will excite both sides of the ribbon equally resulting in no response to the direct wave, but now we have reflections. Sound will be reflected from the walls back into the sensitive side of the mike and, although reduced in intensity, will add to the 10 units of sound from source A. The difference now between the two sound sources is not only one of intensity, but ratio of reflected to direct sound. Naturally this ratio is greater for sound source B than for source A.

It is, of course, obvious that there is no such thing in broadcasting as a perfectly dead studio. This example is, therefore, rather crude, but serves to illustrate the basic idea of how acoustical treatment influences the polar action upon a microphone. This understanding is imperative from an operational point of view.

While we are on the subject of response patterns, it is well to be sure that it is clear as to just what information they are meant to convey. Fig. 3 shows the polar response pattern of a Western Electric 639 A cardioid, plotted for four different frequencies against the ideal cardioid curve. This shows that there is a narrower response-angle at higher frequencies than at lower ones because high frequencies tend to travel in beams.

The curves at the left in Fig. 3 illustrate, for example, that on 50 cycles the response is down about 5 db at 90° with respect to its response for the same distance at zero degrees. It will also be noted that for 9,000 cycles the response at 90° is down about 6 to 7 db. Also, at 90° the 9,000-cycle response is about 11 db lower than the zero-axis, 50-cycle response. Since, in broadcasting, we are concerned not only with wanted and unwanted sound, but all shades in between, the individual response patterns of a mike prove extremely useful if utilized properly. From this discussion we have four fundamental operating points for the microphone:

1. If a sound source must be moved about the mike, loss of response can be compensated for, if desired, by moving the mike closer to the source.

2. The ratio of reflected to direct sound can be raised in a sufficiently live studio by using greater angles from the zero axis of the mike. This is especially true of bi-directional microphones.

3. The more live the studio, the greater will be this effect.

4. High-frequency sound sources must be more nearly on beam for a given distance to achieve the same intensity of response as lower-frequency sound sources.

To this latter point should be added the note that a bi-directional microphone has a greater deviation in angular response between high and low frequencies. Also, the cardioid or uni-directional instrument has a much wider angle of response at all frequencies than a bi-directional mike.

This is shown in Fig. 3, where the 1,000-cycle curve of an RCA 77 B uni-directional mike (solid curve) is super-imposed on the 1,000 cycle curve of an RCA 44 BX velocity (bi-directional) microphone. It should be observed that at 60° the relative response of the 77 B is down only about 2.5 db, whereas the 44 BX at 60° is down about 6 db. The very great difference at 90° is clearly shown. This is typical of all makes of broadcast microphones when comparing the uni-directional and bi-directional patterns.

Studio Acoustics:

The non-standard acoustical characteristics of broadcast studios are the outstanding factor that has prevented any establishment of definite

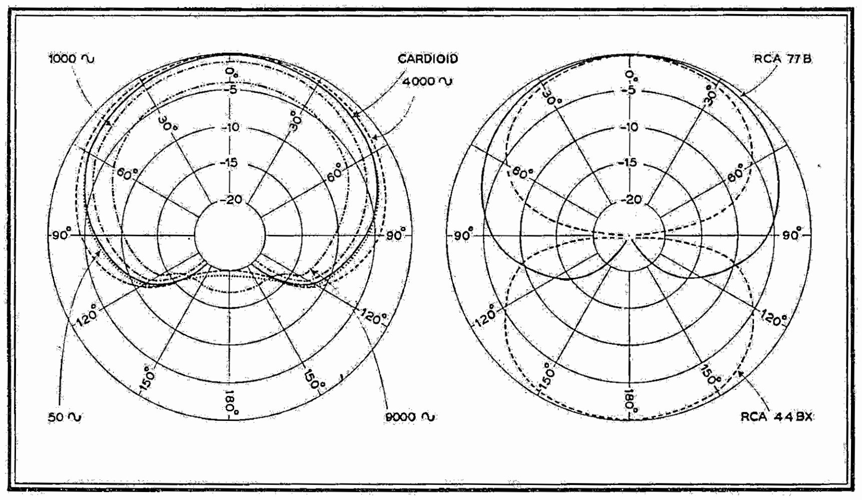

Fig. 3. Left: W. E. Cardiod. Right: Uni-directional and bi-directional RCA types

standards in microphone setups. If there were such a thing as a standard studio, designed to approach as nearly as practicable the ideal condition of sound dispersion, the setup for any particular musical organization would be a simple matter anywhere when once worked out. Naturally, such is not the case. It is probably safe to say that there are no two studios anywhere in the world that are acoustically alike.

Right here, the author feels it advisable to bring out a point so far neglected in the scanty literature on mike technique. This point is that a great number of operators and producers (alas, even in FM) do not have adequately designed and acoustically controlled studios in which to work. This is the most deplorable situation existing in broadcasting, but it is outside the influence of the average operator. All we can do is hope, and then plead, and plead some more until the station owners become cognizant of the extreme importance of modern studio design. It is a sad state of affairs that even some of the most recent FM installations, with completely new studios independent of any AM connection, are neglecting this feature. Most of the good articles appearing thus far on microphone setups have been concerned with well-designed, musically-live studios of the network centers, or the more production-conscious independent owners.

Yet how can an article of this kind be useful to the average operator wanting to do his best in studios of the same design as those built twenty years ago? Since it is the purpose of this paper to present practical information, the writer will attempt to show some definite rules that can be used to meet any acoustical conditions. High-fidelity techniques for a good, modern studio can not be applied to most of the studios in use around the country. We shall have to plant our feet squarely on the ground and face the situation as it really exists.

In the modern studio, the walls are live to musical sounds, but are broken up acoustically in some manner to avoid standing waves, while still achieving a maximum response to diffused, poly-phased high-frequency sound. Under those conditions, good tonal brilliance can be obtained with a minimum of microphone and control-board manipulation. This is the type of design we need to really practice proper microphone technique.

The Single Announcer

The single announcer is the logical starting point for our discussion of mike technique. It must be understood that we are not concerned at present with announcing over a background of music or any form of dramatic presentation. You may ask: "What is there to discuss, then? The announcer just steps up to the mike, or sits down in front of one, and talks!"

You're right. He does. But the fact of an error does not justify it.

Announcing alone occupies a considerable portion of the broadcaster's schedule. Correct voice transmission is so very important to radio because we do not have the sense of sight to help our impressions; the voice is the complete medium of expression. The intake of the breath, the most subtle inflections, the style of delivery, the original voice timbre, all are necessary to the listener. Any or all of them may be severely affected by the announcer's relation to the mike.

Those of us in the technical end of radio must at some time of our careers realize that engineering training does not develop an appreciation of artistic values or sense of showmanship. We are very apt to lose sight of the real reasons behind the keys, faders and perfectly matched impedances of the control system. They are designed this way so that the electrical impulses may correspond to the thunderous crescendos of Wagner's Die Walküre or the light, delicate strings of Debussy's Festivals. It is with the intangible qualities of human experience, expressed by great artists through the moods of music, that the wires and switches and dials and knobs of the technical department are concerned. As we grow ever more conscious of this, we come to see our work in its true relation to radio's service to the listening audience.

But to get back to the announcer and our starting point in high-fidelity microphone techniques. Mugging the mike is a strongly entrenched and deeply-rooted habit. But the announcer must be kept away from the microphone. He will probably object. He may refuse. Some announce desks have microphones permanently mounted on them; and if the announcer moves back he must hold his script in an uncomfortable position, without an arm rest. This is plainly and simply an engineering error. Like fingerprints, there are no two voices exactly alike. For every voice there is a definite relationship to the microphone which allows the most natural and pleasing reproduction.

For voice work alone, 2 ft. is the minimum distance from the face to the microphone that will assure naturalness of that voice! A distance of 2 ft., incidentally, should only be used for the softest voices. Compare this with your own studio operating practice. What is the distance your announcers use? Probably somewhere between 4 and 12 ins. Voice waves at this distance from the mouth and throat cavities do not create electrical impulses that correspond to the natural character of that particular voice.

Do this as an experiment: On rehearsal, set up microphones (in addition to the regular announce mike) 3, 6, and 9 ft. away. Stagger them enough so that no mike will be in front of another. If the regular microphone is immediately in front of the announcer's face, you will have to move it lower or to one side. Don't tell him what you are doing. Just say you are testing mikes. If he is told to start reading at a distance from one microphone and then to move closer, he will subconsciously alter his volume and you will not get a natural check.

Now turn only one microphone on at a time. Start with the farthest mike. Unless you are so conditioned to hearing the mugging voice that you can accept no other, you will find a new experience in naturalness of voice transmission. A distance will be found where the voice begins to sound hollow in a live studio, or thin in a dead one. Use the distance just a shade closer than this point.

In small announce booths, where space is at a premium and where distance would create a barrel effect due to the proximity of the studio wall to the microphone, the mike should be suspended over the announcer's head at a distance of several feet. We are borrowing this technique from television, where the mike must be kept out of the range of the camera, and is often as much as 15 ft. from a performer. It should be borne in mind, however, that the sound in television is only a secondary expression to the picture. In other words, all of the intent and meaning need not be embraced in sound, as in audio broadcasting.

This same technique is most convenient when two persons are seated at a table and using a single mike as illustrated in Fig. 4. The table top should be deadened by acoustical treatment or by the use of a heavy cloth cover to prevent reflected sounds from entering the mike.

HOW MANY MICROPHONES?

In an approach to the study of specific types of program setups, the question of the number of mikes will invariably arise, especially on a large show. Keep always in mind the previous discussion on the monaural character of

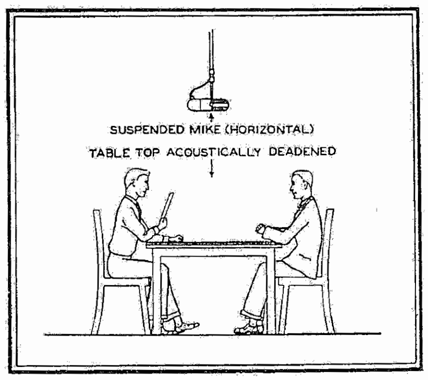

Fig. 4. Setup for interview in small room

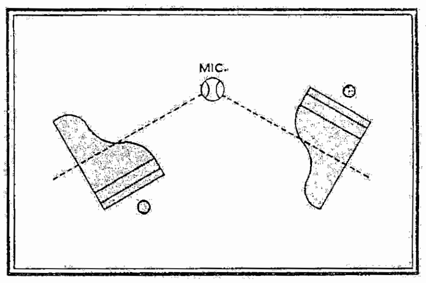

Fig. 5. Orientation for a two piano-team

the system and the lack of focusing power unless deliberately used as a means of concentrating the attention.

The greatest weakness in mike setups for large shows has always been the use of too many microphones. There is bound to be some distortion, however slight, in multiple-mike setups due to time lag of sound waves which create phase additions and subtractions at the various pickup positions. This source of distortion, however, is only minor compared to the other faults of this technique. Aside from the operational difficulties of handling a large number of channels on the mixing panel, with greatly increased chances of error, the control man and his board take the place of the conductor. All the dramatic interest, the emotional pattern written into the original score, plus even the conductor's interpretation, is placed in the ratio of fader adjustments and the reactions and psychological temperament of the operator. In other words, too many variables are injected between the performers and the listeners.

Let's establish a foundation upon which we can build a workable structure to determine the correct number of microphones for a given show. Let's also be practical, and realize that many operators have neither the very latest studios nor an adequate amount of rehearsal time.

1. Whenever it is possible in the time allotted for setting up, arrange your performers about a single microphone (following suggestions given later for each type of show) so that the overall balance is correct. If you do this, you achieve balance by proper positioning, rather than by mixing various sources on the control board.

2. Perhaps your time is running out, and you are still having trouble with a particular section in obtaining balance. This is more apt to occur in a dead studio than in one which is acoustically live. Another mike will have to be used on the troublesome section. However, use of the second pickup can probably be limited to particular spots in the show, such as rhythm-

section accentuation of an orchestra as required by the score.

3. Some of the more complex shows do require more than one microphone. Take, for example, a variety show consisting of a drama cast, a chorus, and an orchestra. Remember that the microphone does not focus attention as your ears do in the studio. Of course, it is likely that a program of this type will originate at one of the larger stations, having modern studios. The more rehearsal time you have, the less the number of microphones necessary, to the point where the absolute minimum is reached. So much for the basic theory of microphone technique. Let's go on with more specific program setups.

Picking up the Piano:

The single piano pickup is the simplest setup, since it is perhaps the least affected by acoustical nature of the room. No matter how softly the pianist plays, as long as he is unaccompanied by other instruments, the microphone should not be placed up under the lid. The distortion arising from the close proximity of large physical objects to the microphone is well known. It should then be obvious that a pickup under the lid and close to the sounding board will not allow natural transmission of the piano tones. Close-miking almost any instrument makes it necessary to hold down the volume, with great loss of musical brilliance. It is true that many operators have become so accustomed to this type of piano setup that, as is the case with the close-talking announcer, the sound may be familiar, but it is not natural reproduction. We must remember that it is the business of the broadcaster to transmit the natural sound of the original program content.

Here is the best method of determining the setup for a single piano: Start with a distance of 20 ft., head high. In a dead studio, this distance will probably result in a thin response, especially on low passage. In live studios, the sound may be too reverberant for clear-cut transmission. Move the microphone in on a line drawn through the center of the sounding board until the tones are full-bodied, with just the right amount of reverberation. This distance is seldom less than 8 ft., and will allow tonal brilliance and balance between lows and highs that is sacrificed in close-up technique.

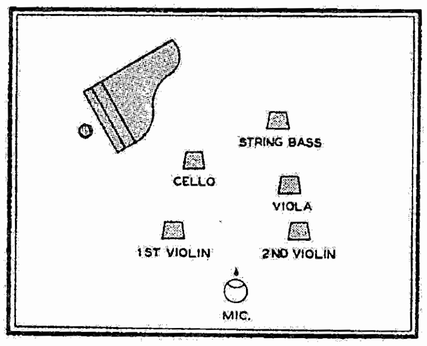

Fig. 6 Orientation for a salon orchestra

Now comes the final check for balance between bass and treble, in other words, the check of the player's left and right hand pressures. If the pianist has a heavy right hand, and bass response is somewhat weak in relation to the highs, keep the mike at the same distance, but swing it toward the tail of the piano. This method increases the response from the bass strings. If the pianist has a heavy left hand, with a relative loss of highs (this lack of highs is also apt to occur in dead studios), the mike should be moved toward an imaginary extension of the keyboard. This will increase response to the treble strings and decrease that from the bass strings.

The twin piano team imposes only slight additional requirements. Fig. 5 illustrates the most satisfactory orientation. The temperaments of the pianists must, of course, be considered and the lead piano given prominence by moving the mike closer to that piano if the accompanist is heavy-handed. The procedure for bass and treble balance should be followed not by moving the microphone, but the piano itself.

Vocalist and Piano:

The distance of the vocalist from the mike, the distance of the mike from the piano, and the distance of the vocalist from the piano must be considered in this setup. There is no excuse for using more than one microphone unless the time allowed for rehearsal is zero.

The first requirement is to listen to the vocalist in the studio. Does he or she sing out with the chest muscles, using a large volume and dynamic range? Or is the vocalist the crooner type, using only the larynx and throat for emphasis? Every vocalist is in one or the other general category.

One who sings out with full volume should be placed 12 to 6 ft. from the microphone when a piano alone is used as background. On rehearsal, always start with the greatest distance. The goal is to use the lowest volume and the highest volume without having to ride gain on the fader control. This part of the balance can always be achieved by careful rehearsal checks. The balance between vocalist and piano accompaniment is not always so simple. A good pianist or one familiar with a particular vocalist's style will automatically adjust his volume to the pianissimo and crescendo of the singer.

A bi-directional mike should be placed about 8 ft. from the piano, with the vocalist on the opposite side at the distance determined by trial.

It is well to point out here that a very common error is in taking a vocal solo too literally. The presence of the piano must always be there, with only slight emphasis on the voice. It should be a blend with, of course, the voice always a little predominant, but not with a weak background of piano tones, as is often heard.

When the pianist insists on playing so loudly that the accompaniment smothers the vocalist (some pianists cannot alter their volume and still play well) the dead side of the vocalist's microphone must be turned toward the piano. This is almost always the only adjustment necessary from the 8-ft. distance between mike and piano. If the studio is very live and the piano tones are still too prominent, turn the piano around so that the lid opens toward a wall of the studio. Don't at any time, move the vocalist closer to the mike if it is necessary to ride gain on the natural dynamic range of the voice.

Setup for Small String Groups:

Small salon groups, string quartettes, or hillbilly groups, playing in intimate style, require good instrumental definition. This calls for comparatively close mike setups, but not too close! The author is always hesitant about using the word "close" since the reader is apt to take it as meaning directly into the face of the instrument.

Consider a small salon orchestra, generally consisting of several violins, a viola, a cello, a string bass and sometimes a piano. Due to their comparative volume they are usually placed in that order from the microphone. Fig. 6 illustrates the general orientation of such a group with the mike.

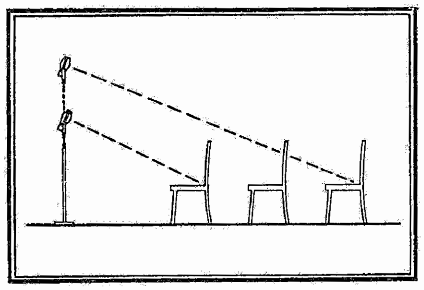

Fig. 7. Adjusting microphone for balance

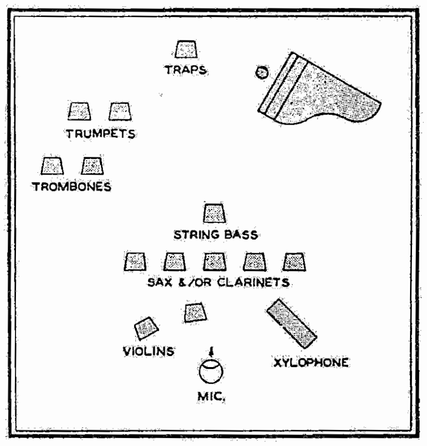

Fig. 8. Conventional one-microphone plan

Now assume that the approximate distances are violins 4 ft., viola 6 ft., cello 8 ft., and string bass 10 ft., with piano somewhat off-mike at 10 ft. If the violins are too predominant for proper sectional balance, usual practice is to move the violins further back or to one side in a less sensitive zone, or else to bring the other instruments in closer.

However, remembering the focusing power principle, it is clear that quite a range of sectional balance can be obtained by the simple expedient of adjusting the microphone height and tilt, as shown in Fig. 7. If the violins are too predominant, raise the mike and focus on the other instruments. If the violins are weak, the mike should be lower and focused on them. This is a better method than moving the predominating instruments to one side in a less sensitive area of the mike, since the higher frequencies containing the over-tones are very important for wide-range pickups.

There is little difference in pickup for this type of musical organization between dead and live studios, since the setup must be intimate in character, with high direct-to-reflected sound ratio, minimizing the effect of the acoustical nature of the studio.

Small Orchestra Setups:

A large number of organizations presenting popular, serious, or variety music comes within the small-orchestra category. There may be a combination of brass, strings, and reeds, numbering anywhere from 4 to 15 musicians.

Again the first step is to visualize the instruments in comparative power output. From softest to loudest, they are:

1. Violins, trumpets or trombones (muted), guitar.

2. Clarinets, saxophones, xylophone, vibraphone.

3. String bass.

4. Piano.

5. Trombone (open belled), trumpets (open belled).

6. Traps, bass drums, guitar (electrically amplified).

These are the most likely instruments to be encountered in such a setup. As before, the conventional approach is to arrange them in the order listed from the single microphone.

Look now at Fig. 8. This is a live-studio practice, starting with the violins at about 8 ft. When initially checking the balance of sections, remember the height and tilt adjustment for focusing power to obtain proper blend. Then, and not until then, try moving a troublesome section.

If you think it necessary to move instruments, keep these principles in mind: A predominating section may be too loud not because the relative distances between instruments is incorrect, but because the microphone is too close to the entire outfit. Move the mike back.

A weak section may be too soft not because it is too far from the mike, but because all instruments are too close to the microphone. Move the mike back.

In other words, the farther back a microphone is placed (within the limits of acoustically allowable distance) the better the chance of a good balance between all sections.

Another very important item is the treatment afforded muted trumpets or trombones. When their bells are muted, the instruments must be very close to the microphone. This means really close, about 2 ft. If the players cannot or will not step from their regular positions to one immediately in front of the mike, a separate microphone must be spotted just in front of that section. Obviously it is to be used only when they are muted.

Now consider a studio of older design, with dead characteristics. When the musicians number around 12 to 15, it is often difficult to get good sectional balance with the plan shown in Fig. 8. Even though the farther instruments may contribute about the same number of volume units, the pickup may be thin, because the dead studio does not reinforce the harmonics and overtones. Also the mike must be a little closer in a dead studio, emphasizing the discrepancy in sectional presence.

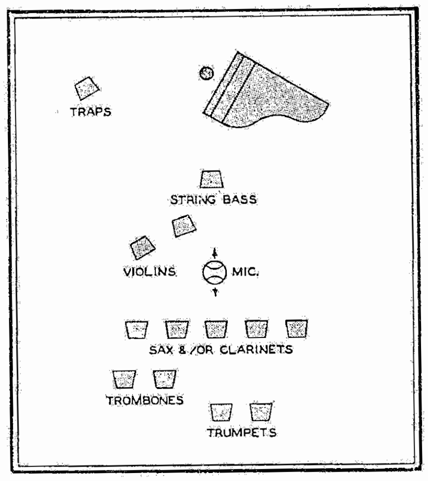

Fig. 9. Improved pickup location

In nearly every case, however, best results are obtained with one mike, and by using the setup illustrated in Fig. 9. The mike is a bi-directional ribbon type. The instruments must be more nearly on beam, due to the narrower angle of response in comparison to the uni-directional mike employed in Fig. 8 and to the dead acoustics of the studio. More time and movement of players will be necessary to get this setup exactly right, but it is far better than the usual procedure of using additional microphones.

How to Add a Vocalist:

A vocalist with the type of musical program just described imposes special problems unless the organization is thoroughly trained in prior broadcast production technique.

The ideal arrangement would be for the vocalist to stand in front of the orchestra, facing the mike at a distance of several feet. This arrangement, however, is often not practical as, for instance, when the mike is raised and slanted so as to obtain proper balance of instrumental tones.

Where such a situation occurs, there is no alternative but to use a second microphone for the vocalist, preferably a uni-directional type, with dead side toward orchestra. This mike can be used for the announcer, also.

Symphony Orchestras:

The large symphony orchestra setup should be based upon exactly the same principles as heretofore discussed, except that the grouping of instruments, the use of a chorus, and the type of musical score make the problems much broader in scope. Fortunately, such programs are not usually attempted in an inadequately-designed studio.

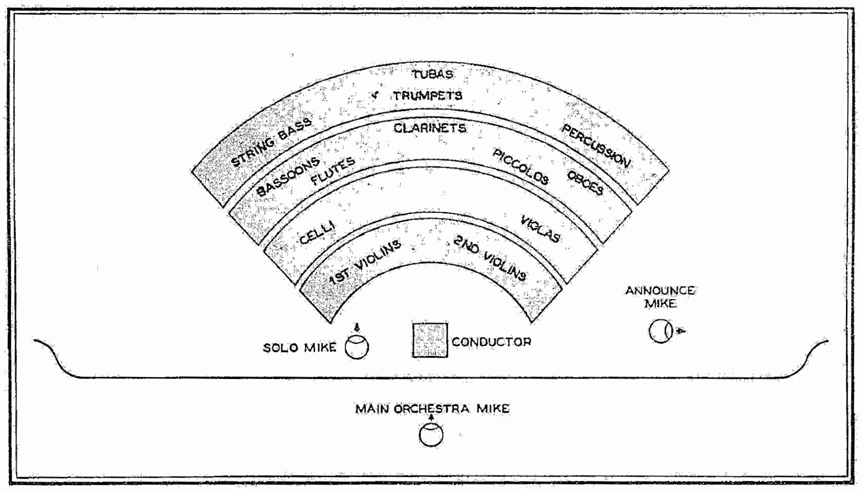

Fig. 10. Symphony orchestras may require solo and section mikes for highest fidelity

Generally, the usual physical arrangement of the orchestra for regular audience listening will be satisfactory for broadcast purposes. On the initial trial of the main orchestra microphone, several mikes should be suspended at the most likely pickup positions in order to avoid the confusion of moving mikes. The general area for these microphones is 15 to 20 ft. high, and 15 to 25 ft. in front of the violin section. Fig. 10 illustrates a typical grouping of a large symphony orchestra.

If a vocalist or instrumental solo is called for, it is almost always necessary to use a second mike to achieve proper sound perspective. Remember that vocal or instrumental solos are *not* to be entirely predominant; the orchestral accompaniment must be very much present. Always try to get the conductor to listen to a monitor speaker for a final check on balance, or get some responsible member of the organization to pass on it. The very best control and production men do this simply because every symphony organization has its own arrangement of score or possibly a distinct interpretation of the original score. This individuality must be conveyed to the listeners. Thus, many times, a slight rearrangement of instruments in relation to the microphone is found necessary.

In spite of the usually superior results obtained with a single, well-orientated microphone, it is often necessary to deviate from this practice for true high-fidelity pickup. The main orchestra microphone, back far enough to obtain the proper blend of all instrumental tones, will faithfully pick up the delicate, distant tonal beauty of the violin passages in *Clair de Lune*, for example. Most music lovers, however, criticize this same microphone setup for such numbers as the Strauss waltzes, where the tonal perspective of the strings should be closer and more strident in quality.

Many leading conductors and producers of symphony broadcasts insist on an added mike suspended directly over the strings or other sections, to be turned up only on cue. This procedure supplies the missing psychological factor of microphone-to-sound perspective. When a choir is used with a symphony orchestra, it is also necessary to use a separate mike due to the spread of the total combination, in order to obtain focusing power.

Notes on Field Setups:

Field setups are an entirely different matter. The musical instruments must usually be placed more in a straight line due to lack of depth of the platform. Added to this is the inevitable noise and confusion about the point of origin. It may be argued that really high-fidelity transmission is impossible from such a broadcast, and this is true. But FM listeners especially will appreciate any pains taken to improve the pickup.

Multiple microphone arrangements are absolutely necessary for such an orchestra of even medium size.

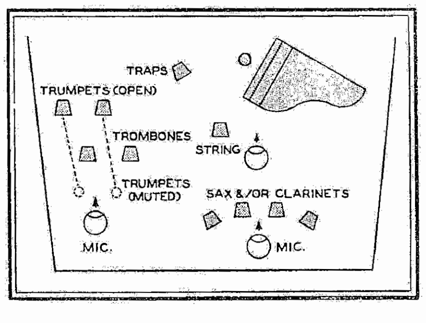

Fig. 11. Mike setup for dance orchestra

Fig. 11 illustrates a 3-microphone treatment of a typical dance orchestra playing in the open. Always strive to obtain a good balance with the minimum number of mikes, taking into consideration the usually heavy background noise. It is almost always necessary to use a microphone for each group of instruments, as shown in Fig. 11, to give the section playing the lead or melody at any particular time the highest intensity. But remember that the presence of the other sections is important.

Conclusion:

It is apparent to the experienced broadcast man that many types of program setups have not been treated in this article. Indeed, such a treatment would require a full-size book. The author has attempted to take up only those situations most commonly encountered.

There will, no doubt, be some who have honest disagreements with parts of this paper. The author extends an invitation of correspondence to any reader who may care to offer comment or criticism. Progress in any line of endeavor can be achieved only by the inquisitive mind, and earnest doubts of traditional practices. Microphone set-up techniques in particular, and operational engineering in general are in need of exhaustive, wide-open discussion.

ROOM ACOUSTIC DESIGN

PREFACE

The third edition of the NAB Engineering Handbook contained a reprint of a paper presented by Mr. John E. Volkmann before the Acoustical Society of America in New York, entitled "Polycylindrical Diffusers in Room Acoustic Design." Mr. Volkmann, who is Manager, Theatre Equipment Engineering, Sound Engineering Section, RCA Victor Division, Camden, N. J., has prepared another paper entitled "Acoustic Recommendations for Small Combined Studio, Scoring Stage and Review Room" which although it was prepared originally for Motion Picture Studios provides sufficient data believed, of interest in the construction of Television Studios, especially in view of the trend toward large screen television.

The NAB Department of Engineering, with the permission of Mr. Volkmann and RCA, has combined the two papers to provide informational data useful in the design of studios used for AM and FM broadcasting as well as for television broadcasting and recording.

BASIC REQUIREMENTS AND DEFINITIONS

Although large scale recording and projection operations usually require separate rooms for optimum acoustic results, economic considerations frequently necessitate the combining of these facilities into one studio and the primary purpose of these recommendations is to provide the basic information necessary to accomplish this objective. The proper planning and acoustic design of a studio before construction has received ever-growing attention by architect and engineer, and has eliminated much of the necessity for the curative treatment of acoustic defects.

Good Acoustics:

The requirements for good acoustics have been stated by Wallace Clement Sabine, pioneer in architectural acoustics, as follows: ". . . it is necessary that the sound should be sufficiently loud; that the simultaneous components of a complex sound should maintain their proper relative intensities; and that the successive sounds in rapidly moving articulation, either of speech or music, should be clear and distinct, free from each other and from extraneous noises. These three are the necessary as they are the entirely sufficient, conditions for good hearing." (1)

The criterion most frequently used for judging the acoustic excellence of a room is its optimum reverberation time. Other factors and acoustic phenomena with which the scientist or acoustical engineer is concerned in determining the acoustical performance of a room include the following: echo, resonance, interference pattern, sound absorption, reflection, transmission, diffuse sound, rate of decay, intensity distribution, sound level, and noise level.*

Experience in rooms with wood-paneling and sound-diffusing surfaces indicates that the manner in which the reverberant and other after-sounds in a room are distributed has possibly as much to do with the acoustical excellence of a room as the actual time of decay in the room, and from time to time we have heard discussions on the diffusion of sound in rooms not only in regard to fulfilling the assumptions in the Sabine reverberation formula, but in regard to its subjective effect.

Physical Design Factors:

The major physical factors with which the architect is concerned in the acoustical design of a studio are as follows:

1. Choice of Site.
2. Size of Studio.
3. Shape and Proportions of Studio.
4. Selection and Placement of Absorbing Materials.
5. Sound and Vibration Isolation.

The broadcasting and recording engineers are also interested in the placement of microphones and loudspeakers and in variable reverberation control.

GENERAL CONSIDERATIONS

Since most room acoustic difficulties occur due to reverberation and other reflection phenomena, a brief discussion of these general factors and their control is considered important to a proper understanding of acoustical design.

Reverberation:

The action of sound when confined in an enclosure is much more complex than its action in free air. In free air only the direct sound from the source can be heard. In a room, however, the sound one hears is composed of both the direct and the reflected waves. The sound wave generated in the room proceeds or expands spherically from its source until it meets the boundary surfaces where it suffers partial reflection, absorption, and transmission. The reflected wave from each surface continues its travel inside the room only to again suffer partial reflection, absorption, and transmission

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*See Appendix I for definitions of acoustic terms.

each time it strikes a boundary surface. This process is repeated until the sound energy is completely dissipated. In an ordinary room with plaster walls and ceiling, an ordinary sound may suffer from 200-300 such reflections before its energy is completely dissipated by absorption or transmission. If we recall that sound travels 1120 feet per second it is readily seen that the duration of this prolonged after-sound, called the reverberation time of the room, may be several seconds for a large room where the mean free path is long. The effect of excessive reverberation is to blur speech and music due to the overlapping of successive sounds.

The reverberation time of a room, which is the time required for a sound of given initial intensity to die away to the threshold of minimum audibility, depends directly on the intensity level of sound and the size of the room, and depends inversely on the absorption in the room. The standard reverberation time of a room is the time obtained when using an initial intensity of 1,000,000 times the intensity at minimum audibility. Increasing the size of a room increases the reverberation period due to the fewer number of reflections which occur in it during a unit interval of time.

Optimum Size of Studio:

Just as tradition and experience have taught that an optimum reverberation time exists for each size room, so we have learned that an optimum size of studio exists for each type and size of orchestra. The data (2) in Table I is based on commercial practices and subjective listening tests. The column labeled "auditorium practice" shows the minimum number of orchestral instruments recommended for auditoriums, (3) while the column labeled "broadcast practice" gives the maximum number of instruments sometimes crowded into broadcast studios. The recommended practice for recording and scoring studios is shown in the column labeled "optimum." Experience in Hollywood indicates that a volume of 50,000 cu. ft. constitutes about the minimum requirements for scoring studios. Several typical Hollywood scoring stages are shown in Figures 16, 17, and 18. Another acoustical factor which, in the case of sound-picture projection, is related to the size of room is the acoustical power required for proper sound reproduction. Most sound movie reproducing equipment is more than adequate for the ordinary size review room; however, in those cases where the picture is reviewed in a theatre, the reproducing equipment should meet the "Minimum Power Requirements for Theatres" established by the Academy Research Council.(4)

From the standpoint of reverberation time, minimum power requirements, air-conditioning system requirements, etc., the theatre, which is used for reviewing purposes, should be made a minimum in size compatible to its seating capacity and architectural treatment. Small review rooms and scoring rooms on the other hand should be made as large as possible to most nearly approximate auditorium and hall conditions. For normal theatre sizes the Academy of Motion Picture Arts and Sciences recommends 125 cu. ft. per seat. For small rooms, say with a seating capacity of only 20, this figure may be increased to as much as 500 cu. ft. per seat.

TABLE I

Size of Orchestra VS. Room Size

Volume of Room in Cubic Feet	Auditorium Practice	Optimum	Broadcast Practice
10,000	--	6	12
20,000	6	9	25
50,000	9	21	50
100,000	19	38	130
200,000	31	70	250

Room Resonances--Effect of Parallel Reflecting Surfaces:

When sound is generated between two parallel reflecting surfaces, "standing waves" are set up at certain frequencies depending on the distance between the surfaces. The lowest frequency at which the standing wave resonance will be set up is that frequency for which its wavelength equals twice the distance. The parallel surfaces will likewise cause resonance for harmonic frequencies. Thus, for a flat reflective ceiling whose height is 11.2 ft. above floor level, resonance may occur at 50 cycles, 100 cycles, 150 cycles, etc. (frequency equals 1120 divided by wavelength, that is, by twice the ceiling height).

The effect of these resonances is twofold: first, it introduces frequency discrimination in the form of peaks and dips in the response characteristic, and second, it introduces a persistence or handing-on effect in the sound for frequencies at or near resonance. This frequency discrimination and persistence gives a hollow sounding characteristic especially if the resonance frequencies are widely separated (surfaces close together), which is generally the case in small rooms.

The effects of standing wave resonances may be minimized or controlled by any or all of three methods: (1) by absorption treatment, (2) by changing the position or spacing of the surfaces, and (3) by changing the shape of the reflecting surfaces. In general, the first method is not satisfactory in small studios, since the absorption required for damping the resonances is too great from the viewpoint of optimum reverberation and hence causes the room to sound "dead," and if the absorption material is se-

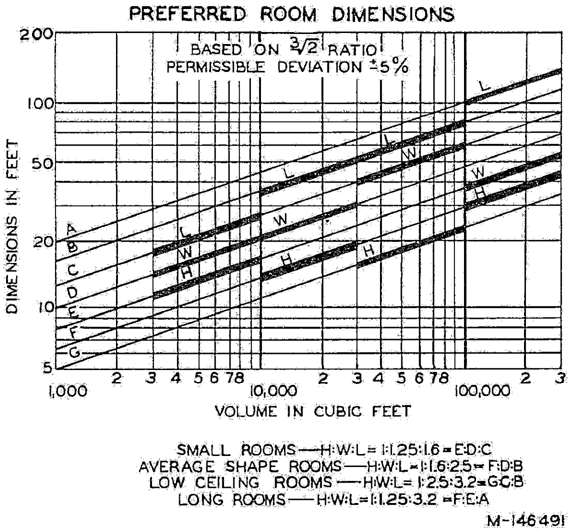

Figure 1--Preferred studio dimensions.

lective may cause it to sound too "bassy." The other two methods will be discussed under separate paragraphs later.

Structural Resonance:

The phenomena of resonance, or the ability to vibrate most easily at certain frequencies, may occur in structures as well as in the air in rooms. Structural resonance usually is not harmful unless the resonant body is closely coupled mechanically to the source of sound. As discussed later, structural resonance is sometimes a virtue; for example, the resonance of wood paneling in an auditorium is often a factor which improves its acoustic excellence, especially for music. In this case, however, the fundamental and harmonic resonances do not occur at any one frequency, but rather over many different frequencies due to variations in size and construction of the individual wood panels.

Shape of Studio--Preferred Proportions: (5)

To minimize the frequency discrimination effect caused by the standing wave systems set up between parallel surfaces in a room, it is desirable to choose major dimensions which are not integral to each other. By proportioning the three major dimensions of a room in the ratio of the cube root of two (or in ratio of multiples of the cube root of two), a good distribution of the natural resonance frequencies is obtained. For small rooms the preferred ratio of height to width to length is 1 to 1.25 to 1.6 as shown in Figure 1. For the average size and shape studio the preferred dimensions should be in the ratio of 1:1.6:2.5. In all cases the dimensions shown are derived from the ratio of the 3/2. Stating it another way the major dimensions should be separated 1/3 octave with respect to each other, or, other ratios derived from this fundamental ratio by shifting any or all of the dimensions by one or more integral octaves may be used. Other preferred proportions for less common shaped rooms are also shown in Figure 1.

We next deal with the shape and treatment of individual surfaces.

Shape of Reflecting Surfaces--Sound Diffusion: (5)

A pleasing reverberation characteristic depends not only on the proper reverberation time, but on a uniform rate of decay of sound. This requires a diffuse distribution of the after sound in a room and may be obtained by shaping and paneling the walls, ceiling, and other surfaces so as to disperse their reflections in all directions. Many rooms and halls known for their acoustic excellence have such wood paneling and sound-diffusing surfaces. For example, in the Philadelphia Academy of Music, which has several tiers of boxes and wood paneling throughout the auditorium, the dimensions of the projecting surfaces, being comparable to the wavelength of sound, tend to disperse the reflections and give a more diffuse distribution of sound. An important point regarding sound diffusion is that it does not lessen the total energy in the room, but merely tends to increase the number of reflections which occur per unit time and hence lessens the intensity level of the individual reflections. Because the difference in intensity level between reflections is less, the decay of sound in the room tends to be smoother and more uniform and hence more pleasing to a listener. This factor is also of practical importance in making the placement of microphones less critical.

A second factor of importance in these famous music rooms is that the wood paneling also helps to give a diffuse distribution of sound by virtue of the fact that the energy incident on its surface which is not absorbed is reradiated. (6)

This reradiated energy does not follow the regular law of equal angle of incidence and

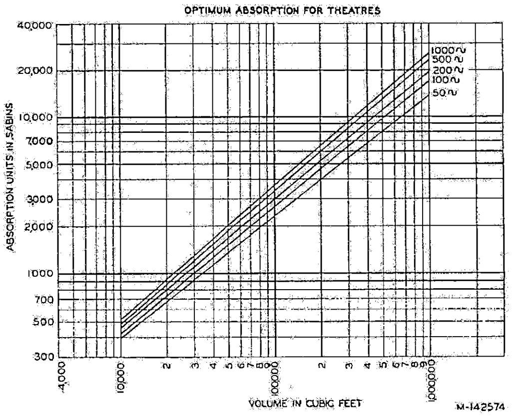

Figure 2--Optimum absorption.

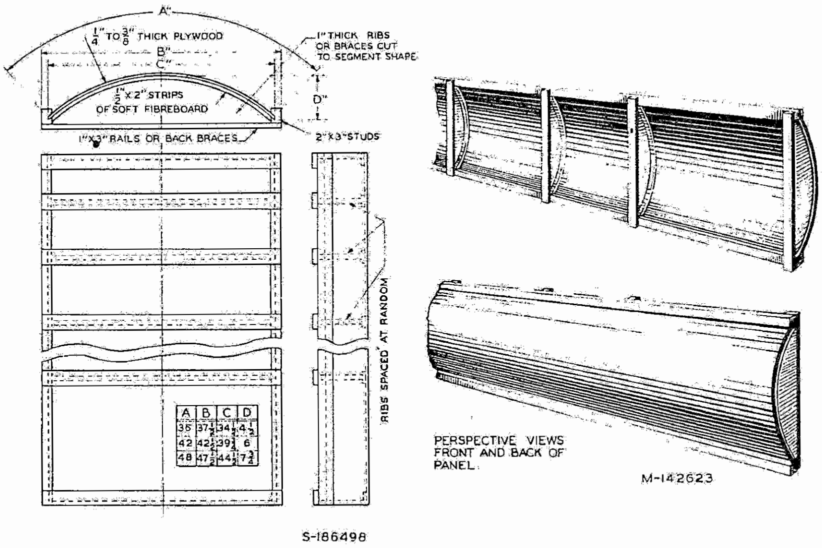

Figure 3--Convex wood panel.

reflection, but due to panel vibration acts more like a loudspeaker diaphragm and, therefore, for most frequencies either disperses the incident energy over a wide angle or changes its direction. A third important factor in these rooms is that the radiation from the wood paneling occurs further due to direct mechanical or "telephonic" transmission from the original sound sources on the stage through the wood flooring and panel-mounting structure. Since the wood panels are more or less of different sizes and shapes, the panel resonance frequencies are not selective. Two other interesting facts to note here are that panel radiators have a decay time of their own, and that the transmission time in wood is much shorter than in air.

This "diffuse" distribution of energy coming from many random directions and from a great number of small sources of sound in the room has, in addition to giving a more uniform distribution and decay of sound, the important psychological effect created by enveloping the listener with sound which gives a certain feeling of body or depth to the sound. This "feeling" effect of sound is further enhanced by the transmission of sound through the wood floor to the seats and from other structures in the room.

Sound Absorption Characteristics:

The sound-absorbing materials used for the control of reverberation and delayed reflections should have adequate efficiency at the low frequencies to give the optimum reverberation characteristic for the specified room size (see Figure 1).

The optimum sound-absorption units versus frequency corresponding to the optimum reverberation times for various room volumes is shown in Figure 2.

Convex Wood Panels: (5)

Figure 3 shows a typical curved wood panel consisting of 4' x 8' plywood formed over curved segment braces and fitted at the edges with 2" x 3" strips which have been routed and held together with 1" x 3" back braces. The segment braces which must be spaced at random have strips of 1/2" Celotex or other soft fiberboard placed between them and the paneling to prevent rattles. It is felt that convex cylindrical wood paneling is particularly suited to meet the aforementioned requirements of a good sound diffuser because it disperses sound energy not only by reflection from its curved surface but by radiation due to its resonance action or panel vibration which, as already mentioned, is set up, either by direct transmission from the original source of sound or by partial absorption and reradiation of the aerial sounds impinging on its surface.

Dispersion by reflection depends on the size and curvature of the panel and their relation to the wavelength. Dispersion characteristics for a curved and flat panel plotted on polar coordinates are shown in Figure 5. This shows clearly the diffusing action of convex curved surfaces and was obtained by rotating the panel about its long axis while keeping the source and directional pickup constant.

It should be noted here that the apparent source for convex reflectors is always behind the surface. The value of such surfaces in reducing the interference effect of first reflections as compared to flat surfaces is shown in figure 5. Note that the aiding effect at some frequencies and the cancellation at others is less for the convex panel. The relative intensity levels of the direct and reflected waves used in obtaining the interference curves are shown in figure 6. The reduction in interference effect is significant when we remember that the total energy content of the reflected wave from either surface remains the same. As mentioned earlier, the reduction of the interference effect of first reflection is important in studio microphone pick-up in allowing greater freedom of placement. For remote surfaces in large rooms a similar diffuse reflection may be obtained by means of concave reflectors providing the focus point or the apparent source of sound, which in this case is in front of the reflector, is not within or near the seating area or any other critical area.

Dispersion of sound by means of panel resonance depends on the modes of vibration set up. For areas of motion small compared to the wavelength radiated, the distribution will tend to be non directional. For vibrations normal to the surface, a convex cylindrical panel would tend to set up a cylindrical wave front as compared to the plane wave front in the case of flat panels. The resonance frequencies and response of a panel depend on a great number of factors such as the damping coefficient of the material, thickness, spacing of braces, method of mounting of the entire panel, etc. It is interesting to note that due to the added stiffness introduced by bending, a smaller panel thickness may be employed for curved panels for the same frequency. The decay time or "resonance time" of a typical panel excited at one of its modes of vibration is shown in figure 7. In addition to the dispersion and radiation action, the vibration of the wood paneling aids the absorption efficiency over more rigid materials.

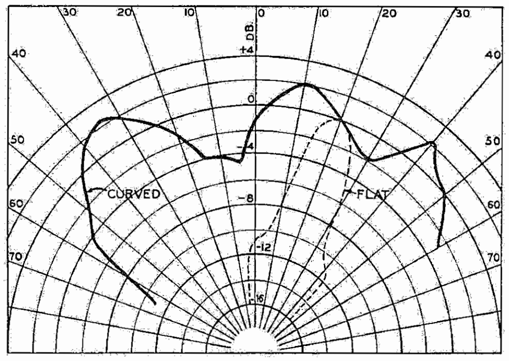

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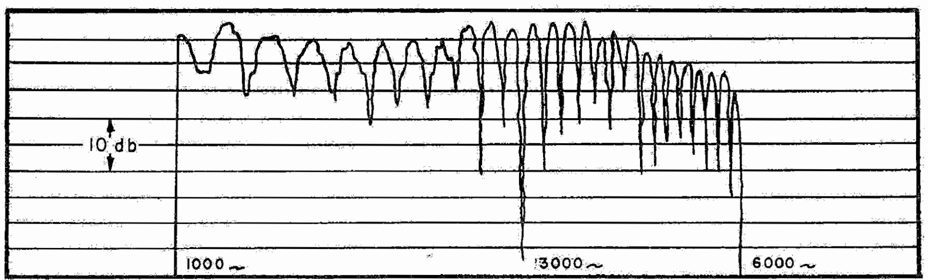
Figure 4--Polar distributions from convex and flat panels.

Preliminary data indicates a coefficient of approximately 0.18 over a considerable band of frequencies.

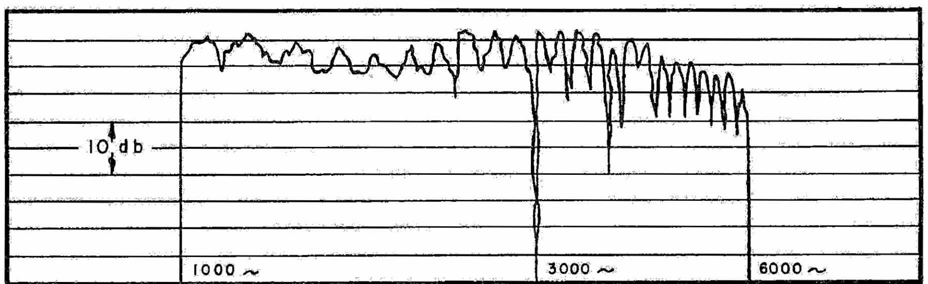
This absorption characteristic in conjunction with its dispersing action makes convex wood paneling particularly ideal for small studios.

Objectionable Shapes--Interference Phenomena:

Flat, untreated surfaces, if very extended in area or if close to the recording microphone, may give rise to objectionable interferences due to the phase difference between the direct and reflected sounds. The "loud" and "dead" spots caused at various frequencies when the direct and reflected waves are comparable in intensity level can become very pronounced and, due to

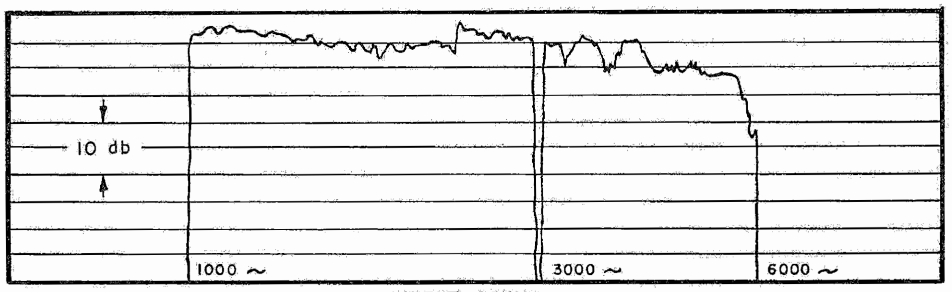


INTERFERENCE WITH FLAT PANEL

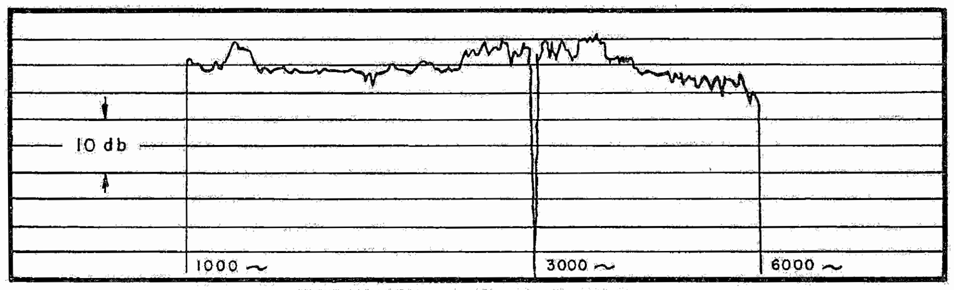


INTERFERENCE WITH CURVED PANEL

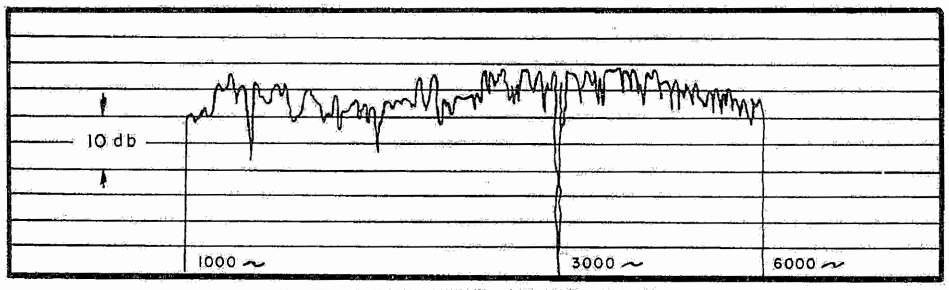
Figure 5--Interference between direct and reflected waves for convex and flat panels.



DIRECT SOUND



REFLECTION FROM FLAT PANEL



REFLECTION FROM CURVED PANEL

Figure 6--Direct and reflected response of panels used in interference measurements.

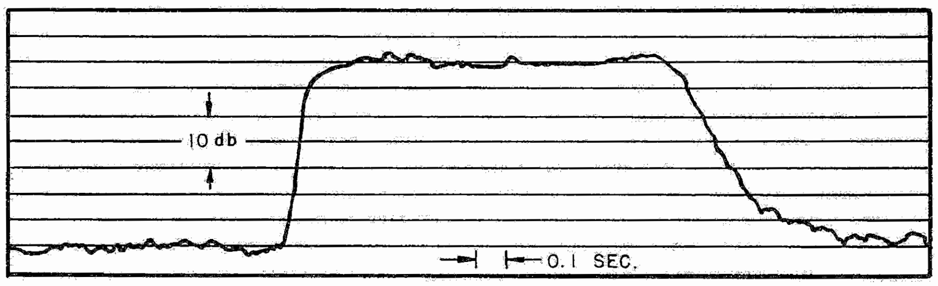


Figure 7--Growth and decay curve of vibrating convex panel.

frequency discrimination, give a hollow sounding effect. In view of this effect it is desirable that all flat surfaces in the studio have some absorption or else be shaped to disperse the reflection.

Concavely curved surfaces, even though treated with absorbing material, should be avoided. Due to their focusing effect, such surfaces accentuate the interference problems already discussed.

In larger rooms, reflections from concave surfaces and from large untreated flat areas give further trouble due to the echo or time delay effect and therefore should definitely be avoided.

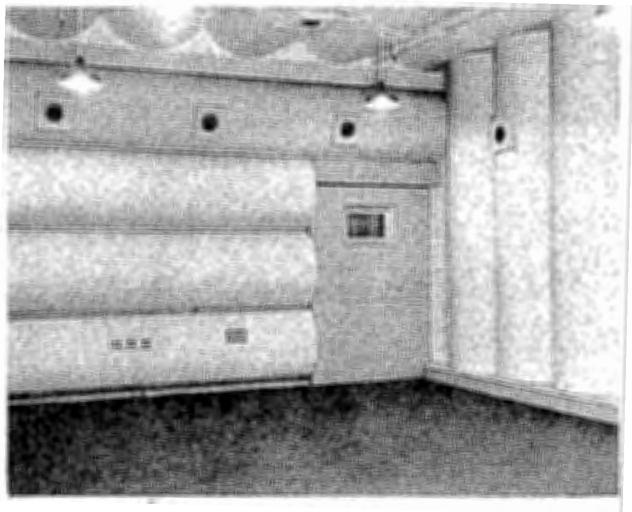


Figure 8--Polycylindrical studio.

Studio With Polycylindrical Wood Panels:

Figure 8 is a photograph of one of the studios in the RCA Sound Engineering Laboratories at Indianapolis, Indiana, used for listening purposes. In addition to the cylindrical wood diffusers already mentioned, these studios have several other features of interest in room acoustic design.

In order to obtain maximum diffusion in the three orthogonal planes, axes of the polycylindrical surfaces were placed mutually perpendicular to each other. Three different sized panels and curvatures were used further to obtain asymmetry in the pattern of diffused sound. The high degree of diffusion obtained in this type of room is evidenced in the smoothness of decay curves. The reverberation curve for this studio is shown in figure 9, which conforms reasonably well with published optimal times. (10)

Another feature of the studio is the choice of the major dimensions (12.5 ft. x 20 ft. x 32 ft.) which progress in two-third octave steps in order to avoid the "piling up" of room resonances. This is especially important as already pointed out for small rooms where the frequency gap between fundamental and harmonic resonances is oftentimes great, and unless guarded against may lead to a room response with wide hollows in the audible range. In the present case this consideration was important because the use of large convex diffusers effective in the range below approximately 300 c.p.s. was not considered practical. In large rooms consideration should be given to the use of low-frequency diffusers as well as for the mid range and high frequencies.*

*See Appendix II.

Orchestra Shell:

Occasionally regular sound stages (usually treated over the entire wall area with 4" rockwool and over the entire ceiling area with 2" rockwool to eliminate all reflections) are employed for musical recordings, in which case it is necessary to "liven" the stage.

An application of polycylindrical design to an orchestra shell which was used for recordings by Leopold Stokowski on the Walt Disney Live Action Stage, and which is here reproduced through their courtesy, is shown in figure 10. The multiplicity of dispersing surfaces and resonant panels not only minimized the microphone placement problem, but gave a more pleasing reinforcement of sound to the conductor and musicians. The platforms were for elevating and reinforcing the bass instruments in the orchestra.

Exclusion of Noise: (7)

A review room with good acoustics has its walls insulated against the transmission of outside noises into the studio. The transmission of sound is of two kinds: (a) aerial, and (b) structural. Small openings due to doors, windows, portholes, etc., transmit sound to a great degree. Thus, all the joints between walls, doors, windows, etc., should be made as air-tight as possible.

Likewise, transmission of sound through structures, such as the noise from vibrating motors and machinery, should be minimized by using massive walls and floors and by separating all vibrating bodies from their supporting structures by sound-insulating materials such as cork, lead, or rubber.

Massive walls are not always necessary to obtain sufficient sound insulation. A double wall of fairly light construction will give good sound insulation provided the two walls are not closely coupled mechanically by nails or cross-members, that is, provided the walls are kept isolated or separated from each other.

Projection Booth:

Treating of ceilings and walls inside the projection booth lowers the noise level, permitting more accurate control of volume and quality.

Treating ceilings and walls, by lowering the noise level, reduces the sound transmitted into the studio. Insulation of machines from floor by heavy blocks mounted on cork or Keldur is often necessary to reduce transmitted sound.

Double or triple optical glass in ports is usually used to reduce the sound conducted by the air through these ports.

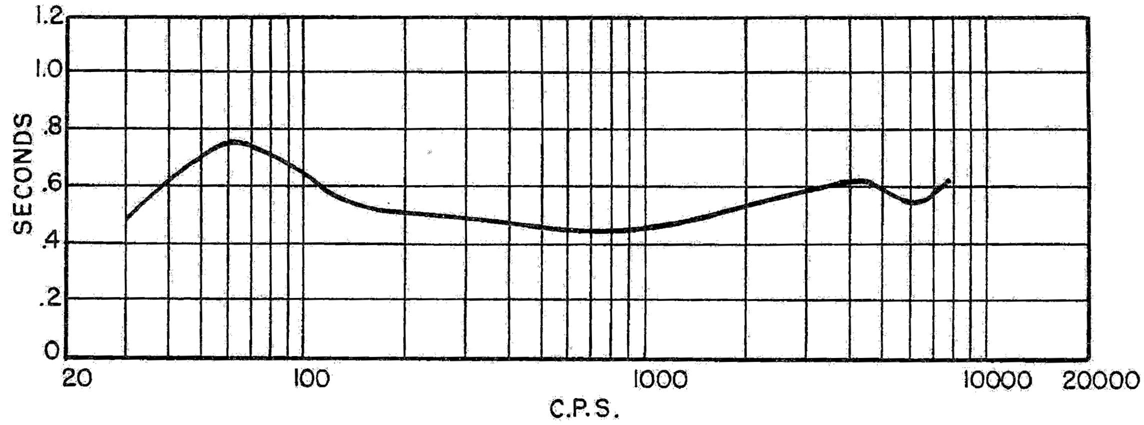


Figure 9--Reverberation time of listening studio.

The projection booth walls should be either very heavy (12" brick walls), or should be constructed with double walls which are isolated from each other so as not to conduct sound, by supporting them on some such material as felt and avoiding any construction which will tie the walls rigidly together.

Figure 11 shows a cross section of a typical projection booth window of the sound retarding type in accordance with principles outlined by National Broadcasting Company engineers and others. Figure 12 shows a cross section of a typical sound retarding door such as the Riverbank door used by broadcasting stations and other sound studios. Figure 13 illustrates construction of a typical sound insulating wall in accordance with principles stated by Bagenal and Wood in their book "Planning for Good Acoustics." (6)

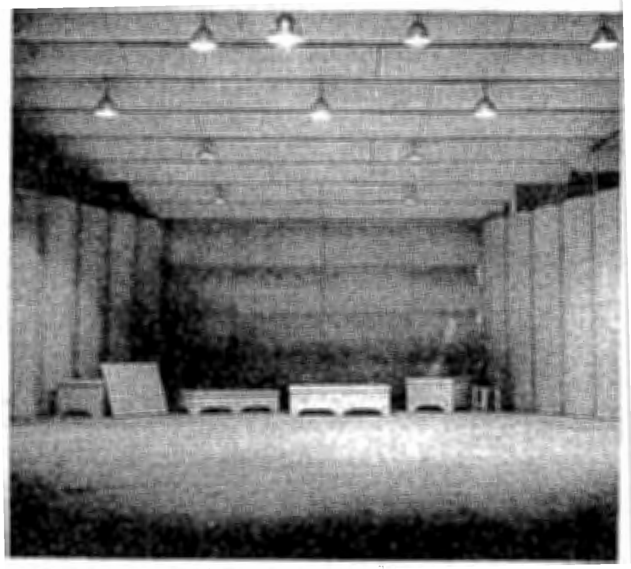


Figure 10--Polycylindrical Orchestra Shell.

ACOUSTIC DESIGN OF A TYPICAL SMALL COMBINATION STUDIO

The following specification covers, in accordance with the principles already outlined, recommendations on the acoustic treatment of a typical small combination studio.

Purpose of Studio:

The general purpose of this studio is to provide the following recording functions:

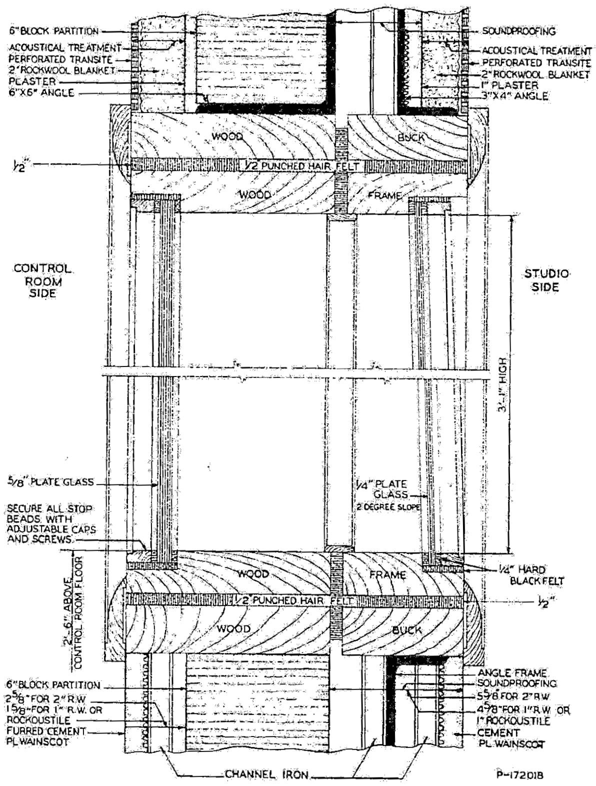


Figure 11--Sound insulated window.

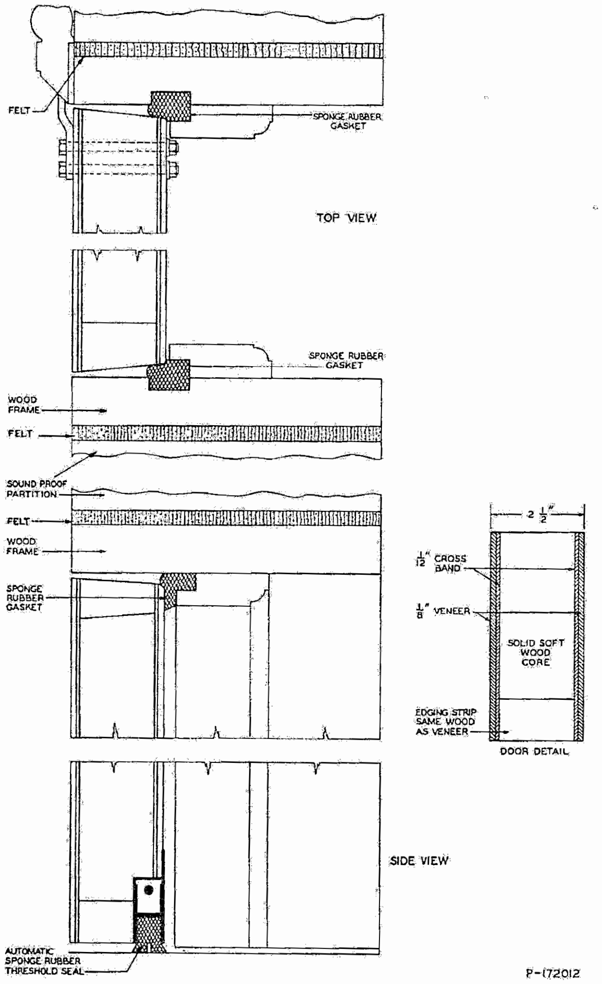


Figure 12--Sound retarding door.

1. Straight Recording (Voice or Music).
2. Pre or Post Scoring of Small Orchestras.
3. Rerecording.
4. Reviewing and Projection.
5. Monitoring and Control.

A plan view showing the general arrangement of rooms and equipment to serve these functions is given in figure 15.

Size and Proportions of Studio:

The studio proper shall have a minimum volume of 16,000 cubic feet and shall have its average height, width, and length proportioned in the ratio of 1: 1.6: 2.5, namely,

Height, 16 feet
Width, 25 feet
Length, 40 feet

A deviation in average dimension not greater than plus or minus 5% is permissible. (See figure 1.)

It should be noted that this studio constitutes the very minimum requirements commercially acceptable with regard to size, and will produce optimum results for orchestras of 10 pieces or less. When larger size orchestras are contemplated, the minimum requirements of 50-65,000 cubic feet (25'x40'x65') as practiced in Hollywood should be adhered to.

Shape of Studio:

The general shape of the studio shall be such that all reflecting surfaces in the room shall be convexly curved to disperse the sound in random directions. The proposed design with polycylindrical surfaces shown in figure 14 is recommended for this purpose.

Walls: All walls shall be treated with convex wood panels, general specifications for which are given in figure 3. The panels on the rear wall should be disposed horizontally while the front and side wall panels should be vertical an indicated in figure 14. All walls shall be

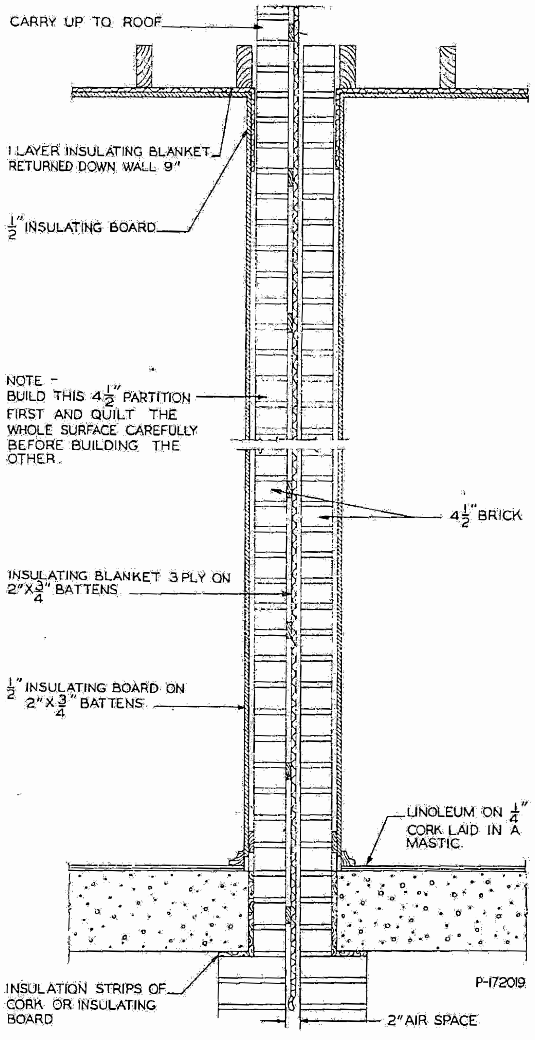


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Figure 13--Sound insulating wall construction.

insulated as specified later. Double-wall construction is particularly important between the projection booth and studio. A blanket of 1" rockwool or other sound-absorbing material should line the entire back of the wood paneling.

Ceiling: The entire ceiling shall be treated with convex wood panels which are disposed lengthwise to the room as shown in figure 14. The attic side of the ceiling should receive a granulated fill of rockwool or similar acoustical material to "load" the ceiling and to afford partial sound insulation against external noises.

Floor: The entire floor area (except in orchestra or bandshell area) should be covered with a good grade of reinforced carpeting with padding underneath. All seats should be of the heavily upholstered type.

Loudspeaker Chamber and Screen: The loudspeaker preferably should be mounted flush with the screen wall and all openings between them caulked to give a seal-tight chamber. The ceiling and back side of the screen wall forming the loudspeaker chamber should be lined with 4" of rockwool.

Sound Absorption Treatment:

No additional absorption is required for reviewing and scoring purposes to that provided inherently by the wood paneling and floor treatment. For dialogue recording it is desirable to furnish adjustable drapes on any nearby wall in the microphone area.

Exclusion of External Noise and Building Vibration:

A noise survey of the proposed studio site should be made by a qualified acoustical engineer before plans are drawn. From the viewpoint of excluding outside noises and building vibrations it is strictly essential that during erection all joints and openings between panels, etc., shall be fully caulked to give a continuous and seal-tight enclosure and further that the entire wall structure shall be floated on cork, rubber, or other suitable material, in order to completely isolate the walls from the main building structure. This precaution is extremely important since a single mechanical bridge or solid connection between the inner and outer shells caused by nails, pipes, ducts, etc., can almost completely nullify the sound insulation by setting the inner structure into vibration. Any bracing between inner and outer walls which may be necessary for structural reasons should receive individual isolation treatment to break the continuous mechanical connection. The various methods employed in building practice and patented methods for vibration isolation are too numerous to treat in these specifications. The underlying principle for preventing transmission through solids is to avoid a continuous medium (reinforced concrete, brick, etc.) or solid connection (wood, metal, etc.) by interposing a resilient or less dense material (cork, rubber, felt, air, etc.) in the link between the source of vibration and the reception point. In general the greater the number of such discontinuities (dense to less dense medium and vice versa) the greater the isolation effect. The ceiling and floor structure should receive similar caulking and vibration isolation treatment.

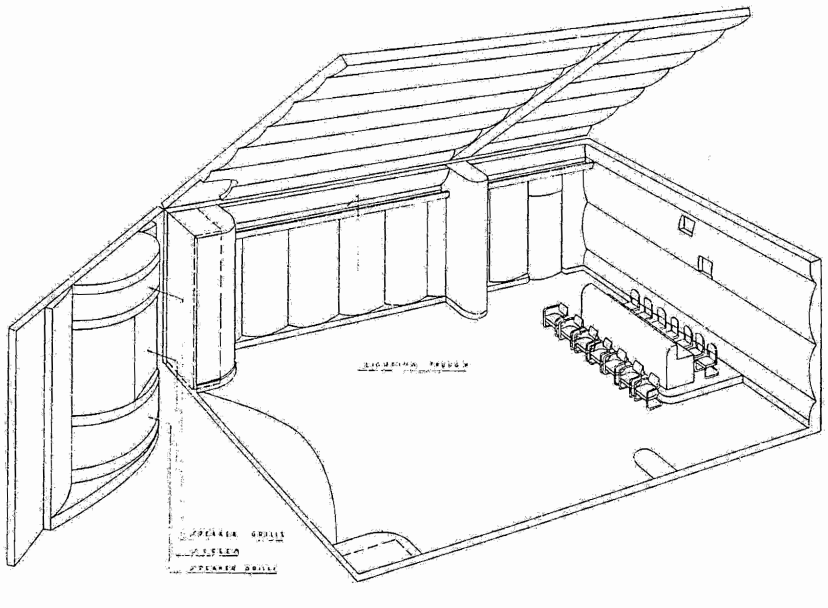

Figure 14--Proposed Design of a Small Film Recording Studio and Review Room.

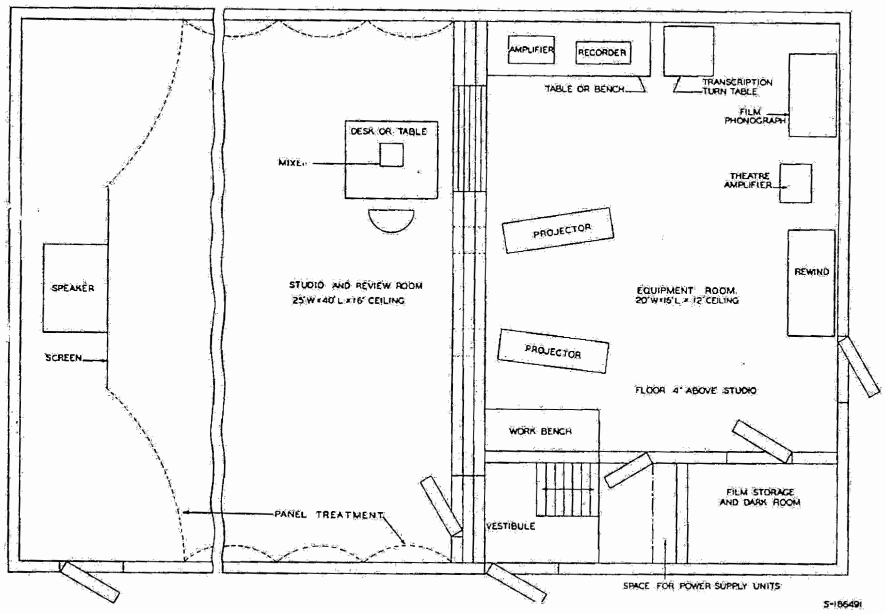

Figure 15--Proposed arrangement of a small studio.

Exclusion of Air-Borne Noises--Ventilation Ducts:

In addition to caulking all openings around pipes, ducts, etc., entering the studio with a mastic felt or other soft material, it is essential to line all ventilator ducts on the inside with an efficient sound-absorbing material and on the outside with a sound deadener for a sufficient distance from the point where they enter the studio so that the ambient noise level in the studio is not greater than 30 db. above the standard reference level of 10^{-16} acoustic watts per square centimeter, using a noise-level meter set on the 30 db. loudness contour characteristic. In general this represents a distance equal to at least 10 times the duct diameter. Extra precaution should be taken at the point where the duct enters the studio that outside noise does not leak through due to the canvas or other isolation coupling.

Recording and Projection Booth:

The size of the projection and recording equipment booth should be 20 ft. x 16 ft. with a 12-ft. ceiling height as indicated in the floor plan in figure 15. The booth floor should be elevated 4 ft. above the studio floor level and should be properly insulated according to the principles already outlined. (Either the entire booth floor or the areas under all vibrating equipment or both should be isolated from direct connections to the main building structure with 1" felted asphalt mastic.)

The ceiling and exposed wall areas above wainscot level should be treated with a fireproof sound absorbent.

Vestibule:

Entrance to the studio should be made only through double sound-insulating doors or a sound lock. Details of a typical sound-insulating door are shown in figure 12. The vestibule or sound-lock area may be used for storage or other non-noise-producing purposes as shown in the proposed studio floor plan.

SUMMARY

Acoustic Treatment Guide:

The main design features and considerations essential to good acoustics in small studios may be summarized as follows:

1. The size of the studio should be commensurate with the number and kind of musical instruments to be recorded. For a ten-piece orchestra the minimum volume would be 16,000 cubic feet. (For orchestras of other size refer to table I.)

2. The major dimensions of the studio should be proportioned to give a preferred ratio of average height, width, and length of 1 to 1.6 to 2.5. For a volume of 16,000 cubic feet this would represent a studio 16 ft. x 25 ft. x 40 ft.

3. Large parallel surfaces should be avoided.

4. All reflecting surfaces should be shaped, preferably convex, to give a

Figure 16--Scoring Stage of Republic Studios--
Courtesy Republic Productions, Inc.

diffuse distribution of sound and to minimize the effects of standing wave resonances.

5. If polycylindrical surfaces are used, their size and shape should be varied, and their axes disposed to be mutually perpendicular in the three orthogonal planes.

6. The characteristics of convex wood paneling are particularly well suited for supplying the necessary absorption and diffusing properties for studios. The panel bracings should be spaced at random distances to avoid selective resonance. The panels should be varied in size and backs lined with rockwool or other absorbing blanket.

7. The floor should be covered with a good grade of reinforced carpeting with padding underneath.

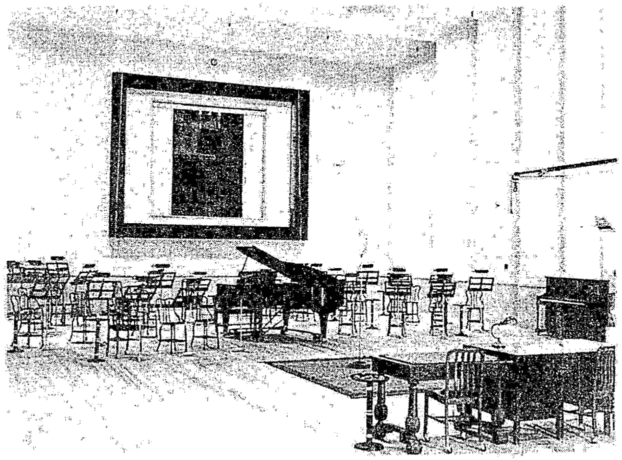

Figure 17--Scoring Stage of Columbia Studios--
Courtesy Columbia Pictures, Inc.

8. Seats should be of the upholstered type.

9. Adjustable drapes should be provided on wall surfaces in the microphone area for speech recordings.

10. Studio walls, ceiling, and floor should possess sufficient sound isolation to prevent the transmission of extraneous noises into the studio.

11. The projection booth should be treated with a fireproof sound absorbent on ceiling and exposed wall areas above wainscot level.

12. All machinery and vibrating equipment such as arc generators, voltage regulators, ventilators, etc., should be acoustically isolated from the studio.

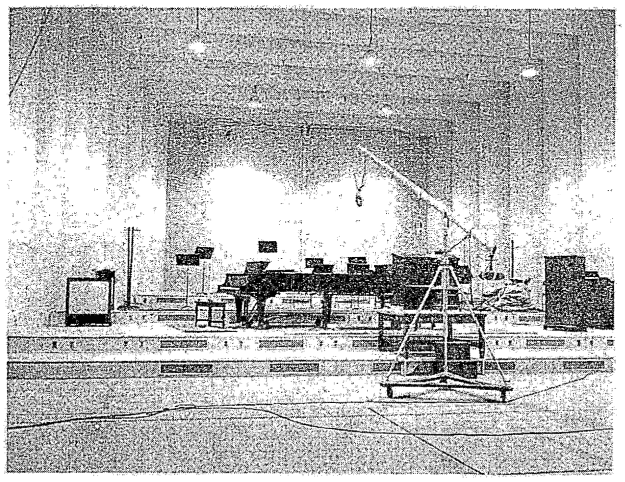

Figure 18--Walt Disney Scoring Stage--Courtesy Walt Disney Productions, Ltd.

13. Air-conditioning equipment should be of the low-velocity type and preferably operated to give full volume at half speed. All ducts should be provided with acoustic baffling and lined for a distance of at least 10 diameters from the studio.

The foregoing general recommendations on the acoustic treatment for studios are offered merely as a guide. The services of a qualified acoustic engineer should be relied upon for the exact specifications relative to the proper type, amount and location of acoustic treatment, shape of studio, and type and amount of sound insulation required. The trend today is toward close cooperation between architect and engineer, and toward functional styling based on the acoustic design. In the case of new studios, a noise survey of the proposed site and complete acoustic specifications for the studio are desirable and worth while.

In illustrations or references to acoustical insulating materials, in this book, specific materials mentioned are listed only as an example of a material suitable for the purpose described. Any equivalent acoustical material may be used with similar results.

REVERBERATION CHAMBERS FOR RERECORDING*

In the recording of sound on film or wax it is frequently desirable to add a reverberatory quality to the recording after its completion. This may be accomplished by reproducing the sound in a highly reverberant room--the so-called reverberation chamber--and "mixing" the output from a microphone in this room with the original recording in a process known as "dubbing" or rerecording.

Surprisingly, when the electrical level of the reverberated signal is as much as 20 db below the electrical level of the original recording at the mixing panel, the combined reproduced signal conveys a strong impression of reverberation in every syllable of speech, and chord or passage of music.

Unlike in other means, electrical or mechanical, of adding a reverberatory note to a recording, the chamber method provides both the proper growth characteristic and the decay quality of sound in a live enclosure. Delay networks, magnetic tape recordings, and other devices for achieving synthetic reverberation usually permit only provision for the decay characteristic; no attempt is made to introduce the growth characteristic, since the latter is less essential in an approach to total reverberation.

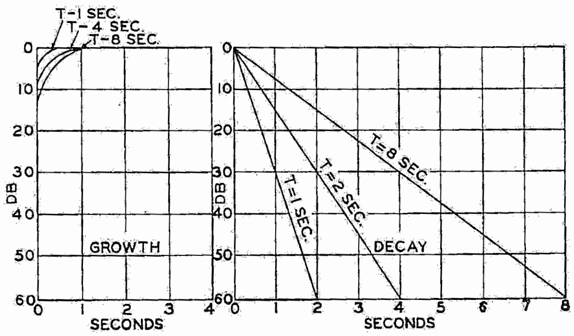

Figure 19--Curves Illustrating Growth and Decay of Sound in Different Rooms.

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*Condensed from article in Journal of the Society of Motion Picture Engineers. Vol. 45, No. 5 (November, 1945), Pp. 350-357.

It is interesting to plot the growth curves of sound and the corresponding decay characteristics for a number of rooms. Figure 15 shows the sound-growth and sound-decay curves for rooms that have different reverberation times, namely 8 sec., 4 sec., and 1 sec. The curves are plotted on the assumption that the power output of the source in the different rooms is such as to provide the same value of steady-state energy-density in each enclosure.

For the addition of reverberation to recordings made on film, a room with a reverberation period of approximately 4 sec. appears adequate. A chamber of 4000 cu ft volume, with walls and ceiling made of concrete, answers this purpose very well. If the mean dimensions for the height, width, and length of the enclosure are 12.5 ft, 16 ft, and 20 ft, respectively, the total interior surface comes to approximately 1540 sq ft. Crediting concrete, 6 in. thick, with an absorptivity of 0.03 sabine at 1000 cycles, the total absorption comes to 46.2 sabines, and the reverberation time therefore to 4.33 sec. Mean dimensions are indicated because the preferred shape of the enclosure is nonrectangular, in order to avoid flutter echoes.

It is interesting to consider the decay characteristic of the sound which actuates the microphone in the chamber. The sound which was originally recorded in the recording stage is itself characterized by the reverberation of the studio. During reproduction the decay characteristic of the chamber is superimposed upon that of the stage. Thus, superimposing a reverberation period of 4 sec. upon sound recorded in a room with a reverberation period of one second has, for short initial intervals, the effect of sound decaying in a room of 8 or more sec. reverberation. This may account for the low electrical level of the reverberated signal necessary (at the mixing console) for its combination with the original recording to obtain the desired reverberatory character in the rerecording.

Another reason for the low electrical level required of the reverberated signal is the fact that the microphone represents only one ear. In binaural hearing, the ear is to some extent capable of suppressing unwanted sound, whether direct or reflected, and to concentrate only on the desired sound. This can be readily demonstrated by closing one ear in the reverberation chamber, in which case the reverberation appears considerably prolonged.

Another reason for maintaining a low electrical level for the reverberated signal is, of course, an attempt to preserve as much as possible the intelligibility of the dialogue. The ear is evidently able to judge the reverberation of a room by the audible, slow trailing-off of the sound intensity at the end of *words*.

In the case of music, where longer reverberation tends to provide a richer or more pleasing quality, the electrical level of the reverberated signal is kept higher. Still, the definition of instruments can be preserved remarkably well by this means. One may indeed consider whether this type of reproduction does not supply a superior rendition, unattainable in any other way, since both clarity of instruments and an undertone of prolonged decay exist simultaneously.

Different recordings call for the addition of different amounts of reverberation, and sometimes, *for the addition of different types*, of reverberation characteristics. The reason for this is that, in sound-on-film recordings, a large number of different sound effects have to be included. If the dialogue is recorded in a cellar, tunnel, hull of a ship, etc., but the incident sounds (footfalls, jack-hammer drives, engine noise) are not included in the original recordings, it is desirable to match the character of these effects with the "room-tone" existing in the surrounding in which the speech was recorded. Some variation in the reverberation characteristic can be effected by changing the distance between loudspeaker and microphone, since this will change the ratio of direct-to-reflected sound at the microphone.

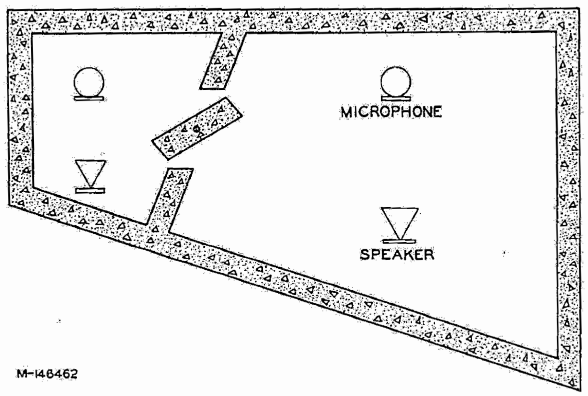

Figure 20--Double Reverberation Chambers

The character of the reverberated signal may be altered more perceptibly by dividing the chamber into a small and a large room. Since the number of normal modes and their spectral distribution is decidedly different in two such enclosures a considerable variation in the quality of the signals may be effected by using one or the other of the two rooms. If a door is provided in the partition between the two chambers, a further variation will result by placing the speaker in one of the rooms and the microphone in the other. Such a door acts like an acoustic high-pass filter, thereby making the reverberation characteristic of each room a function of the door opening.

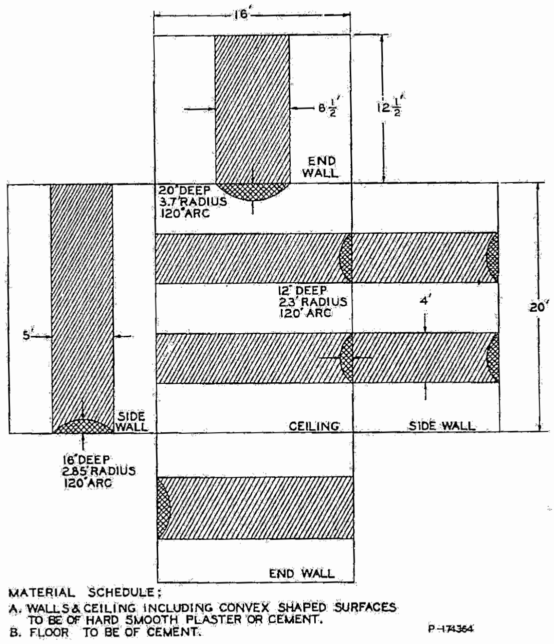

Figure 21--Developed Plan for Echo Chamber

A change in the character of the sound picked up in the chamber can, of course, also be secured by using different microphones.

Figure 20 represents the plan of a dual reverberation chamber, of which the walls as well as the ceiling and the floor were kept at a slant to avoid echoes. The dual chambers employ concrete for the material of the walls, the ceiling, and the floor. A massive door, 5 ft wide and 6 ft high, weighing approximately 450 lb, may be rotated by means of a knob located at the rerecording console; in this manner it is possible to change the reverberation characteristics of the rooms easily and quickly. Where it is not feasible to employ dual chambers and slanting walls an alternative plan for the chamber is shown in figure 21.

Figure 22 shows a block schematic of a recording channel employing a reverberation chamber.

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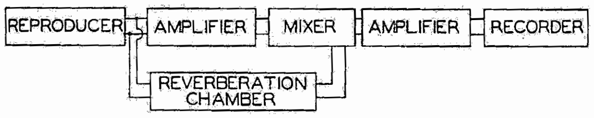

Figure 22--Block Diagram of Rerecording Channel

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APPENDIX I

Definitions in Architectural Acoustics From ASA Standards Z24.1

Acoustic Absorptivity (Absorption Coefficient)

The acoustic absorptivity of a surface is equal to one minus the reflectivity of that surface.

Acoustic Reflectivity (Reflection Coefficient)

The acoustic reflectivity of a surface not a generator, is the ratio of the rate of flow of sound energy reflected from the surface, on the side of incidence, to the incident rate of flow. Unless otherwise specified, all possible directions of incident flow are assumed to be equally probable. Also, unless otherwise stated, the values given apply to a portion of an infinite surface thus eliminating edge effects.

Acoustic Transmittivity

The acoustic transmittivity of an interface or septum is the ratio of the rate of flow of transmitted sound energy to the rate of incident flow. Unless otherwise specified, all directions of incident flow are assumed to be equally probable.

Cycle (∿)

One complete set of the recurrent values of a periodic quantity comprises a cycle.

Diffuse Sound

Sound is said to be in a perfectly diffuse state when in the region considered, the energy density, averaged over portions of the region large compared to the wavelength, is uniform and when all directions of energy flux at all parts of the region are equally probable.

Echo

An echo is a wave which has been reflected or otherwise returned with sufficient magnitude and delay to be perceived in some manner as a wave distinct from that directly transmitted.

Flutter Echo

A flutter echo is a rapid succession of reflected pulses resulting from a single initial pulse. If the flutter echo is periodic and if the frequency is in the audible range, it is called a musical echo.

Frequency (f)

The number of cycles occurring per unit of time, or which would occur per unit of time if all subsequent cycles were identical with the cycle under consideration is the frequency. The frequency is the reciprocal of the period. The unit is the cycle per second.

Intensity Level (IL) (Sound Energy Flux Density Level)

The intensity level, in bels, of a sound is the logarithm to the base ten of the ratio of the intensity I of this sound, to the reference intensity I_0. Intensity level may also be expressed in decibels.

Interference Pattern

An interference pattern is the resulting space distribution of pressure, particle velocity, energy density or energy flux when sound waves of the same frequency are superposed.

Mean Free Path

The mean free path for sound waves in an enclosure is the average distance sound travels in the enclosure between successive reflections.

Multiple Echo

A multiple echo is a succession of separately distinguishable echoes from a single source.

Noise

Noise is any undesired sound.

Rate of Decay (of Sound Energy Density) (S)

The rate of decay of sound energy density is the time rate at which the sound energy density is decreasing at a given point and at a given time. The practical unit is the decibel per second.

Reverberation

Reverberation is the persistence of sound, due to repeated reflections.

Reverberation Time (T)

The reverberation time for a given frequency is the time required for the average sound energy density, initially in a steady state, to decrease, after the source is stopped, to one-millionth of its initial value. The unit is the second.

Sabin

The sabin is a unit of equivalent absorption and is equal to the equivalent absorption of one square foot of a surface of unit absorptivity; i.e., of one square foot of surface which absorbs all incident sound energy.

Sound

(a) Sound is an alteration in pressure, particle displacement or particle velocity propagated in an elastic material or the superposition of such propagated alterations. (b) Sound is also the sensation produced through the ear by the alterations described above.
Note--In case of possible confusion, the Term "Sound Wave" may be used for concept (a), and the term "Sound Sensation" for concept (b).

Wavelength (λ)

The wavelength of a periodic wave in an isotropic medium is the perpendicular distance between two wave fronts in which the displacements have a phase difference of one complete cycle.

APPENDIX II

Sectional Dimensions For Convex Cylindrical Panels

α	60°			90°			120°			180°
Arc (a)	Width (c)	Depth (b)	Radius (r)	Width (c)	Depth (b)	Radius (r)	Width (c)	Depth (b)	Radius (r)	Radius (r)
16"	15-1/4	2	15-1/4	14-3/8	3	10-3/8	13-1/8	3-7/8	7-5/8	5-1/8
32"	30-5/8	4-1/8	30-5/8	28-3/4	6	20-3/4	26-1/4	7-5/8	15-1/4	10-1/8
48"	45-7/8	6-1/8	45-7/8	43-1/4	9	31-1/4	39-1/2	11-1/2	22-7/8	15-1/4
64"	61-1/8	8-1/4	61-1/8	57-5/8	12	41-5/8	52-5/8	15-1/4	30-1/2	20-3/8
80"	76-3/8	10-1/4	76-3/8	72	15	52	65-3/4	19-1/8	38-1/8	25-1/2
96"	91-5/8	12-1/4	91-5/8	86-3/8	18	62-3/8	78-7/8	23	45-3/4	30-1/2
112"	107	14-3/8	107	100-3/4	21	72-3/4	92-1/8	26-3/4	53-3/8	35-5/8
128"	122-1/4	16-3/8	122-1/4	115-1/4	23-7/8	83-1/4	105-1/4	30-5/8	61	40-3/4
144"	137-1/2	18-3/8	137-1/2	129-5/8	26-7/8	93-5/8	118-3/8	34-3/8	68-5/8	45-3/4