Section 6. Sound Indoors

Section 6 discusses the behavior of sound indoors. The foundation for this discussion was laid in Section 5 ('Sound Outdoors").

In comparison with its behavior in free field, sound indoors exhibits quite complex behavior. Decades of acoustics research have been spent attempting to quantify the effects of a reverberant field. These efforts have yielded a body of equations that are used to describe how sound behaves in a given room.

We make no effort to present those calculations in detail here. Such effort would not serve our present purpose. Instead, this chapter presents the basic principles of indoor acoustics, using simple mathematics where necessary.

6.1 Boundaries

The walls, ceiling, and floor of a room are, to some extent, both flexible and porous to sound. Figure 6-1 shows what happens when a sound wave strikes such a boundary surface.

Part of the wave energy is reflected, as shown in Figure 6-1(a). The percentage of the energy that is reflected is related to the stiffness of the surface.

Wave energy that is not reflected enters the boundary. Part of this energy is absorbed (b) by the boundary material through conversion into heat. The remainder (c) is transmitted through the boundary. Both effects (a) and (b) are related to the flexibility and porosity of the boundary.

When sound strikes a smaller obstacle (not a wall or ceiling, but perhaps a podium or pulpit), it bends around that object. This is known as refraction, and is shown in Figure 6-2.

Refraction, reflection, transmission, and absorption are all dependent on the frequency of the sound wave and the angle at which it strikes the boundary. The percentages are not generally dependent on the intensity of the sound.

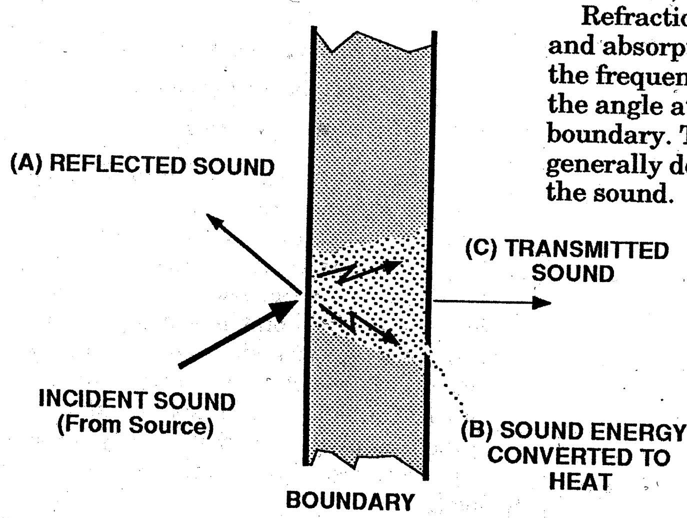

Figure 6-1. Effect of boundary surfaces on sound transmission and reflection

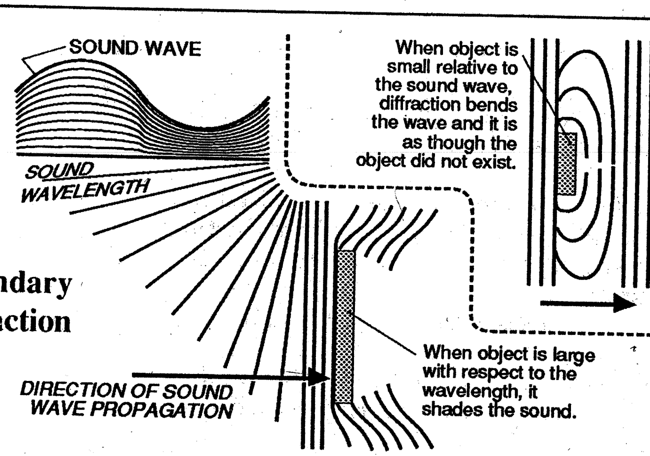

Figure 6-2. Effect of boundary surfaces on sound refraction

6.1.1 The Absorption Coefficient

In architectural acoustics, a measure of the energy lost when a sound wave strikes a given material is specified by the absorption coefficient of the material. The concept of the absorption coefficient was developed by Dr. Wallace Sabine, who is regarded as the father of modern architectural acoustics.

Sabine defined an open window — which does not reflect any sound — as the perfect absorber, assigning it a coefficient of 1 (100%). Similarly, he gave the perfect reflective surface a coefficient of 0. The absorption coefficient of any material is thus a number between 0 and 1, which is readily converted into a percentage.

The relationship between the absorption coefficient of a boundary material and the intensity of the reflected sound wave is therefore a simple one. For example, consider that a given boundary material has an absorption coefficient of 0.15. To determine the effect of the boundary on a sound wave:

1) Convert the coefficient into a percentage

$$0.15 = 15\%$$

15% of the sound is absorbed by the material.

2) To get the amount reflected, subtract from 100%

$$100 - 15 = 85\%$$

85% of the sound is reflected.

3) Finally, the conversion to dB is a 10 log function

$$10 \log 0.85 = -0.7 \text{ dB}$$

The reflected sound pressure is 0.7 dB lower than the incident sound pressure.

This computation can be reduced to a single equation relating the reduction in sound level to the absorption coefficient:

$$\text{ndB} = 10 \log (1 \div (1 - a))$$

where 'ndB' = the reduction in sound level, and 'a' = the absorption coefficient.

Table 6-1 gives absorption coefficients for a number of different typical boundaries. The coefficients for each material are given at different frequencies, since the effect varies with frequency.

MATERIAL	Frequency (Hz) 125	1k	4k
Brick Wall (18" Thick, unpainted)	.02	.04	.07
Brick Wall (18" Thick, painted)	.01	.02	.02
Interior Plaster (On metal lath)	.02	.06	.03
Poured Concrete	.01	.02	.03
Pine Flooring	.09	.08	.10
Carpeting (With pad)	.10	.30	.70
Drapes (Cotton, 2x fullness)	.07	.80	.50
Drapes (Velour, 2x fullness)	.15	.75	.65
Acoustic Tile (5/8", #1 Mount*)	.15	.70	.65
Acoustic Tile (5/8", #2 Mount*)	.25	.70	.65
Acoustic Tile (5/8", #7 Mount*)	.50	.75	.65
Tectum Panels (1", #2 Mount*)	.08	.55	.65
Tectum Panels (1", #7 Mount*)	.35	.35	.65
Plywood Panel (1/8", 2" Air space)	.30	.10	.07
Plywood Cylinders (2 Layers, 1/8")	.35	.20	.18
Perforated Transite (w/Pad, #7 Mount*)	.90	.95	.45
Occupied Audience Seating Area	.50	.95	.85
Upholstered Theatre Seats (Hard Floor)	.45	.90	.70

* #1 Mount is cemented directly to plaster or concrete, #2 Mount is fastened to nominal 1" thick furring strips, #7 Mount is suspended ceiling w/ 16" air space above.

Table 6-1. Approximate absorption coefficients of common materials

Note that for mid and high frequencies an occupied audience area has an absorption coefficient very close to 1 (complete absorption). This is why the presence of an audience can have an enormous effect on the acoustics of a hall, and will provide a significant contrast with unupholstered seats on a hard floor. As we shall see in subsequent chapters, this fact is very important to the design of indoor sound systems.

6.2 Standing Waves

One significant effect of hard boundary surfaces is the formation of what are called standing waves.

Figure 6-3 shows what happens when a continuous sound, at one frequency, strikes a reflective boundary head-on. The reflected sound wave combines with subsequent incoming waves. Where the wave crests (maximum pressure) coincide, they combine and reinforce one another. The troughs (minimum pressure) also combine.

The result is a stationary pattern in the air, consisting of zones of low pressure (called nodes), alternating with zones of high pressure (called antinodes). The phenomenon is known as a standing wave.

Walking through such a standing wave zone, you can easily identify physical places where the sound is very loud, and others where the sound is very soft. Note that these alternate maximal and minmal air pressure zones are spaced at distances of 1/2 wavelength. Their position in space depends on the frequency of the sound.

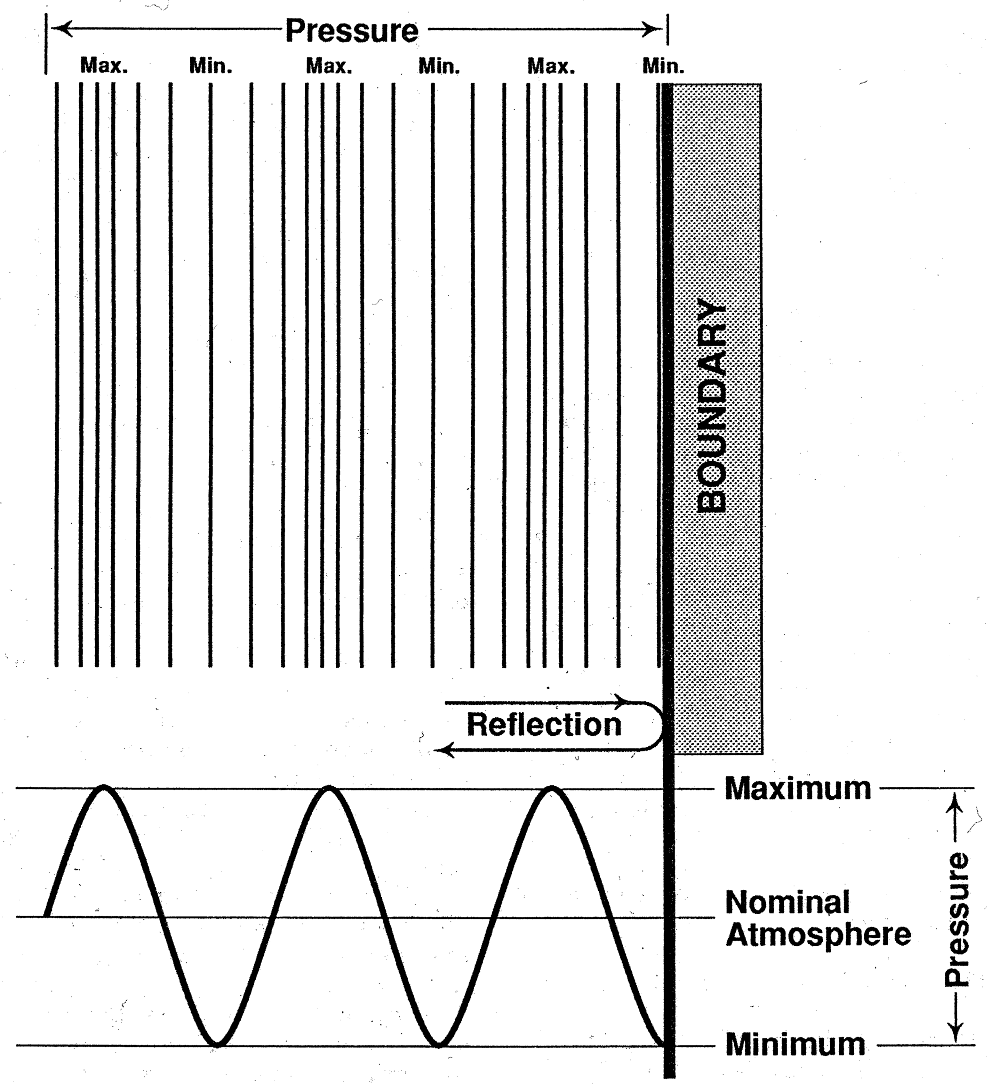

Figure 6-3. Formation of a standing wave by reflection at a boundary

6.2.1 Standing Waves in a Room

Now consider Figure 6-4, which shows two parallel walls. The walls are assumed to be highly reflective. In the center, we place a point source of sound.

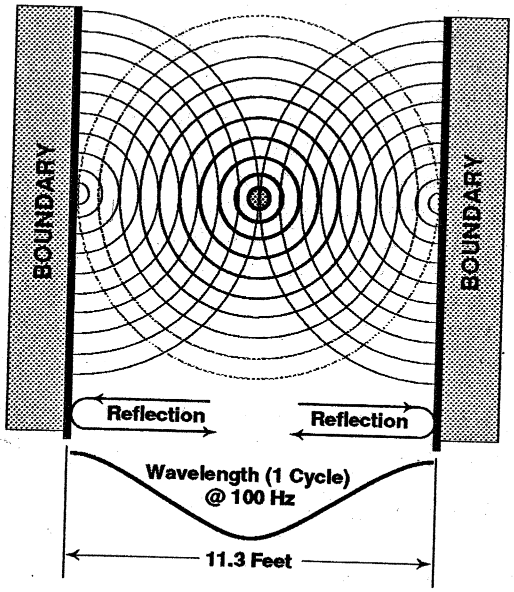

Figure 6-4. Formation of a standing wave in a room

Suppose our point source emits a brief tone. The sound waves travel out in all directions, with those propagated to the sides eventually reaching the walls. Some energy is absorbed by the walls; most is reflected back. Reflected waves from each wall travel to the other wall, reflecting again. The process continues until the energy of the sound is completely dissipated by absorption in the air and the walls.

In this situation, standing waves will be formed if – and only if – the wavelength of the sound "fits" the distance between the walls. Such standing waves are also called room resonances, natural frequencies, or modes.

For example, the wavelength of a 100 Hz tone is:

1130 ft/sec. ÷ 100 cycles/sec. = 11.3 ft./cycle

If the walls in Figure 6-4 are 11.3 feet apart, then successively reflected waves will reinforce each other, forming stationary nodal and antinodal points in the room. The same effect will occur at frequencies that are integral multiples of 100 (200 Hz, 300 Hz, etc.). This is illustrated in Figure 6-5.

Suppose that the area between the walls held an audience, and that the 100 Hz tone was a bass note in a musical passage. Those audience members seated at nodal points might have trouble hearing the note, while those seated at antinodes would hear it as unnaturally loud! Obviously, standing wave phenomena can drastically affect the quality of sound in a room.

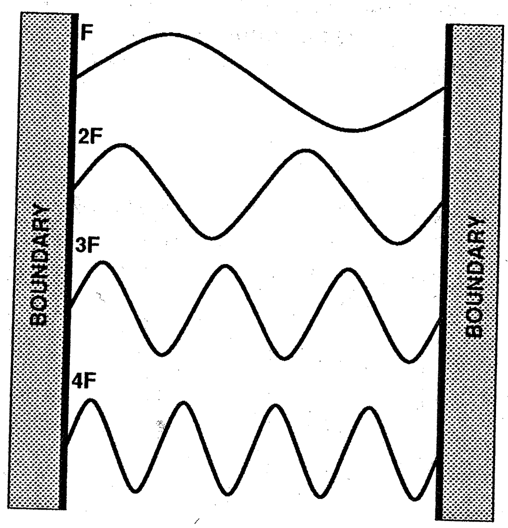

Figure 6-5. Room modes at harmonic frequencies

In rooms, the situation is much more complicated than what we have described here. There are three such simple resonance systems (one between each set of opposite walls, and one from floor to ceiling), and two more complicated ones (one involving all four walls, and one involving all six surfaces). Consequently, a given room will exhibit many such resonances at different frequencies.

Good acoustical design takes room resonances into account, and strives to minimize them through use of non-parallel walls and various types of absorptive treatments. One of the most effective and simple treatments is to hang drapes.

Notice in Figure 6-3 that the zone spaced 1/4 wavelength from the reflecting surface is a node. In this zone, the pressure is minimum but the air particle velocity is maximum. Absorptive material hung at this point will have far greater effect on the standing wave than its absorption coefficient would indicate.

6.3 Reverberation

Another substantial and much discussed effect of boundary reflection is reverberation. Reverberation is modeled as described below.

Imagine a point source of sound at the center of an enclosed room. We turn on the source, and sound radiates outward in all directions, eventually striking the boundaries of the room. Some of its energy is absorbed, some is transmitted through the boundary, and most is reflected back into the room.

After a certain time, enough reflections have occurred that the space is essentially filled with a random field of sound waves. If the source remains on, the system reaches a state of equilibrium such that the energy being introduced by the source exactly equals the energy dissipated in boundary transmission and absorption.

Acousticians describe this equilibrium in statistical terms. Ignoring for the moment any standing-wave resonances or focused reflections, we can say that the sound pressure is the same at all points that are not too near the source.

Let's say that we now turn off the source. The remaining sound waves in the room continue to ricochet from boundary to boundary, losing energy with each reflection. At some point, all the remaining energy in the system dissipates, and the sound ceases.

This decaying of the sound is what we perceive as reverberation. The amount of time that it takes for the acoustical energy to drop by 60 dB is called the decay time, or reverberation time, abbreviated RT_{60}.

The length and spectral characteristic of the decay – together with any resonances – form the acoustic signature, or characteristic sound, of a room. These factors are obviously determined by the absorptive qualities of the room boundaries, and by the volume and shape of the room.

NOTE: There are several related equations, named after the people who developed them, for calculating the reverb time based on absorptivity of a given environment: Sabine, Norris-Eyring, and Hopkins-Stryker. The unit of absorption in Sabine's equation is the Sabin. It is beyond the scope of this book to go into details of these equations, which are shown below.

	MKS units: S = Surface area in m^2 V= Volume in m^3	English units: S = Surface area in ft^2 V= Volume in ft^3
Sabine Gives best correspondence with published absorption coefficients where $\bar{\alpha}<0.2$.	$T = \dfrac{0.16V}{S\bar{\alpha}}$	$T = \dfrac{0.49V}{S\bar{\alpha}}$
Eyring Preferred formula for well-behaved rooms having $\bar{\alpha}\gtrsim 0.2$.	$T = \dfrac{0.16V}{-S \ln(1-\bar{\alpha})}$	$T = \dfrac{0.49V}{-S \ln(1-\bar{\alpha})}$
Fitzroy For rectangular rooms with poorly distributed absorption. α_x, α_y and α_z are average absorption coefficients of opposing pairs of surfaces with total areas of x, y and z.	$T = \dfrac{0.16V}{S^2}\left(\dfrac{x^2}{X\alpha_x}+\dfrac{y^2}{Y\alpha_y}+\dfrac{z^2}{Z\alpha_z}\right)$	$T = \dfrac{0.49V}{S^2}\left(\dfrac{x^2}{X\alpha_x}+\dfrac{y^2}{Y\alpha_y}+\dfrac{z^2}{Z\alpha_z}\right)$
	T = Decay time (in seconds) for 60 dB level reduction.	

Figure 6-6. Reverberation time equations

6.4 Critical Distance

Relatively short to moderate reverberation with a smooth spectral characteristic is perceived as pleasant, natural, and musical. Excessive reverberation makes it difficult to understand speech, and can destroy the texture and impact of music. Most of us have strained to understand an announcement in a large, hard-surfaced gymnasium, arena or transportation terminal... not because it wasn't loud enough, but because there was too much reverberation. To be successful with indoor sound systems we must be careful in handling reverberation.

We have said that the reverberant field in a room is the same intensity everywhere in the space. What relevance does the inverse square law have indoors, then?

In order to answer the question, we must distinguish between the direct sound (the initial sound emitted by the source, before reflection), and the reverberant sound.

For the direct sound, the inverse square law is valid indoors just as it is outdoors. In terms of pure sound pressure once the reverberant field is activated, it adds a second pressure component.

Figure 6-7 shows an omnidirectional loudspeaker radiating sound in a reverberant field. The direct sound of the loudspeaker propagates into the space, diminishing in intensity according to the inverse square law. Initially,

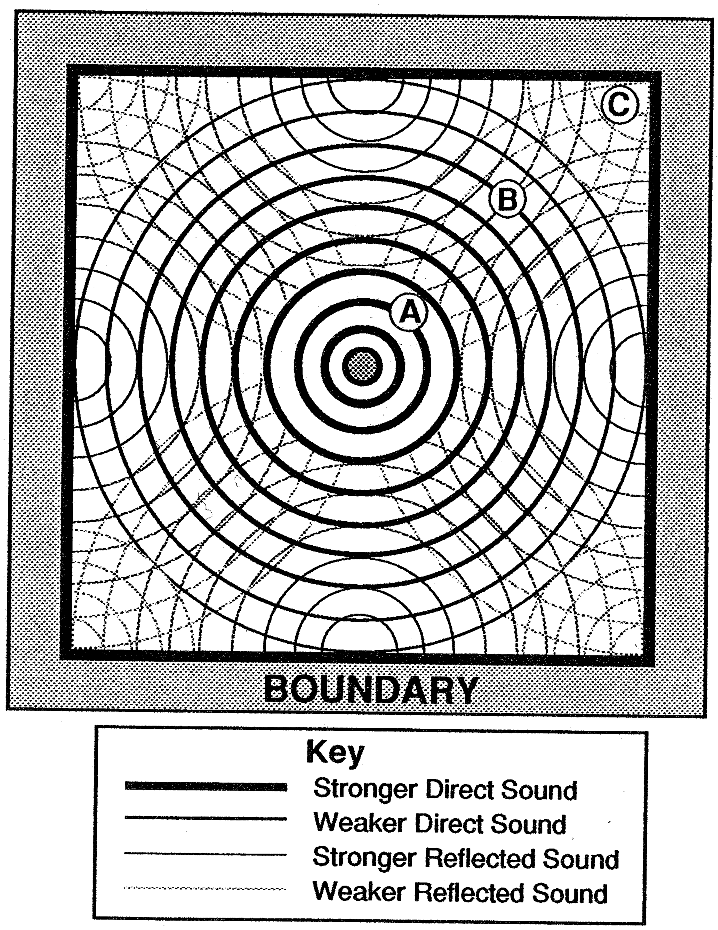

Figure 6-7. Development of a reverberant field from a theoretical point source of sound in the center of an acoustic environment

there is primarily direct sound all over, as can be seen in Figure 6-7(a). At a certain distance from the loudspeaker, and after the sound has existed long enough to reverberate, the intensity of the direct sound equals that of the diffuse reverberant sound (b). Ultimately, at a sufficient distance from the loudspeaker, the reverberant sound is predominant and swamps out the direct sound (c).

The distance from the acoustic center of the source to the point at which the intensity of the direct sound equals that of the reverberant field is called the critical distance.

As we move further away, beyond the critical distance, into the region where the reverberant field predominates, the intensity of the sound levels off to a statistically constant value, assuming the sound source continues to excite the room at the same level.

It follows logically and intuitively that we can increase the critical distance by using a directional loudspeaker. If we concentrate the power of the system along a given axis which corresponds to an absorptive area (such as an audience), the direct sound will predominate over a longer distance along that axis. This is not only because the sound energy is concentrated more in a forward direction, but also because there is less energy radiated and reflected toward the sides, and hence less energy to reflect from walls, ceiling, floor, etc. so that the reverberant field does not receive as much of the loudspeaker's energy. Since a directional loudspeaker's power decreases the farther we go off-axis, we must be aware that we will gain critical distance on-axis at the expense of the off-axis critical distance. Figure 6-8 illustrates this situation.

The real benefit of the horn is that it increases the direct level in a portion of the environment.

Since the direct sound follows the inverse square law, and we assume that the reverberant field's intensity is equal everywhere, it also follows that the ratio of direct-to-reverberant sound is an inverse square relationship. In other words, given that the ratio of direct to reverbant sound is 1:1 at the critical distance, then at twice the critical distance the direct sound will be 6 dB below the reverberant sound (half the level).

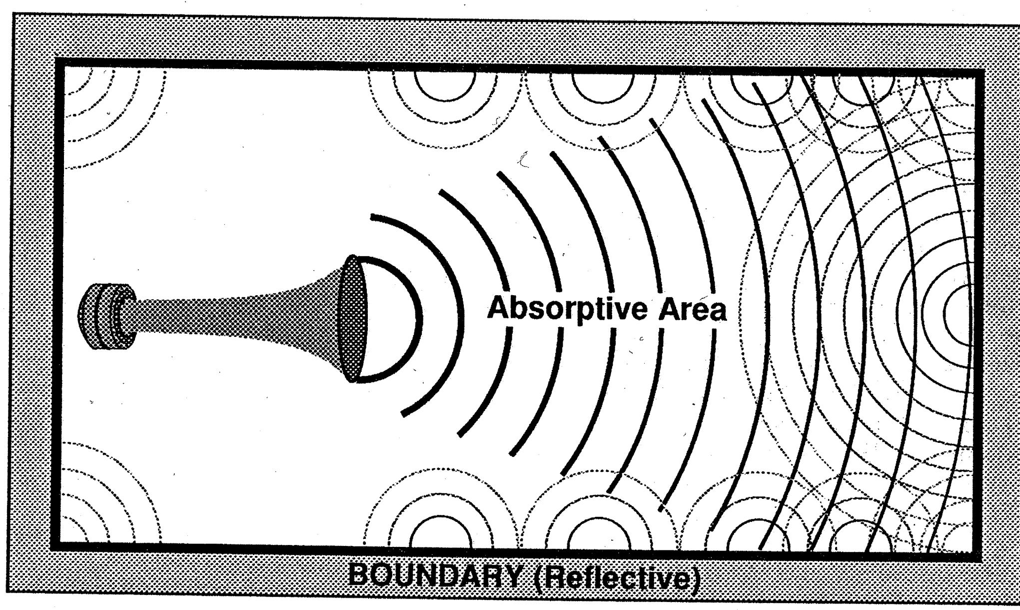

Figure 6-8. Directional radiator in a reverberant field

6.4.1 Implications For Sound Reinforcement

The preceding discussion reveals a fundamental reason why directional elements are the norm in sound reinforcement. Obviously, we don't want the direct sound of the system to be swamped by the reverberant sound. If we can increase a system's critical distance, then we have a better chance of maintaining clarity (or intelligibility) over greater distances.

A theoretical structure – buttressed by experimental evidence – has been constructed by acousticians, relating calculated critical distances and direct-to-reverberant ratios to intelligibility in speech reinforcement. This area is somewhat complex, and good modeling requires a solid mathematical background – and benefits from some help in the form of a well-programmed computer. For this reason, we will not go into detail in this basic handbook.

In any case, using such data, acousticians have developed methods for calculating the behavior of speech reinforcement systems in relation to hall acoustics. These methods are used routinely by consultants and contractors to design such systems.

Once we get into the field of music reinforcement, all bets are off, and such quantitative judgments are much more difficult to make. Here we enter a realm in which individual taste and subjective impressions reign.

Let us take it as given that we want the sound our system delivers to be as clear as possible, and that we wish to have as much control over the sound as we can. These two criteria imply that we wish to maximize the system's critical distance — that is, minimize the extent to which we excite the reverberant field — so that the audience hears mostly direct sound. The specific techniques for doing so are presented in Sections 17 and 18.