Engineering News

ALTEC LANSING

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TECHNICAL LETTER NO. 182

ENGINEERING LOUDSPEAKER LOCATIONS

By Don Davis

To appreciate the engineering reasons that govern the proper location of a loudspeaker or loudspeakers it is necessary to first consider what is expected of the loudspeaker in a successful sound reinforcement system.

Criteria Governing Loudspeaker Locations

To insure that all listeners with normal hearing can clearly hear a sound reinforcement system, it is necessary to place the loudspeaker (or loudspeakers) where it is possible to achieve the following effects:

1. Sufficient acoustic gain. Sufficient acoustic gain is defined as the ability to achieve the same sound pressure level (SPL) at the most remote listener's ears as is generated at a microphone diaphragm two feet in front of the talker's mouth.

2. Even distribution (coverage) of the generated SPL to all listener locations. Even distribution is defined as no variation greater than plus or minus 3 dB from any point within the boundaries of the listening area to any other point, as measured with a sound level meter incorporating a one octave bandpass filter centered on 4 KHz, when the input signal to the system is a white-noise generator.

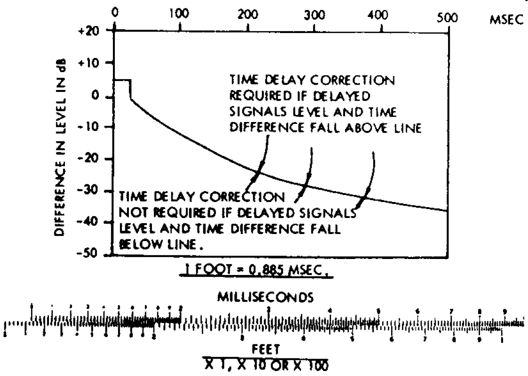
3. No harmful time delay relationships generated by the loudspeakers. Time delay is determined by two factors which must be taken into consideration when evaluating the effect of time delay:

   (a) Total delay.
   The first parameter, total delay, is found by measuring the path length from each source of possible interference (other sources carrying intelligence pertinent to the program) and converting the difference in path length into milliseconds (see Figure 1).

   (b) Relative loudness of near sound to far sound.
   The second parameter is the measurement or calculation of the difference in level of the two sources under consideration. This can be accomplished by calculating the D_2 and D_o losses in dB or by actual in-the-field measurement of an existing system.

4. It is possible to generate the required acoustic level that calculations reveal are necessary, without exceeding the loudspeakers' electrical-input power ratings (power handling).

Any proposed loudspeaker location must satisfy these four basic requirements. Meeting these required conditions provides the basis for the engineering design of the remainder of the system.



EXAMPLE: D_o = 120', D2 = 60' - LEVEL FROM TALKER = 44 dB-SPL, LEVEL FROM LOUDSPEAKER = 80 dB-SPL, TIME DIFFERENCE = 120' -60' = 60', 60' = 53 + MSEC. DELAYED SIGNAL (80 -44) = -36 dB.

TIME DELAY IN MILLISECONDS

Figure 1

Potential Acoustic Gain Before Feedback

The same formula is used for the calculation of potential acoustic gain before feedback in both single source and in distributed sound reinforcement systems.

NOTE

In playback systems, acoustic gain is limited only by loudspeaker efficiency and power handling capacity. In reinforcement systems, the limiting factor is almost always feedback. Acousta-Voiced T.M. sound systems have the capability of achieving unity gain before feedback.

Unity gain is the total loop gain that allows the SPL generated by the loudspeaker, as measured at the microphone diaphragm being used, to equal the SPL generated by the talker at the same microphone diaphragm. The gain formula is the mathematical expression of the acoustic gain achieved at a remote listener position with the sound system as compared to no sound system.

The gain formula, however, differs in each case. In combination systems (combined single source high level with distributed low level) the engineering is treated first as a single source and then separately as a distributed system.

Single Source Locations

Figure 2 illustrates the parameters governing acoustic gain in a source location.

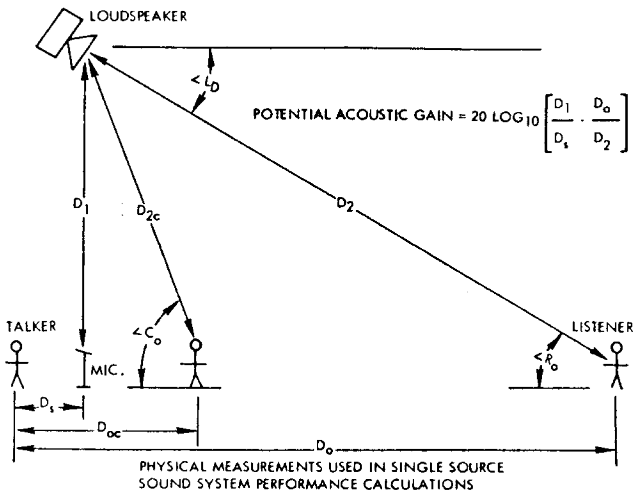


Figure 2

This formula reveals that to achieve high acoustic gain it is necessary to meet the following requirements:

1. Place the microphone as close to the talker as is possible under the functional circumstances.

2. Separate the microphone and the nearest loudspeaker by the greatest distance consistent with the requirements of coverage and time difference.

3. Minimize the separation between the loudspeaker and the listener.

In applying the gain formula to the design of loudspeaker locations, make D_s (the distance from the talker to the microphone) two feet. The accompanying figures will make this assumption and there are additional charts to allow conversion to other D_s distances.

The only other major factor affecting the potential acoustic gain is the number of microphones turned on at any one time. Figure 3 illustrates the number of dB that must be subtracted from the calculated potential acoustic gain for each additional microphone used.

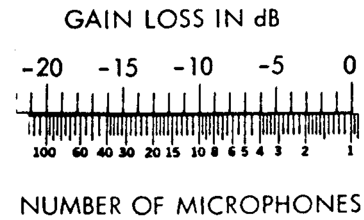


Figure 3

Deciding How Much Gain is Required

The first step in applying the gain formula is to select, from experience, what you feel would be a workable location for both the microphones and the loudspeaker array, considering coverage and time differences. After selecting the proposed loudspeaker and microphone locations, measure the following distances:

D_1, the distance between the proposed loudspeaker location and the nearest proposed microphone location.

D_2, the distance between the proposed loudspeaker location and the most remote listener location.

D_o, the distance between the proposed talker location and the most remote listener location.

Because these locations vary widely in different spaces and often are inaccessible to the measurer, Figures 4 through 7 are included to assist in the calculation of these distances.

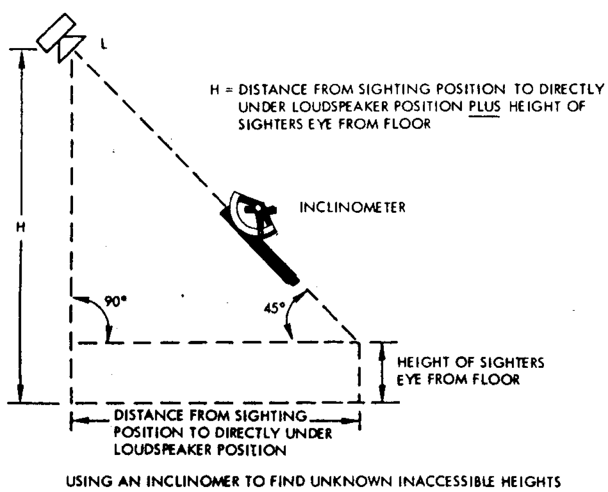


Figure 4

When an inclinometer is set for 45 degrees and the bubble is centered when the proposed location is viewed through the view-finder, the distance from the viewing location to the point directly below the proposed loudspeaker location plus the eye height of the viewer is equal to the height of the proposed loudspeaker location (isosceles triangle theorem). Only rarely are microphones and loudspeakers placed in a direct vertical relationship; therefore, the distance between them is most often the hypotenuse of a right triangle.

Having found the height of the proposed loudspeaker location and being able to measure the distance from the proposed microphone location to a point directly under the proposed loudspeaker location, use the Pythagorean Theorem (the sum of the square of the sides of a right triangle are equal to the square of the hypotenuse) to determine distance D_1 as shown in Figure 5.

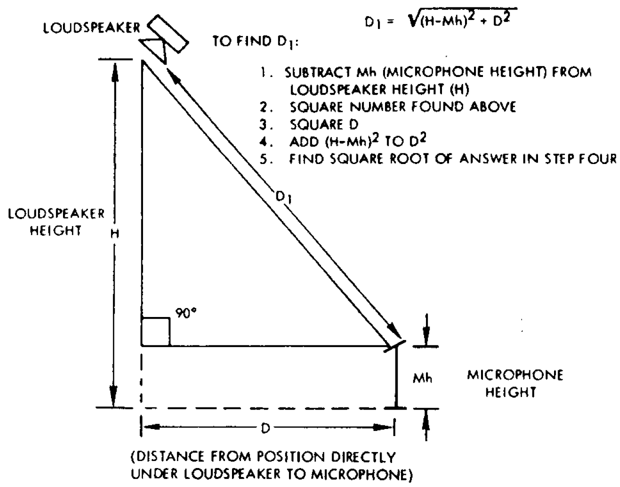
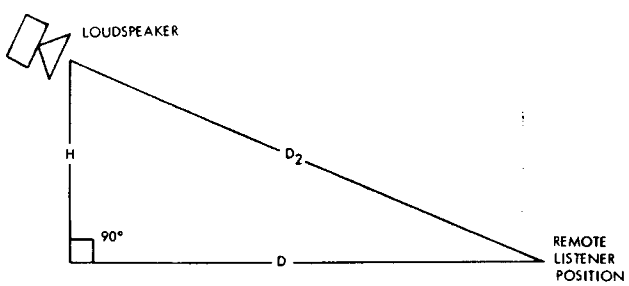


Figure 5

Finding the distance D_2 follows a similar pattern.



$H^2 + D^2 = (D_2)^2$

WHERE: H = HEIGHT OF LOUDSPEAKER

D = DISTANCE FROM POSITION DIRECTLY UNDER LOUDSPEAKER TO REMOTE LISTENER POSITION

Figure 6

To assist in the conversion of the sides into squares and of their sum into square root, refer to Figure 7.

The top scale is the number to be squared and the bottom scale is the number after being squared. To find the square root of a number, find the number on the bottom scale and directly above it on the top scale is its square root.

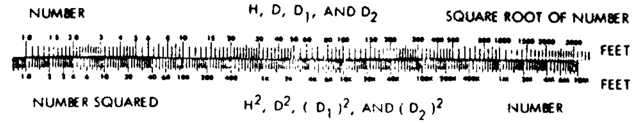


Figure 7

From these measurements, determine the required gain first. Using Figure 8, find D_o on the bottom scale and just above it on the top scale is the inverse square law loss in dB for this distance. Obviously, this is the number of dB acoustic gain required if this loss is to be overcome and if the most remote listener is to hear the same SPL as that appearing at the microphone diaphragm.

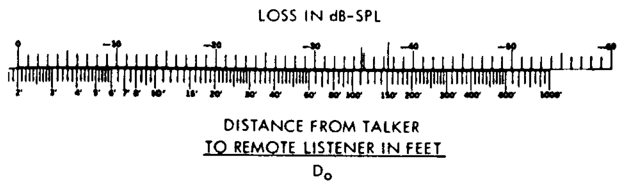


Figure 8

Direct and Reverberant Sound Fields

It must be remembered that inverse square law attenuation of the acoustic signal is valid only when the listener is in the direct sound field — where the direct sound energy predominates over the reflected sound energy. The direct sound field is dependent upon the directional characteristics of the loudspeakers but not upon the acoustics of the room. The reverberant sound field — depends upon the room acoustics and only upon the total radiated power characteristics of the loudspeakers, without regard to their directivity. It is of concern when the reflected sound energy predominates over the direct sound energy. In most rooms that are not either very small or very absorptive, the reverberant sound pressure predominates after a critical distance from the loudspeaker is exceeded.

A human listener, however, has the capabilities of localization included in his signal processing equipment and the direct sound field continues to exert significant influence until it drops more than -12 dB below the reflected sound.

The sound system engineer should always remember it is highly desirable, in system design, to increase the ratio for direct-to-reverberant sound over that which would have prevailed without sound reinforcement. Single source arrays can accomplish this through the use of directional horns and stacking of low-frequency loudspeakers. Distributed sound systems accomplish the same result by reducing the distance between the loudspeaker and the listener. All efforts in this direction materially aid speech intelligibility.

Inverse square law attenuation of the acoustic signal is a valid criterion, considering the above conditions, until the direct sound field falls more than -12 dB below the reverberant sound field.*

*Additional losses occur above 1000 Hz due to the effect of air absorption (refer to CE Technical Letter AV-1).

Calculation of a Loudspeaker's Critical Distance

The distance from a loudspeaker located in a semi-reverberant space (the type of space in which the majority of sound reinforcement systems are located) where direct sound energy is equal to reflected sound energy occurs at $D_c = 0.14 \sqrt{\bar{a}\ A}$

Where D_c = the critical distance from the loudspeaker for equality of direct-to-reflected sound energy in feet.

$\bar{a}$ = the absorption coefficient of the material on surfaces of the space.

A = the surface area of the enclosed space in square feet.

0.14 a constant.

Because of the ability of human hearing to detect direct sound energy down to -12 dB, compared to the reflected sound energy, for the purpose of this discussion the formula for D_c can be changed to:

$$D_c = 0.56 \sqrt{\bar{a}\ A}$$

Two Limits Calculated

To illustrate the use of this formula, consider two extremes. A space 100 feet long, 60 feet wide and 40 feet high with all surfaces hard (marble). The absorption coefficient of marble is 0.01. D_c is therefore $0.56 \sqrt{24{,}800 \times 0.01} = 8.8$ feet.

In the other extreme, the same dimensioned wall surfaces are a highly absorbent, perforated tile material with an absorption coefficient of 0.99. D_c then becomes

$$0.56 \sqrt{24{,}800 \times 0.99} = 87.9 \text{ feet.}$$

In actual jobs in semi-reverberant spaces, the half-way point between these two extremes is the value usually encountered. Thirty-five to 40 feet marks the critical distance of the loudspeaker(s) in a majority of cases. This distance also correlates with the ideal distance to put between the sound system microphone, whenever possible, and the sound system loudspeaker to achieve a high potential acoustic gain. And this is also the distance beyond which path-length differences can begin to cause detrimental time delay, so consider the importance of this parameter during the design process.

Always remember, while the reverberant field remains relatively constant beyond the critical distance, it does not mean the inverse square law attenuation can be ignored since inverse square law attenuation does continue, for the direct sound energy, all the way to the rear of the building. Even though its attenuation cannot be conveniently observed, all measures available must be taken to keep the ratio of direct-to-reflected sound as high as possible. For all of these reasons, it is best to treat loudspeaker layouts as if inverse square law attenuation were inviolable.

Considerations in Assigning the Loudspeaker an Initial Location for Calculation Purposes

In selecting an initial test location (on paper) for the loudspeaker so the potential acoustic gain available can be calculated, consider the following check list:

1. The loudspeaker should be far enough from the nearest microphone to take advantage of the critical distance of the loudspeaker.

2. The loudspeaker should not be so far from the talker's position that the difference in the path length from the talker to any listener and the path length from the loudspeaker to any listener is greater than 40 to 50 feet. This can vary in cases where a great difference in acoustic levels attend the path differences. Example: If the loudspeaker is 15 feet from the listener and the talker is 65 feet from the listener, but the SPL from the loudspeaker is 25 dB greater than the level arriving from the talker, the sound from the loudspeaker predominates (see Figure 1).

3. The loudspeaker location should allow an unobstructed sight line to all audience seats and preferably with as little difference in distance to all seats as possible (usually height provides a better angle for this purpose). It should also be noted that centerline locations offer:

   (a) Most symmetrical coverage potential.
   (b) Highest acoustic gain potential.
   (c) Minimum coupling with normal room modes.

Considering the above factors, it is always desirable to maximize the distance from the microphone to the loudspeaker while simultaneously minimizing the distance from the loudspeaker to the listener. Figure 9 illustrates the loss in level with distance from the loudspeaker to any distance point.

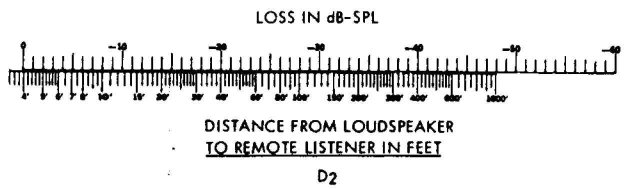


Figure 9

Making the Potential Acoustic Gain Calculation (see Figure 10)

SOUND SYSTEM ACOUSTIC GAIN BEFORE FEEDBACK (IF ACOUSTA-VOICED) EQUALS

$$20 \text{ LOG}_{10} \left[\frac{D_1}{D_s} \cdot \frac{D_o}{D_2} \right]$$

WHERE:

D_1 = DISTANCE FROM PROPOSED MICROPHONE LOCATION TO PROPOSED LOUDSPEAKER LOCATION

D_s = DISTANCE FROM TALKER TO MICROPHONE (FOR INITIAL CALCULATIONS D_s IS ASSUMED TO EQUAL TWO FEET)

D_o = DISTANCE FROM TALKER TO PROPOSED LOCATION OF MOST REMOTE LISTENER

D_2 = DISTANCE FROM PROPOSED LOUDSPEAKER LOCATION TO PROPOSED LOCATION OF MOST REMOTE LISTENER

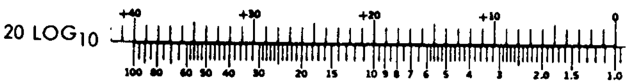
Figure 10

For purposes of illustration, assume a situation with the following conditions:

D_1 = 35 feet
D_o = 80 feet
D_2 = 70 feet
D_s = 2 feet (always start with this at 2 feet in initial calibrations because some common reference point is necessary and 2 feet is a realistic typical microphone distance).

First, $D_1 \times D_o = 2800$. $D_2 \times D_s = 140$. $2800 \div 140 = 20$. Using Figure 11, find 20 on the bottom scale. On the top scale, the corresponding potential acoustic gain is 26 dB. Referring to Figure 8, notice that a D_o of 80 feet equals a D_o loss of -32 dB. The loudness at the remote listener (which should be equal to that of the talker at the microphone diaphragm) is therefore -32 dB + 26 dB = -6 dB below the desired SPL.

POTENTIAL ACOUSTIC GAIN IN dB-SPL



$$\left[\frac{D_1}{D_s} \cdot \frac{D_o}{D_2}\right]$$

Figure 11

If this were a church and the microphone distance in the pulpit could be easily adjusted to 1 foot instead of two feet, the loudness at the remote listener would then equal that which would have occurred at the microphone diaphragm from the talker if the distance had remained two feet. In other words, the same effect can be obtained by "cheating" on microphone-to-talker distance (see Figure 12).

DISTANCES UNDER TWO FEET

GAIN RE LEVEL AT 2'

DISTANCE FROM TALKER TO MICROPHONE IN INCHES

(NORMAL D_s = 2')

Figure 12

If, instead of a church pulpit situation, this is a theatre in the round and it is desired to work the microphone at 10 feet; then (-6 dB) + (-14 dB) or 20 dB more gain potential is needed to satisfy the remote listener loudness requirement (see Figure 13). The use of multiple microphones can really complicate this type of situation.

The requirement to produce more than 26 dB acoustic gain (the maximum normally expected from a single source system), immediately suggests examination of the possiblity of using a distributed loudspeaker system. If the D_o loss is -32 dB, inspect how far away from the microphone the nearest loudspeaker can be placed. Example: Assume this D_1 distance is 35 feet. Then $20 \log 10 \frac{35}{10} \cdot \frac{35}{x} = 32$ dB enables the required height to be determined. In Figure 11, find the number below 32 dB (40). This means $\frac{35}{10} \cdot \frac{35}{x} = 40$, or $x = \frac{1225}{400}$ or 3 feet. These figures quickly show that a distributed loudspeaker system must be used.

DISTANCES OVER TWO FEET

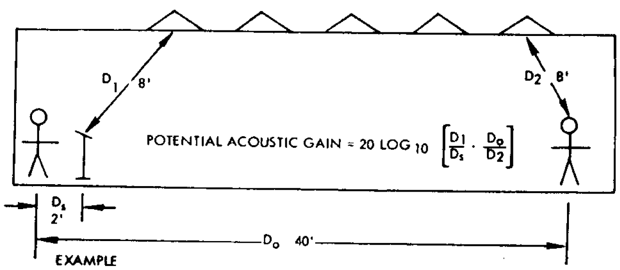
LOSS IN dB-SPL

TALKER TO MICROPHONE DISTANCE IN FEET

Figure 13

Handling the Loudspeaker Locations in Distributed Sound Reinforcement Systems

As will be quickly seen, a properly designed distributed-sound-reinforcement system costs a minimum of 50 percent more than an equally well designed single source system. But as just illustrated, there are cases where there is no possible way to achieve the necessary gain except through loudspeaker distribution. The same familiar dimensions occur in distributed sound reinforcement systems as are found in single source, but with the variations in meaning shown in Figure 14.



$20 \text{ LOG}_{10}\left[\frac{8}{2} \cdot \frac{40}{8}\right] = 26$ dB

D_o LOSS = 26 dB

D_1 = DISTANCE FROM MICROPHONE TO CLOSEST LOUDSPEAKER
D_2 = DISTANCE FROM LISTENER TO CLOSEST LOUDSPEAKER
D_o = DISTANCE FROM TALKER TO LISTENER
D_s = DISTANCE FROM TALKER TO MICROPHONE (2 FEET)

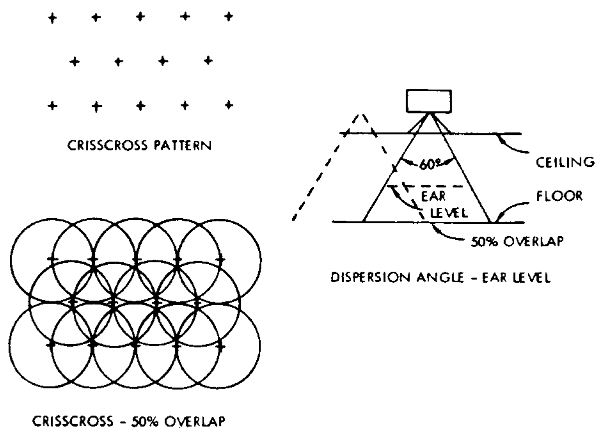
PHYSICAL MEASUREMENTS USED IN DISTRIBUTED SOUND SYSTEMS PERFORMANCE CALCULATIONS

Figure 14

In the case of distributed systems in small absorbent rooms (such as conference rooms) all the odds can very easily be stacked against the sound engineer. The advantage of the loudspeaker critical distance is not available, speakers and microphones are closer than desired for acceptable potential gain figures, and talkers are often far removed from microphones. The ultimate absurdity in this type of situation is the installation of both microphones and loudspeakers 3 feet apart, side-by-side in the ceiling, with D_2 and D_s distances of 10 feet each. Usually this unfortunate technical blunder is accompanied by equally poor installation and testing techniques and the result, to be charitable, is a disaster. By very careful control of loudspeaker-to-microphone distances and by hanging the microphones as close to the talkers as possible, some usable gain can be achieved. (Out-of-phase arrays with attention to symmetrical reflection patterns allows the microphones to be placed in null regions with attendent increase in acoustic gain). Remember, however, that this type of installation uses multiple microphones and Figure 3 is a good reminder of why these jobs require as much political influence with the customer as they do engineering planning.

Acceptable Distribution Criteria

The crisscross dispersion angle pattern shown in Figure 15 and the coverage radius versus ceiling height shown in Figure 16 enables quick calculation of the usable coverage area at 60 degrees for differing ceiling heights to listener ear levels.



DISTRIBUTED LOUDSPEAKER DATA

Figure 15

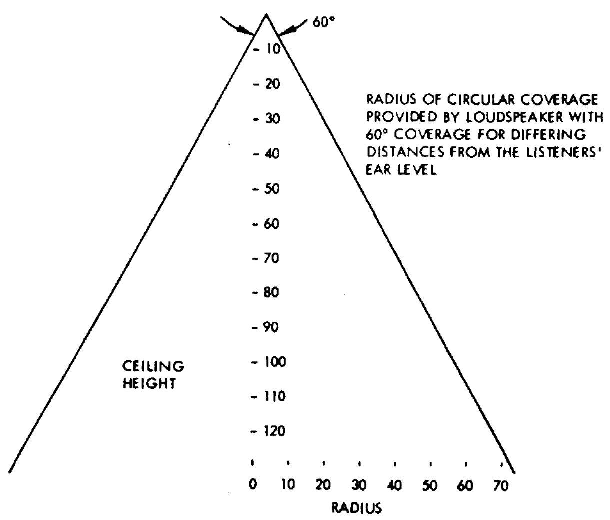


Figure 16

It is very important that these minimum distribution requirements be achieved. Any experienced contractor who has Acousta-Voiced large distributed systems will positively confirm that these requirements are on the conservative side. In laying out distributed sound systems, careful and continuous reference to Figure 1 insures awareness of the need for time delay apparatus if either distance or level parameters exceed those desired. Field experience has shown that 60 degrees included angle and 50 percent overlap is violated at the contractor's hazard. At this time, we are unaware of any manufacturer that can safely ignore them if uniformity of response throughout the listening area is expected.

In the very special case of industrial paging systems in noisy (over NC 60), highly reverberant (over 10 seconds at 500 and 1000 Hz), vast areas (1000' x 300') with high ceilings (80' to 100'), distributed sound systems can be made to operate very satisfactorily for voice paging. High power drivers on multicellular horns, achieving 200 to 400 percent overlap of rated patterns is conservative design. Naturally, each driver is allotted full power.

These systems cannot be made to work without Acousta-Voicing, and even though the rest of the system design is flawless, the system without benefit of tuning will only drive the room to a loud, uncontrolled roar.

Multicellular horns with from two to ten cells may be used with preference for fewer cells per horn and more horns.

In this extreme case for industrial systems, the only parameter to be calculated is density of coverage, and it can be stated from experience there has never yet been a system installed with too many loudspeakers.

Electrical Power Requirements per Driver

The final calculation before final determination of the loudspeaker location(s) is to verify the chosen drivers can handle the electrical power required to generate the sound pressure levels necessary to match that required at the remote listener's seat.

To again use the example of a single source system in a church, where the compromise of working the microphone at one foot can be accepted, it was found the system has a potential acoustic gain of 26 dB. Measure the pastor's sound pressure level with a sound level meter placed two feet in front of him. Typically, 75 to 80 dB-SPL is found to be a realistic figure. To be conservative then, assume delivery of 80 dB-SPL at the most remote listener's ears is desired. Part of the means of accomplishing this is moving the pastor closer to the microphone so that 81 to 86 dB-SPL then appears at the microphone diaphragm to enable delivery of 80 dB-SPL at the rear seat with a system capable of +26 dB acoustic gain but with a D_0 loss of -32 dB.

Referring to Figure 9, notice the D_2 loss from the loudspeaker to the most remote listener is 70 feet or approximately -25 dB. Therefore, 80 dB + 25 dB = 105 dB-SPL at 4'* must be generated to achieve 80 dB-SPL at 70 feet.

It is at this point that some engineers penalize themselves 10 dB and experience the consequent disappointment with the finished system. A +10 dB peaking factor must be allowed because all of the levels discussed up to this point are the kind read by an RMS sound level meter and this does not show the true peaks present. Therefore, at 4' from the loudspeaker, a capability of 105 dB + 10 dB = 115 dB-SPL is actually required to satisfy the power requirement.

If the system is to be Acousta-Voiced, the power must be raised an additional limit of not less than 10 dB to an approximate maximum of 20 dB, but that is a separate story. The question

*Used because Altec loudspeaker efficiency ratings are given at this distance.

now is, how much electrical power is needed to drive the loudspeaker array to this level — 115 dB-SPL? That, of course, depends on the efficiency of the loudspeaker array itself. Consider the following examples:

1. An 844A.

2. An A-7.

3. Multicells with 288Ds plus 515B low-frequency loudspeaker in an 825 enclosure.

Table I. Comparative Efficiency of Systems

SPL at 4'	844A	A-7	Multicell Array
Efficiency	99.0 dB	101.5 dB	103.0dB

Using Figure 17 allows rapid conversion of the 4' efficiency rating into a required electrical power input.

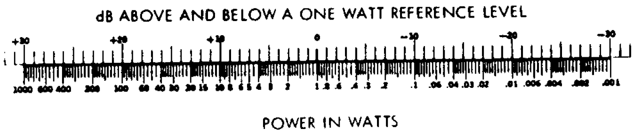


Figure 17

Referring to Table I, it can be quickly calculated that to meet requirements:

1. The 844A needs 115 dB - 99 dB = 16 dB-SPL more acoustic level at 4 feet.

2. The A-7 needs 115 dB - 101.5 dB = 13.5 dB-SPL more acoustic level at 4 feet.

3. The multicell array needs 115 dB - 103 dB = 12 dB-SPL more acoustic level at 4 feet.

Using Figure 18, the power requirements shown in Table II can be found.

Table II. Power Required to Reach 115 dB-SPL at 4 Feet

844A	A-7	Multicell Array
40 watts	22 watts	16 watts

To successfully Acousta-Voice the multicell array, an additional 10 dB or 160 watts would require bi-amplification with not less than two 80-watt power amplifiers, and sufficient drivers would have to be available. Typically; two 515Bs in a 211, plus two multicells — a near and far horn — each with two 288D drivers would meet this requirement.

One final example can quickly illustrate what is required if a large industrial paging system is properly engineered. This example, by the way, is restricted to the frequency range of 500 Hz to 3150 Hz where only short pages are the program material, but these short pages must be heard.

A 290E on a 203B multicellular horn has a true efficiency rating of 109 dB-SPL at 4' from 1 watt. If the ceiling height is 87 feet, the D_2 loss is approximately -27 dB (see Figure 9). The noise level through the frequency band quoted is approximately 65 dB-SPL. To be understood, this noise must be exceeded by not less than 12 dB, and in the interest of intelligibility in such a reverberant space, 20 dB would be of real value. Then 65 dB-SPL + 20 dB = 85 dB-SPL as the desired acoustic level, and 85 dB-SPL + D_2 loss of 27 dB = 112 dB-SPL as the required acoustic level measured 4' from the horn. Then adding 10 dB for peaking factor increases the required acoustic level to 122 dB-SPL. Using Figure 17, 112 dB-SPL - 109 dB = 13 dB is found to be approximately 20 watts of electrical input power to the driver. Again, looking at the figures, notice that 100 watts (the power rating of the driver) would allow 20 dB - 13 dB = 7 dB of Acousta-Voice equalization, so 100 watts should be allotted to each unit to provide a program level at the listener's ears of 65 dB + 12 dB or 77 dB because this driver is known to require more than 8 dB of correction.

To achieve the desired 200 percent minimum overlap (400 percent is noticeably better) meant 100 units in this space. 100 units x 100 watts = a 10,000 watt sound system. (Experience has shown that the necessity to raise the ratio of direct-to-reflected sound levels demands clustering of the drivers and horns in each area covered, often as many as four to a single area.)

Conclusion

These, then, are the necessary and sufficient conditions to consider in the engineering of a successful location for the loudspeaker(s) in a sound reinforcement system:

1. Sufficient acoustic gain.

2. Adequate coverage.

3. Freedom from time delay.

4. Capability of handling required electrical power to generate calculated acoustic levels.

Experience with literally several hundred sound systems in the last two years has shown that when these criteria are satisfied in full, the sound system always is a completely useful, adequate and easily sold system. Experience has again indicated that where the system is compromised up to -10 dB below these goals, the customer will buy the system and be happy IF he hasn't been exposed to a first class system. When the compromise exceeds -10 dB, you enter that shadowy area where WHO is selling the system has more bearing than WHAT he is selling, and I am always impressed by those who have mastered the art of repeatedly selling "the emperor's new suit", albeit to a necessarily different customer each time. This little manual of engineering loudspeaker locations is dedicated to those of you who feel REPEAT business is profitable.