Chapter 6: Behavior of Sound Systems Indoors

Introduction

The preceding five chapters have provided the groundwork on which this chapter is built. The "fine art and science" of sound reinforcement now begins to take shape, and many readers who have patiently worked their way through the earlier chapters will soon begin to appreciate the disciplines which have been stressed.

The date at which sound reinforcement grew from "public address by guesswork" to a methodical process in which performance specifications are worked out in advance was marked by the publication in 1969 of a paper titled "The Gain of a Sound System," by C. P. and R. E. Boner (4). It describes a method of calculating potential sound system gain, and that method has since become a fundamental part of modern sound system design. The following discussion is based on the Boner paper. Certain points are expanded, and examples are given that require calculations more complicated than those in the original study. Also discussed is the relation between theoretically achievable system gain and practical operating parameters of typical indoor sound systems.
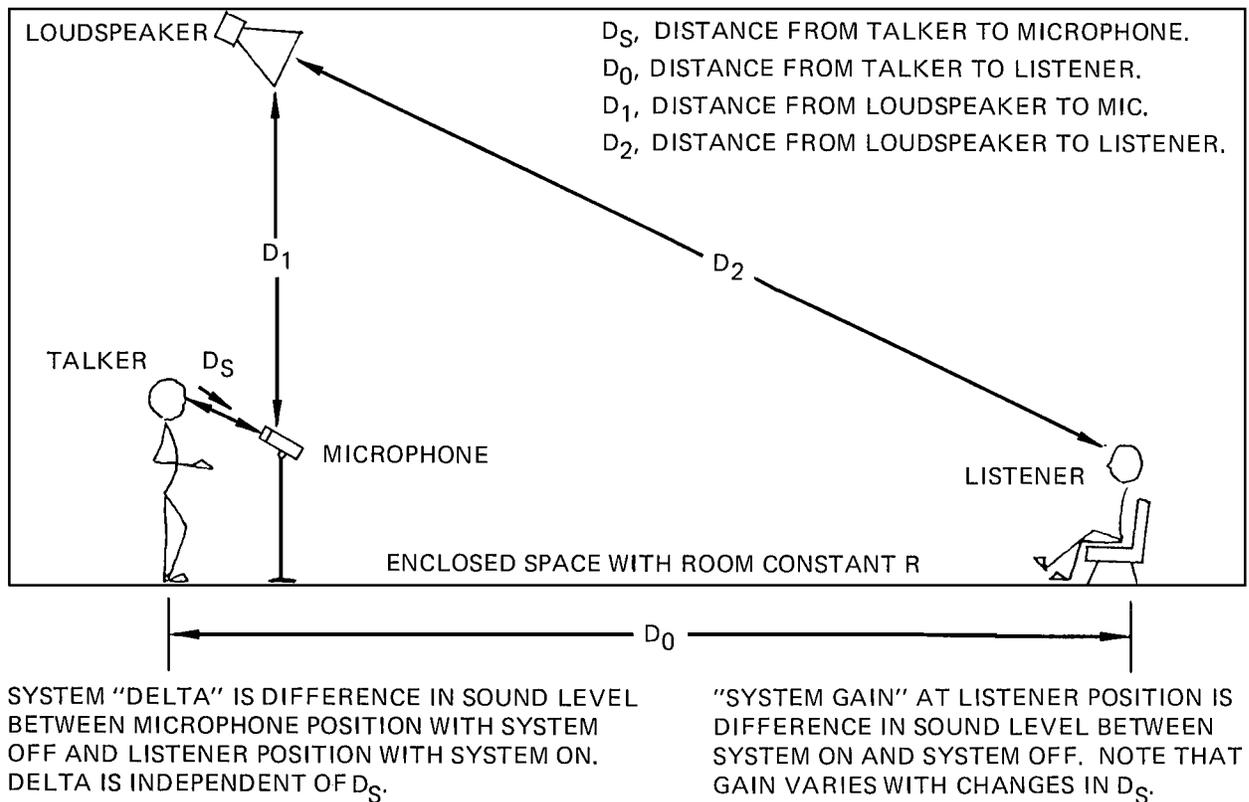

D_S, DISTANCE FROM TALKER TO MICROPHONE.
D_0, DISTANCE FROM TALKER TO LISTENER.
D_1, DISTANCE FROM LOUDSPEAKER TO MIC.
D_2, DISTANCE FROM LOUDSPEAKER TO LISTENER.

SYSTEM "DELTA" IS DIFFERENCE IN SOUND LEVEL BETWEEN MICROPHONE POSITION WITH SYSTEM OFF AND LISTENER POSITION WITH SYSTEM ON. DELTA IS INDEPENDENT OF D_S.

"SYSTEM GAIN" AT LISTENER POSITION IS DIFFERENCE IN SOUND LEVEL BETWEEN SYSTEM ON AND SYSTEM OFF. NOTE THAT GAIN VARIES WITH CHANGES IN D_S.

Figure 6-1.   An indoor sound system

Acoustical Feedback and Potential System Gain

Just as in the outdoor case studied earlier, if we have a microphone/amplifier/loudspeaker combination in the same room and gradually turn up the gain of the amplifier to a point approaching sustained feedback, the electrical frequency response of the system changes with the gain setting. The effect results from an acoustic feedback path between the loudspeaker and the microphone. As a person talks into the microphone, the microphone hears not only the direct sound from the talker, but the reverberant field produced by the loudspeaker as well.

The purpose of using high-quality loudspeakers and microphones having smooth response characteristics, and sound system equalization (apart from achieving the desired tonal response) is to smooth out all of the potential feedback points so that they are evenly distributed across the audible frequency range. When this has been done, there should be as many negative feedback points as positive feedback points, and the positive feedback points should all reach the level of instability at about the same system gain.

We might expect this to average out in such a way that the level produced by the loudspeaker reaching the microphone can never be greater than that produced by the talker without causing sustained oscillation. In other words, we assume that the extra gain supplied by all the positive feedback spikes is just balanced out by the loss caused by all the negative feedback dips.

If the Boner criteria for optimum system geometry are followed, the microphone will be close to the talker so that it hears mostly direct sound from the talker. It will be far enough from the loudspeaker to be well into the reverberant field of the loudspeaker, so that direct sound from the loudspeaker is not an appreciable factor in triggering system feedback. Assuming that listeners are also in the reverberant field of the loudspeaker, it follows that the sound level in the listening area with the system turned on cannot be greater than that of the unaided talker at the microphone position with the system turned off. Using the Boner concept of system *delta*, the situation at maximum gain corresponds to a delta of unity. (Delta is defined as the difference in decibels between sound level at the system microphone with system off and the level in the audience area with system on. See Figure 6-1).

Although we have described these as conditions of maximum potential system gain, it is possible in practice to achieve a delta greater than unity. For example, if a directional microphone is used it can discriminate against the reverberant field

and allow another 3 to 4 dB of system gain. Another possibility is to place the listener in the direct field of the loudspeaker, allowing a further increase in system gain. If the level of the reverberant field is lower in the performing area than in the listening area, additional system gain also results. This situation is described by the Boners as a room constant in the microphone area different from that in the seating area. Similar results may be noted in rooms having large floor areas, relatively low ceilings, and substantial sound absorption. In such rooms, as we have seen, sound from a point source tends to dwindle off beyond D_C at a rate of 2 or 3 dB for each doubling of distance rather than remaining constant in level.

Still another way to increase gain is to electrically suppress the positive feedback frequencies individually with very narrow bandwidth filters. If one could channel all energy into the negative feedback frequencies, the potential system gain would theoretically become infinite! Unfortunately, the acoustic feedback path is not stable enough to permit this degree of narrow-band equalization.

In all other situations, a gain setting is reached at which sustained oscillation occurs. By definition, maximum system gain is reached just below this point. However, the system cannot be operated satisfactorily at a point just below oscillation because of its unpleasant comb-filter response and the prolonged ringing caused by positive feedback peaks. To get back to reasonably flat electrical response and freedom from audible ringing, it usually is recommended that a properly equalized system be operated about 6 dB below its maximum gain point. Even an elaborately tuned system using narrow-band filters can seldom be operated at gains greater than 3 dB below sustained oscillation.

Sound Field Calculations for a Small Room

Consider the room shown in Figure 6-2. This is a typical small meeting room or classroom having a volume less than 80 m³. The average absorption coefficient $\bar{a}$ is 0.2. Total surface area is 111 m². The room constant, therefore, is 28 m².

From the previous chapter, we know how to calculate the critical distance for a person talking (nominal directivity index of 3 dB). In the example given, D_C for a source having a directivity index of 3 dB is 1 meter.

The figure also shows geometrical relationships among a talker, a listener, the talker's microphone and a simple wall-mounted loudspeaker having a directivity index of 6 dB along the axis pointed at the listener. The microphone is assumed to be omnidirectional.

Step 1: Calculate relative sound levels produced by the talker at microphone and listener.

We begin with the sound system off. Although the calculations can be performed using only relative levels, we will insert typical numbers to get a better feel for the process involved.

The microphone is .6 meter from the talker, and at this distance, the direct sound produces a level of about 70 dB. Since D_C for the unaided talker is only 1 meter, the microphone distance of .6 meter lies in the transition zone between the direct field and the reverberant field of the talker. By referring to Figure 6-3, we note that the combined sound levels of the reverberant field and the direct field at a distance of .6 meter must be about 1 dB greater than the direct field alone. Therefore, since we have assumed a level of 70 dB for the direct field only, the total sound level at the microphone must be 71 dB.

Next, we use a similar procedure to calculate the sound level at the listener's position produced by an unaided talker:

The listener is located 4.2 meters from the talker, more than 3 times the critical distance of 1 meter, and therefore, well into the reverberant field of the talker. We know that the sound level anywhere in the reverberant field is equal to that produced by the direct field alone at the critical distance. If the level produced by direct sound is 70 dB at a distance of .6

meter, it must be 4.6 dB less at a distance of 1 meter, or 65.4 dB, and the level of the reverberant field must also be 65.4 dB. The sound level produced by the unaided talker, at the listener's position, therefore is 65.4 dB.

At this point, let us consider two things about the process we are using. First, the definition of critical distance implies that sound level is to be measured with a random-incidence microphone. (For example, we have chosen a non-directional system microphone so that it indeed will "hear" the same sound field as that indicated by our calculations). Second, we have worked with fractions of decibels to avoid confusion, but it is important to remember that the confidence limits of our equations do not extend beyond whole decibel values, and that we must round off the answer at the end of our calculations.

Step 2: The sound field produced by the loudspeaker alone.

Now let us go back to our example and calculate the sound field produced by the loudspeaker. Our system microphone is still turned off and we are using an imaginary test signal for the calculations. We can save time by assuming that the test signal produces a sound level at the microphone of 71 dB — the same previously assumed for the unaided talker.
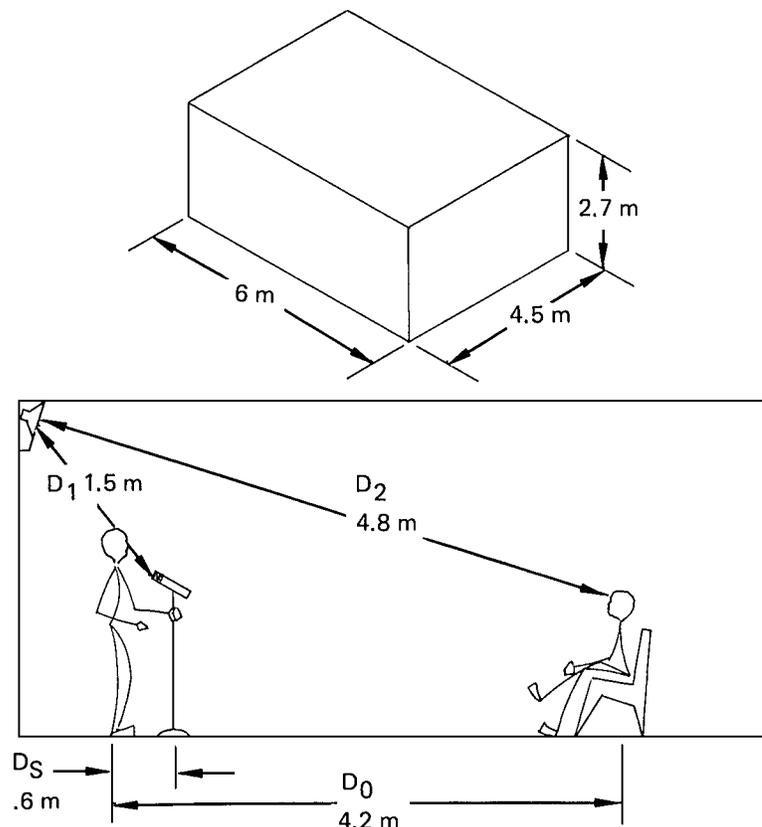

Figure 6-2. Indoor sound system gain calculations

The loudspeaker is mounted at the intersection of wall and ceiling. Its directivity index, therefore, is assumed to be 6 dB. In this room, the critical distance for the loudspeaker is 1.4 meters. This is almost the same as the distance from the loudspeaker to the microphone. Since the microphone is located at the loudspeaker's critical distance, and since we have assumed a level of 71 dB for the total sound field at this point, the direct field at the microphone must equal 71 dB minus 3 dB, or 68 dB.

The listener is 4.8 meters from the loudspeaker (more than 3 times the critical distance) and therefore, well into the reverberant field of the loudspeaker. We know that the level in the reverberant field must equal the level of the direct field alone at the critical distance. The sound level at the listener's position produced by the loudspeaker must, therefore, be 68 dB.

Step 3: Potential acoustic gain is now considered.

Since we deliberately set up the example to represent the condition of maximum theoretical gain for a properly equalized system, we can use these same figures to calculate the difference in level at the listener's position between the unaided talker and the talker operating with the system turned on. We have calculated that the unaided talker produces a level at the listener's position of 65.4 dB. We have also calculated that the level produced by the loudspeaker at the listener's position is 68 dB. The

acoustic gain of the system for this specific set of conditions must be the difference between the two, or only 2.6 dB. Obviously such a sound reinforcement system is not worth turning on in the first place.

Note that system acoustical gain is dependent upon the distance from the microphone to the talker. A more general concept is that of system delta. According to the Boner paper, the maximum theoretical D of a properly equalized system is unity. In our example, D works out to be -3 dB. Why?

The Boners emphasize that for maximum system gain the microphone must be in the direct field of the talker and in the reverberant field of the loudspeaker. But in our example, the microphone is not quite in the direct field of the talker and is located at the critical distance of the loudspeaker! To achieve more gain, we might move the microphone to a distance .3 meter from the talker and use a more directional loudspeaker. This would result in a 3 dB increase in D and a potential acoustic gain at the listener's position of about 9 dB.

In practice, however, we cannot operate the system at a point just below sustained feedback. Even if we modify the system as described above, our practical working gain will only be about 3 dB. Our calculations merely prove what we could have guessed in advance: in a room this small, where an unaided talker can easily produce a level of 65 dB throughout the room, a sound system is unnecessary and of no practical benefit.
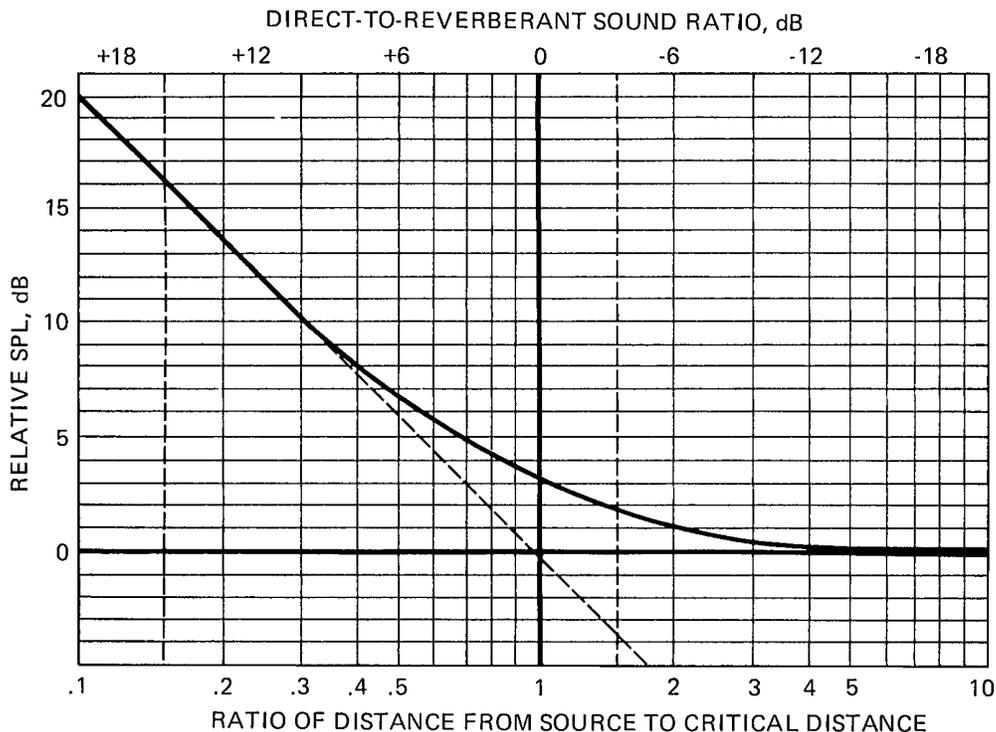

Figure 6-3. Relative SPL vs. distance from source in relation to critical distance

Calculations for a Medium-Size Room

Consider a more typical (and more complicated) situation in which the sound system is used in a larger room and in which a directional microphone is employed. Figures 6-4 and 6-5 show a room having a volume of 918 m³, a total surface area of 630 m² and $\overline{a} = 0.15$.

The first step is to calculate the room constant, and we would do well to examine the actual distribution of absorptive material in the room. Chapter 5 explains why the effective room constant R' in a particular situation may vary substantially from the figure given by the equation $R = S\overline{a}/(1 - \overline{a})$. Rather than complicate the example, however, assume that the equation really does work and that the room constant is about 110 m².

The next step is to calculate critical distances for the talker and the loudspeaker. Since the loudspeaker does not have a uniform radiation pattern, we must calculate its critical distance at the particular angle in which we are interested. Figure 6-5 shows the distances involved and the geometrical relationships between talker, microphone, loudspeaker and listener.
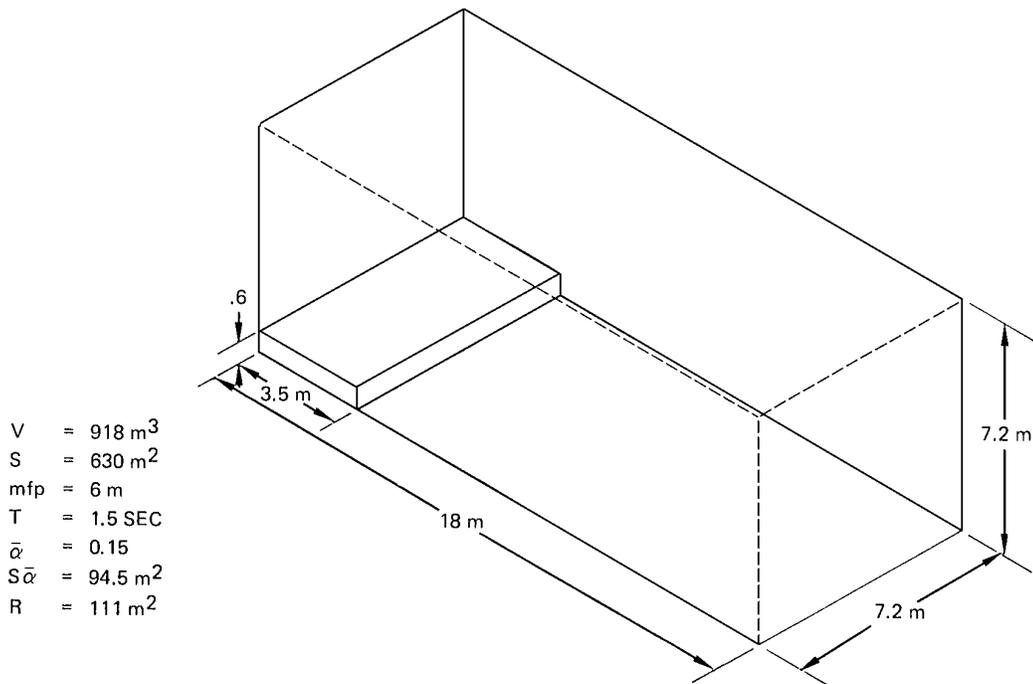

$$
\begin{aligned}
V &= 918\,m^3 \\
S &= 630\,m^2 \\
mfp &= 6\,m \\
T &= 1.5\,SEC \\
\overline{\alpha} &= 0.15 \\
S\overline{\alpha} &= 94.5\,m^2 \\
R &= 111\,m^2
\end{aligned}
$$

Figure 6-4.   A sound system in a medium-size room

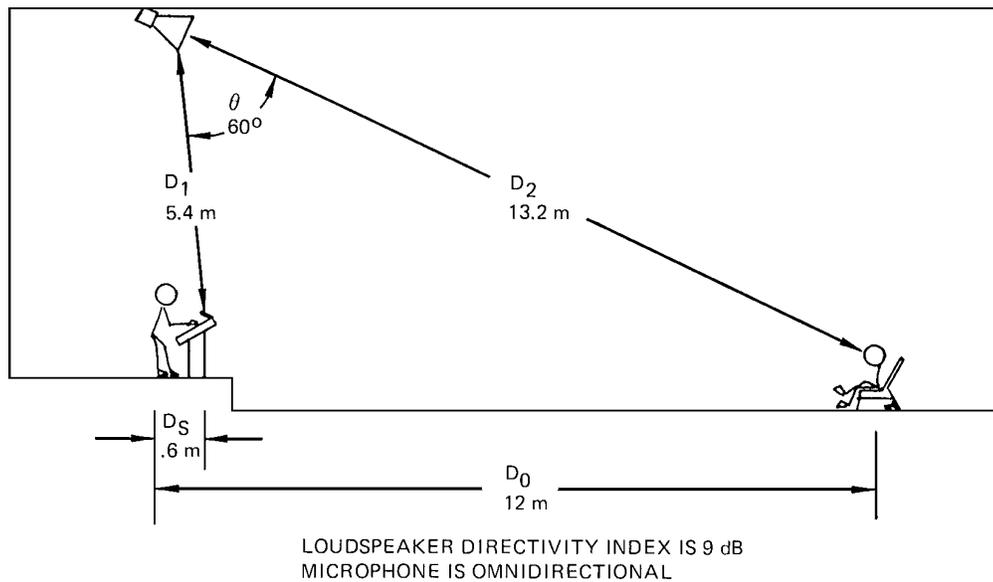LOUDSPEAKER DIRECTIVITY INDEX IS 9 dB
MICROPHONE IS OMNIDIRECTIONAL

Figure 6-5. Sound system in a medium-size room, gain calculations

In the frequency range of interest, the loudspeaker is assumed to have a directivity index along its primary axis of 9 dB. From Figure 6-6 we find the corresponding critical distance of 4.2 meters. The loudspeaker's directivity index at a vertical angle of 60° is assumed to be -3 dB, with a corresponding critical distance of 1 meter. The unaided talker has a directivity index of 3 dB and his critical distance must therefore be 2 meters.

Our next step in calculating system gain is to find the difference in level produced by an unaided talker at the listener position as contrasted with that at the microphone position. In this example the listener is 12 meters from the talker and the microphone again is .6 meters away.

The talker's critical distance of 2 meters is more than 3 times the microphone distance. Therefore, the microphone is well in the direct field of the talker. The listener is more than 3 times the critical distance and is well into the reverberant field of the unaided talker. Setting the level produced by the unaided talker at 70 dB for a distance of .6 meters, we calculate that the direct field at D_c must be 60 dB, and since the reverberant field must also equal 60 dB, the level produced by the unaided talker at the listener's position is 60 dB.

The third step is to make similar calculations for the loudspeaker alone. The listener is located on the major axis of the loudspeaker and is more than 3 times the critical distance of 4.2 meters. The microphone is located at a vertical angle of 60 degrees from the loudspeaker's major axis, and also is more than 3 times the critical distance (at this angle) of 1 meter. Both the listener and the microphone are located in the reverberant field of the loudspeaker.

If the sound level produced by the loudspeaker at the microphone can be no greater than 70 dB (the same level as the talker) then the level produced by the loudspeaker at the listener's position must also be 70 dB, since both are in the reverberant field.

Having established these relationships we know that the talker produces a level at the listener's position of 60 dB with the sound system off and 70 dB with the sound system on, or a maximum potential gain of 10 dB. Allowing 6 dB headroom in a properly equalized system, we still realize 4 dB gain at the listener's position, and the sound system can be said to provide a small but perceptible increase in sound level.

DIAGONAL LINES: CRITICAL DISTANCE IN METERS (OR FEET)

EQUATIONS:
CRITICAL DISTANCE = $0.14\sqrt{QR}$
DIRECTIVITY INDEX IN dB = $10 \log Q$

NOTE: EQUATIONS AND CHART CAN BE USED WITH ENGLISH OR METRIC UNITS. TO CONVERT CHART SCALES TO MORE CONVENIENT VALUES FOR METRIC CALCULATIONS, DIVIDE CRITICAL DISTANCES BY 10 AND ROOM CONSTANTS BY 100.
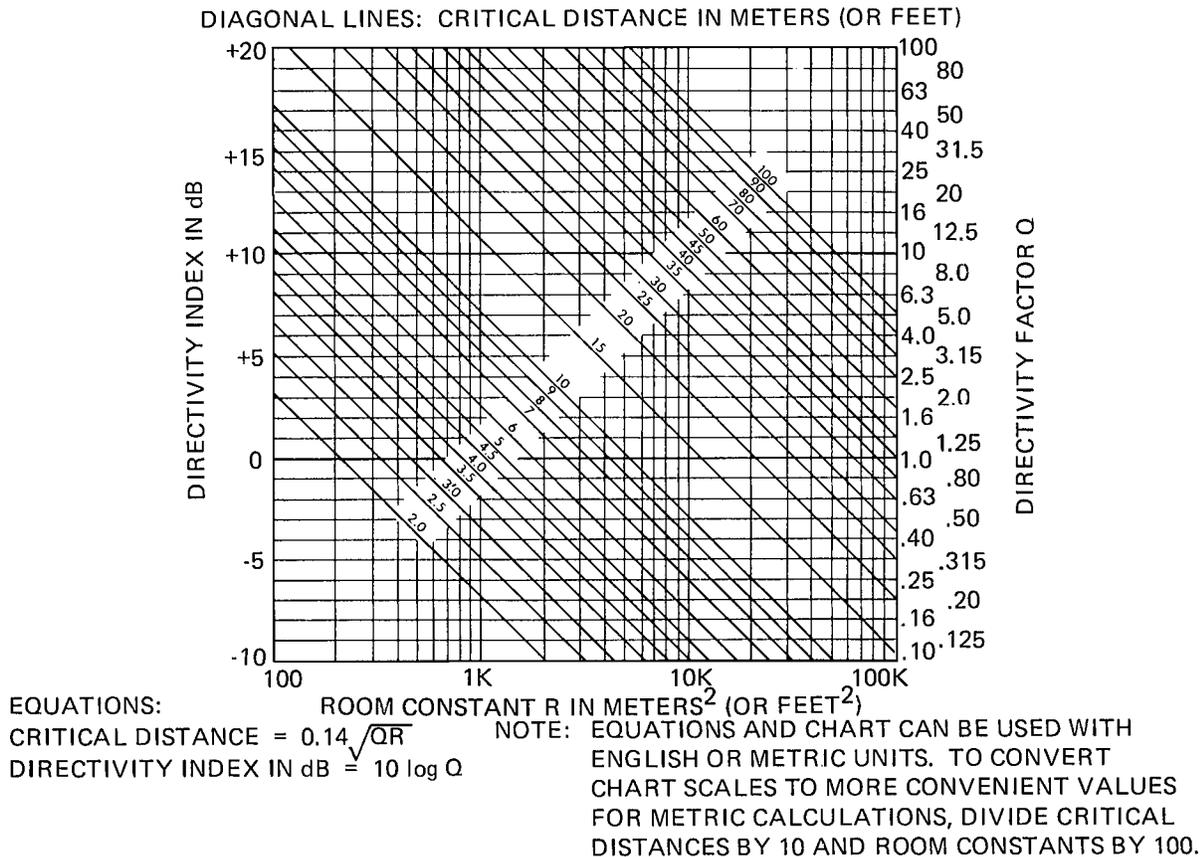
Figure 6-6. Critical distance as a function of room constant and directivity index or directivity factor

However, all of the preceding calculations have assumed that the microphone is an omnidirectional unit. What happens if we substitute a directional microphone? Figure 6-7 shows the additional geometrical relationships needed to calculate the increase in gain produced by a directional microphone.

Note that the distance from talker to microphone is still .6 meters and that the talker is assumed to be located along the major axis of the microphone. The loudspeaker is located 5.4 meters from the microphone along an angle of 75° from the major axis.

Figure 6-7 also shows a typical cardioid pattern for a directional microphone. The directivity index of such a microphone along its major axis is about 5 dB.

Since the talker is located on the major axis of the microphone, it "hears" his signal 5 dB louder than the random incidence reverberant field. In theory this should increase potential system gain by a factor of 5 dB.

But we must also consider the microphone's directional characteristics with relation to the loudspeaker. If the directivity index of the microphone at 0° is 5 dB, the polar pattern indicates that its directivity index at 75° must be about 3 dB. This tells us that even though the loudspeaker is 75° off the major axis of the microphone, it still provides 3 dB of discrimination *in favor* of the direct sound from the loudspeaker.

We know that the loudspeaker's directivity index is -3 dB along the axis between the loudspeaker and the microphone. We also know that the microphone's directivity index along this axis is +3 dB. The combined directivity indices along this axis must therefore, be 0 dB and we can find the equivalent critical distance from Figure 6-6.

The combined critical distance of loudspeaker and microphone along their common axis is about 1.3 meters. Since the distance between the two is more than 3 times this figure, the microphone still lies within the reverberant field of the loudspeaker. Using the directional microphone should therefore allow an increase in potential system gain before feedback of about 5 dB. (In practice, little more than 3 or 4 dB of additional gain can be achieved.)
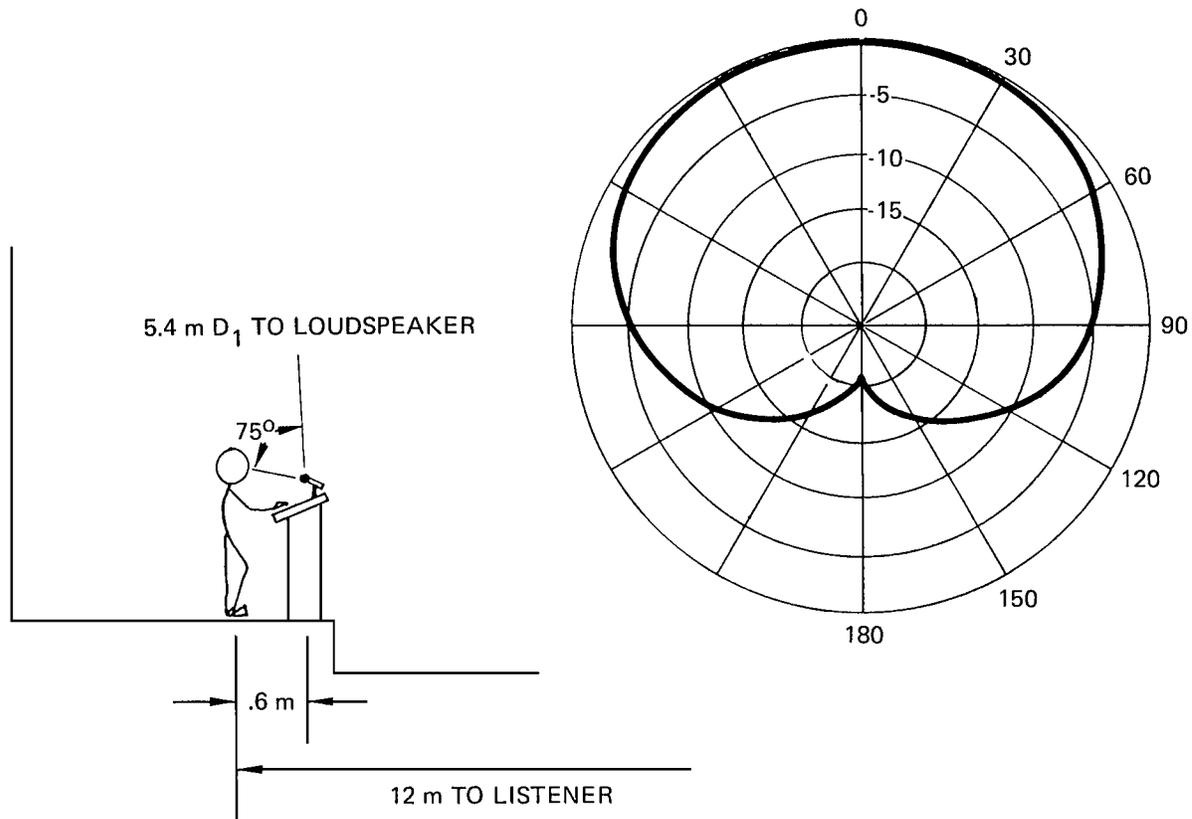
ASSUME A DIRECTIONAL MICROPHONE IS USED; ITS POLAR RESPONSE AT 1 kHz IS SHOWN ABOVE.

Figure 6-7. Characteristics of a cardioid microphone

Calculations for a Distributed Loudspeaker System

Figure 6-8 shows a moderate-size meeting room or lecture room. Its volume is 485 m³, surface area is about 440 m², and $\bar{a}$ is 0.2 when the room is empty. For an unaided talker in the empty room, R is 110 m². However, when the room is fully occupied, $\bar{a}$ increases to 0.4 and the corresponding room constant is 293 m². We calculate the critical distance for the unaided talker (directivity index of 3 dB) to be 2 meters in an empty room and 3.4 meters when the room is full.

The room is provided with a sound system diagrammed in Figure 6-9. Forty loudspeakers are mounted in the ceiling on 1.5 meter centers to give smooth pattern overlap up into the 4 kHz region. Coverage at ear level varies only 2 or 3 dB through the entire floor area.

The usual definitions of critical distance and direct-to-reverberant ratio are ambiguous for this kind of loudspeaker array. Here, however, we are interested only in potential acoustic gain, and the ambiguities can be ignored. We already have stated that the loudspeaker array lays down a uniform blanket of sound across the room. The relative directional and temporal components of the sound field do not enter into gain calculations.

An omnidirectional microphone is located .6 meters from the talker, less than 1/3 D_c. No matter how many people are present, the microphone is in the direct field of the talker.

The farthest listener is 9 meters from the talker, more than three times D_c when the room is empty, and just about three times D_c when the room is full.

If the unaided talker produces 70 dB sound level at the microphone with the system off, and if the amplified sound level can be no greater than 70 dB at the microphone with the system on, then the maximum level is 70 dB everywhere in the room.

$V = 485 \text{ m}^3$

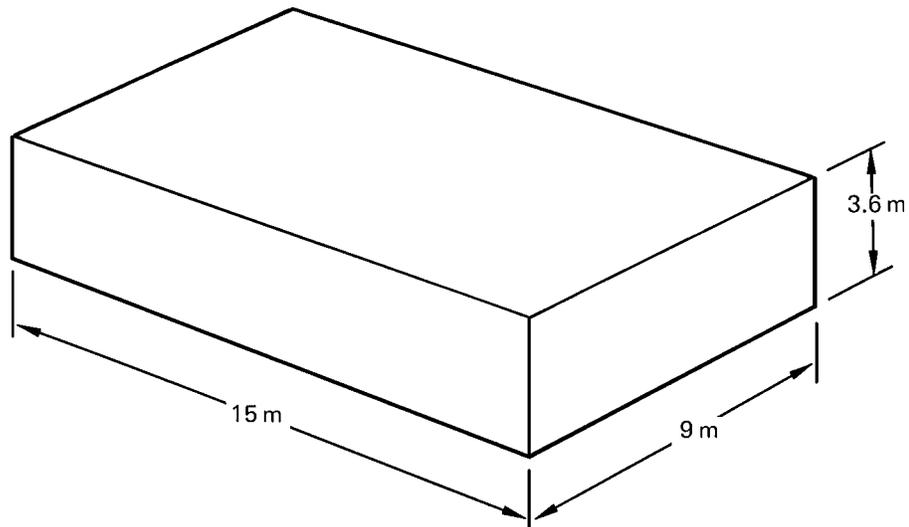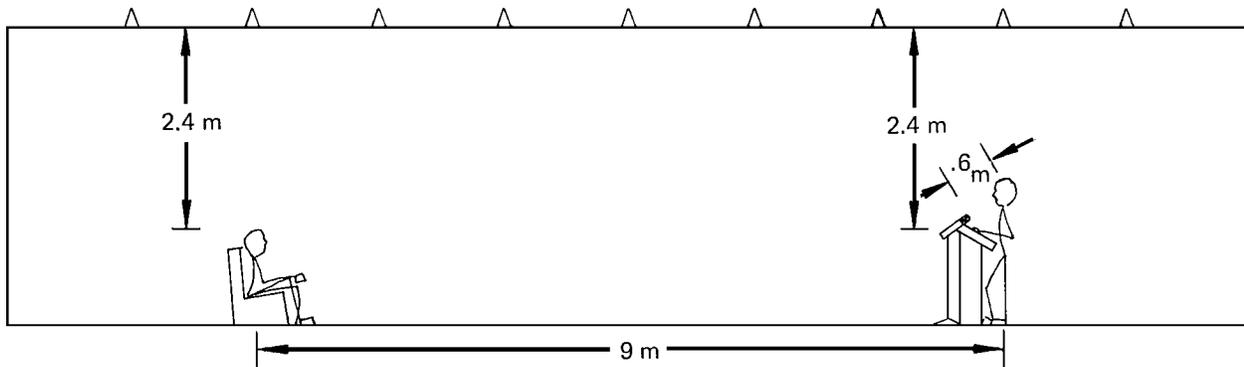$S = 440 \text{ m}^2$

$\bar{\alpha} = 0.2$

$R = 110 \text{ m}^2$

3.6 m

15 m

9 m

Figure 6-8.   A moderate-size lecture room

2.4 m

2.4 m

.6 m

9 m

MIC. IS OMNIDIRECTIONAL.
SPKRS. PRODUCE EVEN SOUND FIELD AT EAR LEVEL.

Figure 6-9. Sound system in a medium-size lecture room

From our calculations of critical distances, we see that the unaided talker will produce a sound level at the listener of 59 dB in an empty room and about 55 dB with a full audience. For a usable working delta of -6 dB, the calculated acoustic gain at the listener's position is about 5 dB in an empty room and about 9 dB when full.

Can we get more gain by turning off the loudspeaker directly over the microphone? Not in a densely packed array such as this. The loudspeakers are mounted close together to produce a uniform sound field at ear level. As a result, the contribution of any one loudspeaker is relatively small. However, by turning off *all* the loudspeakers in the performing area and covering only the audience, some increase in system gain may be realized.

In the example just given, each loudspeaker is assumed to have a directivity index in the speech frequency region of +6 dB at 0°, +3 dB at 45°, and 0 dB at 60°. Suppose we use only the 25 loudspeakers over the audience and turn off the 15 loudspeakers in the front of the room. In theory, the increase in potential gain is only 1 dB with a single listener or 2 dB when the audience area is filled. Even if we allow for the probability that most of the direct sound will be absorbed by the audience, it is unlikely that the gain increase will be more than 3 dB.

The calculations required to arrive at these conclusions are tedious but not difficult. The relative direct sound contribution from each of the loudspeakers at microphone and listener locations is calculated from knowledge of polar patterns and distances. By setting an arbitrary acoustic output per loudspeaker, it is then possible to estimate the sound level produced throughout the room by generally reflected sound (reverberant field) and that produced by reflected plus quasi-direct sound.

System Gain vs. Frequency Response

In the preceding examples we have not defined the frequency range in which gain calculations are to be made. In most sound systems the main reason for worrying about system gain is to make sure that the voice of a person talking can be amplified sufficiently to reach a comfortable listening level in all parts of the seating area. Therefore, the most important frequency band for calculating gain is that which contributes primarily to speech intelligibility: the region between 500 and 4000 Hz.

Below 500 Hz the response of the system can be gradually shelved, or attenuated, without seriously degrading the quality of speech. Above 4 kHz sound systems tend to take care of themselves, due to the increase in overall acoustical sound absorption. At very high frequencies, most environments are substantially absorptive, the air itself contributes considerable acoustical absorption and loudspeaker systems tend to become directional. These factors make it highly unusual to encounter feedback frequencies much above 2500 Hz.

To make sure that a sound reinforcement system will successfully amplify speech, it is a good idea to make gain calculations in at least two frequency bands. In a well-designed system, if calculations are made for the regions centered at 1 kHz and 4 kHz, chances are that no unforeseen problems in achieving desired system gain will be encountered.

However, the region below 500 Hz cannot simply be ignored. The room constant and the directivities of the loudspeaker system and the microphone should be checked in the 200 - 500 Hz range to make sure that there are not substantial deviations from the calculations made at 1 and 4 kHz. If the room has very little absorption below 1 kHz, and if the loudspeaker system becomes nondirectional in this region, it may be impossible to achieve satisfactory system gain without severely attenuating the mid-bass region. The result is the all too familiar system which provides satisfactory speech intelligibility, but which sounds like an amplified telephone.

The Indoor Gain Equation

From the foregoing discussions, we can appreciate the complexity of indoor system gain analysis and the need for accurately calculating the attenuation of sound along a given path, from either talker or loudspeaker, noting when we leave the direct field and make the transition into the reverberant field. If we were to attempt to establish a general system gain equation, we would have a very difficult task. However, in the special case where the microphone is in the talker's direct field, and both microphone and listener are in the loudspeaker's reverberant field, then the system gain equation simplifies considerably.

Let us consider such an indoor system, first with the system turned off, as shown in Figure 6-10. If the talker produces a level *L* at the microphone, then the level produced at the listener will be:

Level at listener = $L - 20 \log (D_{ct}/D_s)$, where D_{ct} is the critical distance of the talker. The assumption made here is that the level at the listener is entirely made up of the talker's reverberant field and that that level will be equal to the inverse square component at D_{ct}.

Now, the system is turned on, and the gain is advanced until the loudspeaker produces a level L at the microphone. At the same time, the loudspeaker will produce the same level L at the listener, since both microphone and listener are in the loudspeaker's reverberant field.

Subtracting the levels at the listener between the system on and the system off, we have:

$$\text{Difference} = L - \left[L - 20 \log (D_{ct}/D_s)\right]$$

or:

$$\text{Gain} = 20 \log D_{ct} - \log D_s$$

Finally, adding a 6 dB safety factor:

$$\text{Gain} = 20 \log D_{ct} - 20 \log D_s - 6$$

Note that there is only one variable, D_s, in this equation; D_{ct} is more or less fixed by the directivity of the talker and the acoustical properties of the room.

Of course there are many systems in which the microphone may be placed in the transition zone between the talker's direct and reverberant fields, or where the listener is located in the transition region between the loudspeaker's direct and reverberant fields. In these more complicated cases, the foregoing equation does not apply, and the designer must analyze the system, both on and off, pretty much as we went stepwise through the three examples at the start of this chapter.

Measuring Sound System Gain

Measuring the gain of a sound system in the field is usually done over a single band of frequencies. It is normally specified that system gain shall be measured over the octave-wide band centered at 1 kHz. Another common technique is to use pink noise which is then measured with the A-weighted scale. A typical specification for sound system gain might read as follows:

"The lectern microphone shall be used in its normal position. A small loudspeaker shall be mounted on a stand to simulate a person talking approximately .6 meters from the microphone. The response of this test loudspeaker shall be reasonably flat over the range from 250 - 4000 Hz.

"With the system turned off, the test loudspeaker shall be driven with a pink noise signal to produce a sound level of about 80 dB(A) at the system microphone. This level shall be measured with a precision sound level meter, using the "A" scale, with its microphone immediately adjacent to the sound system microphone.

"After noting the sound level at the system microphone with the sound system turned off, the sound system shall be turned on and its gain advanced to a point just below sustained oscillation. The amplified sound level shall be measured with the same sound level meter in the central part of the auditorium.

"The D of the sound system shall be calculated by subtracting the measured SPL at the microphone (system off) from the measured SPL in the auditorium (system on)."

The gain of the system is of course measured at some point in the auditorium and is the level difference at that point produced by the test loudspeaker before and after the system has been turned on. Details of the measurements are shown in Figure 6-11.

$L - 20 \log (D_{ct}/D_s)$

$[L]$

D_s

L

$[L]$
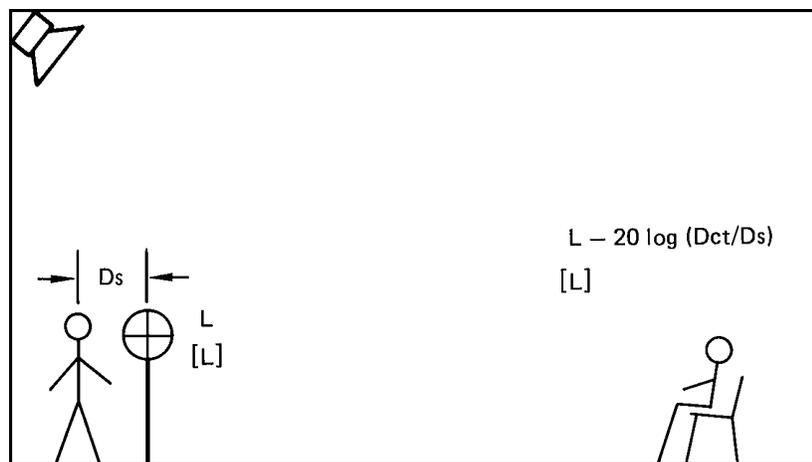
BRACKETS INDICATE LEVELS WITH SYSTEM ON.

Figure 6-10. Conditions for the indoor system gain equation

General Requirements for Speech Intelligibility

The requirements for speech intelligibility are basically the same for unamplified as for amplified speech. The most important factors are:

1. Speech level versus ambient noise level. Every effort should be made to minimize noise due to air handling systems and outside interferences. In general, the noise level should be 25 dB or greater below the lowest speech levels which are expected. However, for quite high levels of reinforced speech, as may be encountered outdoors, a noise level 10 to 15 dB below speech levels may be tolerated.

2. Reverberation time. Speech syllables occur three or four times per second. For reverberation times of 1.5 seconds or less, the effect of reverberant overhang on the clarity of speech will be minimal.

3. Direct-to-reverberant ratio. For reverberation times in excess of 1.5 seconds, the clarity of speech is a function of both reverberation time and the ratio of direct-to-reverberant sound.

In an important paper (8), Peutz set forth a method of estimating speech intelligibility which has found considerable application in sound system design. The Peutz findings were compiled on the basis of data gathered over a period of years. The data and the method used to arrive at the published conclusion are clearly set forth in the paper itself. The conclusions can be summarized as follows:

1. In practice, the articulation loss of consonants can be used as a single indicator of intelligibility. Although the original research of Peutz was in Dutch speech, the findings seem to be equally applicable to English.

2. As would be expected, the researchers found wide variations in both talkers and listeners. However, a 15% articulation loss of consonants seems to be the maximum allowable for acceptable speech intelligibility. In other words, if articulation loss of consonants exceeds 15% for the majority of listeners, most of those people will find the intelligibility of speech to be unacceptable.

3. Articulation loss of consonants can be estimated for typical rooms. Articulation loss of consonants is a function of reverberation time and the direct-to-reverberant sound ratio.

4. As a listener moves farther from a talker (decreasing the direct-to-reverberant sound ratio) articulation loss of consonants increases. That is, intelligibility becomes less as the direct-to-reverberant ratio decreases. However, this relationship is maintained only to a certain distance, beyond which no further change takes place. The boundary corresponds to a direct-to-reverberant ratio of -10 dB.
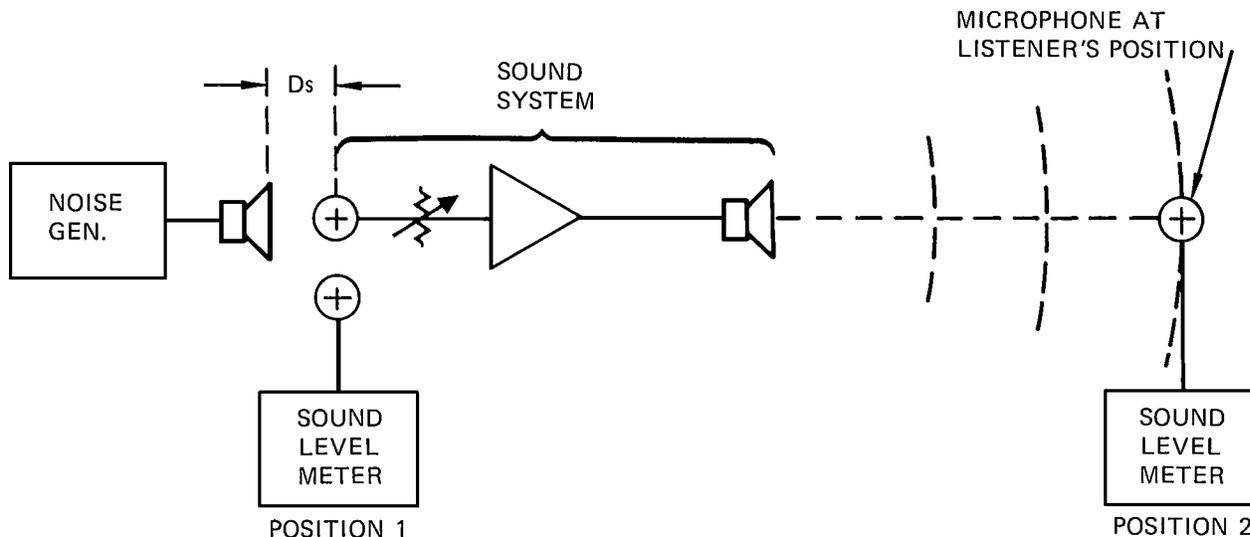
SYSTEM GAIN = LEVEL, POSITION 2, SYSTEM ON − LEVEL, POSITION 2, SYSTEM OFF

SYSTEM Δ = LEVEL, POSITION 2, SYSTEM ON − LEVEL, POSITION 1, SYSTEM OFF

Figure 6-1 1. Measurement of sound system gain and delta (D)

The last point is illustrated graphically in Figure 6-12, adapted from the Peutz paper. Each of the diagonal lines corresponds to a particular reverberation time. Each shelves at a point corresponding to a direct-to-reverberant sound ratio of -10 dB. Note that the shelf may lie above or below the 15% figure depending upon the reverberation time of the room. This agrees with other published information on intelligibility. For example, Rettinger points out that in rooms having a reverberation time of 1.25 seconds or less, direct sound and early reflections always make up the greater portion of the total sound field. Intelligibility in such rooms is good regardless of the direct-to-reverberant sound ratio at any given listening position. Conversely, anyone who has worked in extremely large reverberant spaces such as swimming pools or gymnasiums knows that intelligibility deteriorates rapidly at any point much beyond the critical distance. According to the chart, a 15% articulation loss of consonants in a room having a reverberation time of 5 seconds corresponds to a direct-to-reverberant sound ratio of only - 5.5 dB.

Problems associated with speech intelligibility in enclosed spaces have received a great deal of attention prior to the publication of the Peutz paper. The virtue of Peutz' method for estimating speech intelligibility is its simplicity. It must be remembered, however, that a number of contributing factors are ignored in this one simple calculation. The chart assumes that satisfactory loudness can be achieved and that there is no problem with interference from ambient noise. It also postulates a single source of sound and a well behaved, diffuse reverberant sound field.

The data from the Peutz paper have been recharted in a form more convenient for the sound contractor in Figure 6-13. Here we have arbitrarily labeled the estimated intelligibility of a talker or a sound system as "satisfactory", "good", or "excellent", depending upon the calculated articulation loss of consonants.

There often is a dramatic difference in the acoustical properties of a room depending upon the size of the audience. Calculations should be made on the basis of the "worst case" condition. In some highly reverberant churches particularly, it may turn out that there is no practical way to achieve good intelligibility through the entire seating area when the church is almost empty. The solution may involve acoustical treatment to lessen the difference between a full and an empty church, or it may involve a fairly sophisticated sound system design in which reinforced sound is delivered only to the forward pews when the congregation is small (presuming that a small congregation can be coaxed into the forward pews).
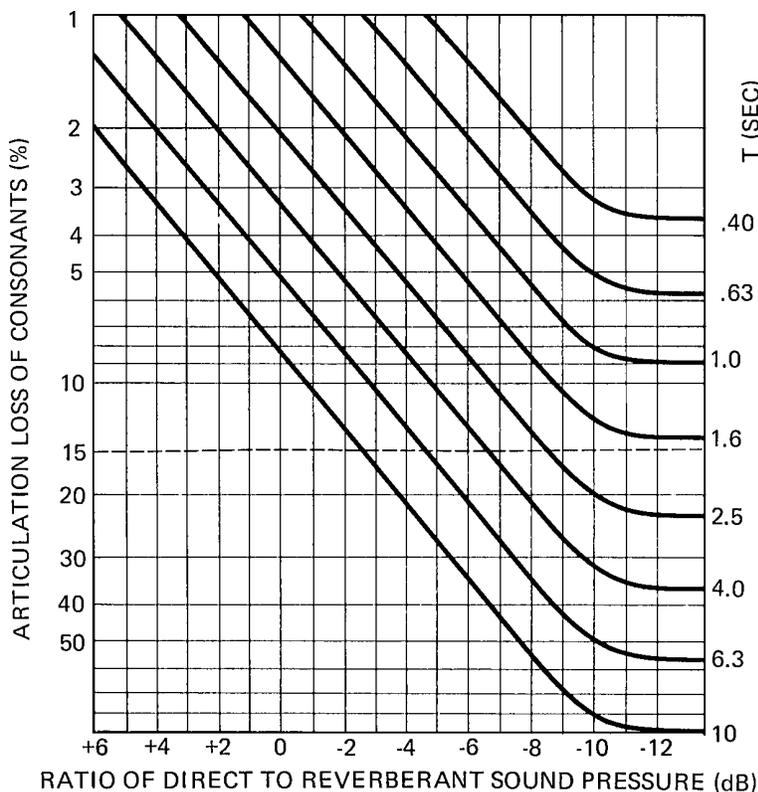

Figure 6-12. Probable articulation loss of consonants vs. reverberation time & direct-to-reverberant sound ratio

Also, local acoustical conditions may exist which are not taken into account by statistical theory and, therefore, not covered by the Peutz findings or any of the other equations we have studied. Such localized dead spots or zones of interference may not be discovered until the sound system is installed. In large reverberant spaces, sufficient flexibility should always be built into the sound system design to allow for such surprises.

The effect of masking by unwanted background noise has been touched on only briefly in this section. Such unwanted noise may be produced by sound from the outside environment, by noisy air handling equipment, by noisy backstage mechanical equipment or by the audience itself. For good listening conditions, the level of ambient noise as measured on the "A" scale should be at least 10 dB below the desired signal. Since the optimum level for reproduced speech in the absence of strong background noise is 65 - 70 dB(A) this means that background noise with a full audience should not exceed 55 dB(A). In auditoriums and concert halls, acoustical designers normally attempt to reduce background noise in an empty house to a level not exceeding 25 dB(A). In a church or meeting hall, the maximum tolerable background noise for an empty room is about 40 dB(A).

A sound reinforcement system cannot be turned up indefinitely. In many situations it is difficult enough to achieve a useful operating level of 60 - 65 dB(A) without feedback. It is easy to see, therefore, that the presence of excessive background noise can render an otherwise good sound reinforcement system unsatisfactory.

As an example of how the Peutz analysis can dictate the type of sound system to be used, let us consider a reinforcement system to be used in a large reverberant church. Details are shown in Figure 6-14.

Let us assume that the reverberation time is 4 seconds at mid-frequencies and that the designer's first choice is a single-point loudspeaker array to be placed high above the chancel. Coverage requirements pretty much dictate the directional characteristics of the array, and let us assume that the array will consist of two JBL Bi-Radial horns: 20° by 40° for far coverage, and 90° by 40° for near coverage. What we wish to calculate is the direct-to-reverberant ratio at selected points in the audience area to determine if the Peutz criteria for acceptable intelligibility can be met. The most direct way of doing this is to calculate the total reverberant level in the room for a given power input to each horn and compare it with the direct sound coverage provided by each horn over its coverage angle.
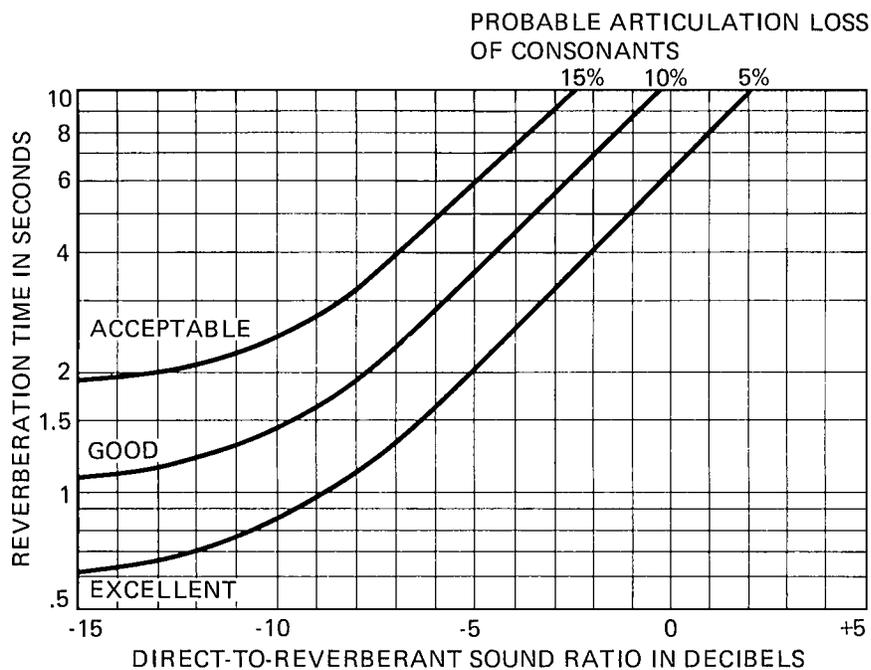
PROBABLE ARTICULATION LOSS
OF CONSONANTS

Chart developed from data published by V.M.A. Peutz in
Journal of the AES, Vol. 19, December, 1971

Figure 6-13. Probable intelligibility as a function of reverberation time
and direct-to-reverberant sound ratio

The analysis shown in Figure 6-14 indicates that when each of the two horns is powered by one watt, the reverberant field in the room (read directly from Figure 5-21) is 94 dB-SPL. The direct field level provided by each horn over its coverage angle is about 85 dB-SPL. This produces a direct-to-reverberant ratio of -9 dB, and an inspection of Figure 6-13 tells us that the system will have marginal intelligibility. Note that for 4 seconds of reverberation time, the direct-to-reverberant ratio should be no less than about -7 dB if acceptable intelligibility is to be expected. This simple analysis has told us that, on paper, we have designed a sound system which will likely fail to satisfy the customer.

Had the system consisted of a single horn, knowledge of its on-axis DI and Q could have led quickly to a determination of critical distance, and the direct-to-reverberant ratio could have been scaled from D_C. However, for the composite array analyzed here, there is no single value of DI or Q which can be used, and a direct calculation of the overall reverberant level, using what we know about the efficiency of the transducers, and making a comparison with the direct field, based on the sensitivities of the transducers, is the quickest way to solve the problem.

But the question remains: What kind of system will work in this large resonant room? Clearly, a distributed system is called for. In such a system, a number of lower-powered loudspeakers are placed on columns on each side of the church, each loudspeaker covering a distance of perhaps no more than 5 or 6 meters. In this way, the direct-to-reverberant ratio can be kept high. If such a system is further zoned into appropriate time delays, the effect will be quite natural, with subjective source localization remaining toward the front of the listening space. Details of this are shown in Figure 6-15.

Again, we calculate the total reverberant level and compare it with the longest throw each loudspeaker will be called upon to handle. There are 14 loudspeakers, 7 on each side. Let us assume that the efficiency of these loudspeakers is 1.2% and that their sensitivity is 95 dB, 1 watt at 1 meter. Feeding one watt into each loudspeaker results in a total acoustical power of 14 x .012, or 0.17 watt. Again using Figure 5-21, we observe that the reverberant level will be 92 dB-SPL. The longest throw each loudspeaker has to cover is, say, 4 meters. Since the 1-watt, 1-meter sensitivity is 95 dB, the direct field for each loudspeaker will be 12 dB lower, or 83 dB.
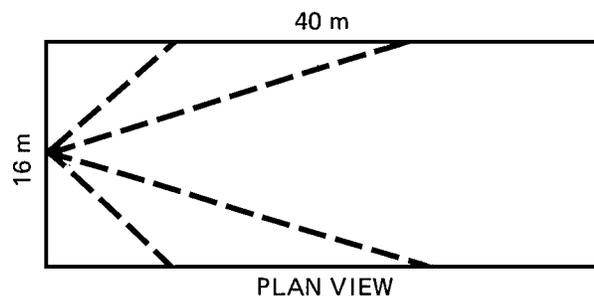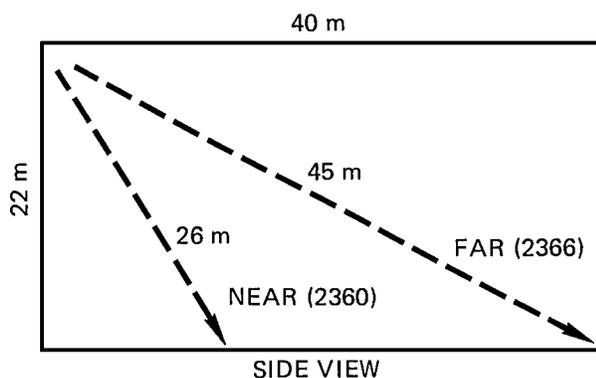
$$V = 14,000 \text{ m}^3$$
$$S = 3,740 \text{ m}^2$$
$$RT = 4 \text{ sec}$$
$$\bar{a} = .14$$
$$R = 610 \text{ m}^2$$
HF eff (%) = 20%

$$\text{REVERBERANT SPL} = 10 \log\left(\frac{2 \text{ watts x } .2}{610}\right) + 126$$
$$= \underline{94 \text{ dB - SPL}}$$

DIRECT FIELD DATA:

1. 2366 at 45 m: 1 watt produces 85 dB-SPL
2. 2360 at 26 m: 1 watt produces 85 dB - SPL

∴ Direct-to-reverberant ratio = 85-94 = 9 dB

Figure 6-14.   Analysis of intelligibility criteria

Thus, the direct-to-reverberant ratio will be 83-92, or -9 dB. This is still not good enough, but we must remember that more than half the listeners will be closer to a loudspeaker than 4 meters. Another very important point we have not yet considered is the fact that the distributed loudspeakers are aimed almost totally into the audience, with its absorption coefficient considerably greater than $\overline{a}$ of .12. This is the appropriate time to use R' instead of R in our calculations.

Calculating R' based upon an a' of .95 for the audience area in the 1 kHz band:

$$R' = S\overline{a}/(1 - a') = 375/.05 = 7500 \text{ m}^2.$$

Recalculating the reverberant level from Figure 5-21, we get 80 dB-SPL. The new direct-to-reverberant ratio is 83 - 80, or +3 dB, and the system will be quite workable.

Will the reverberant level really be only 80 dB? In actuality, we might observe something a little higher than 80 dB, but not enough to alter our analysis significantly.

We can also ask the question of whether our analysis using R' would have materially affected the performance of the central array system. A rigorous analysis would be a little tedious, but we can make a simplifying assumption. Let us assume that half of the direct sound from the central array was incident on the audience with its .95 absorption coefficient. Let us round this off and call it 1.0 instead, resulting in no sound at all being reflected from the audience. This would only lower the reverberant level in the room by 3 dB, hardly enough to make the direct-to-reverberant ratio workable.

More than any other we have carried out in this chapter, this analysis points up the multi-dimensional complexity of sound system design. Again, we state that there are no easy solutions or simple equations. Instead, there is only informed rational analysis and thoughtful balancing of many factors.
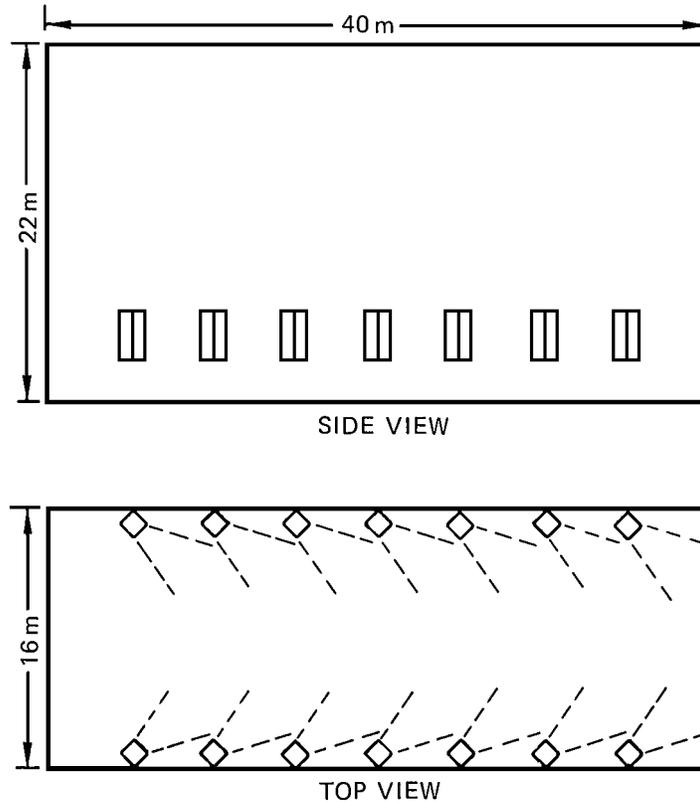

Figure 6-15. A distributed system in a large church

The Role of Time Delay in Sound Reinforcement

The preceding example mentioned time delay as a means of preserving naturalness in a distributed system. This comes about by way of the Haas (or precedence) effect (5), which is illustrated in Figure 6-16. If two loudspeakers are fed the same signal, a listener mid-way between them will localize the source of sound directly ahead (A). At *B*, we have introduced a delay in one of the otherwise identical channels, and the listener will clearly localize toward the *earlier* loudspeaker. At *C*. the leading signal has been reduced in level, resulting in an effect of equal loudness at both loudspeakers. This has the approximate effect of restoring the apparent localization to the center. While this tradeoff is not an exact one, the values shown in the graph at *D* indicate the approximate trading value between level and delay for equal loudness at both loudspeakers.

Figure 6-16E shows how delay is typically implemented in sound reinforcement. Here, that portion of the audience seated under the balcony does not get adequate coverage from the central array. Small loudspeakers placed in the balcony soffit can provide proper coverage only if they are delayed so that the sound arrives at the listeners in step with that from the central array. In this way, the listener tends to localize the source of sound at the central array — not at the soffit loudspeakers. If the soffit loudspeakers are not delayed, listeners under the balcony would localize sound directly overhead, and those listeners just in front of the balcony would be disturbed by the undelayed sound. In practice, the delay is usually set for an additional 20 msec in order to minimize comb filtering in the overlap zone between direct and delayed sound fields.

The ready availability of solid state digital delay units has made time delay an indispensable element in sound system design.
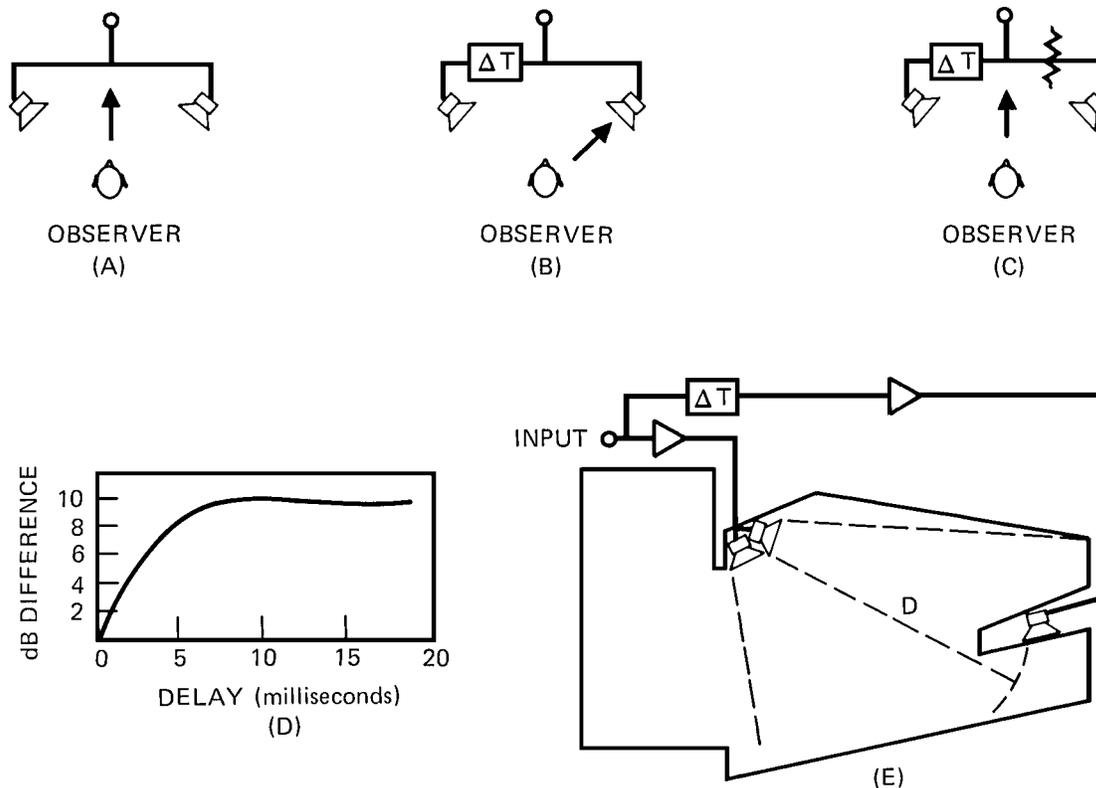
TIME DELAY = D/344 seconds
(D MEASURED IN METERS)

Figure 6-16. The Haas, or precedence, effect

System Equalization and Power Response of Loudspeakers

It is customary to equalize all professional sound reinforcement systems for two reasons: overall response shaping and control of feedback. The overall response may be made smoother for a more natural effect through the use of broadband equalization and through the proper choice of drive components themselves. Where high system gain is required, narrow-band notch filters may successfully remove the tendency of the system to "ring" at certain frequencies. We will examine the requirements of broad-band equalization first.

A sound system is equalized by feeding pink noise (equal power per octave) into the system and adjusting the system's response to fit a preferred contour at some point in the middle of the house. This procedure is shown in Figure 6-17A. The response contour most often used today is shown at *B*.

At the point in the house where the measurement is made, the reverberant field predominates, and what we are shaping with the equalizer is actually the power response of the loudspeaker as influenced by boundary absorption in the room. If the loudspeaker's power response is smooth to begin with, then all is well. However, if, as in some older designs, the system's power response is irregular, then equalization will usually make things worse, as shown in Figure 6-18.
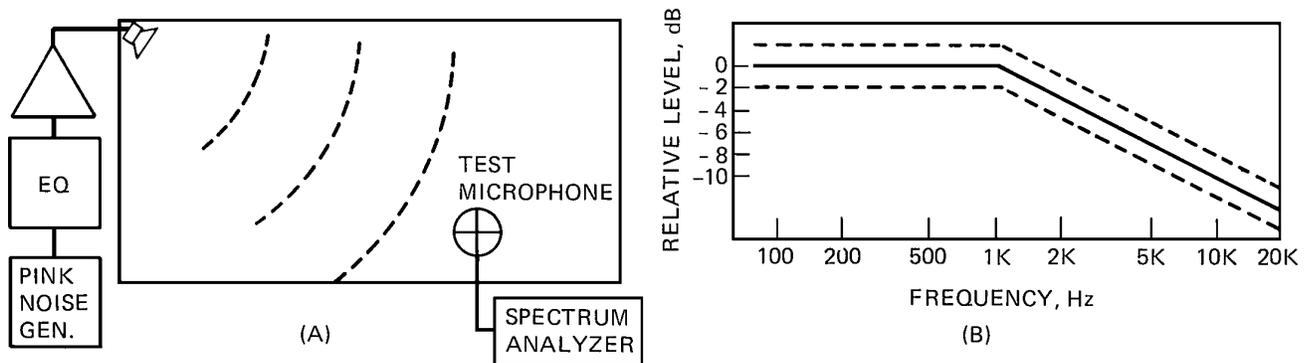

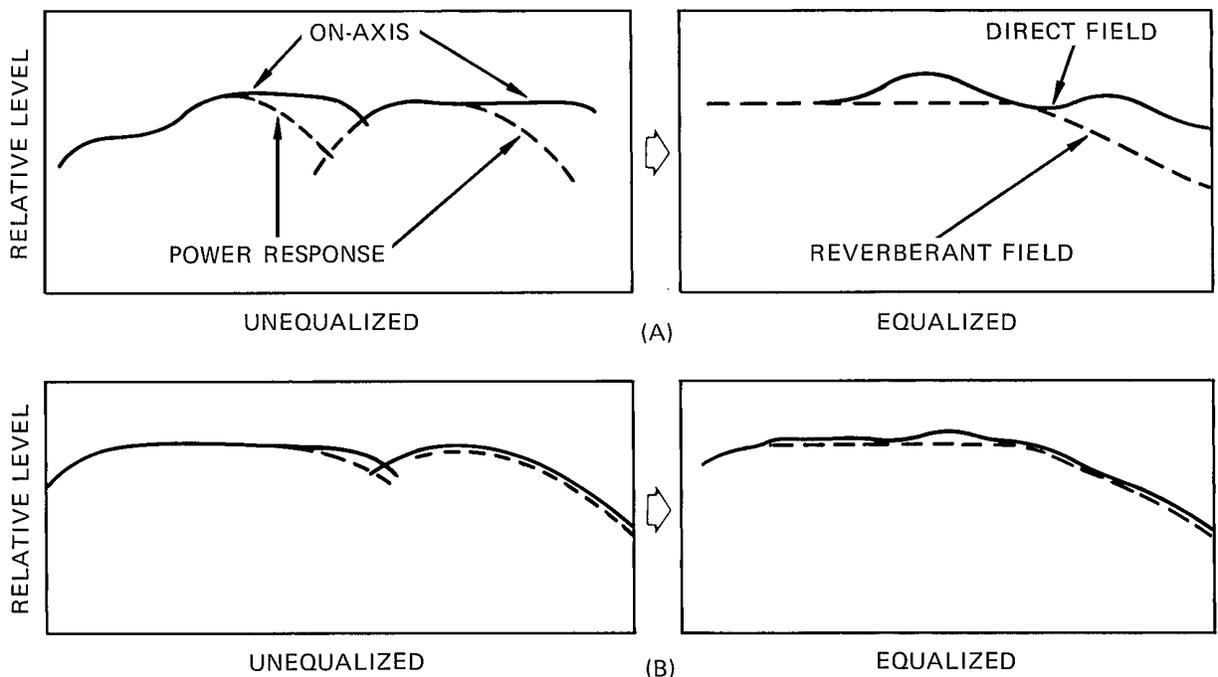Figure 6-17. Sound system equalization procedure

Figure 6-18. System equalization

At *A*, we see the on-axis (solid curve) and power (dotted curve) response of a 2-way system making use of a ported LF horn unit and an older type HF radial horn. When such a system is equalized for smooth power response, as in the case of the standard mid-house equalization procedure, then the on-axis, or direct field response of the system will have a couple of "bumps" in its response. This will have the effect of making both speech and music sound unnatural.

Now let us examine the case at *B*. Here, the LF part of the system consists of a single 380 mm (15") LF driver in a ported enclosure, and the HF horn is a JBL 2360 Bi-Radial. Note that the power response and on-axis response very nearly lie over each other. Thus, the adjustment of the system out in the house will result in both reverberant field response (power response) and direct field response (on-axis response) tracking each other closely. Such a system can often be broad-band-equalized merely through the proper choice of components, dividing network and transducer drive levels, requiring little, if any, added electronic equalization.

The graph shown in Figure 6-19 shows this clearly. Here, we have plotted the variation in R over the frequency range for a large auditorium. The room we have chosen has the following characteristics:

$$V = 13,500 \text{ m}^3$$

$$S = 3538 \text{ m}^2$$

$$RT_{125 \text{ Hz}} = 1.5 \text{ sec} \qquad R_{125 \text{ Hz}} = 1774 \text{ m}^2$$

$$RT_{1 \text{ kHz}} = 1.2 \text{ sec} \qquad R_{1 \text{ kHz}} = 2358 \text{ m}^2$$

$$RT_{4 \text{ kHz}} = 0.8 \text{ sec} \qquad R_{4 \text{ kHz}} = 3989 \text{ m}^2$$

This spread between reverberation times at low, mid, and high frequencies is typical of a good auditorium. When we calculate the room constant as a function of frequency and plot it, along with the sound level that would be produced by one acoustic watt in the room, we see that the total variation in SPL is only about 3 dB. The importance of this observation is that, if we had a loudspeaker system exhibiting flat power response, then it would produce a reverberant SPL in this auditorium that would vary no more than the inverse of the curve shown in Figure 6-19. Obviously, the smoother the power response of a loudspeaker, the less equalization it will require and the more natural it will sound on all types of program.

Another use of equalization is in controlling feedback. As we have stated many times, a sound reinforcement system should be operated at least 6 dB below the point of feedback if it is to be stable. Through careful and selective use of narrow-band notch filters, the first several ring modes of a sound system can be minimized, and the overall system gain can be increased perhaps 3 or 4 dB. The practice of narrow-band equalization is complex, and it is best left to those who have been trained in it.
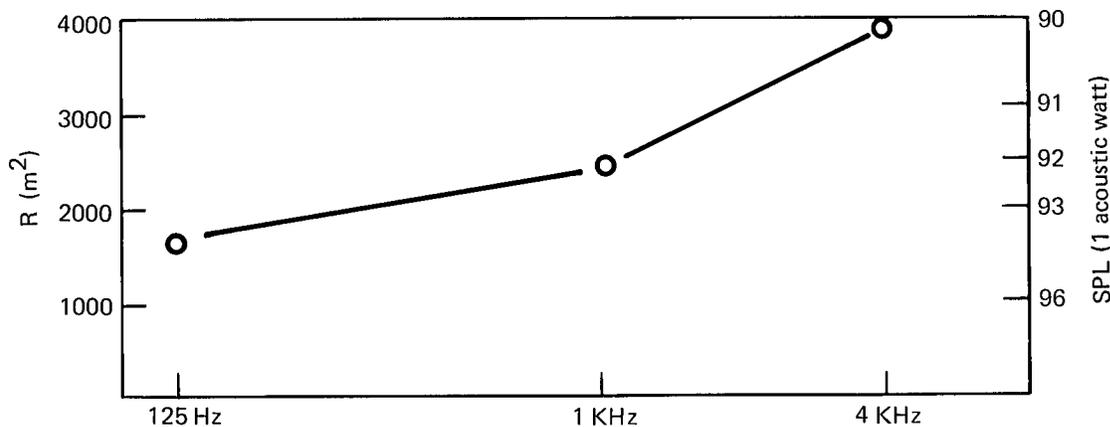

Figure 6-19. Variation in R and reverberant level with frequency

System Design Overview

There is a rational approach to indoor sound reinforcement system design, and it can be broken down into the following steps:

1. Lay out the coverage requirements, generally starting with a central array. Determine the drive requirements for each element in the array.

2. Calculate both direct field and reverberant field levels at various parts of the audience area, and then determine if their ratios, in combination with the reverberation time of the room, will result in adequate intelligibility. These calculations are most important in the 1 kHz range, but they should also be made in the 125 Hz and 4 kHz ranges as well. Determine the requirements for adequate gain, noting the value of D_S that will be required in normal operation.

3. If the intelligibility criteria are met, then the system can be completed. If the intelligibility criteria indicate an inadequate direct-to-reverberant ratio, consider the possibility of increasing R through the addition of acoustical absorption in the room. In existing rooms, this may not be possible; however, for rooms still in the design phase, it may be possible to increase the amount of absorption.

4. If a recalculation of the room parameters indicates that a central array will work, then the design can be completed. If not, the next step is to determine the nature of a distributed system that will satisfy the requirements of intelligibility. A central array can often be designed to cover just the front part of a room, with delayed loudspeakers covering the rear of the room. In marginal cases, this is likely to be more satisfactory than an all-out distributed system.

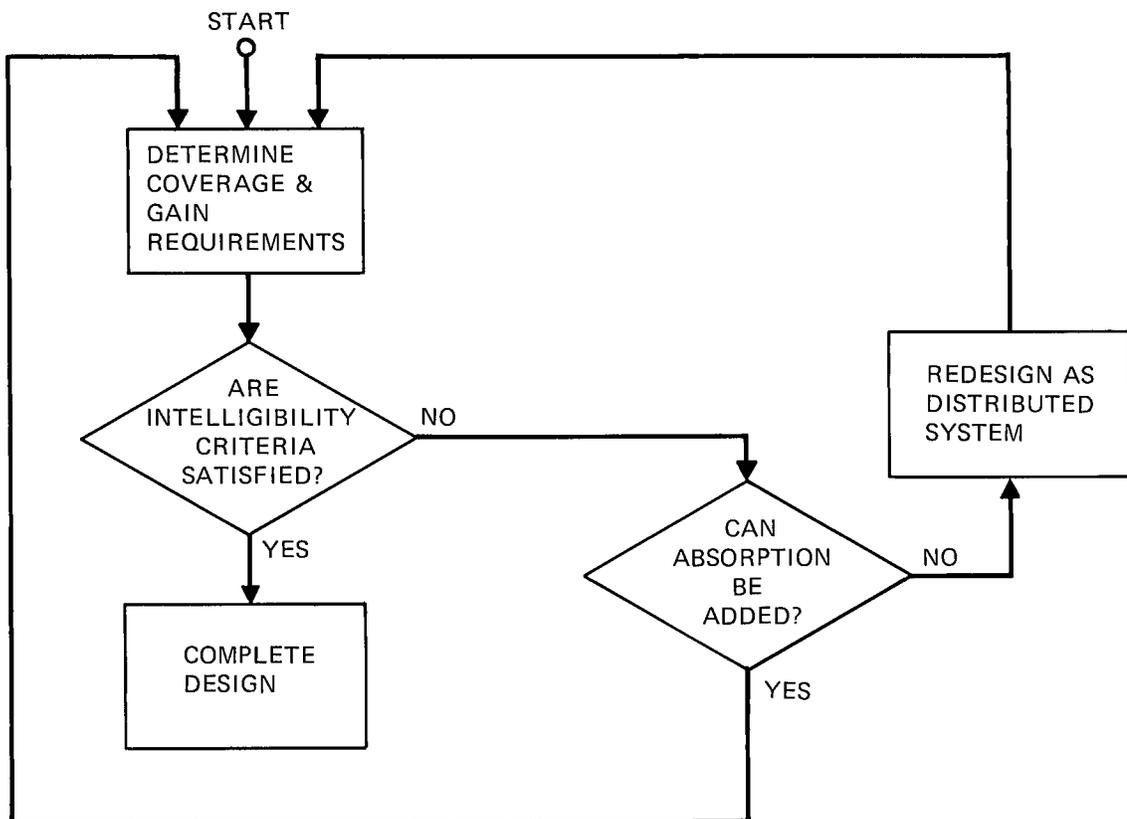The entire process described above has been reduced to the flow chart shown in Figure 6-20.

Figure 6-20. Flow diagram for system design

Chapter 7: System Architecture and Layout

Introduction

Just as the building architect interprets a set of requirements into flexible and efficient living or working spaces, the designer of a sound reinforcement system similarly interprets a set of requirements, laying out all aspects of the system in an orderly fashion. A full sound system specification will detail almost *everything*, including all equipment choices and alternatives, rack space requirements, wire gauges and markings, and nominal signal operating levels. In addition, the electroacoustical aspects of the system will have been worked out well ahead of time so that there will be few surprises when the system is turned on for the first time.

The consultant or design engineer lays out the broad system parameters, but it is the sound contractor who is responsible for all component layout and orderly completion of the system, along with documentation for usage as well as maintenance. System architecture also addresses signal flow and nominal operating levels, consistent with the requirements of the system. The best designs are usually the simplest and most straightforward ones.

In this chapter we will cover several design projects, beginning with basic design goals and fundamental performance specifications. We will then move on to system descriptions and layout, suggesting ways that the specification can be met. We will concentrate on the electroacoustical problems that are fundamental to each case study. By way of review, we will first discuss a few basic audio engineering subjects, beginning with an abbreviated signal flow diagram for a relatively simple speech reinforcement system.

Typical Signal Flow Diagram

Assume that we have the following requirements:

1. Up to ten microphones may be needed at different locations.

2. The system is to be used primarily for speech reinforcement.

3. The system shall be able to produce peak levels up to 85 dB-SPL in all parts of the house under all speech input conditions, including weak talkers. The room noise level is about 25 dB(A).

The most basic interpretation of these requirements tells us the following:

1. A small Soundcraft or Spirit console should suffice for all input configurations and routing control.

2. A single central array is the preferred system type, based on the desire for most natural speech reproduction. The array may be specified using individual HF and LF components; alternatively, an appropriate full-range system with integral rigging capability may be specified, as we will show here.

3. Both biamplification and system response equalization are recommended, and this suggests that a digital loudspeaker controller be used for frequency division, time alignment, and system response equalization.

Note that there are many points in the system where we can set or change gain. There is always considerable *gain overlap* in the electronic devices used in sound system work. The purpose of this is to allow for a great variety of input conditions as well as to allow the equipment to be configured in different ways, as required. It is critical that the designer specify a nominal setting of each gain control, locking off, when possible, those controls that will not — or should not — be altered during normal system use. This important setting of gain relationships should be based on the absolute requirement that the input noise floor of the system should not be degraded later in the chain, and that no early stage of amplification should overload before the output power amplifier overloads. In our exercise here, we

will simplify things by considering only a single microphone path through the system to a single loudspeaker.

For the moment, let us consider only the abbreviated console flow diagram shown in the upper part of Figure 7-1A. Microphone ratings in use today state the unloaded output voltage when the unit is placed in a sound field of 94 dB SPL. Normal speech level at an operating distance of .5 meter is about 72 dB SPL; If we are using a microphone with a sensitivity of 10 mV/Pa, the microphone's nominal voltage output in the 72 dB sound field will be:

$$E = 10^{22/20} \times 10 \text{ mV} = .8 \text{ mVrms}$$

Step One:

Set a reference input of .8 mVrms at 1000 Hz at one of the microphone inputs on the console. With the input and output faders at their nominal "zero" markings, set the microphone's input trim control for a console output of 0.4 Vrms. (Alternatively, a stable sound pressure level of 72 dB may be generated at the microphone, and the microphone trim setting adjusted for 0.4. Vrms output.) In making this setting, the trim potentiometer marker will normally be somewhere between 10 o'clock and 2 o'clock. This

setting represents a nominal operating point for the microphone/console combination, and there is ample flexibility for operating the system above or below this setting, as may be required by weak or loud talkers. Frequency division and system equalization are to be carried out by a digital controller, the JBL model DSC260. The loudspeaker to be used is the JBL model SR4726A, and the recommended amplifier is the JBL model MPX600. Typical operating levels are as shown in the lower portion of Figure 7-1A.

The level diagram shown in Figure 7-1B shows that, at the power amplifier's output, the noise level of the microphone is about 3 dB greater than the noise contributed by the power amplifier. Both of these noise sources will be swamped out by the acoustical noise level in the acoustical space, however. The electrical noise floor is transformed over to an equivalent noise level of -2 dB(A) at a distance of 20 meters, some 25 dB lower than the acoustical noise floor of a typical space. With this calibration procedure, the maximum output level possible in the house is limited by the dynamic range and nominal operating point established for the DSC260. If more output level is desired, the nominal operating points must be reset accordingly.
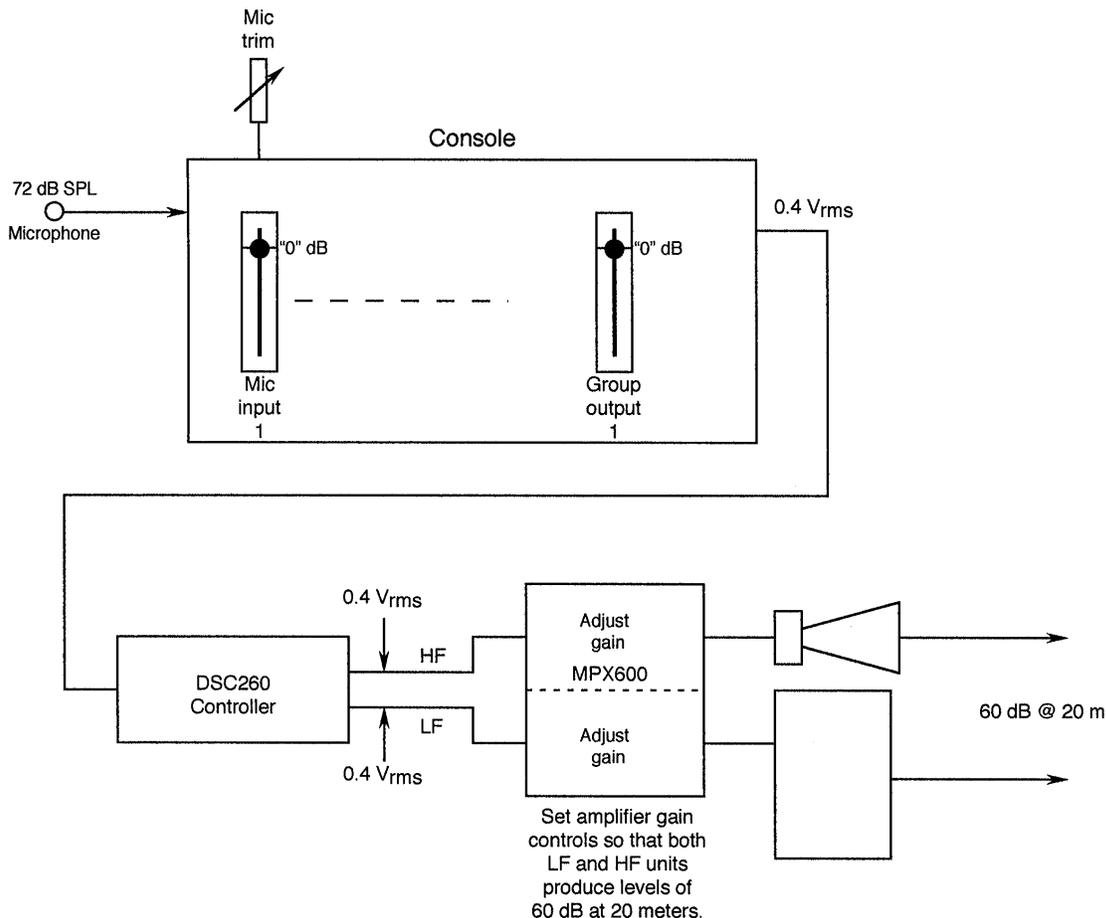

Figure 7-1A. Signal flow diagram for a simple reinforcement system

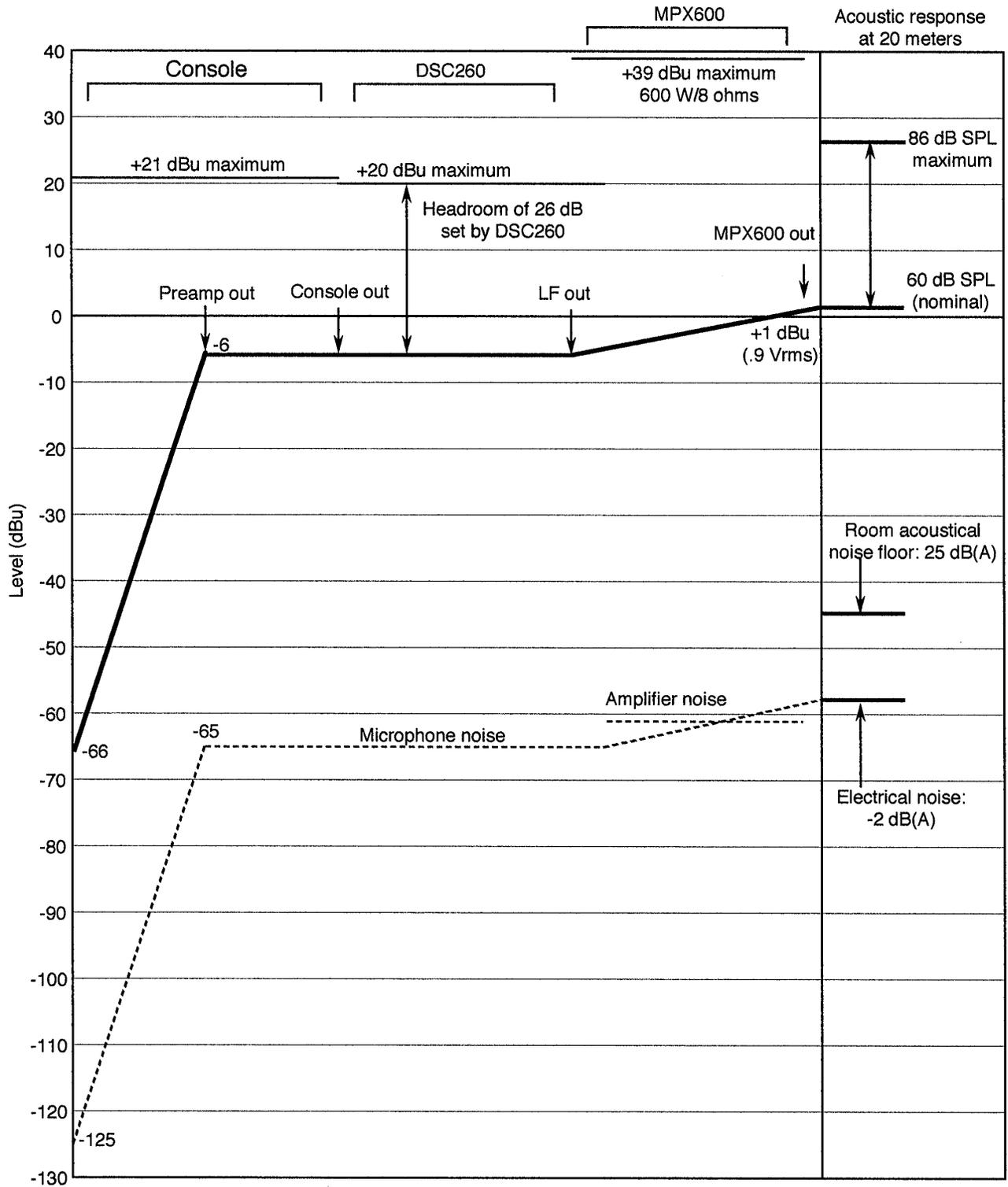## Level Diagram for a Simple Speech Reinforcement System

Figure 7-1B. detailed level diagram showing noise levels, nominal operating levels, and maximum output levels of each device

Step Two:

We now have to determine what the nominal operating level of the system should be for the farthest listeners, which we will assume are some 20 meters away from the loudspeaker. Let us further assume that the reverberation time in the room is no greater than 1.5 seconds in the range from 250 Hz to 2 kHz and that the average noise level room is in the range of 25 dB(A). Referring to Figure 7-2, we can see that for an ambient noise level in the 25 dB(A) range, the EAD for a lowered voice would be about 2 meters, or a speech level of about 60 dB SPL.

For a direct field level of 60 dB at a distance of 20 meters, the LF section of the loudspeaker will require a signal input of 0.1 watts (into 8 ohms). In the biamplification mode the HF section will require considerably less than 0.1 watt input in order to reach the desired level at a distance of 20 meters.
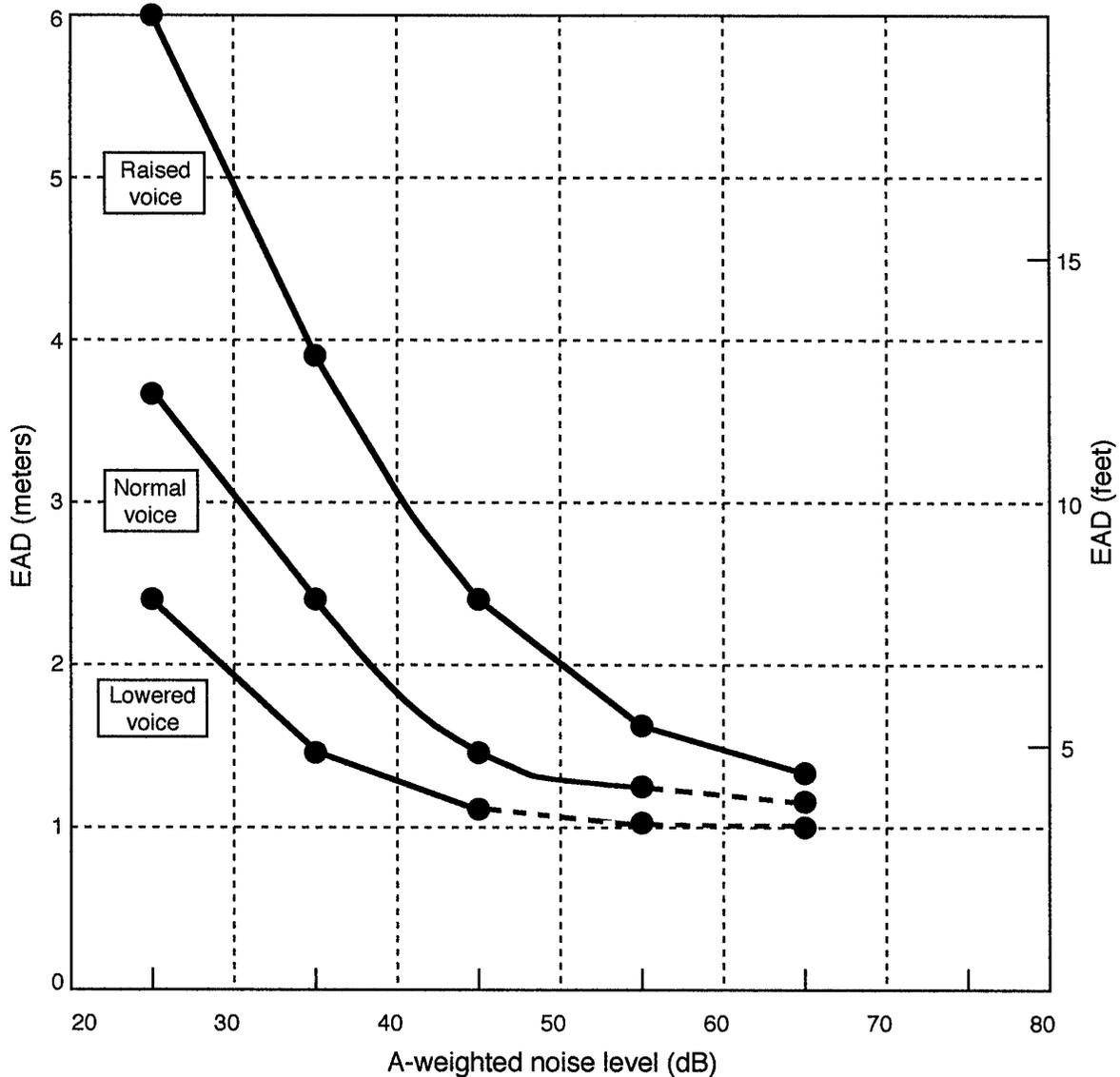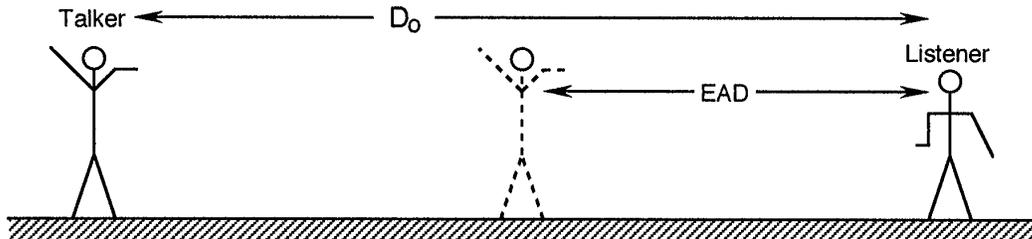

Figure 7-2. EAD versus A-weighted noise levels

Step Three:

For a simulated microphone input of 72 dB SPL, adjust the HF and LF outputs of the DSC260 for nominal levels of 0.4 Vrms. Then, advance the LF gain control on the MPX600 amplifier until a reference level of 60 dB SPL has been reached at a distance of 20 meters. Following this, increase the level of the HF section to reach the same value. Details here are shown in Figure 7-1.

Set up in this manner, there will be adequate headroom, in the console, controller, and power amplifier to handle nominal speech levels as well as levels up to 25 dB higher, should this ever be deemed necessary.

Amplifier and Loudspeaker Power Ratings

A persistent question is: what amplifier power rating do I choose for use with a loudspeaker of a given power rating? The detailed answer is addressed in JBL's Technical Note Volume 1, Number 16A; here, we will only summarize those recommendations:

1. For systems that will be stressed with full amplifier output for long periods of time, we recommend that the amplifier's continuous output rating be chosen to be equal to the loudspeaker's input power rating. Situations of this sort occur primarily in music reinforcement, where a constant, wide-band signal predominates.

2. For applications, such as speech reinforcement, where there is an operator who controls levels carefully, we can confidently recommend an amplifier with output capability that is twice (3 dB greater) than the loudspeaker's continuous rating. The rational here is that peak power requirements, often slightly in excess of the loudspeaker's continuous rating, can be handled with no problem, and it makes sense to provide amplification accordingly.

3. For certain critical monitoring applications, as in recording studios or film postproduction environments, amplifiers may be chosen that can deliver four-times (6 dB greater) power than the loudspeaker can withstand on a long-term continuous basis. The rational here is that the loudspeakers can ordinarily handle midrange and high frequency peaks of short duration that are much higher in instantaneous power than the long-term continuous rating of the loudspeaker.

In most speech reinforcement applications, condition 2 above will apply. Note however that there is no absolute necessity to use the larger amplifier unless high acoustical peak levels are anticipated.

Wire Gauges and Line Losses

In modern sound system engineering it is standard practice to locate power amplifiers as close to the loudspeaker loads as is possible so that line losses become negligible. However, in some applications this is not possible, and the designer must consider line losses, choosing wire gauges that will keep to an acceptable minimum.

Figure 7-3 shows the fundamental calculations. Note that there are actually *two* sources of loss: the loss in the wire itself and the loss due to the impedance mismatch that the long wire run can cause. For example, let us assume an input signal of 8 volts into a nominal load of 8 ohms. With no line losses the power dissipated in the load would be 8 watts (E^2/R_L).

Let us assume that the wire run is 80 meters and that AWG #10 wire is used. Using the table, we can see that the wire resistance in one leg will be:

$$R = 80/300 = .26 \text{ ohms}$$

and the total round trip resistance in the wire run will be twice that value.

The voltage across the 8-ohm load will then be:

$$E_L = 8/\big[8 + (2 \times .26)\big] \times 8 = 7.5 \text{ volts,}$$

and the power dissipated in the load will be:

$$P_L = (7.5)^2/8 = 7 \text{ watts}$$

The power loss is then:

$$\text{Loss (dB)} = 10 \log (7/8) = 0.58 \text{ dB}$$

The general equation for the loss in dB is:

$$\text{Loss dB} = 20 \log\left[\frac{R_L}{R_L + 2R_1}\right]$$

where R_1 is the resistance in each of the two wire legs, and R_L is the resistance of the load.

As given here, the loss consists of *two* terms: the actual loss generated in the wire run and the added loss incurred due to the impedance mismatch between the intended load and the actual load.

Good engineering practice dictates that losses at the load be held to 0.5 dB or less.

AMERICAN WIRE GAUGE (AWG)	RESISTANCE PER 300M (1000 ft), COPPER
10	1 ohm
12	1.6
14	2.5
16	4
18	6.3
20	10

SAMPLE CALCULATION:

FIND POWER LOSS IN AN 80m RUN OF
AWG NO. 10 WIRE TO AN 8-ohm LOAD.

$$R = 80/300 \times 1 \text{ ohm} = .26\,\Omega$$

$$E_L = \frac{8}{8 + (2 \times .26)} \times 8 = 7.5 \text{ volts}$$

$$P_L = (7.5)^2/8 = 7 \text{ watts}$$

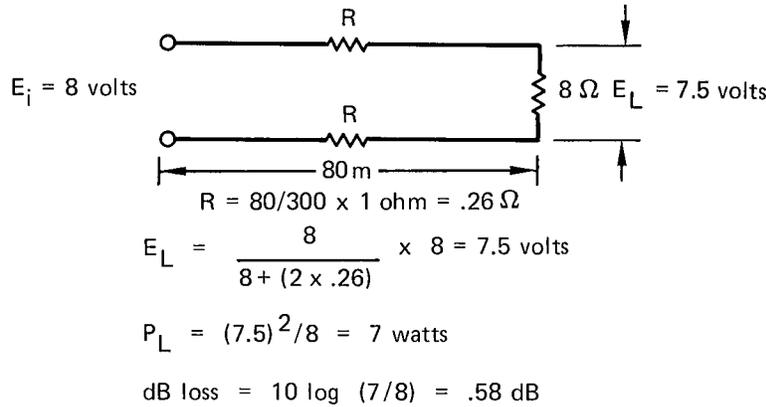$$\text{dB loss} = 10 \log (7/8) = .58 \text{ dB}$$

Figure 7-3. Calculation of resistance in wire runs

Constant Voltage Distribution Systems (70-volt lines)

Many distribution systems in the United States make use of the 70-volt line for powering multi-loudspeaker paging systems. In Europe the 100-volt line is common. In either system, the full output power of the driving amplifier is available at a line voltage of 70 Vrms or 100 Vrms, respectively.

In placing loads across the line, the design engineer simply keeps a running count of the number of watts of power drawn from the line. There is no need to calculate the aggregate load impedance at any point in the process. When the total number of watts drawn from the line is equal to the power rating of the amplifier, then the line is fully loaded and properly matched.

Figure 7-4 shows details of a 70-volt distribution system. The maximum load on the amplifier is transformed so that the applied voltage will be 70 Vrms. This then will correspond to a total transformed load impedance, Z_L, equal to $5000/P_O$, where P_O is the maximum power output of the amplifier.

Individual loads are placed across the amplifier in parallel using line-to-loudspeaker distribution transformers that have a 70-volt primary and a tapped secondary designated in watts. The system

designer (or installer) merely has to keep a running tally of watts drawn from the line, and when the number of watts equals the continuous output power rating of the amplifier, then the system is fully loaded. Ordinarily, no additional loads will be placed across the line, but there is some leeway here.

The alternative to 70-volt distribution is to laboriously keep track of combined load impedances in parallel, a big task. Details of a 70-volt transformer are shown in Figure 7-5.

In Europe, a 100-volt transmission system, derived in a similar manner, is used.

Low Frequency Augmentation — Subwoofers

Whether in the cinema or in open spaces, LF augmentation systems are becoming popular for special effects. For indoor applications many acoustical engineers calculate the reverberant sound pressure level that can be produced by a transducer, or group of transducers, operating continuously over an assigned low frequency band, normally from 25 Hz to about 80 Hz. The equation for determining the reverberant level is:

$$L_{REV} = 126 + 10 \log W_A - 10 \log R,$$

where W_A is the continuous acoustical power output from the transducer and R is the room constant in m^2.

In using this equation, we assume that the space is fairly reverberant at very low frequencies and that the value of absorption coefficient at 125 Hz (the lowest value normally stated for materials) will be adequate for our purposes.

Some design engineers prefer to make actual direct field calculations for one or more subwoofer units at a distance, say, of two-thirds the length of the enclosed space. In large motion picture spaces, both sets of assumptions yield results that are usually within 5 dB of each other.

The phenomenon of mutual coupling always comes to our aid in increasing the power output of combined subwoofer units. Figure 7-6A shows the
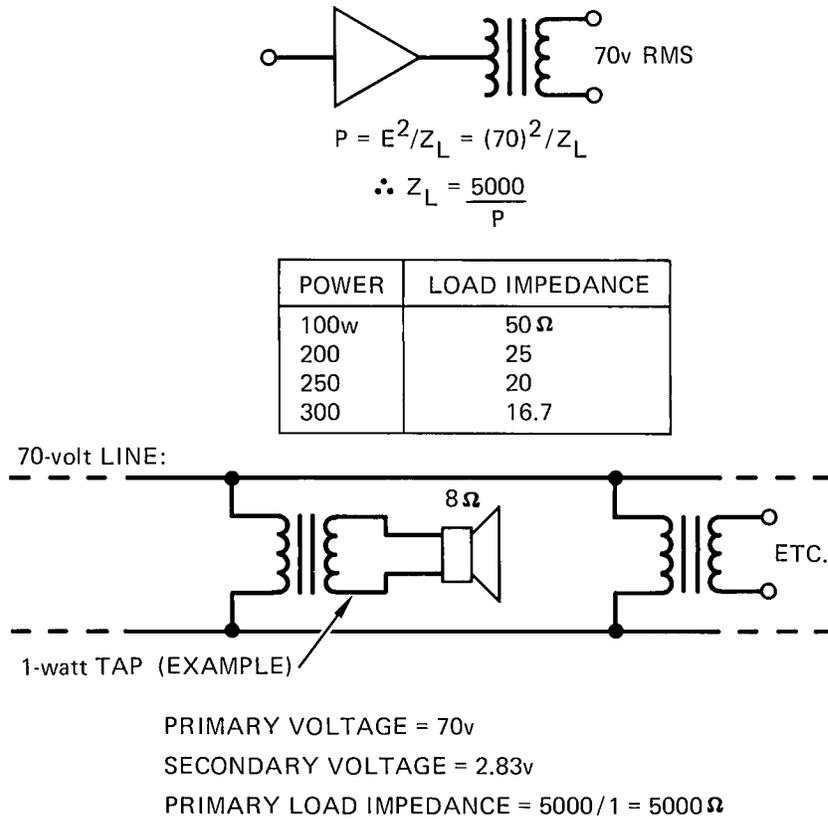

$$P = E^2/Z_L = (70)^2/Z_L$$

$$\therefore Z_L = \frac{5000}{P}$$

POWER	LOAD IMPEDANCE
100w	50 Ω
200	25
250	20
300	16.7

70-volt LINE:

PRIMARY VOLTAGE = 70v

SECONDARY VOLTAGE = 2.83v

PRIMARY LOAD IMPEDANCE = 5000 / 1 = 5000 Ω

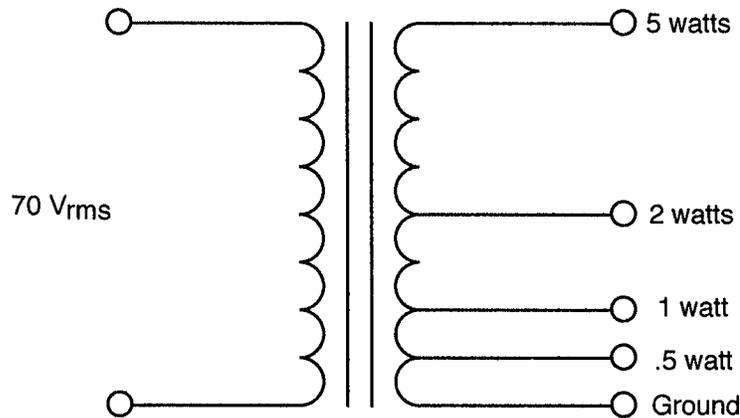**Figure 7-4. Details of a 70-volt transmission system**

Figure 7-5. Details of a typical 70-volt distribution transformer

transmission coefficient for a direct radiator as a function of cone diameter. The solid curve is for a single unit, and the dotted curve is for two units positioned very close to each other. In addition to the double power handling capability afforded by the two units, the dotted curve shows a 3 dB increase in transmission coefficient at low frequencies. This is due basically to the tendency for the two drivers to behave as a single unit with a larger cone diameter, and hence higher efficiency. Thus, at B, we see the relative response of a single woofer (solid curve) compared to two such radiators (dashed curve). Note that the upper frequency transition point for the pair is 0.7 that of the single unit. The more such units we combine, the lower the effective cut-off frequency below which mutual coupling is operant.

As an example, let us pick a large cinema with the following physical parameters:

$$V = 14,000 \text{ m}^3$$
$$S = 3700 \text{ m}^2$$
$$T_{60} = 1.2 \text{ seconds}$$
$$R = 2500 \text{ m}^2$$

We will use the JBL 2242H LF transducer. Taking into account its power rating and its dynamic compression at full power, we note that its power output in acoustic watts will be:

$$W_A = (W_E \times \text{reference efficiency})10^{-dB/10}$$

where W_E is the transducer's continuous power rating (watts) and -dB is the transducer's power compression at full power.

Substituting the values of W_E of 800 watts, reference efficiency of .004, and power compression of 3.3 dB, we get the value of 15 acoustical watts.

The reverberant level in a space with a room constant of 2500 is then:

$$L_{REV} = 126 + 10 \log 15 - 10 \log 2500 = 104 \text{ dB SPL}$$

We can now construct the following table:

Number of Units	Maximum Level	Power Input
1	104 dB	800 W
2	110 dB	1600 W
4	116 dB	3200 W

We cannot continue this process much beyond that shown here. What happens is that the frequency below which mutual coupling takes place falls below the nominal cutoff frequency of the system, and eventually all we see is a simple 3 dB increase per doubling of elements.

For multiple subwoofers outdoors, it is best to assume that levels fall off according to inverse square law.

$$f = \frac{2\pi r}{\lambda}$$
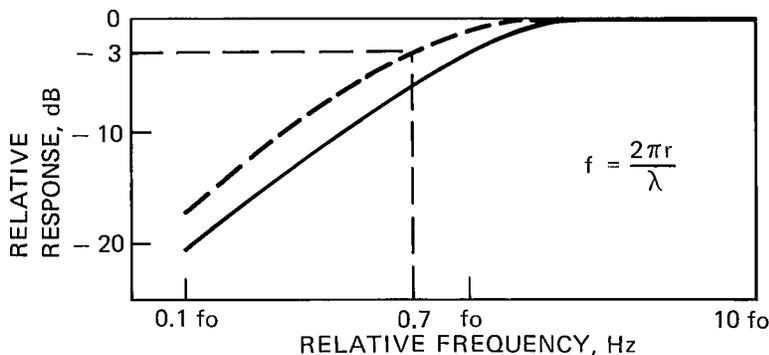
A. TRANSMISSION COEFFICIENT FOR A DIRECT RADIATOR
(SOLID CURVE, SINGLE UNIT; DASHED CURVE, TWO UNITS)
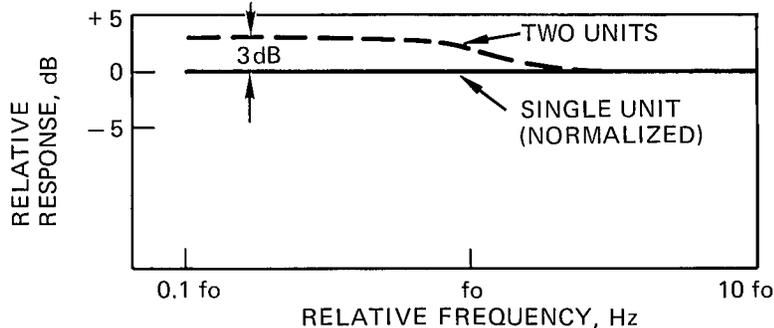
B. INCREASE IN OUTPUT DUE TO MUTUAL COUPLING

Figure 7-6. Details of mutual coupling

Case Study A: A Speech and Music System for a Large Evangelical Church:

1. Basic Description and Specifications:

The fan shaped architectural design shown in Figure 7-7 is common for modern evangelical churches in that it accommodates many people, all seating positions with good sightlines. The major acoustical problem is likely to be the curved front *fascia* of the balcony and the curved back wall itself. If not properly treated, these surfaces can cause severe reflections back to the platform. In many cases, such spaces are designed with an overabundance of absorbing material, making the room quite dead. There is then a need for a very robust speech-music reinforcement system to provide a feeling of ambience and acoustical envelopment.
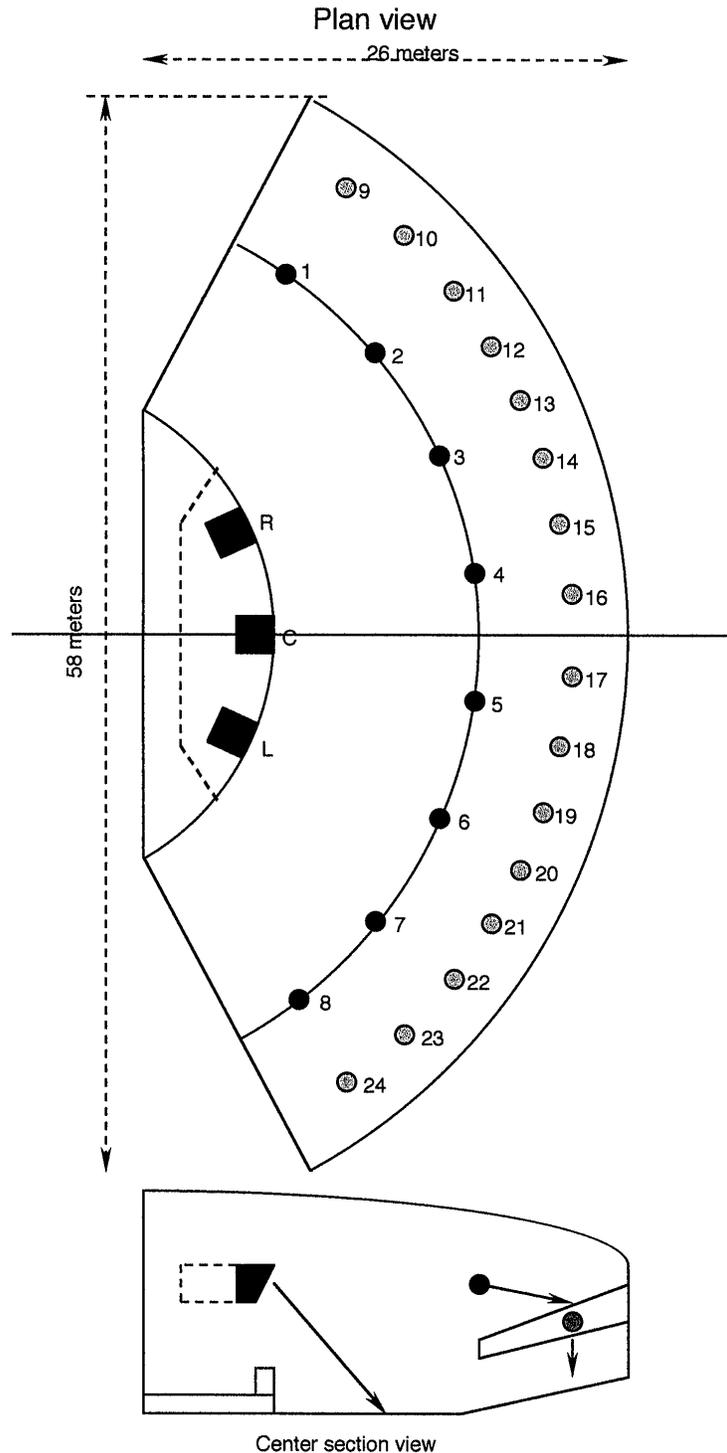

Figure 7-7. Evangelical church, plan and sections views

The system consists of a central array of left, center, and right stereophonic music channels; speech will be reinforced over the center channel only. Delayed coverage for the balcony area will be provided by a ring of seven flown loudspeakers, and under-balcony coverage will be augmented by a ring of fifteen soffit mounted loudspeakers. The main array over the platform should be designed for nominal horizontal coverage in excess of 120 degrees. If a CADP2 analysis shows they are needed, side and front fill loudspeakers may be added to increase coverage in those areas of the main floor.

The main stereo array is intended to cover primarily the main floor and first few rows under the balcony. Coverage of the balcony area will be essentially monophonic, with the same signal fed to all of those loudspeakers.

The main stereo loudspeaker systems should be capable of extended bandwidth at both low and high frequencies. Music levels of 105 dB are expected on the main floor.

2. Exercises:

2.1 Main Arrays:
    1. Specify the elements in each of the three main arrays and determine the power needed.
2.2 First Delay Ring:
    1. Specify the elements needed, power required, and distribution method.
2.3 Second Delay Ring:
    1. Specify the elements needed, power required, and distribution method.

3. Suggested Answers to Exercises:

3.1 Main Arrays.
    There are two basic approaches in designing the main arrays. A completely custom system may be assembled with individually fabricated low and mid-frequency enclosures. The benefit in this approach is that the HF components can be kept tightly clustered together with a minimum of interference among them. However, the attendant costs of rigging may make this approach unreasonably expensive.
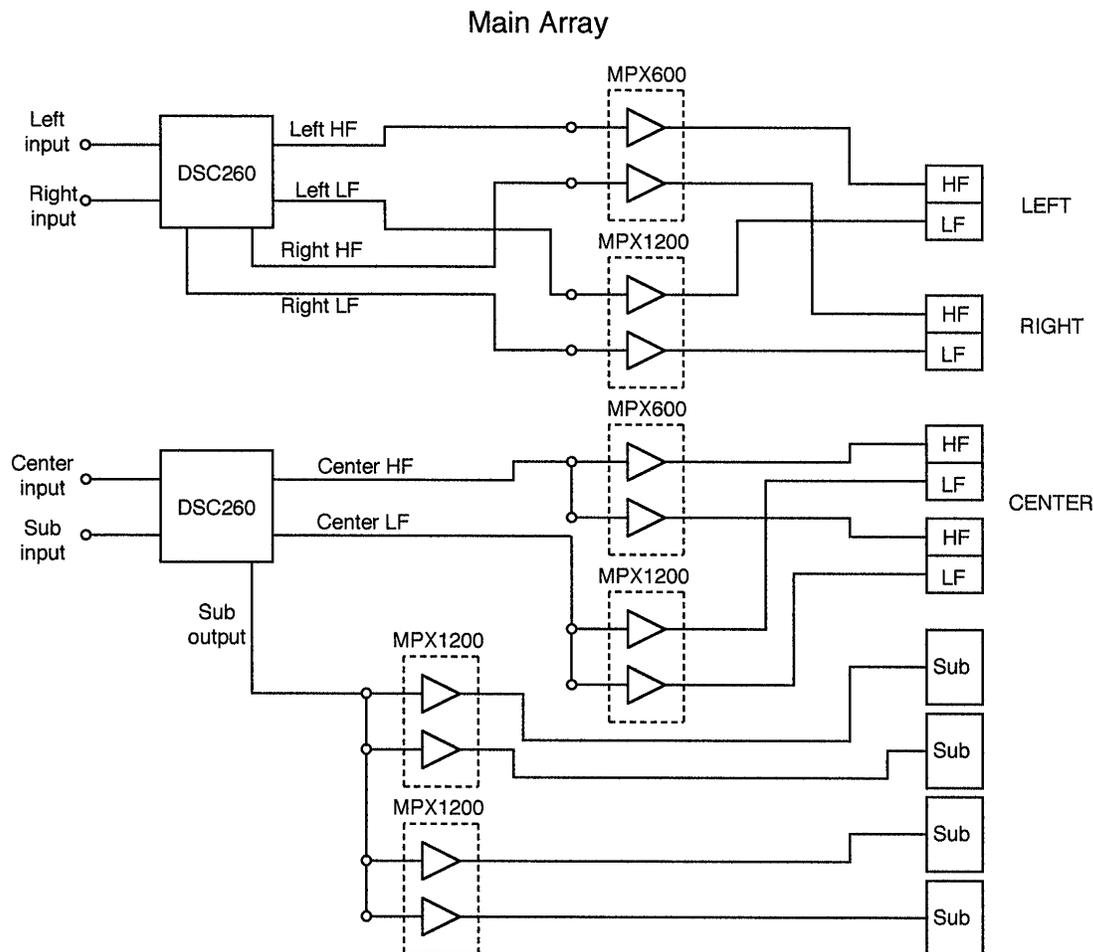
Main Array

Figure 7-8. Electrical diagram of the main array

The other approach is to use pre-existing building blocks, such as the SP225-9 system. Specifically, four of these systems can be arrayed with sufficient space between them for good stereo presentation and splayed to produce an included coverage angle of 135°. Rigging is integral in the SP Series, so that problem is solved. A center pair of SP225-9 units should be located side by side and splayed along their common back angle to give 120° coverage. This channel is primarily for speech, but may be used as well as a stereo center channel for music. The remaining two SP225-9 units will be used for left and right stereo presentation.

All units will have a nominal downward elevation angle of about 60°, and the average throw to the main floor is 10 meters. Level calculations are as follows:

Level	Power	Distance
100 dB	1 W	1 meter
131 dB	1200 W	1 m
111 dB	1200 W	10 m

In this case, we are powering the two LF units in each SP225-9 as a parallel (4 ohm) load to be driven by one section of a MPX1200 amplifier.
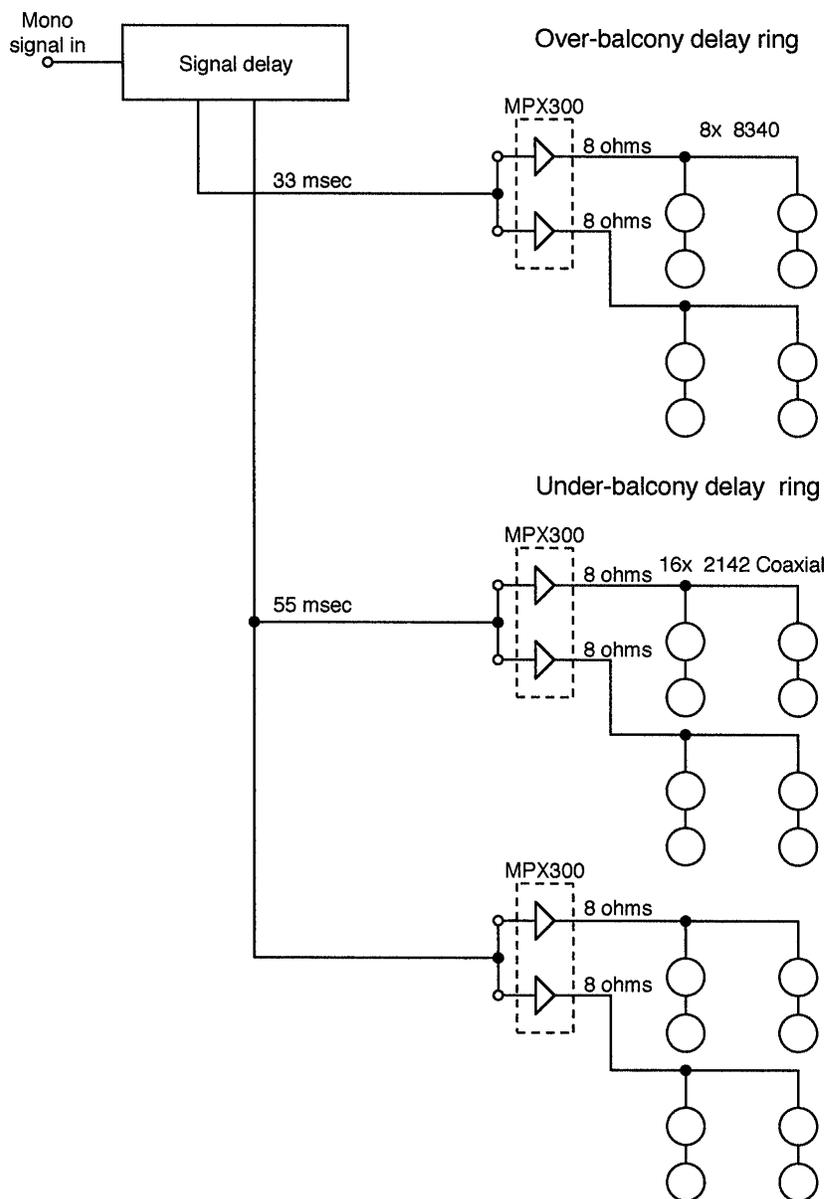
Delay Systems

Figure 7-9. Electrical diagram of delayed systems

The three HF sections in each main array should be powered by one section of an MPX600 amplifier (200 watts into 16 ohms).

Each one of the four main arrays should also have an SP128S subwoofer module. These would be powered the same way as the LF sections of the SP128S units. Level calculations are given:

Level	Power	Distance
102 dB	1 W	1 meter
133 dB	1200 W	1 m
113 dB	1200 W	10 m

The electrical diagram for the main array is shown in Figure 7-8.

3.2 Delay Rings:

The first delay ring of loudspeakers should be composed of fairly robust loudspeakers, such as the 8340A. Each one of these eight loudspeakers has a far throw range of about 6 meters and can accommodate 150 watts input. We can calculate the level output of each one as follows:

Level	Power	Distance
96 dB	1 W	1 meter
118 dB	150 W	1 m
102 dB	150 W	6 m

As an ensemble, these eight loudspeakers will provide added high frequency coverage of the balcony seats, ensuring good articulation.

The under balcony soffit system consists of 16 transducers located 2 meters above the seated listener's ear height. The model 2142 has a sensitivity of 97 dB and a power rating of 90 watts. A seated listener directly under one of these transducers, if it is powered by 1 watt, will hear a level of 91 dB. A listener mid-way between a pair of them will hear a level of about 90 dB. The 90-watt per transducer rating means that peak levels of about 110 dB can be developed under the balcony.

Case Study B: A Distributed Speech Reinforcement System for a Large Liturgical Church

1. General Information and Basic Performance specifications:

The system to be studied here is typical of what may be found in just about every large religious edifice in Europe and in many large cities in the U. S. The plan and front section views are shown in Figure 7-10. The building under consideration here has an internal volume of 12,000 cubic meters and surface area of 4000 square meters. A mid-band empty house reverberation time of 2.5 seconds indicates a value of 800 square meters of absorption units (S$\bar{a}$), as extrapolated in Figure 5-10.

Our major concerns with a system in such a space as this are the net speech direct-to-reverberant ratio and the reverberation time itself. Each loudspeaker will contribute to the overall reverberant level behind the amplified speech, and our first step is to determine the number of loudspeakers that will be required to cover the entire seating area. Studying the plan view of the building, we can see that 8 loudspeakers will cover the transept seating, while 10 systems will cover the nave seating.

2. Analysis:

The longest "throw" that will be required of any single loudspeaker is to cover a listener seated at the center aisle, a distance of about 7 meters. Let us now specify a JBL Control 28 and power it to produce a level of 85 dB at a distance on-axis of 7 meters.

We can do this directly by setting up the familiar level/power/distance chart as follows:

Level	Power	Distance
92 dB	1 W	1 meter
75 dB	1 W	7 m
85 dB	10 W	7 m

We now want to make an estimate of the reverberant level that will exist in the room when the direct sound from a single loudspeaker at the listener is 85 dB. To do this, we must determine the efficiency of the loudspeaker. Taking data from the Control 28 specification sheet, and averaging the DI over the 200 to 2000 Hz range, we now use the following equation:

Sensitivity (1 W @ 1 m) = 109 + DI + 10 log Efficiency.

Plan view

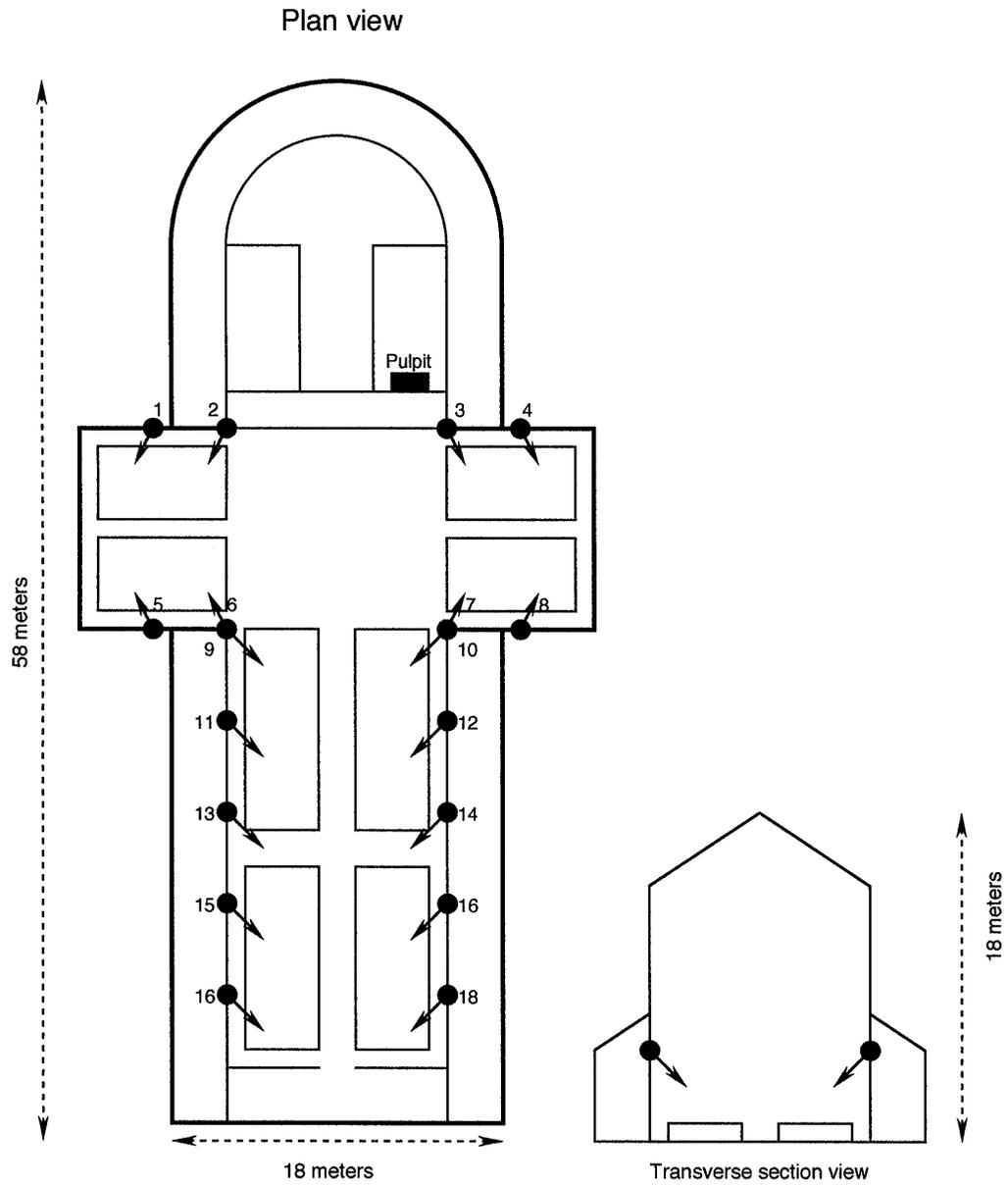Figure 7-10. Plan and front elevation views of large liturgical church

Side Section View

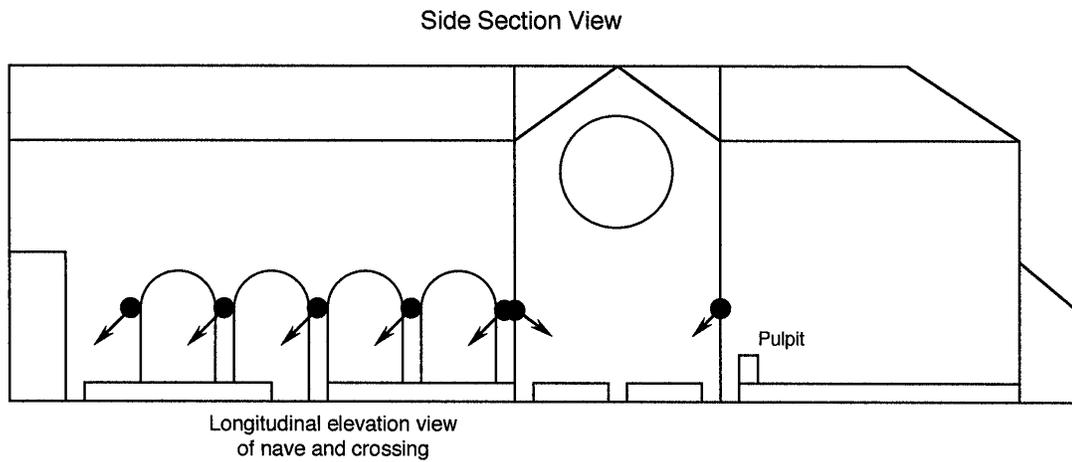Longitudinal elevation view
of nave and crossing

Figure 7-1 1. Side elevation view of large liturgical church

Entering values and rearranging:

$$10 \log \text{Efficiency} = 92 - 109 - 5 = -22$$

The efficiency is then $10^{-22/10} = 10^{-2.2} = .63\%$

The total contribution to the reverberant field will be from all 18 loudspeakers working at once. We can then calculate the radiated acoustical power as: $18 \times 10 \times 0.0063 = 1.134$ acoustical watts. Here, 18 is the number of individual loudspeakers, 10 W is the electrical power applied to each of them, and 0.0063 is the efficiency.

For the next step in the analysis we need to determine the resulting reverberant level in the room.

$$L_{rev} = 126 + 10 \log W_A - 10 \log R$$

In relatively live spaces, $S\bar{a}$ and R are virtually identical; therefore,

$$L_{rev} = 126 - 0.6 - 29 = 96 \text{ dB SPL.}$$

We have now reached a point in our analysis where we can estimate the overall system performance regarding speech intelligibility. We know the following:
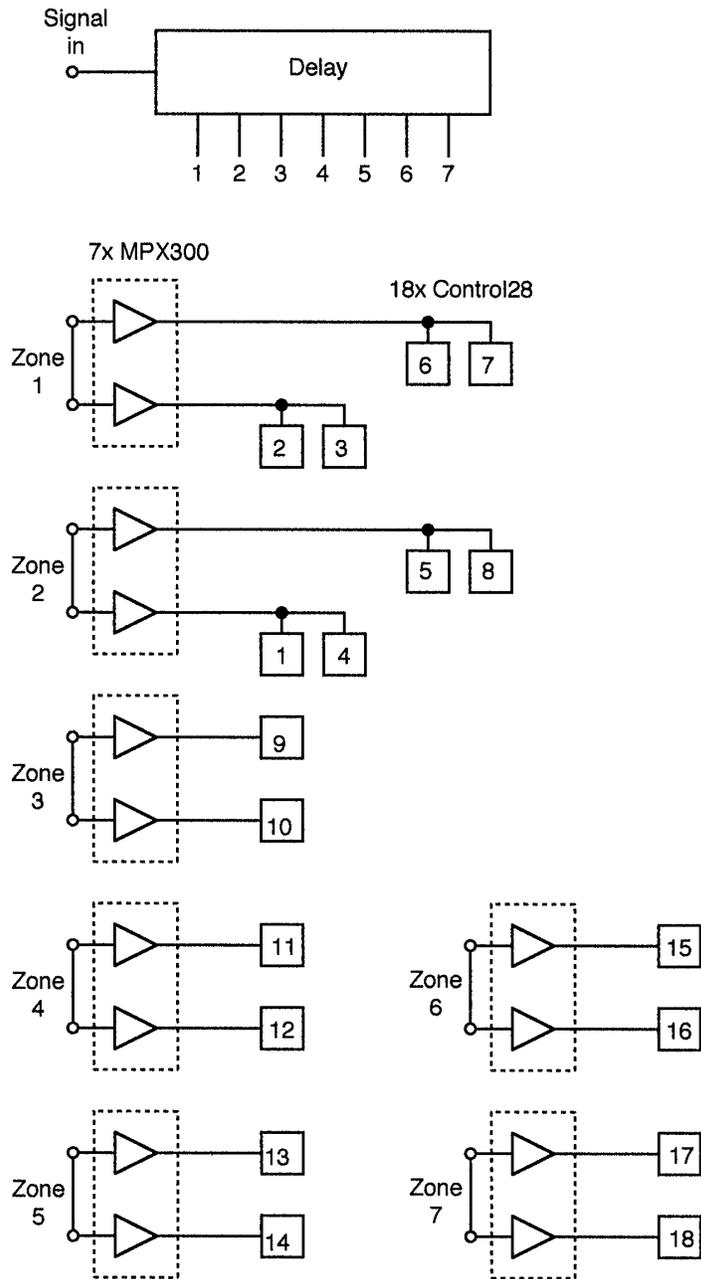

Figure 7-12. Electrical diagram of delay zoning system

1. That when a single loudspeaker produces a level of 85 dB SPL at the farthest listener, the resulting reverberant level is 96 dB SPL.

2. That the mid-band reverberation time in the room is 2.5 seconds.

As our final step in the analysis, we can check the probable system speech intelligibility performance, according to Peutz' Articulation loss of consonants (Al$_{cons}$) by means of the chart shown in Figure 6-13.

Locating a direct-to-reverberant ratio of -11 dB along the bottom axis, and then moving up to the horizontal line corresponding to a reverberation time of 2.5 seconds, we see that the system's performance is slightly above the borderline of 15% Al$_{cons}$. This indicates that we have barely succeeded in our design goal of acceptable performance. In other words, our analysis has shown that we have marginal acceptability in terms of speech intelligibility.

In reality, there are three factors that will work in our favor:

1. The Peutz criteria are based on acoustical relationships at 2 kHz. With mid-band (500 Hz) reverberation at 2.5 seconds we can reasonably assume that reverberation time at 2 kHz will be about 2.2 seconds due to excess air attenuation, as can be read from Figure 5-13.

2. We have not yet considered that the surface area (as opposed to the average absorption coefficient) on which most of the initial acoustical power is aimed is more absorptive than the average absorption coefficient would indicate. At 2 kHz, the audience area will have an absorption coefficient conservatively estimated at about 0.5, and this indicates that the actual reverberant level generated by the reinforcement system could be a good 3 dB less than our calculations indicate. This would easily move our data point in the graph of Figure 6-13 very close to the "GOOD" zone. Of course we must take into account the actual number of persons present in the audience area before this assumption can be made. On any given Sunday, if all the attendees at a service can be coaxed into the front pews, the loudspeakers behind them are unnecessary and can be turned off, further reducing the level of the reverberant field.

3. The final factor working in our favor would be the reduction of reverberation time in the space due to the presence of the congregation. Remember that the reverberation time of 2.5 seconds is for the empty room.

3. Delay Zoning:

Suggested delay settings are:

Zone 1. Loudspeakers 2, 3, 6, and 7	10 msec
Zone 2. Loudspeakers 1, 4, 5, and 8	22 msec
Zone 3. Loudspeakers 9 and 10	40 msec
Zone 4. Loudspeakers 11 and 12	55 msec
Zone 5. Loudspeakers 13 and 14	70 msec
Zone 6. Loudspeakers 15 and 16	85 msec
Zone 7. Loudspeakers 17 and 18	100 msec

4. General comments:

The system described in this section emphasizes the complex inter-relations between acoustics and electroacoustics that are inherent in basic sound reinforcement design in large, live spaces. We strongly urge that all of the basic relationships presented here be carefully studied and understood. The fundamental principles we would like to stress are:

1. Whenever possible, use distributed loudspeakers that cover the intended seating area, but that have rapid cutoff beyond their nominal coverage angles; in other words, keep the on-axis DI as high as possible consistent with required coverage.

2. Try to minimize the longest throw distance within a given loudspeaker zone. Loudspeakers have been placed in overhead chandeliers in the attempt to do this. Pewback systems take this approach to the limit.

3. Seat the congregation toward the front of the room and turn off unnecessary loudspeakers.

4. Many large spaces were designed during a time when few people cared about speech intelligibility, and many liturgical spaces are simply too live for modern requirements. A careful assessment should be made here, and no live liturgical space should be altered acoustically without the advice and counsel of an experienced acoustical consultant.

Case Study C: Specifications for a Distributed Sound System Comprising a Ballroom, Small Meeting Space, and Social/Bar Area.

1. General Information and Basic Performance Specifications:

1.1 Ballroom Description: The size of the space is 33 meters long, 22 meters wide, and 8 meters high. A stage is located at the center of one short side, and the room may be used for banquets, displays, and social events such as dancing.

A distributed (ceiling) system will be used for general speech/music purposes, as well as amplification of stage events. For this purpose the system should be zoned for delay. Reinforced levels up to 100 dB SPL will be expected, and coverage should be uniform within 1.5 dB up to a frequency of 2 kHz. The space is normally carpeted, except for dancing. Reverberation time is minimal.

1.2 Meeting Space Description: This space is typical of many that will be found in convention and meeting areas. The size is 8 meters by 5 meters and 3 meters high. A distributed ceiling system is to be designed, uniform within 1.5 dB up to 2 kHz. Normal maximum levels are expected to be 85 dB SPL.

1.3 Social Area: This space is of irregular shape, as shown in the diagram. A foreground stereo music system is to be specified for this space; no paging will be required. The system should be capable of producing levels of 85 dB SPL. There is also a disco/dance floor area, and a four-loudspeaker installation should provide levels of 105 dB at the center of the dance floor.

2. Exercises:

Study the attached figures that detail the layout of distributed systems in general, and pick either the square or hexagonal layout.

2.1 Ballroom System:

1. Determine quantity and placement of ceiling loudspeakers that will meet the specification.

2. Determine the power allocation for each loudspeaker and describe the power distribution system (70-volt or low-Z).

3. Determine the minimum number of workable zones for signal delay for stage events.

2.2 Meeting Space System:

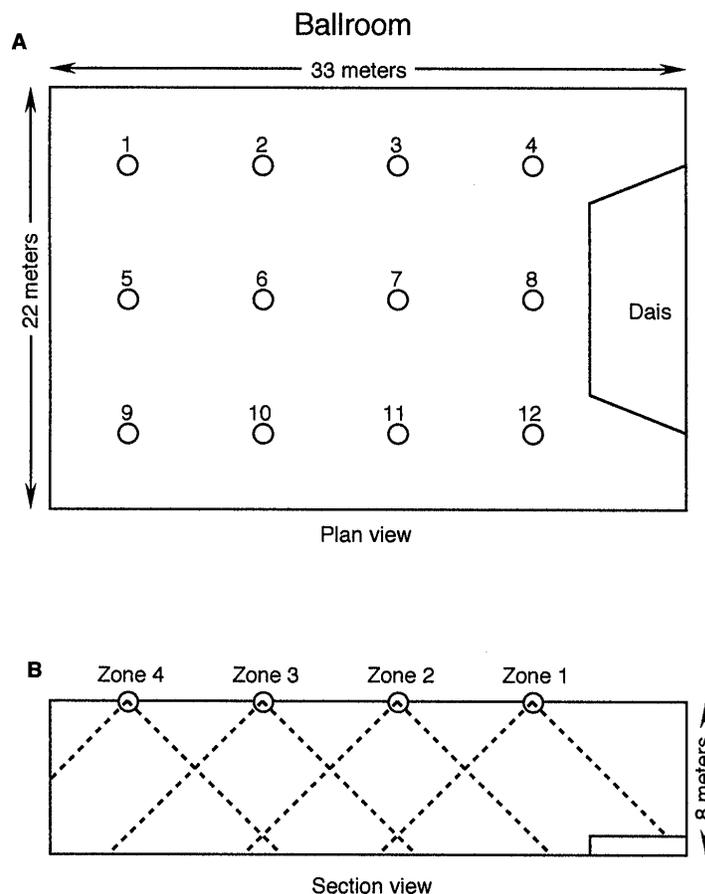1. Determine the model loudspeaker required and the spacing density in the ceiling.

Figure 7-13. Ballroom layout. Plan view (A); side section view (B).

2. Determine the power allocation for each loudspeaker.

2.3 Social Area System:
1. Suggest a stereo layout of loudspeakers that will provide all patrons with satisfactory sound.
2. Determine power requirements and distribution method.
3. Specify disco components that will produce a level of 115 SPL dB in the middle of the dance floor.

3. Answers to Exercises:

3.1 Ballroom System:
1. Use the square array, with center-to-center overlap. Reasons: results in easier zoning requirements and fits the rectilinear design of the room better. Designing for seated ear height (1 meter) results in 12 loudspeakers.
2. Use JBL 2155 coaxial loudspeakers. With sensitivity of 102 dB and power rating of 150 watts, a

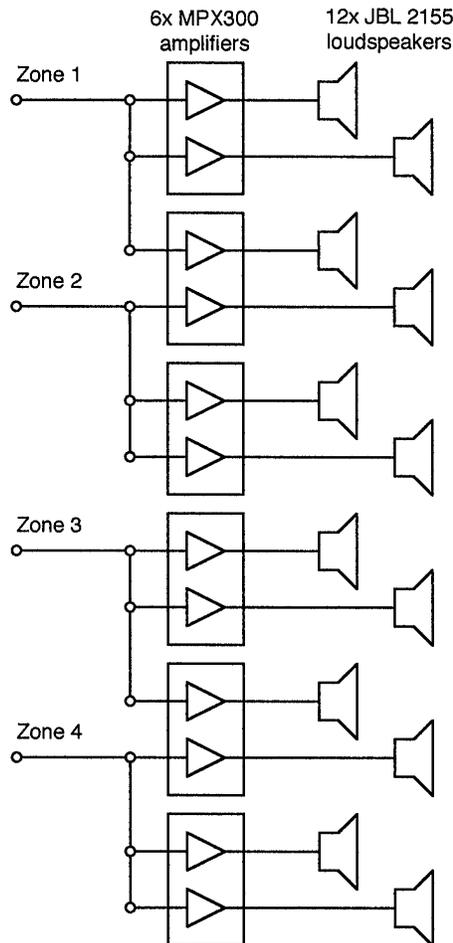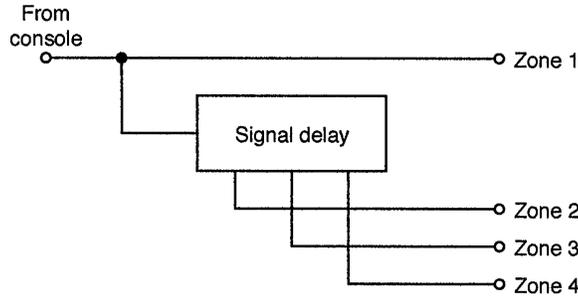Signal flow diagram for Ballroom System:

Figure 7-14. Ballroom system, signal flow diagram.

Meeting space

A

Plan view

B
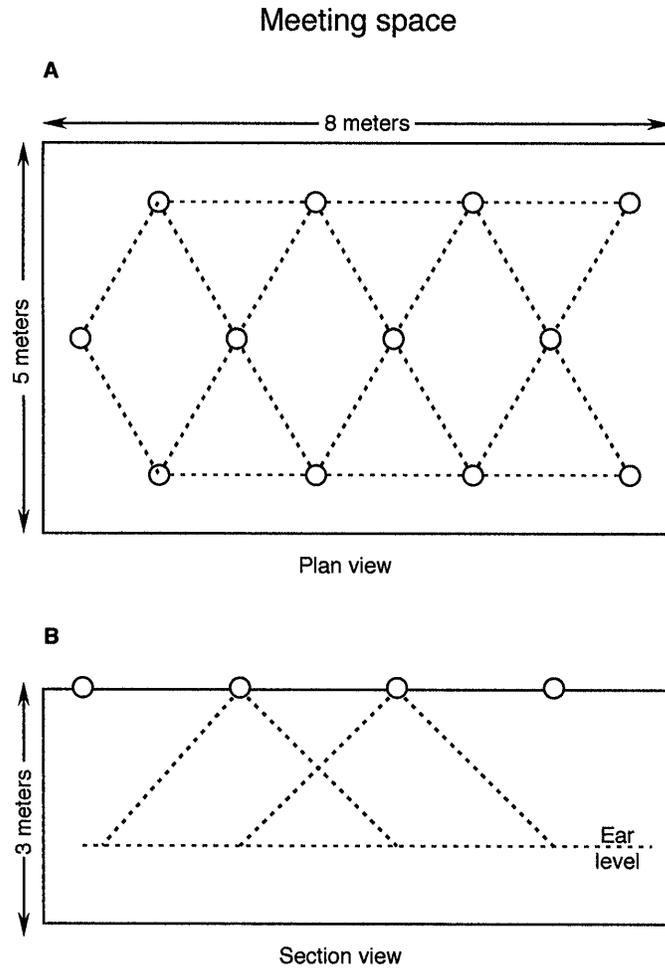
Section view

Figure 7-15. Meeting space layout. Plan view (A); side section view (B)

Distributed System for Meeting Space

One section of
MPX300 amplifier

12x JBL Control26C loudspeakers
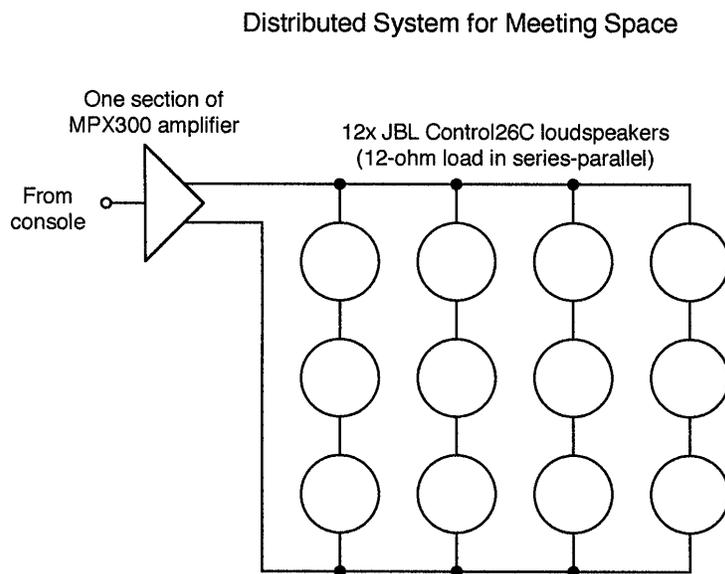(12-ohm load in series-parallel)

From
console

Figure 7-16. Meeting space system, signal flow diagram.

single loudspeaker will, at a distance of 7 meters, produce a level of 105 dB. The added contribution of the eight neighboring loudspeakers will increase this by 3 dB, making a maximum level capability of 108 dB. Level variations will be 1.4 dB.

Because of the wide-band capability of the loudspeakers and relatively high power required, a low impedance distribution system should be used. Each 8-ohm loudspeaker should be driven from a section of a JBL MPX 300 amplifier, making a total of 6 amplifiers. This will provide 200 watt capability into each loudspeaker, which will more than exceed the specification.

JBL Professional provides a program for determining layout density for distributed ceiling loudspeakers. It is called Distributed System Design, version 1.1, and runs on Windows 95 and is available from JBL Professional.

3. Zoning requirements: Measure the average distance from center stage to a center listening position directly under each zone. Subtract from that the value of 7 meters. For each meter difference, calculate 3 milliseconds of delay:

Zone	Difference	Delay
1	negligible	0 msec
2	12 meters	36 msec
3	20 meters	60 msec
4	26 meters	78 msec

In normal cases, the calculated delay values will have to be adjusted slightly on-site for best overall sound quality.

3.2 Meeting Room System:

1. Use the hexagonal array, with center-to center overlap, for best coverage. Twelve loudspeakers will be required, and the JBL Control 26C/CT will be specified because of its nominal coverage angle of 110°. With a sensitivity of 89 dB, one watt input will produce a level of 83 dB at a distance of 2 meters (ear height). With one watt fed to the six nearest neighboring loudspeakers, the level will increase to 87 dB, which is 2 dB higher than the design requirement.

It is obvious that normal usage will require only about one watt per loudspeaker. However, we should provide the system with a nominal 10 dB headroom factor for undetermined applications. The total power in this case would be 120 watts, and a single section of a JBL MPX300 amplifier, operating in series-parallel, would more than meet this requirement. The resulting load impedance of 12 ohms could easily accept 120 watts from one section of the MPX300 amplifier.

3.3 Social Area System:

1. The foreground stereo system is shown as alternating L and R loudspeakers around the main bar area. The JBL Control 5 would be a good candidate for this application. In order to see if 12 of the units can meet the specification, we will pick a point midway in the room (marked X) and sum the individual levels of the loudspeakers at that point. Taking the 1-watt, 1 meter sensitivity as a reference point, we can set up a table as follows for summing the individual contributions:

Spkr	Distances	S	-	20 log Dist	=	Net level				
1	10	89	-	20	=	69				
							>73			
2	8	89	-	18	=	71				
								>77.2		
3	7	89	-	17	=	72				
							>74.5			
4	8	89	-	18	=	71				
									>81	
5	8	89	-	18	=	71				
							>75.7			
6	6	89	-	15	=	74				
								>76.3		
7	5	89	-	14	=	72				
							>77.5			Total: 82 dB
8	6	89	-	15	=	74				
9	8	89	-	18	=	71				
							>74.5			
10	7	89	-	17	=	72				
								>77.5	>77.5	
11	8	89	-	18	=	71				
							>74.5			
12	7	89	-	17	=	72				

The total level at the test position is thus 82 dB when all 12 loudspeakers are powered with 1 watt, or 85 dB with 2 watts per loudspeaker. With 20 watts per loudspeaker, we would have a comfortable 10 dB margin over our target value of 85 dB SPL. The system will be "coasting" most of the time, and a single stereo amplifier, with loads wired in series-parallel, will suffice.

2. Disco system. There are a number of possibilities here. A very high-end approach would be to specify two JBL 4892 Array Series systems at each corner. They should be mounted near the ceiling and angled down so that the horns point toward the center of the dance floor.

The 4892 is a biamplified system, and its overall sensitivity of 98 dB is limited by its LF section. The LF power rating is 600 watts. The distance from

each corner to the center of the dance floor is 7 meters:

Level	Power	Distance
98 dB	1 W	1 meter
126 dB	600 W	1 m
109 dB	600 W	7 m

A total of 8 4892 systems would increase the output by 9 dB, producing level capability of 118 dB at the center of the dance floor. The amplifier requirement would be 4 MPX600 units and 4 MPX300 units. Note carefully that the MPX600 has a maximum output capability of 400 watts into 8 ohms. This is approximately 2 dB less than 600 watts, and we would have to derate the system's overall output capability by that amount.
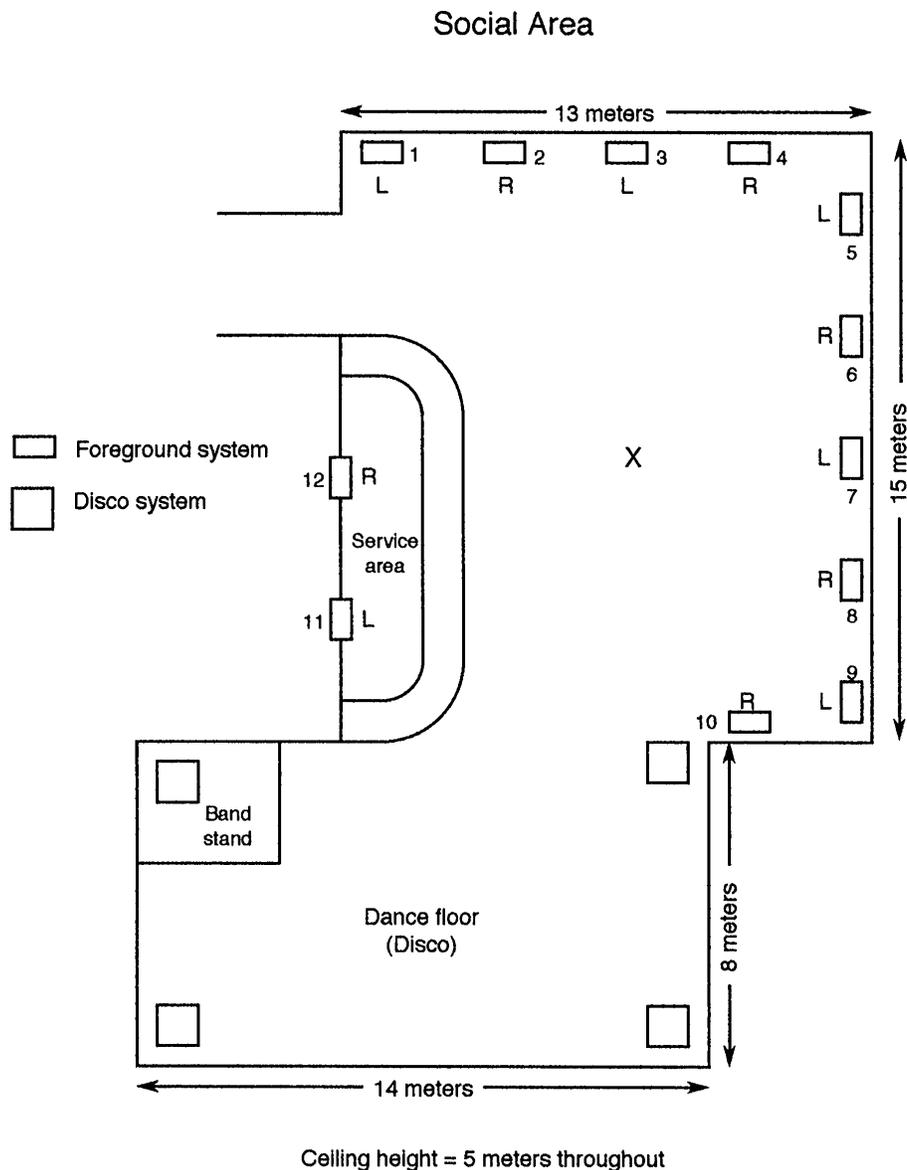
Social Area

Ceiling height = 5 meters throughout

Figure 7-17. Plan view of social area and disco area.

For a stereo system, we would need one DSC260 digital controller for frequency division and other signal processing.

Subwoofer requirements can be met with four (one in each corner) JBL 4645B systems. Drive requirements would be two MPX1200 amplifiers and one section of a DSC260 controller. The amplifiers can deliver 800 watts continuous power per channel into 8 ohms. With their half-space reference efficiency of 2.1%, the four subwoofer systems can produce a total power of 60 acoustic watts.

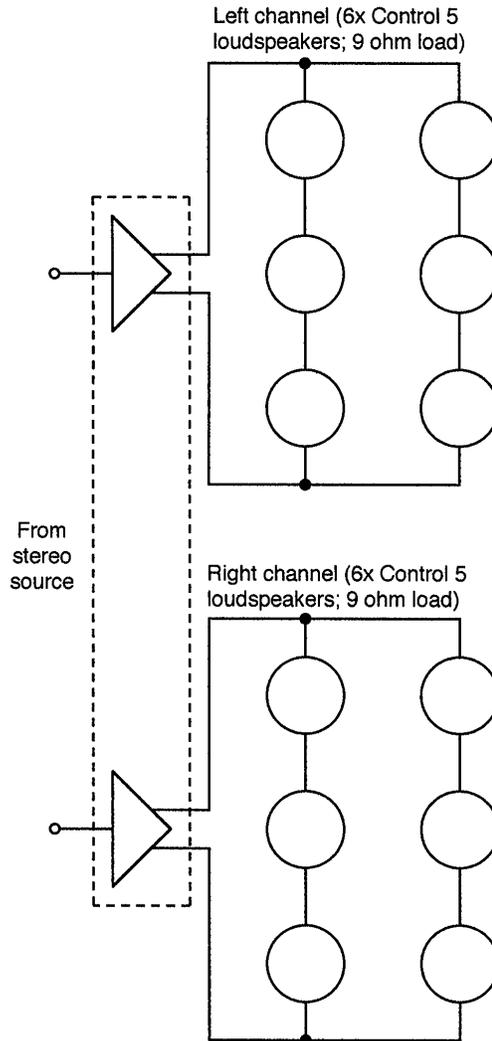## Systems for Social Area
Foreground music system:

Left channel (6x Control 5 loudspeakers; 9 ohm load)

From stereo source

Right channel (6x Control 5 loudspeakers; 9 ohm load)

Figure 7-18. Social area system, signal flow diagram.
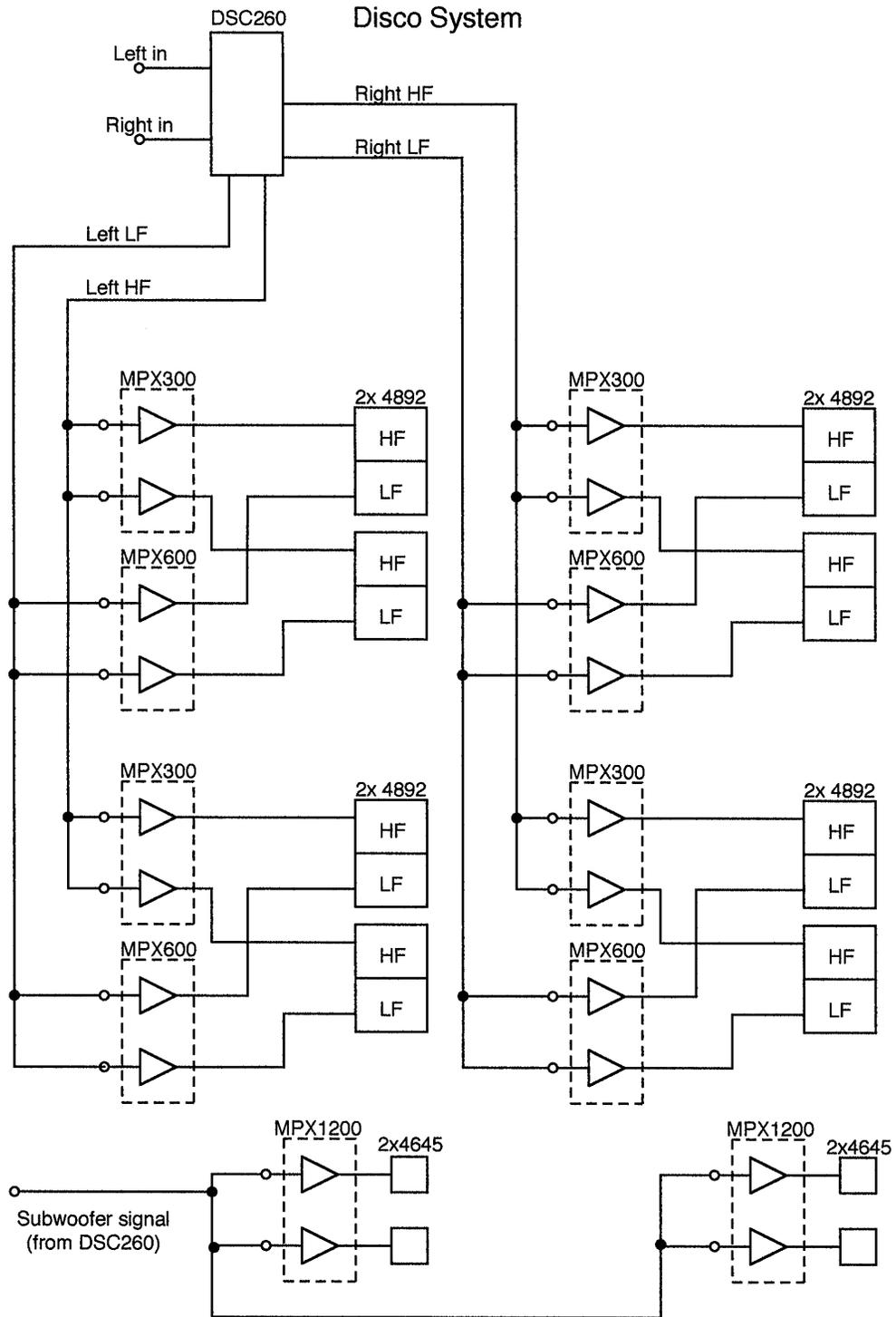
Figure 7-19. Disco system, signal flow diagram.

Recommended Reading:

Books and Anthologies:

1. D. and C. Davis, *Sound System Engineering*, second edition, Howard F. Sams, Indianapolis, 1987.

2. J. Eargle, *Electroacoustical Reference Data*, Van Nostrand Reinhold, New York, 1994.

3. Various, *Sound Reinforcement*, an anthology of articles on sound reinforcement from the Journal of the Audio Engineering Society, volumes 1 through 26. (Available from the AES)

Papers:

4. C. P. and R. E. Boner, "The Gain of a Sound System*," J. Audio Engineering Society*, volume 17, number 2 (1969).

5. H. Haas, "The Influence of a Single Echo on the Audibility of Speech," *J. Audio Engineering Society*, volume 20, number 2 (1972).

6. H. F. Hopkins and N. R. Stryker, "A Proposed Loudness-Efficienty Rating for Loudspeakers and the Determination of System Power Requirements for Enclosures," *Proceedings of the IRE*, March 1948.

7. C. T. Molloy, "Calculation of the Directivity Index for Various Types of Radiators," *J. Acoustical Society of America*, volume 20, pages 387-405 (1948).

8. V. M. A. Peutz, "Articulation Loss of Consonants as a Criterion for Speech Transmission in a Room," *J. Audio Engineering Society*, volume 19, number 11 (1971).

9. V. M. A. Peutz, "Quasi-steady-state and Decaying Sound Fields," *Ingenieursblad*, volume 42, number 18 (1973, in Dutch).

10. Various, "Loudspeaker Arrays — Design and Performance*," J. Audio Engineering Society*, volume 38, number 4 (1990).