LOUDSPEAKERS—Can we measure what we hear?

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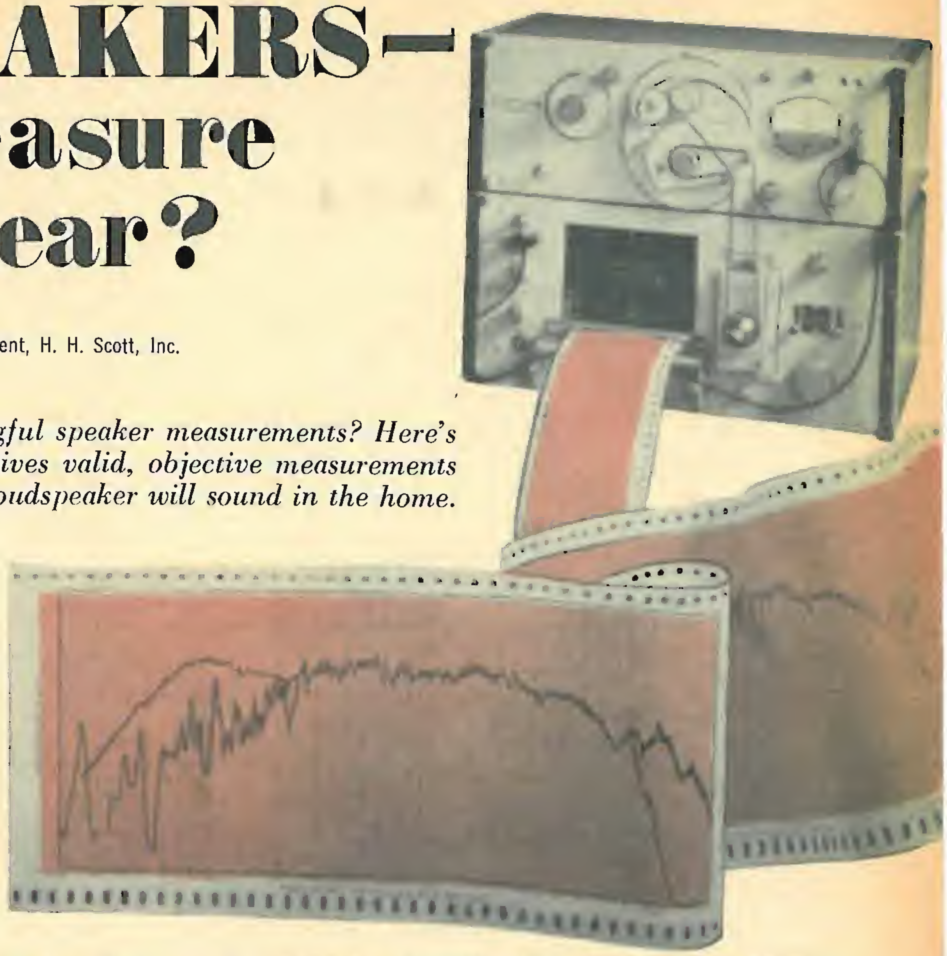
Is it really possible to make meaningful speaker measurements? Here's a reverberant-room technique that gives valid, objective measurements that are useful in predicting how a loudspeaker will sound in the home.



TO the novice, measurement of the frequency response of a loudspeaker system appears to present no more problems than the measurement of an amplifier. Considerable irritation is sometimes expressed at the aura of mystery and the atmosphere of controversy which surround loudspeaker measurements. There is often a feeling that the "experts" are going out of their way to make matters unnecessarily complicated. Is this criticism justified? Is the subject really so complicated? Or is it possible to make relatively straightforward measurements that convey meaningful information? This article will attempt to shed some light on these questions.

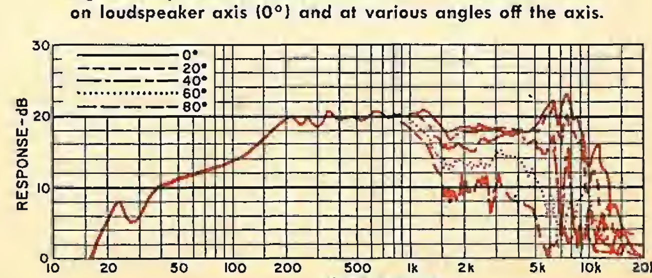
Let us begin by sidestepping temporarily the question of the relationship between the measurement and what a speaker sounds like. All we want to know at this point is the frequency response of the loudspeaker. We immediately encounter the difficulty that the loudspeaker, unlike an amplifier, has no output terminals to which we can conveniently attach a measuring instrument, because the output that we want to measure is not electrical in nature, but acoustical. Consequently, we have to use a microphone to make the measurements. This in itself presents no particular problem; there are microphones available that have extremely wide range, flat response, and can be obtained accurately calibrated. When such a microphone is connected to a suitable amplifier and indicating voltmeter or recording instrument, we have a suitable system for measuring the acoustic output of the loudspeaker.

The difficulties begin when we attempt to use this measuring system. In what kind of environment shall we operate the loudspeaker and where shall we place the microphone? Taking up the second question first, we know of course that the output from a loudspeaker varies with the distance, so we have to decide upon a distance at which to make our measurement. At first glance this presents no particular problem because, remembering our elementary physics, we know from the inverse square law of sound propagation that it is always possible to relate the sound intensity at any given point to that at any other distance from the source. We are not quite so likely to remember that this applies only to a point source, but if we do, we conclude that it is merely necessary to make the measurement at a location sufficiently far from the loudspeaker to insure that it behaves like a point source.

If loudspeakers radiated sound equally well in all directions at all frequencies, there would be no problem involved in deciding whether to place the measuring microphone on-axis or at some point off the axis of the loudspeaker. But practical loudspeakers do depart from perfect omnidirectionality to a considerable extent; consequently, some kind of a decision has to be made. In the days of monophonic sound reproduction it was often argued that practically all listening was done on or near the axis of the loudspeaker, and that consequently the most significant measurement was the one made on the axis. There may have been a little difficulty in defining the axis exactly (except for a perfectly symmetrical loudspeaker) as for example in the case of a two-way system, but the problem could be minimized by making the measurement at a sufficiently great distance from the loudspeaker so that it did not make very much difference. This line of reasoning does not apply to today's use of loudspeakers for stereophonic reproduction. Obviously, if the listener is positioned midway between the left and right speakers, he is not on the axis of either one of them.

Although we started out with the assumption that we would not allow subjective considerations to influence our

Fig. 1. Response measurements taken in an anechoic room on loudspeaker axis (0°) and at various angles off the axis.



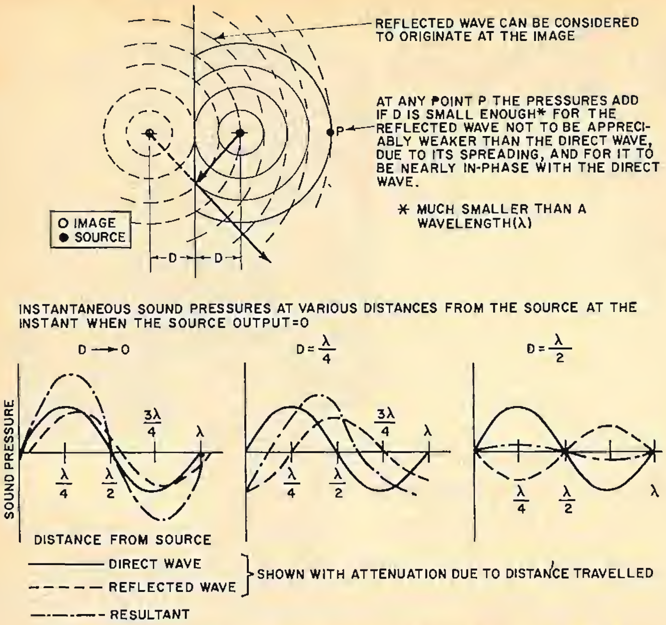


Fig. 2. With reflecting surface placed immediately behind the speaker, the results are as shown in this illustration.

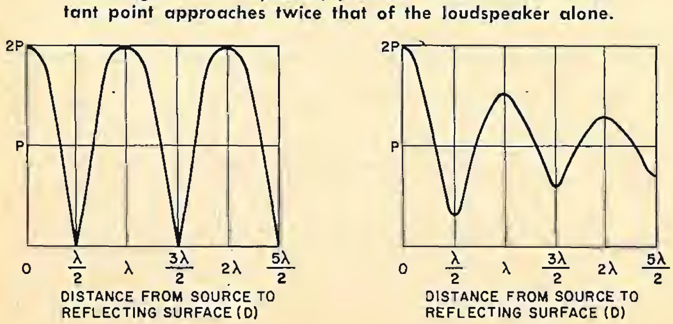
judgment, we suddenly find ourselves entangled in a discussion of where the listener is positioned with respect to the loudspeakers. The listener sneaked in while we were not looking and there does not seem to be any way of getting rid of him.

The Acoustic Environment

Perhaps it would be the better part of valor to retreat to a consideration of the other topic that we started with—the acoustic environment. The previous discussion has assumed that all the sound received by the measuring microphone comes directly from the loudspeaker. This would hold true in a space free of reflecting surfaces, such as a wide-open space well removed from the ground or, more practically, an anechoic chamber. It must be obvious to the reader that we are now heading towards a repetition of the old statement, "but a loudspeaker is not listened to in open space, nor in an anechoic chamber." This is inconsistent with our starting premise that the subjective aspect was going to be disregarded. We had stated that we were interested only in the frequency response of the loudspeaker. Why don't we decide that the measurement is going to be made in an anechoic chamber, or its equivalent, and let it go at that? Matters become considerably simplified if this decision is made, except for the fact that we are then driven to a reconsideration of the question of where to place the measuring microphone.

The entire problem can be solved by means of a bold step: measurements will be made at many points, both on-axis and off-axis, and the results presented as a series of curves showing frequency response on-axis and at various angles off-axis (Fig. 1). This is a perfectly logical scheme and there is no question but that it completely defines the frequency response of a loudspeaker in an anechoic environment. The question is: of what use is this information to the user of the loudspeaker and to the loudspeaker designer?

Fig. 3. When reflecting surface is much less than half a wavelength behind speaker, pressure of the sound at distant point approaches twice that of the loudspeaker alone.



The response curves obtained in the manner outlined provide full information regarding the smoothness of response of the loudspeaker and are useful in comparing one loudspeaker with another. This is strictly true only in the case of single loudspeaker units. With multi-speaker systems it is usually not practicable to make measurements at a distance sufficiently great to avoid errors caused by the measuring microphone's being on the axis of one loudspeaker and off the axis of another, or somewhere in between. In the regions of crossover between speakers, such as between the woofer and a midrange speaker, there is considerable interference between the two speakers in the frequency range where their outputs overlap. This can result in peaks or valleys in the response curve. This may occur on-axis and not at some angles off the axis, or *vice versa.* The designer is often confronted with the dilemma that while he can design for smooth response on-axis at the expense of rough response off-axis, or the reverse, he has no way of telling which is the preferred condition. In multi-speaker systems the off-axis responses tend to become quite irregular because of the finite size and spacing of the radiating elements, which cause interference effects. These are also extremely difficult for the designer to evaluate.

When the user examines these curves he is apt to be rather surprised to find that all loudspeaker systems, even the very best, show a progressive decrease in response as the frequency goes down below several hundred hertz. Is it really true that all loudspeakers are deficient in bass response? Yes, it is—in an anechoic environment; however, when operated in rooms, as they are in practice, the bass response is considerably better. It is possible, but not particularly easy, to determine the bass response of a loudspeaker when operated in a room by calculation, starting with the frequency response curves made in an anechoic chamber. This is certainly beyond the scope of the average hi-fi fan. It would be useful to have a method of measurement which would provide this information directly.

Effect of Reflecting Surfaces on Bass

Since it appears to be impossible to avoid considering the effects of the listening room, an outline of the way it affects the performance of a loudspeaker might lead to some conclusions regarding a suitable method of measurement. Consider what happens when making a measurement in free space, if a large, perfectly reflecting surface is placed immediately behind the loudspeaker. To simplify the discussion, assume that the loudspeaker is perfectly omnidirectional.

The sound from the rear of the loudspeaker is reflected by the surface so that the sound direction is reversed and it reaches the measuring microphone in addition to the sound arriving directly from the loudspeaker. Instead of one sound wave reaching the microphone, two identical waves fall upon it. The sound pressure at the microphone is doubled, or increased by 6 dB. Since intensity, or acoustic power per unit area, is proportional to the square of the sound pressure, the intensity is multiplied by four.

Note that the area into which the speaker radiates has been reduced by a factor of two by the surface that has been introduced. The power output of the speaker itself is equal to the intensity (power per unit area) times the area; this must be doubled when a reflecting surface is immediately behind the speaker. This can be visualized by considering that the diaphragm must move twice as fast in a smaller volume, *i.e.*, in a half-sphere rather than a full sphere.

It is necessary to define what is meant by "immediately behind the speaker." If the reflecting surface is spaced from the radiating element by a very small fraction of a wavelength, the reflected sound travels from the radiating element to the surface and then back out again so that it is only slightly delayed with respect to the direct sound. It is convenient to think of the reflected sound as originating from an image, spaced behind the reflecting surface as shown in Fig. 2.

If the source and image are very close together, their outputs add up to only slightly less than double their original values. As the distance from source to surface is increased, one wave becomes displaced in space with respect to the other, and they reinforce each other to a lesser and lesser extent until finally, when the displacement is equal to one-half wavelength, they cancel each other, so that their sum is equal to zero. (See Fig. 3A.) When the distance is still further increased, the waves begin to reinforce each other again, with the sum reaching a maximum when the distance is one wavelength, and again decreasing to zero when it is one and a half wavelengths. The pattern is repeated endlessly. By the way, it is assumed that the distance to the observation point is so much greater than the distance from the source to the wall that the direct and reflected waves have essentially equal amplitudes. If the reflected wave has less amplitude than the direct wave, the resultant is as shown in Fig. 3B.

Fig. 3A shows that for a distance less than half a wavelength from source to reflecting surface, the sound pressure at a given point is always increased, but that for greater distances it varies from double its original value to zero. Consequently, the term "immediately behind the speaker" means that the distance must be a small fraction of a wavelength. This is generally the case at low frequencies.

For a given distance from source to reflector, the frequency response at a point on axis looks like Fig. 4. Responses off-axis look similar. The directivity patterns are shown in Fig. 5, in which the horizontal line at the bottom of the illustration represents the axis through the source and its image. The total power is obtained by integrating the values for all the lobes. For frequencies equal to and above those corresponding to Distance (between source and image) = $\lambda/2$, the directivity index (D.I.) for the axis of any lobe is 3 dB. (The directivity index is equal to 10 times the logarithm of the ratio of the intensity to the intensity that would exist if the source were omnidirectional and radiating the same total power as the directional source.) Since all the lobes have maxima that are equal in strength to the response obtained under omnidirectional conditions (distance equal to a small fraction of a wavelength), it follows from the value of the directivity index that their total power must be 3 dB below the power obtained when the wavelength is large.

We have already pointed out that at low frequencies the sound intensity increases by 6 dB. At higher frequencies it is 3 dB less. *The net result is a boost of 3 dB at low frequencies.*

A second reflecting surface at right angles to the first provides a *net gain* in intensity of 6 dB. If a third reflecting surface is added, the intensity is increased by 9 dB. In Fig. 6 these conditions correspond to a loudspeaker in free space, a loudspeaker on the floor in the center of a room, one placed on the floor against the wall, and finally a loudspeaker in the corner of a room.

Fig. 7 shows the results of measurements on a mid-range loudspeaker in the different positions indicated. The increase in sound pressure level at the lower frequencies accords with the theory, and the curves illustrate how the effect decreases as the frequency is increased. It is also apparent, when the shapes of these curves are compared to anechoic-chamber measurements of the same speaker, the latter do not correctly depict the low-frequency performance of loudspeakers when they are used in rooms.

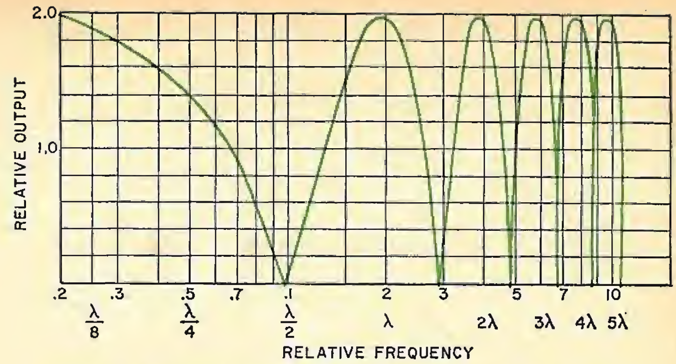


Fig. 4. Relative frequency response at a point on the loudspeaker axis for a given distance from source to reflector.

Multiple Reflections

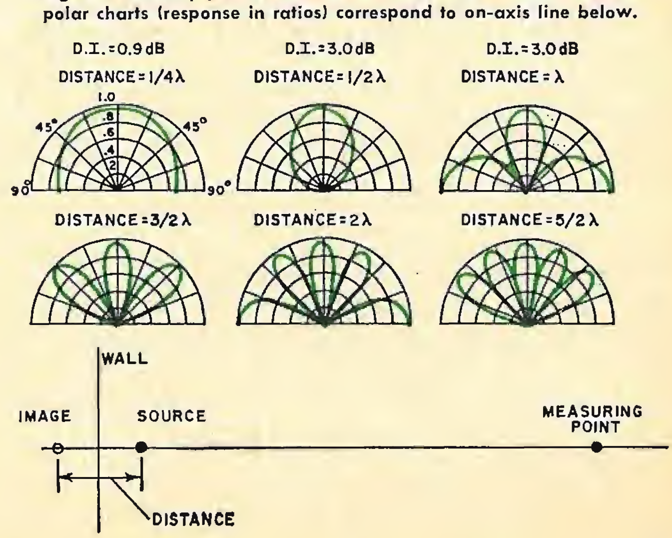
Let us now consider the effect on loudspeaker performance of multiple reflections in rooms. Fig. 8 shows a plan view of a room containing a loudspeaker at S, and a listener at *O*. The sound traveling directly from the speaker to the listener is shown by the double line marked *D*, for direct sound. Sound also reaches the listener after one or more reflections from the walls.

The diagrams at the right show how the energy reaching the listener is built up following the direct sound, by successive reflections. Curve *A* shows the effect of several rays of sound that reach the listener after one reflection from the walls of the room. The intensity of this reflected sound is somewhat lower than that of the direct sound because there is some loss upon reflection, and also due to the longer paths traversed. At *B*, the build-up of sound is shown for two reflections from the walls. The reflections become progressively weaker; consequently the curve showing the build-up begins to bend over as shown.

It should be kept in mind that the situation is depicted in two dimensions only for the sake of simplicity. Actually, sound is reflected from the floor and ceiling as well. As time goes on, the various sound paths fill the room, crisscrossing in a rather random manner, and the curve levels off to a steady-state condition when all of the sound emitted by the source is absorbed at the walls and through transmission losses in the air. The total reflected sound is referred to as *reverberation.*

Just as the reverberation of a room causes a gradual build-up of a given sound, it also causes a gradual decay when the sound stops. The rate of build-up and decay is determined by the volume and the total absorption of the room. The length of time required for the sound to decrease from

Fig. 5. Directivity patterns for speaker near wall. Vertical axes in polar charts (response in ratios) correspond to on-axis line below.



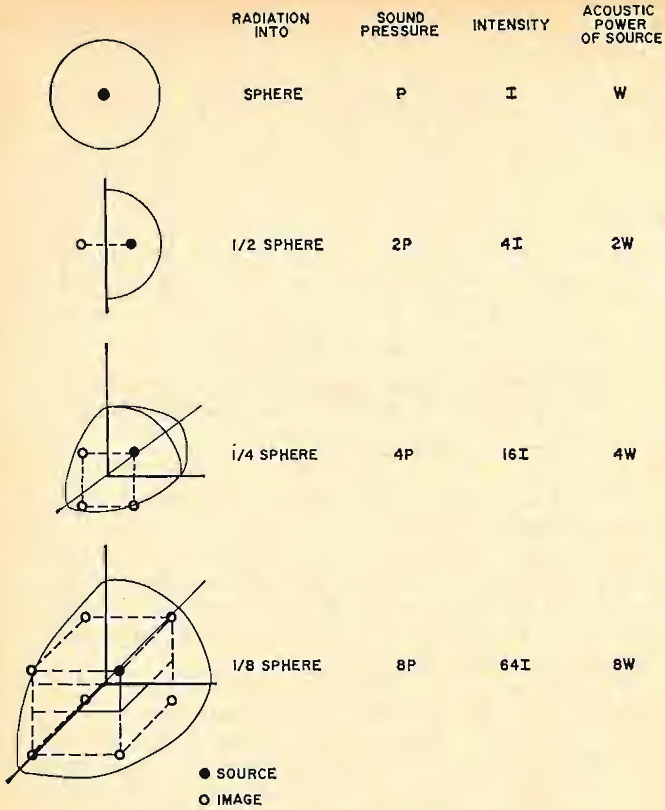


Fig. 6. As the speaker is moved from free space, to a floor in the center of a room (one reflecting surface), to the floor and against a wall (two reflecting surfaces), and finally to the corner of a room (three reflecting surfaces), the sound intensity produced increases proportionally as shown here.

its steady-state value, after the source has been turned off, to a level 60 dB below this value, is called the *reverberation time* of the room. The relationship among the reverberation time, the room volume, and the total absorption, is expressed by the following approximate formula:

$$T = .049\ V/S\alpha \text{ seconds}$$

where V = room volume in cubic feet, S = total surface area in square feet, and α = sound-absorption coefficient.

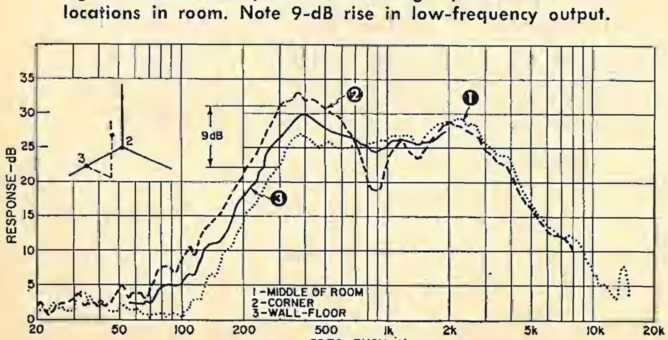
Some interesting conclusions can be obtained by calculating the intensity of the reverberant sound in a typical living room under normal listening conditions, and comparing it to the direct sound. The intensity of the reverberant sound is:

$I_R = 800\ WT/V$ watts per square meter, where W = acoustic power produced by the source.

The direct sound is spread over the surface of a sphere with radius equal to the distance from the measuring point to the source. Assuming this to be d in meters, then $I_D = W / 4\pi d^2$.

Comparing the values for I_R and I_D for a typical living room, one obtains the rather surprising result that at a distance of approximately two feet from an omnidirectional

Fig. 7. Measured response of mid-range speaker in various locations in room. Note 9-dB rise in low-frequency output.



source, the reverberant sound has an intensity approximately equal to that of the direct sound. At greater distances, the direct sound decreases, but the reverant intensity remains constant. *The greater part of what one hears at normal listening distances is the reverberant sound.*

When a source is directional, more of the sound is concentrated in the direction of the listener and less of it reaches the room surfaces to become reverberant sound. With practical loudspeakers located as they normally are in living rooms, the speaker is made more directional by its usual position at the junction of a wall and floor, and of course at the higher frequencies, practical loudspeakers tend to become more directional anyway. As a result, the critical distance (at which the direct and reverberant sound intensities are equal) is more likely to be something over five feet.

What We Hear

While the mathematics of the previous section is convincing, one tends to reject the conclusion. If the reverberant sound predominates—and, of course, this comes from all directions—how is it that we can tell where the sound is coming from? We can do this because of loudness differences and the *precedence effect.* If one listens to monophonic program material on a stereo reproducing system while sitting on the axis of one of the loudspeakers, all of the sound seems to be coming from that loudspeaker. In fact, it is necessary to move near the axis of the other loudspeaker to convince oneself that it is operating at all. What distinguishes the sound of the on-axis loudspeaker is that it arrives earlier and is somewhat louder than the sound from the other loudspeaker. If one sits on the axis between the two loudspeakers, as one would when listening to stereo, the sound appears to come from a point midway between the speakers. Adjusting the balance control moves the virtual sound source nearer one of the speakers, as the level of one speaker is raised with respect to the other. If the difference in level is made great enough, all of the sound appears to come from one loudspeaker. From these experiments, one can conclude that if one is equidistant from two loudspeakers and varies their relative loudness, then beyond a certain point all of the sound appears to come from the louder one.

An interesting experiment is to introduce time delays. Suppose we delay the sound coming from one loudspeaker, either by using a tape loop or some similar delay device or by placing the speaker farther away so that the time taken for the sound to arrive at the listening point is increased. An effect analogous to the results obtained with varying intensities is obtained. For very small time delays the sound source appears to move away from the delayed speaker toward the undelayed speaker. The time differences involved for this effect are somewhere between 0.6 and 1.0 millisecond (or about the time required for sound to travel 1 foot). For longer time delays, all of the sound seems to come from the undelayed speaker.

A fascinating aspect of this precedence effect is that it is possible to use increased loudness to compensate for delay. If we listen to sounds of equal intensity from two loudspeakers, one of which is delayed by a few milliseconds, and we increase the output of the delayed loudspeaker, there is a point beyond which the sound no longer appears to come from the undelayed loudspeaker. More delay requires more level difference for compensation. Strangely enough, this only holds true up to a maximum, in the neighborhood of 15 milliseconds, where it takes something less than 11 dB to make up for the difference. Beyond this point, that is for longer delays, it takes less level increase to make up the difference. As the delay is increased beyond 50 milliseconds or so, the delayed speaker begins to be heard as an echo.

The precedence effect greatly influences what one hears in the presence of both direct and reverberant sound. In general, the reverberant sound is somewhat attenuated in level and is delayed with respect to the source of direct sound as the apparent source. The reverberant sound contributes to the loudness and to our sense of ambience or sensation of being immersed in sound, but it does not confuse us as to where the original sound is coming from.

One might well conclude that the reverberant sound has no significance after all. However, this is not really the case. The reverberant sound contributes to a sense of ambience (which is definitely present in a concert hall) and also to the loudness. Does it also affect one's conclusions about the tonal balance, that is, the frequency response of the loudspeaker? This is a rather ticklish question. Some people maintain that the direct sound is the determining factor, others that the total energy represented by the direct plus reverberant sound is the criterion, and there are views in between. But before attempting to come to our own conclusions, it would be well to decide for ourselves whether the tonal character of the reverberant sound is different from that of the direct sound. If it is not, there is nothing to argue about.

With an omnidirectional source operating in a room whose surfaces reflect sounds without frequency discrimination, the reflected sound reaching the listener has exactly the same tonal balance as the direct sound. For example, if the reflecting surfaces discriminate against the higher frequencies, the reverberant sound lacks high frequencies and the tonal balance is different from that of the direct sound. In rooms that are good acoustically, this is not a very serious consideration. If a source is directional and its directionality increases as the frequency goes up, the reverberant sound contains less and less of the high-frequency range as the frequency increases, so that the tonal balance of the reverberant sound is quite different from that of the direct sound. If one thinks of the sum of the direct and reverberant sound as the acoustical output of the system comprising the loudspeaker and the room, the system response can be considerably different from the on-axis speaker response alone. Consequently we do have to be concerned whether the ear takes the reverberant sound into account when judging tonal balance.

It is probable that in making this judgment the ear sums the direct sound and that part of the reverberant sound produced by the early reflections. This is based partly on the precedence effect and on the fact that the ear tends to listen to varying sounds as if it were a detector with an integrating time of the order of 1/20 second. The sound path corresponding to 1/20 second is approximately 55 feet, which gives some idea of the number of reflections included in the "early sound" in an average living room.

To sum up, the ear follows the precedence effect in establishing the apparent source of a sound and tends to judge its intensity during its integrating time so that the system frequency response referred to above involves summing the direct sound and the early part of the reverberant sound.

When one considers that for stereo listening the direct sound referred to above is not the on-axis sound produced by the speaker and that the degree to which the listener is off-axis is apt to vary considerably from one listening setup to another, it is difficult to conceive of a measuring system for frequency response that would take all these factors into account. It does seem quite clear, however, that the on-axis frequency response is not the sole determining factor and that some method of summing the output of the speaker in all directions would be more meaningful.

Reverberation Chamber

A reverberation chamber is essentially a large room with highly reflecting surfaces in which the level of the reverberant sound is so great that a measuring micro-

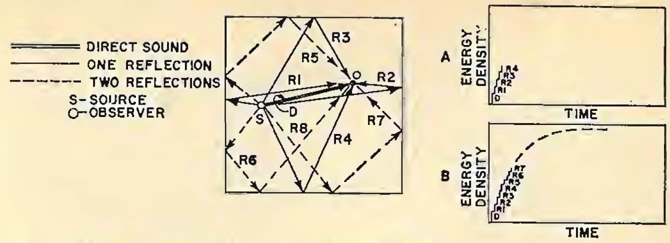


Fig. 8. What we hear in a normal listening room is largely made up of sound that has been reflected one or more times.

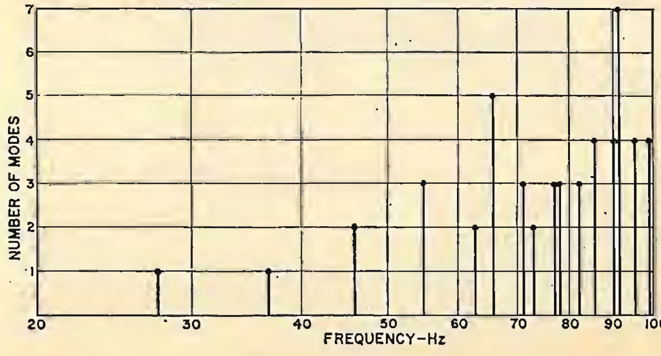
phone, if not placed too close to the loudspeaker, measures the reverberant sound alone. Since the latter is composed of reflections originating as the sound emitted by the speaker in all directions, it is proportional to the total acoustic power output of the loudspeaker. This is essentially what we wish to measure. Originally reverberant chambers were used primarily to measure the total power output of loudspeakers in order to relate it to the electrical input and thereby determine the efficiency. What we are interested in, however, is finding out how the total power output of the loudspeaker varies with frequency. For this purpose, a frequency-swept signal is applied to the loudspeaker and the amplified output of the measuring microphone is fed to a sound level recorder which plots the power frequency response of the loudspeaker automatically. The instrumentation is essentially similar to that used for frequency-response measurements in an anechoic chamber. However, the problems involved in obtaining a meaningful measurement are quite different.

The description previously given of the manner in which a sound field is built up in a room due to multiple reflections assumed a more or less random set of paths for the reflected waves. In actual rooms, however, which possess a certain degree of regularity and symmetry in shape, it frequently happens that the paths of a pair of outgoing and incoming waves are identical. The result is the formation of a standing wave similar to that formed in an organ pipe. In the simplest type of standing wave the pressure is maximum at two opposite parallel walls of the room and zero at the center. If this is plotted in terms of pressure *vs* distance, the shape of the curve is half a sine wave.

Again, as in an organ pipe, there are multiples of the frequency of the lowest mode just described. Since this occurs between each pair of parallel surfaces, there are three sets of these axial modes (or room resonances) in a rectangular room. Standing waves can also be formed along diagonals of a room parallel to the plane of each bounding surface and to diagonals between opposite corners not in the same bounding surface. The standing waves in a room are comparatively sparse in a given bandwidth at low frequencies and become more closely spaced as the frequency increases. Fig. 9 shows a typical distribution.

Suppose the source is emitting sound of a frequency corresponding to the lowest mode (*Continued on page* 74)

Fig. 9. Characteristic frequencies below 100 Hz for a rectangular listening room measuring 15 by 20 by 10 feet high.



Loudspeaker Measurements

(*Continued from page* 29)

for a given pair of opposite parallel surfaces of the room. If a measuring microphone is moved from one wall across the room to the opposite wall, the sound level at the microphone will vary from a maximum, through nearly zero, and then back to a maximum. At double this frequency the pressure will go through two minima and one maximum, the maximum being at the center of the room. At higher frequencies the number of maxima and minima will increase correspondingly. Finally a point will be reached beyond which the spacing between maxima and minima is of the order of magnitude of the dimensions of the microphone, which will consequently average these, and show a more uniform output as it traverses the room.

In this discussion, the effect of other types of modes has been neglected so that in fact the variations in intensity measured by the microphone as its position is changed from one wall to the opposite wall will be extremely irregular. It is rather easy to see that much the same thing will happen if the microphone position is fixed and the frequency is varied. In place of the extreme variations in recorded sound pressure occurring as the microphone is moved across the room, similar extreme variations will be found as the frequency is varied.

Since this irregularity is predicated on a completely uniform output from the sound source, any variations in frequency response of an actual loudspeaker will tend to be obliterated by the extremely irregular response of the room. A frequency response curve obtained in this manner turns out to be a frequency response curve of the room rather than the loudspeaker, except for very broad variations in the speaker frequency response. For example, if the speaker response decreases with increasing frequency, the average of the recorded response curve will reveal this. However, sharp variations in speaker frequency response, in which we are definitely interested, will not be discernible.

The reference to averaging in the previous paragraph gives a hint of the approach which makes speaker frequency response measurements feasible. A warbled frequency might be used so that at any given point in its sweep the recorded trace registers the average of a narrow band of frequencies. Similarly, a narrow band of noise could be swept from one end of the spectrum to the other. The bandwidth must be sufficiently great to obtain a reasonably good average of the room response without making it so broad as to obscure minor variations in the speaker response.

Additional improvement is obtained by *space averaging* as well as frequency averaging. The microphone or the loudspeaker, or both, can be moved continuously while recording the frequency response. If the speaker is moved, this changes the pattern of the standing waves. If the microphone is moved, it rapidly samples the sound pressure along its path and averages it. Each element may be swung back and forth or rotated. Still further smoothing can be obtained by rotating or swinging large reflecting surfaces within the room. These procedures are effective but rather clumsy mechanically. An alternative is to use a number of microphones placed in random positions in the room and sum their output. This is the system that was adopted for use in the reverberation chamber at *H. H. Scott, Inc. (It is also the procedure used by Hirsch-Houck Labs for checking speaker response, but in a normal listening room.—Editors)*

When the outputs of a reasonable number of microphones, say six, are added, some smoothing takes place, but the system is less effective than might be expected. The reason is that the sound pressures at the different microphones vary not only in amplitude but in phase as well. At a given point, the sound pressure might be identical at two microphones but opposite in phase, as a result of which their output sum would be zero. The effect of phase can be eliminated by rectifying the output of each microphone and adding the resulting d.c. outputs. This was tried experimentally, but difficulties were encountered due to nonlinearities of the rectifiers. Incidentally, the ideal system would be to add the power outputs of the microphones. This could be done by squaring the voltage and then adding; however, squaring circuits having wide dynamic and frequency range are not simple.

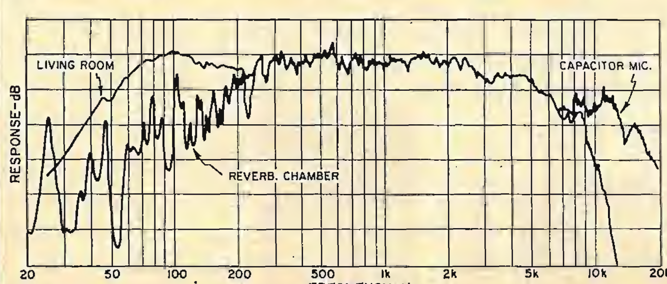


Fig. 10. Speaker frequency response in reverberation chamber. Upper curve at bass end is close-up pressure measurement in typical living-room with speaker on floor against wall.

It was finally decided to sample the microphone outputs one at a time in order to obtain an effective average. This can be done by means of a commutator but again it was felt desirable to stay away from mechanical devices and to use an electronic commutator. This consists of twelve field-effect transistors used as switches. The gates of the FET's are driven by 60-Hz sine-wave signals shifted in phase by multiples of 60° for successive FET's. The commutated signal for each channel has a rectangular envelope, being on for 1/360 second and off for 5/360 second. The choice of six microphones was rather arbitrary. Since smoothness is improved in proportion to the square root of the number of microphones, the law of diminishing returns is involved. For example, the use of twelve microphones would have resulted in 30% better smoothness, but only at high frequencies because of residual room irregularities at lower frequencies.

The *H. H. Scott, Inc.* reverberation chamber was constructed with a height: width:length ratio of 1:1.6:2.5, which is a set of proportions traditionally recommended for the best distribution of modes. Inside dimensions are approximately 23 x 15 x 9 feet. The walls and floor at two diagonally opposite corners are perpendicular to each other to permit testing of corner speaker systems. Three of the surfaces were made non-parallel, with a slope of approximately 5% to improve the diffusion. The room is substantially constructed, the walls being of eight-inch cement block filled with dry sand. The construction provides very stiff walls and ceiling; the sound transmitted through the walls is reduced by approximately 40 dB. Background noise in the room, measured on the C-scale of a sound-level meter is below 40 dB at all times. The reverberation time varies from 4 seconds near 100 Hz to 1.5 seconds near 10 kHz. The frequency response of the room decreases at the higher frequencies due to decreased reflectivity of the room surfaces and losses through transmission in the air itself. The frequency characteristic was measured and corrected for large-scale variations in frequency response by means of an equalizing network. The room is not usable at low frequencies but this is no problem because, due to the omnidirectionality of loudspeakers in this range, an on-axis frequency response measurement is adequate.

The measuring system produces a graph which is a composite of the irregularities of the loudspeaker response and irregularities of the room response. Over the frequency range involved, the speaker response will, in general, be smoother than the room response because of the large number of room modes in a given band of frequencies and because the sharpness of resonance of these modes is considerably greater than that of the speaker resonances. For recorder pen and chart speeds of values that permit accurate tracing of the irregularities in speaker response, the modes of the room, some of which are increasing, some at maximum, and some decaying at a given instant, do not have a chance to build up to very high intensities.

Fig. 10 shows measured curves on a developmental speaker system. Below 300 Hz or so, the room irregularities are extremely great; consequently the low-frequency response of this speaker system is shown as a pressure measurement taken with the microphone close to the speaker in a partially absorbing room. The rather sharp drop at 9 kHz is due to the response of the microphones used in the reverberation chamber. A single microphone measurement on-axis, using a *Bruhl* and *Kjaer* 4133 capacitor microphone, is superimposed on the graph. Since this microphone is flat to 20 kHz, the curve represents the response of the speaker system at high frequencies more accurately. In this range the room modes are so closely spaced that the use of only one microphone is acceptable. The high-frequency response of the speaker system has a downward slope, which is to be expected for anything but a perfectly omnidirectional speaker. In other words, the power response drops much faster than the on-axis frequency response.

The reverberation chamber has been found to be particularly useful in the measurement of the frequency response of speaker systems employing several speakers and crossover networks, and radio-phonograph consoles. Because of the short time required for measurement as opposed to taking a complete set of on-axis and off-axis curves in an anechoic chamber, it is practical to make numerous measurements to determine the influences of baffle sizes and shapes, locations of speakers on the baffle, effect of grille cloth, metallic, wood and molded grilles, frames, and other decorative treatment. As an example of the sensitivity of such tests, it is possible to distinguish the effect of a shift in tweeter location relative to the mid-frequency speaker of only ¼ inch.

Experience with the reverberation chamber has shown that it is an excellent means of detecting lack of smoothness in the frequency response of a speaker system. Listening tests have indicated that irregularities in the curve obtained in the reverberation room are heard as annoying resonances and roughness. While it cannot be claimed that a speaker system having a smooth response in a reverberation room will necessarily sound good, the technique is extremely useful in detecting design defects. ▲