TM

University Sound®

COMMERCIAL
SOUND AND
PUBLIC ADDRESS

INSTALLATION GUIDE

INTRODUCTION

The increasing sophistication of commercial sound and public address equipment and techniques has made advance planning of a sound system very important. This installation guide will assist the sound nstaller in building a system that will give quality performance. The definitions, guidelines and examples, along with the use of the individual product specifications, will give sufficient information to select and utilize the proper equipment to meet a wide variety of system requirements.

University Sound® commercial sound and public address products may be purchased through authorized University Sound dealers located throughout the U.S.A. as well as in many foreign countries. Product line brochures and engineering data sheets are available at no charge from your University Sound dealer or directly from University Sound, 13278 Ralston Avenue, Sylmar, CA 91342. Additional publications are also available from University Sound. The Electro-Voice *Microphone Primer* discusses microphone applications and terminology. The more elaborate *PA Bible thoroughly discusses many aspects of sound einforcement, with particular emphasis on the sound reinforcement of live music. The Microphone Primer* is complimentary. The *PA Bible* is available for a nominal fee.

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UNDERSTANDING SPEAKER SPECIFICATIONS

Axial Frequency Response and Sensitivity

The frequency response displayed on each University engineering data sheet shows how sound pressure level (SPL) varies with frequency directly in front of the speaker system (on axis) at a specified distance. A typical response curve is shown in Figure 1. The sensitivity (sometimes referred to as the SPL rating) is also given on the data sheet, and is defined as the average sound pressure level that the speaker system generates on axis for a specified frequency range, distance and power input. The sensitivity of the example in Figure 1 is 112 dB SPL for a one-watt input at a distance of one meter (3.28 feet).

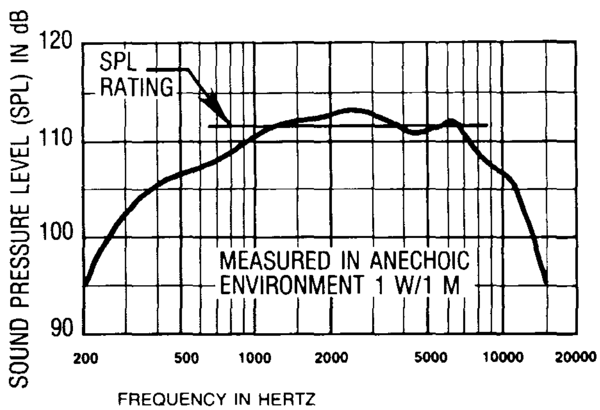


Figure 1
On Axis Frequency Response

The general standard for sensitivity is one watt at one meter; however, other power inputs and measurement distances are occasionally encountered. To allow direct comparisons to be made, conversion to the one-watt-at-one-meter standard must be made. The nomograph in Figure 2 gives the correction factor for the sound pressure level of a speaker versus distance, and the nomograph in Figure 3 gives the input power correction factor.

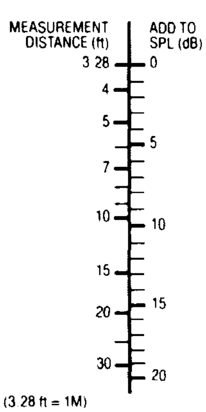


Figure 2
Level Variation with Distance

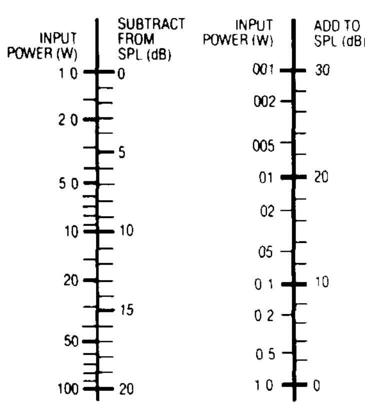


Figure 3
Level Variation with Power

Polar Response

A polar response plot shows how the sound pressure level of a speaker varies as a listener moves off axis from the speaker. Narrow bands (typically octave or one-third octave) of random noise are used to avoid the variations which may occur with single-frequency measurements. These measurements are usually made in both the horizontal and vertical planes as shown in Figure 4. An exception would be the polar plot of a ceiling speaker, where there would be a symmetrical hemispherical pattern.

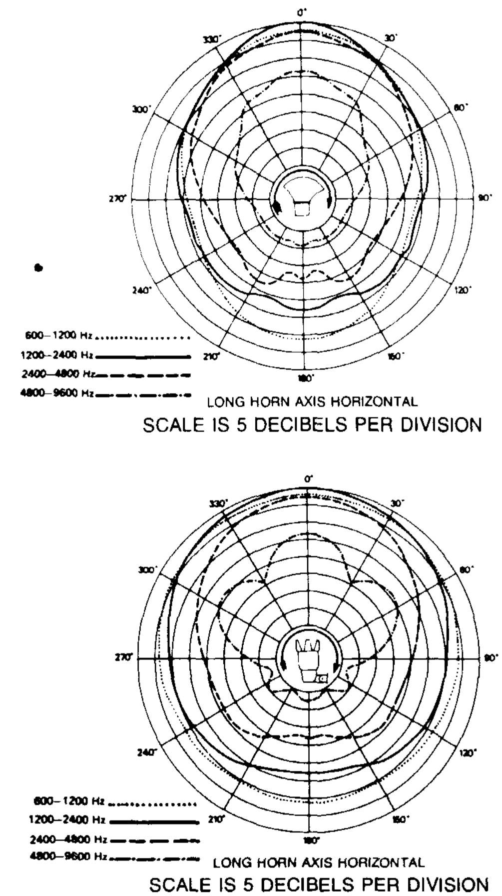


Figure 4
Polar Response
(Octave Bands of Random Noise)

Beamwidth Angle

When designing a system it is helpful to refer to both the axial and polar responses. However, it is often more convenient to represent the complex information of the polar responses (which show the output that a speaker has at any angle) with a single beamwidth angle, as if all speaker output were confined to that specific angle. In reality, the beamwidth angle (sometimes referred to as the coverage angle) identifies the angle within which "most" of the sound from the speaker is confined. Although no absolute standard exists, the beamwidth angle of a given frequency band is most often defined as the angle included by the points on the polar plot where the response is 6 dB less than the on-axis response (see Figure 5).

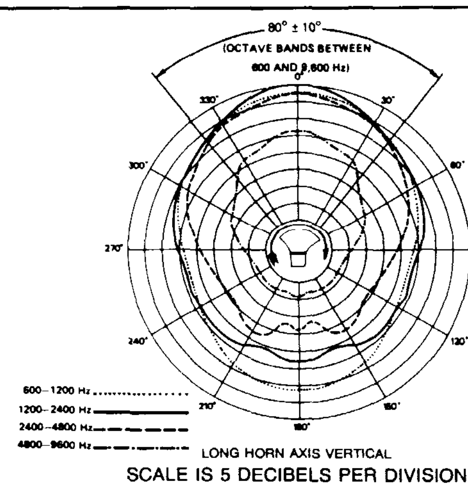


Figure 5
Beamwidth Angle

Ideally, a speaker's beamwidth angle would be nearly the same in every band. High-quality, professional loudspeakers approach this goal, but many general-purpose public address designs fall short, their beamwidth angles narrowing as frequency increases.

In conventional paging systems, the octave band where speech energy is concentrated is centered at 1,000 Hz. However, the next two octave bands, centered at 2,000 Hz and 4,000 Hz, contribute substantially to speech intelligibility, and listeners should be within the beamwidth angle of these bands as well for best results.

Floor-Plan Isobars

Ths sound pattern radiated by a loudspeaker can be described by a set of "isobar" curves (sometimes referred to as "pressure contours"). Floor-plan isobar curves show the "footprint" (the sound pattern created on the floor) of a particular loudspeaker mounted up in the air and pointed down at various angles (see Figure 6). These patterns of sound are created much in the same fashion that patterns of light are created on the ground from headlights on an automobile. The patterns shown in Figure 6 each consist of three lines of contour with an "X" in the center. The "X" marks the spot where the speaker is aimed (on axis), and the contour lines represent locations off axis where the sound pressure level is constant. The inner contour defines where the sound pressure level is 3 dB less than that measured near the center of the contour. The middle contour indicates the coverage area where the SPL is 6 dB below maximum, and the outer contour indicates the 9-dB-down level. It should be noted that a change of 3 dB in level is a minor but noticeable change in level; a change of 6 dB is definitely noticeable; and a change of 9 dB is perceived as approximate halving of loudness. The 6-dB-down contour can loosely be thought of as the two-dimensional result of the beamwidth of the speaker imposed on the floor. The isobar floor plans can be very useful for ensuring even sound coverage when designing a sound system (a design example using floor-plan isobars is given in the Designing for Intelligibility and Adequate SPL section).

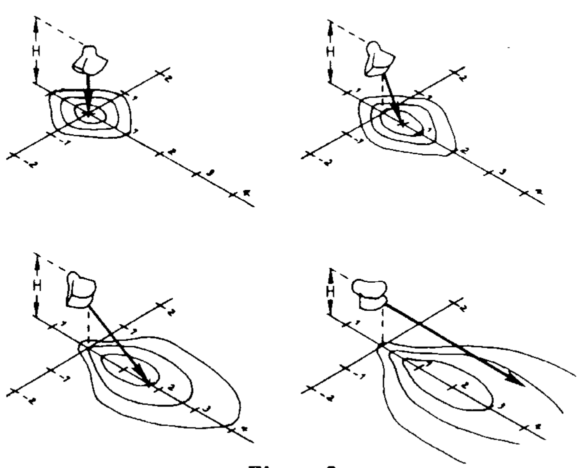


Figure 6
Floor-Plan Isobars

As is the case with beamwidth angle measurements, isobar curves are typically measured using octave-band random noise centered at 1000, 2000, or 4000 Hz. The isobars provide information that is not contained in either the standard polar response plots or the beamwidth angle — that is, the behavior of the speaker in all directions rather than just horizontally or vertically off axis. This information enables a designer to predict the direct sound contribution from each speaker in the system at any place on the floor of a room, or at ear level, allowing optimization of the system design both in cost and performance.

Power Handling

The power rating of a speaker must be known to determine whether a design is capable of meeting the sound pressure level requirements of the system. The power rating combined with the sensitivity will enable a system designer to calculate the maximum sound pressure level attainable at a given distance.

There are numerous methods of specifying power handling in use today. It is important when comparing power ratings by different manufacturers to know how each manufacturer arrives at this rating. We use a random noise input signal because it contains many frequencies simultaneously, just like real voice or music program. Our signal contains more energy at extremely high and low frequencies than typical actual program, adding an extra measure of reliability. The test signal also includes not only the overall "long-term average" or "continuous" level — which our ears interpret as loudness — but also short-duration peaks which are many times higher than the average, again just like actual program. The long-term average level stresses the speaker thermally (heat). The instantaneous peaks test mechanical reliability (cone and diaphragm excursion). Note that the sine-wave test signals sometimes used have a much less demanding peak value relative to their average level. In actual use, long-term average levels exist from several seconds on up, but we apply the long-term average for several hours, adding an extra measure of reliability.

DESIGNING FOR INTELLIGIBILITY AND ADEQUATE SPL

The Basic Idea

Many poor sound systems would have had better performance if the designer had kept the following basic principle in mind. Ideally, speakers with the appropriate coverage patterns should be chosen, aimed and powered to achieve a uniform direct field in the highly absorbtive audience, with no sound aimed at the reflective wall and ceiling surfaces. Where multiple speakers are required in order to achieve a uniform direct field, their coverage patterns should be only slightly overlapped, so that each section of the audience is covered by a single speaker. To the extent this ideal is achieved, reverberation is minimized and intelligibility is maximized.

The following material explains this concept in more detail and illustrates two design approaches using the principle.

What is Reverberation?

Reverberation is the persistence of sound within an enclosure, such as a room, after the original sound has ceased. Reverberation may also be considered as a series of multiple echoes so closely spaced in time that they merge into a single continuous sound. These echoes decrease in level with successive reflections, and eventually are completely absorbed by the room. An open outdoor space is considered to be a non-reverberant environment, as virtually all sound escapes the area without reflection.

Calculating Variations in Level Due to Distance for Non-Reverberant Environments

In a non-reverberant environment, such as outdoors, sound pressure level will be reduced by half (6 dB) every time the distance from the speaker is doubled (this is called the inverse-square law). Figure 7 shows the dB losses to be expected as distance from the speaker is increased from the one-meter (3.28 feet) measuring distance typically used in SPL specifications.

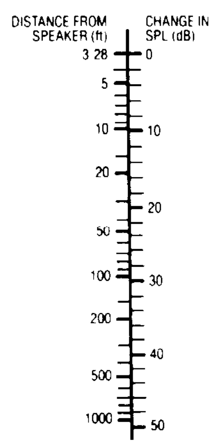


Figure 7
Level Variation With Distance

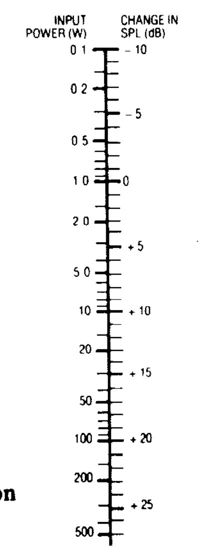


Figure 8
Level Variation With Power

Variations in Level Due to Distance in Reverberant Environments

Where sound is reflected from walls and other surfaces and the environment is reverberant, there is a point beyond which the "reverberant field" dominates and sound pressure level is relatively constant. This distance is typically 10 to 30 feet from the speaker, and is farthest for the least reverberant rooms and speakers with narrow beamwidth angles. Because of the reverberant field, the sound pressure level obtainable in a room is higher and more constant than that predicted by using the inverse-square law alone. However, the frequency and beamwidth information given in the specifications is still necessary in order to obtain satisfactory distribution of the direct sound (or direct field) from the loudspeaker(s), which still follows the inverse-square law. It is the direct signal that contributes to speech intelligibility. This is why the sound system designer should seek a uniform direct field, with as little reverberant field as possible. For example, consider a single speaker with a wide beamwidth angle used to cover a long, narrow, reverberant room. The direct field will be so far below the reverberant field at the back of the room that speech will probably be unintelligible.

Calculating Variations in Level Due to Changes in Electrical Power

In addition to the level variations with distance, each time the power delivered to the speaker is halved, a level drop of 3 dB occurs in any type of environment. The nomograph of Figure 8 shows the change in dB to be expected as the power varies from the one-watt input typically used in SPL specifications.

SPL Guidelines for Typical Applications

For Quiet Environments. In most sound reinforcement and paging situations, the background or ambient noise is low enough that the only question would be how much SPL does the listener expect or find comfortable. Table 1 lists recommended A-weighted average SPL's for a variety of venues and program types. This information may be used in conjunction with the preceding information on speaker sensitivity, level variation with distance and level variation with power in order to assure that the system will provide adequate sound pressure level without distortion. The sound pressure levels listed in Table 1 are based on Electro-Voice measurements and consultation with experts. The figures should be regarded as guidelines, or good starting points, since a particular situation — venue, program material and personal preference — can materially affect the sound level actually required.

Note that "average" used in Table 1 means sound levels that exist for several seconds or longer, such as would be measured on the "slow" scale of a sound level meter. Our ears tend to perceive such average levels as loudness. In actual practice, many of the average levels of Table 1 will exist only during the louder parts of an event. Much of the time, the levels will be 5-to-10 dB under those listed. The short-term peaks which exist in voice and music contribute little to perceived loudness. However, they must be reproduced accurately for a distortion-free sound quality. Short-term program peaks are at least 10 dB above the average.

Choosing the appropriate amplifier power to achieve the recommended sound levels is addressed in the next section.

For Noisy Environments. In situations where the background noise is high — as high, or higher, than what a listener might desire from the sound system in the absence of the noise — the concept of "getting over the noise" with the sound system is valid. Figure 9 gives typical A-weighted average noise levels for a variety of environments as well as sound pressure levels for typical sources. These levels are intended only to give an indications of the levels likely to be encountered. In cases where ambient noise may pose extreme problems, it is suggested that the actual sound pressure levels be measured before the sound system is designed. This typically would require a sound level meter with slow averaging time and "A weighting."

In "getting above" the noise for intelligible voice paging, the average SPL from the sound system should be at least 6 dB above the A-weighted noise. This is only a guideline, however, since the spectral distribution — or "frequency response" — of the noise will affect how much it masks the human voice. In addition, getting above the noise may not be possible. For example, in a sports environment the crowd noise is frequently — though not continuously — very loud: often 100 dBA average, and sometimes as high as 120 dB. Getting even 6 dB above the 100-dB noise may be possible only with a very complex and expensive sound system. A system that could provide uniform coverage in excess of 120 dB would be impractical to achieve, if not impossible. In such cases, the system user is advised not to announce during those brief periods when the crowd is very loud.

Powering to Achieve Both Average and Peak SPL's

The average power that must be delivered to the speaker(s) to achieve the desired average SPL can be determined from the previously presented material on speaker sensitivity, level variation with distance and level

TABLE 1
Guidelines for Maximum A-Weighted Average Sound Pressure Levels for Adequate Sound System Performance in Selected Venues When Background Noise is not a Problem

Venue	Situation	SPL's in dB
Church:	Speech reinforcement only	80-85
	With reinforcement of traditional or moderate-level contemporary music	90
	With reinforcement of rock-type music	95-100
Auditorium:	Speech reinforcement only	80-85
	With contemporary music (not full-tilt, big-concert rock and roll)	95-100
Quiet Factory:	Voice paging	80-85
	Background music	55-60
Office:	Voice paging	65
	Background music	40
Club/Lounge:	Quiet background entertainment, e.g., vocalist with piano or acoustic guitar	85
	Louder background entertainment, e.g., vocalist with pedal bass, keyboard, guitar, synthesizer, etc.	90
	Rock band, recorded dance music	95
	Louder rock band, recorded dance music (full-tilt, big-concert rock can be 110-120 dBA)	100-105
Gymnasium, Arena or Stadium:	(crowd noise likely to exceed capability of sound system)	100-105

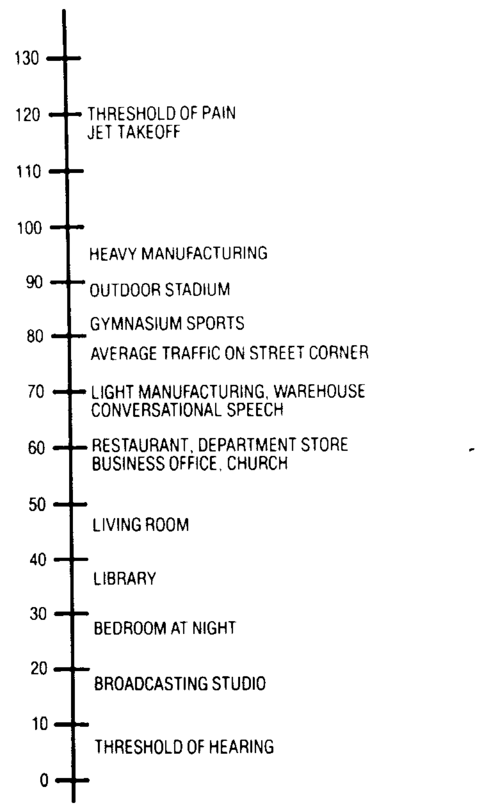


Figure 9
Typical Sound Pressure Levels

variation with power. Enough additional power must be available, however, to reproduce without distortion the short-term peaks that exist in voice and music program. This difference between the peak and average capability of a sound system, when expressed in dB, is often called "peak-to-average ratio," "crest factor" or "headroom." The peaks are large, as noted earlier: at least 10 times the average.

The finest sound systems are typically designed for peaks that are 10 dB above the average, whereas 6 dB of headroom is sufficient for a general-purpose voice paging system. The 10-dB peaks require amplifier power ten times that required for the average sound levels. The 6-dB peaks require four times the power. (Figure 3 shows the power factors associated with different headroom figures.)

An Example. Consider the church system shown in Figure 21, using an EV LR4B line radiator. (Note: Figures 18-21 are shown in the back of this guide.) Assume that services are conventional, so that the 80-dB average level listed in Table 1 is appropriate. Also assume that the maximum distance from listener to speaker is 80 feet and that 10 dB of headroom is desired. Using the LR4B's 1-watt/1-meter sensitivity of 98 dB, and the information in Figure 2 and Figure 3, it may be ascertained that 9.4 watts are required to achieve the 80-dB average level. Figure 2 shows that the 80-foot distance will subtract 27.7 dB from the 1-meter 98-dB rating, for a net level of 70.3 dB. This is 9.7 dB below the desired 80 dB. Figure 3 shows that 9.4 watts, instead of the 1 watt of the sensitivity rating, is required to make up the 9.7-dB shortfall. To provide the desired 10 dB of headroom, 94 watts of amplifier power would be required. A 100-watt amplifier would be ideal.

Maximum Power. In applications that require high sound levels, the amplifier power that can be safely applied to a given speaker must be determined. To efficiently utilize a speaker, amplifier power two-to-four times the long-term average noise power rating is appropriate (see Understanding Speaker Specifications section). This much power is recommended to permit driving the speaker to its long-term average rating with enough reserve power left to handle the short-term peaks, which do not harm the speaker and, in fact, are part of the typical noise power-test signal. However, if system operators are inexperienced and the sound level is increased without regard to the distortion which results from "cutting off," or clipping, a substantial portion of the program peaks, the long-term average capacity of the speaker may be exceeded. The likely result is speaker damage from excessive heat.

A more conservative maximum amplifier power recommendation, which will produce results nearly equal to those of the above professional standards, is power equal to the long-term average noise power rating of the speaker. A very conservative recommendation would be an amplifier 0.5-0.7 times the long-term noise rating. There is a caution associated with these conservative recommendations, however. While thermal damage from the long-term applications of power is unlikely to occur, it is more likely — because of the lower amplifier power — to clip the program peaks. The distortion that results from such peak clipping is in the high-frequency range and may damage the tweeters of full-range speaker systems.

Utilizing Speaker Beamwidth Information for Maximum Intelligibility

Knowing the beamwidth angle of a loudspeaker can aid in providing a uniform direct field in the absorptive audience, with as little sound as possible elsewhere. After selecting a desired speaker location, the beamwidth angle needed to adequately cover the listeners without spilling over to the walls or ceilings must be determined. Once these angles are known, the correct speaker can be found by using catalog specifications. The examples of typical sound system installations given at the end of this guide illustrate this technique.

In multiple-speaker, "central-cluster" installations, it is good practice to have the speakers in close proximity to each other and angled so that the sides of their beamwidth patterns slightly overlap. In distributed systems, where a much higher density of speakers exists, the degree of pattern overlap will affect the variation in SPL encountered at ear level. Maximum and minimum SPL's are created due to addition and cancellation of adjacent sources. A greater degree of pattern overlap will minimize the audibility of this variation. More specific design information is given in the CS410 and CS810 data sheets. Design examples are shown in Figures 18-25.

Using Easy-VAMP™ and Floor-Plan Isobars

Background. In some circumstances, it is desirable to use an approach that is more detailed than using the basic horizontal and vertical beamwidth angles. Environments which have excessive reverberation or high ambient noise levels make it especially difficult to achieve uniform coverage, the desired SPL and high intelligibility. In these situations, floor-plan isobars, introduced in the Understanding Speaker Specifications section, become especially useful.

In recent years, a number of computer-based techniques have been developed to help achieve uniform direct-field coverage. Some of the more complex systems use personal computers, with relatively sophisticated graphics. Simpler systems, such as Electro-Voice's VAMP™ (**V**ery **A**ccurate **M**apping **P**rogram), utilize clear overlays (similar to the floor-plan isobars described earlier) and the Hewlett-Packard HP41 calculator. However, the hardware/software and training investment required to utilize even the simpler systems are not attractive to all sound systems designers. Because of this, University Sound has developed a special adaptation of VAMP, called Easy-VAMP™, which provides a similar design aid without the complexity and cost of the VAMP program. This section explains the application of Easy-VAMP.

Floor-Plan Isobars. Isobars for selected University Sound speakers accompany this guide. (From time to time, University Sound intends to issue more isobars; write to the Advertising Department for information on currently available floor-plan isobars.) These isobars show the 2-kHz octave-band footprint for the speaker aimed at 0°, 30°, 60°, and 90° below horizontal. 2 kHz was selected because this octave band is critical to voice intelligibility. The angles represent a typical range of speaker orientations for practical situations. Intermediate angles may be used if the designer interpolates the information contained in the isobar curves.

Each isobar curve indicates where the SPL on the floor is 3 dB, 6 dB, or 9 dB below the maximum level. If these isobars are scaled to the mounting height of the speaker above ear level, the direct-signal SPL at ear level in a real room may be directly determined.

Each floor-plan isobar is plotted in graphical form with two perpendicular axes corresponding to the horizontal and vertical angles of coverage. Each axis has a series of marks labeled 1XH, 2XH, 3XH, 4XH, etc. The "H" represents the mounting height of the speaker above ear level and, hence,

Figure 10
Scaled Rulers for Typical Mounting Heights

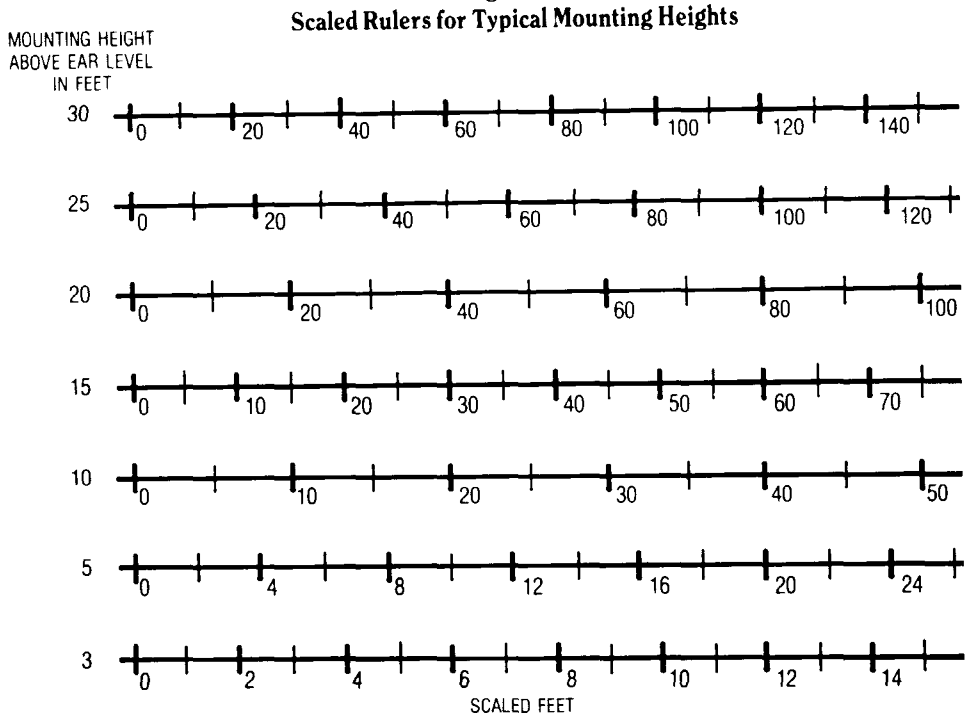
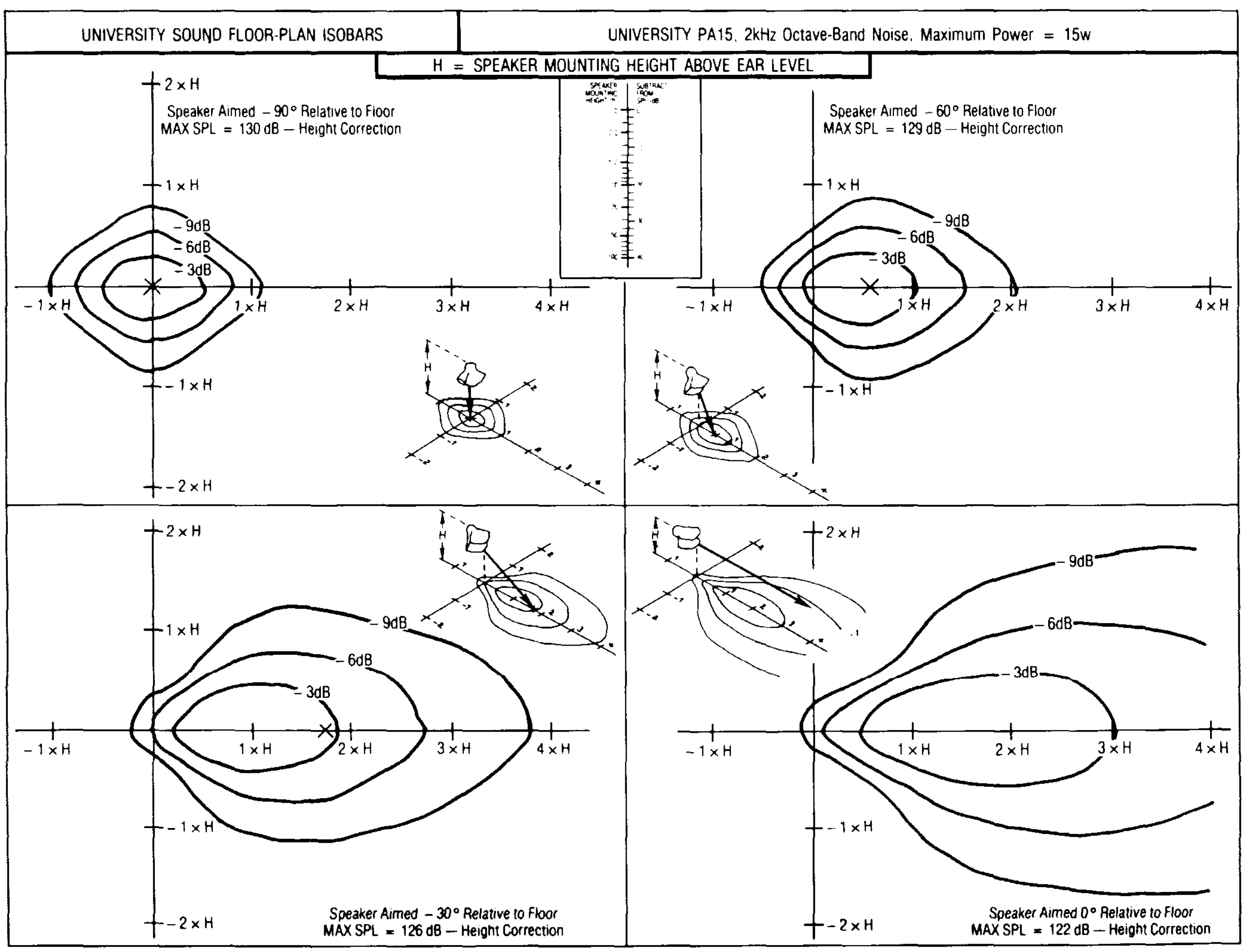


Figure 11
University Sound Floor-Plan Isobars



defines the scale of the floor plan. If the speaker mounting height is 10 feet, the resulting scale distances of the marks are 10 feet, 20 feet, 30 feet, 40 feet, etc. If the mounting height is 15 feet, the marks then become 15 feet, 30 feet, 45 feet, 60 feet, etc. Using this method, the size of the pattern on the floor may be determined for a particular speaker height. If the room is drawn to the same scale, the sound coverage of the room may be predicted by overlaying the floor-plan isobar onto the scaled floor plan of the room.

Using Easy-VAMP. The following five-step process will enable the user to quickly and accurately optimize area coverage using only simple materials.

1) Select a location for mounting the speaker, and determine the height of the speaker above the average ear level of the intended listeners, "H."

2) Draw a scaled floor plan of the room on tracing paper. A simple way to accomplish this is to create a scaled "ruler." Calculate and label the marked distances on the floor-plan isobar (as described above) for the chosen mounting height above ear level. Then lay a blank note card along the longest axis of the floor-plan isobar and transfer to the edge of the card the location of each mark on the isobar. Label these marks on the card with the appropriate distances. The card can now be used as the scaled ruler for drawing the room onto the tracing paper. Scaled rulers for several common mounting heights are given in Figure 10.

3) Experiment with mounting positions and angles for the chosen height by laying the room sketch over various isobars. Ideally, the pattern should cover the entire listening area of the room uniformly, while directing very little sound at the walls and ceiling (reflected sound adds to reverberation, ultimately decreasing intelligibility). Determine the sound pressure level for the direct sound using the nomograph on the floor-plan isobar. Remember that in a reverberant environment the actual resulting SPL may be slightly higher than the calculated level (see Designing for Intelligibility and Adequate SPL).

4) If the coverage and SPL are acceptable, go on to step 5; otherwise revise in the following manner:

   If the isobar pattern does not cover the listening area adequately but the mounting height can be increased, increase the speaker height (increasing the mounting height increases the area of coverage), and repeat steps 2 and 3.

   If the isobar pattern does not cover the listening area adequately but the height cannot be increased, try adding more speakers. Trace additional isobars onto some tracing paper and repeat step 3 using several isobars overlaid on the floor plan of the room. An excellent guideline to follow is to have the -3-dB contours of adjacent speakers touching but not overlapping.

   If the isobar patterns indicate an excessive amount of sound outside the listening area, try lowering the height of the speaker (decreasing the mounting height decreases the area of coverage), and repeat steps 2 and 3.

   If the sound pressure level is not sufficient, try another speaker with a higher sensitivity and/or input power rating, and repeat step 3.

   If the sound pressure level is not sufficient try locating the speaker closer to the listeners (decreasing the distance from speaker to listener increases SPL), and repeat steps 2 and 3.

5) Document the designs. Trace or copy the isobar(s) and room onto a single sheet for comparison to other designs. Also, compute the maximum SPL and note it on the tracings. This information should be enough for the user to choose the best speaker(s) and mounting locations for the project.

Easy-VAMP Example Design. Imagine a 28-foot by 24-foot by 20-foot-high room, which requires a uniformly intelligible paging speaker system with a maximum SPL capability of 90 dB. After investigating beamwidth and power-handling specifications we decide to try a PA15 mounted at the midpoint of one of the walls 2 feet below the ceiling. Using the standard floor-plan isobar for a PA15, shown in Figure 11, we sketch a scale drawing of the room. After some experimentation, we decide to aim the speaker downward at a 60° angle, with the results shown in Figure 12. We see that the PA15 isobar pattern does not cover the far corners of the room very well. Since the coverage angles are greatest along the perpendicular axes, we might suspect that a better mounting position would be near a corner of the room. Figure 13 shows the floor plan and coverage with the speaker at ceiling height near a corner, which shows greater direct signal in the far corners and less signal spilled on the walls. Although there could possibly be adequate sound pressure level in the corners with the first method (because of reverberation in the room), the latter method would result in a higher ratio of direct-to-reverberant sound and would give correspondingly higher intelligibility.

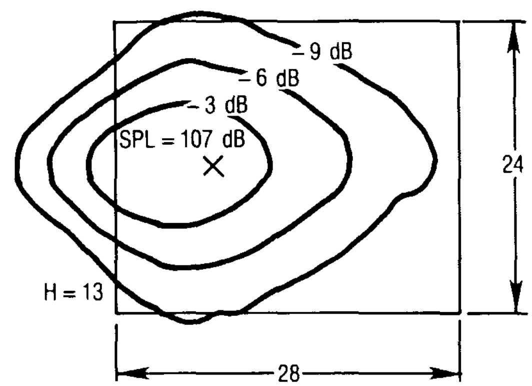


Figure 12
Design Example Using PA15 Mounted on End Wall

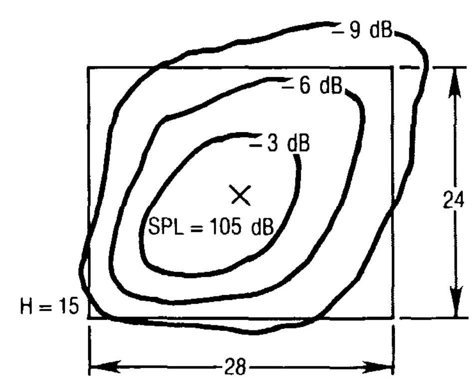


Figure 13
Design Example Using PA15 Mounted on Ceiling

MATCHING AND CONNECTING SPEAKERS

Impedance Matching

The idea that impedances must match when connecting speakers to an amplifier is somewhat fallacious. Often the typical speaker impedance "seen" by the amplifier is very much higher than the true output impedance looking back into the amplifier's output terminals. "Matching" is required, however, in the sense that the speaker load impedance must be suited to the amplifier. Amplifiers (with or without output transformers) are designed to deliver a rated power at a rated distortion figure into the rated load impedance. In multiple speaker designs, the combined speaker impedance should be calculated and "matched" within reasonable limits to the amplifier's rated load impedance.

Impedances higher than rated load will reduce the power delivered to the speakers, although typically distortion and other performance characteristics will be essentially unchanged. Impedances substantially lower than rated load should be avoided, as power at rated distortion will be reduced and possible damage to amplifier output stages or activation of protection circuitry may result.

Low-Impedance Systems

Low-impedance systems use speakers which have nominal impedances of 4, 8, 16 or 45 ohms. In series connection, speaker impedances add:

$$Z_T = Z_1 + Z_2 + Z_3 \ldots + Z_N,$$

where Z_T = total combined load impedance and $Z_1 \ldots Z_N$ are the individual speaker impedances.

In-phase series connection is obtained, as illustrated in Figure 14, by connecting the positive terminal of each speaker in the group to the negative terminal of the next speaker in the group.

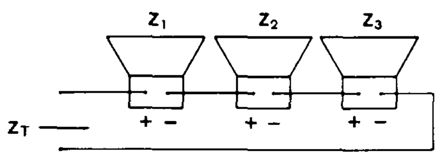


Figure 14
Series-Connected Speakers

In general, series connection should be used only with speakers of the same model and impedance. Otherwise, one speaker in the group may adversely affect the response of the others. Also, series connection is less reliable because a failure which causes any one speaker to become an open circuit will disconnect the others.

In parallel connection, the combined impedance of like units is the impedance of one unit divided by the number of units:

$$Z_T = \frac{Z_1}{N},$$

where Z_T is the total impedance, Z_1 is the impedance of one speaker and N is the total number of speakers.

In unlike units, the total impedance will be:

$$Z_T = \frac{1}{1/Z_1 + 1/Z_2 + 1/Z_3 + 1/Z_N},$$

where Z_T is the total impedance, and $Z_1 \ldots Z_N$ would be the individual speaker impedances.

In-phase parallel connection is illustrated in Figure 15, where all like terminals are connected together.

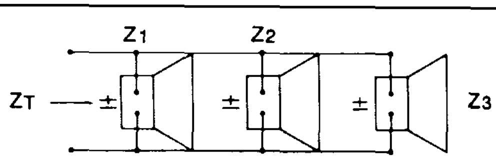


Figure 15
Parallel-Connected Speakers

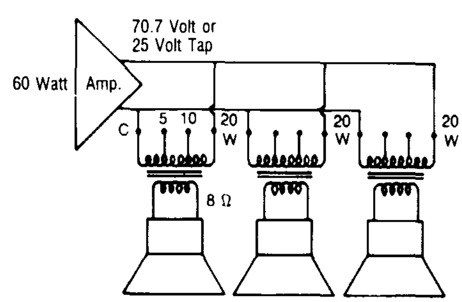
For combination series and parallel, compute the impedance of each similarly wired group of speakers, and then, considering each group as a single speaker, compute the total combined impedance.

In the example of Figure 18 the impedance of the LR4B and the Musicaster™ 100 are both 8 ohms. Therefore, the best approach would be to parallel the two speakers and put them on the 4-ohm tap of the amplifier. "L-pads" may be used to individually adjust the level of each speaker, if desired.

High-Impedance/Constant-Voltage Systems

Systems that have a high-impedance speaker load permit much smaller diameter wire to be used for the speaker lines for a given power loss. Usually, low-impedance speakers are still used in such systems, but transformers are employed at the speaker locations to increase the impedance to the desired value. Multiple transformer taps permit easy adjustment of individual speaker power levels. For many systems, the cost savings in wiring more than compensates for the cost and power loss (insertion loss) of the transformer. Additionally autotransformer level controls may be used to individually adjust the level of each speaker without the power dissipation inherent in L-pads.

Constant-voltage high-impedance systems offer the additional convenience of eliminating the impedance-matching calculations described above by having the full-power output voltage of the amplifier across the speaker load at a constant 25, 70.7 or 100 volts. (Note that the "constant" 25, 70.7 or 100 volts is present only when the amplifier is delivering full rated power to the rated load, and is correspondingly reduced for reduced input signal level.) These systems permit the secondary taps of the transformer to be marked in watts. In system hook-up, the transformers of the speakers are connected in parallel across the amplifier output terminals. When all the power taps add up to the amplifier's rated output, the full rated load impedance is present at the amplifier terminals. See Figure 16 for a typical example.



Each transformer/speaker system connected to 20-watt tap. Note that the total equals the rated power of the amplifer.

Figure 16
Constant-Voltage High-Impedance System

Powers less than rated output are fully acceptable, since the higher-than-rated load impedance that results only decreases the power delivered to the load — precisely the desired effect. Only power totals greater than rated amplifier power should be avoided: load impedance drops below rated load, reducing power delivered at rated distortion and perhaps damaging output stages or activating protection circuitry. A constant-voltage high-impedance system would be used in the distributed system in Figure 19 or the factory example of Figure 20.

Power Loss in Long Lines

For long wire runs, the power losses in the wire become significant, especially for smaller wire sizes. For a 0.1-dB loss in sound pressure level, the total wire impedance should be limited to 1% of the speaker impedance. The somewhat larger losses of 0.5 dB and 1 dB are encountered when the percentages increase to 6% and 12% respectively.

The power supplied from the amplifier terminals to the load is divided between the resistance of the wire and the impedance of the speaker. As an example, see Figure 17. If we assume that the amplifier is delivering 20 watts to the load, then only 17.8 watts is being delivered to the speaker while 2.2 watts is being lost due to line resistance.

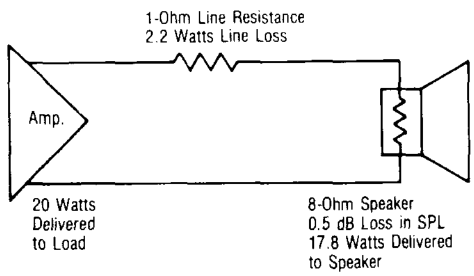


Figure 17
Line Loss

For a 0.5-dB loss in sound pressure level, Table 2 shows the calculated two-wire cable lengths permissible for a number of wire sizes and impedances.

For a 1-dB loss, the wire lengths in Table 2 may be doubled. For a 2-dB loss, the lengths should be multiplied by 4.4.

INSTALLING SPEAKER SYSTEMS

University Sound speakers for commercial sound and public address applications have been designed to mount to a wide variety of surfaces, including walls, ceilings, I-beams, pipes, poles and ac junction boxes. These mounting options give flexibility to the sound system design and time savings in the installation. For specific mounting details of each University Sound speaker consult the individual data sheet.

		Low-Impedance Systems			High-Impedance Systems				
AWG Size	Resistance (ohms/1000 feet)	4 Ω	8 Ω	16 Ω	200 W/100 V 100 W/70.7 V 12½ W/25 V (50 Ω)	100 W/100 V 50 W/70.7 V 6¼ W/25 V (100 Ω)	50 W/100 V 25 W/70.7 V 3⅛ W/25 V (200 Ω)	10 W/100 V 5 W/70.7 V ⅝ W/25 V (1000 Ω)	2 W/100 V 1 W/70.7 V ⅛ W/25 V (5000 Ω)
10	1.00	120	240	480	1,500	3,000	6,000	30,000	150,000
12	1.59	75	150	300	940	1,900	3,800	19,000	94,000
14	2.50	48	96	190	600	1,200	2,400	12,000	60,000
16	4.02	30	60	120	370	750	1,500	7,500	37,000
18	6.39	19	38	75	230	470	940	4,700	23,000
20	10.1	12	24	48	150	300	590	3,000	15,000
22	16.2	7	15	30	93	190	370	1,900	9,300

TABLE 2
2-Wire Copper Cable Lengths in Feet for 0.5-dB Loss in SPL

SOUND SYSTEM EXAMPLES

Below are some examples of places University Sound P.A. products are used. These sketches are presented only as a general guideline in system design. You, a sound contractor, or a qualified audio consultant can work out the details.

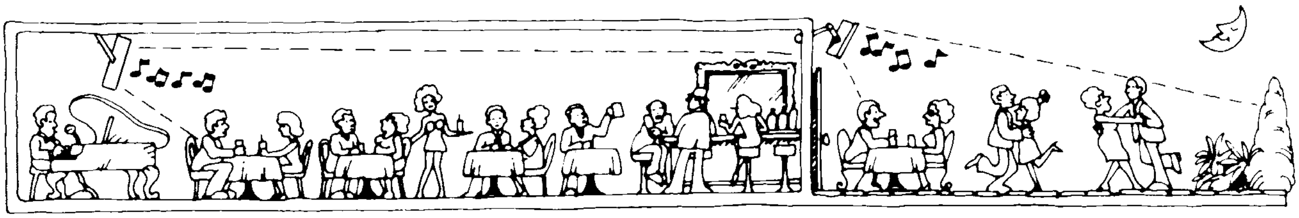


Figure 18. Piano Bar and Patio. LR4B used to project sound from main entertainment area. Weatherproof Musicasters used in outdoor patio system.

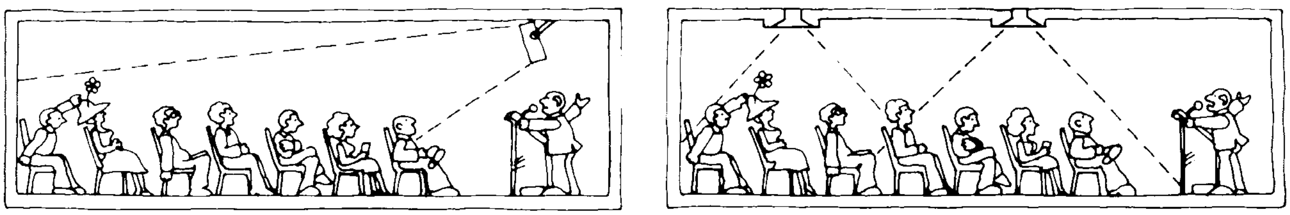


Figure 19. Conference Room. Either a single LR4B or a distributed system of CS810 ceiling speakers may be used. The distributed system might be better suited for rooms that are long with low ceilings.

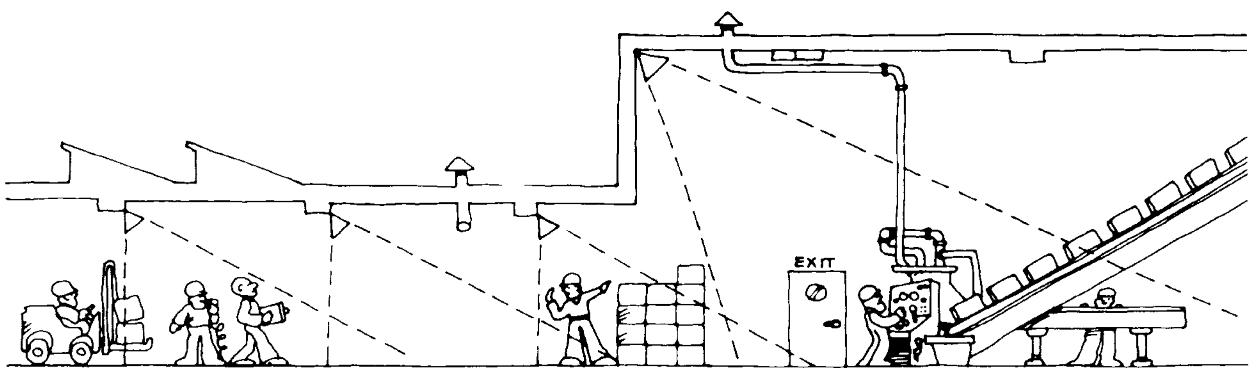


Figure 20. Factory. The size of the paging speaker and/or the power tap selected is determined by the size of the coverage area and the ambient noise in the area. Several candidates would be the PA12, PA15, PA30, CFID15, CFID32 and MILC.

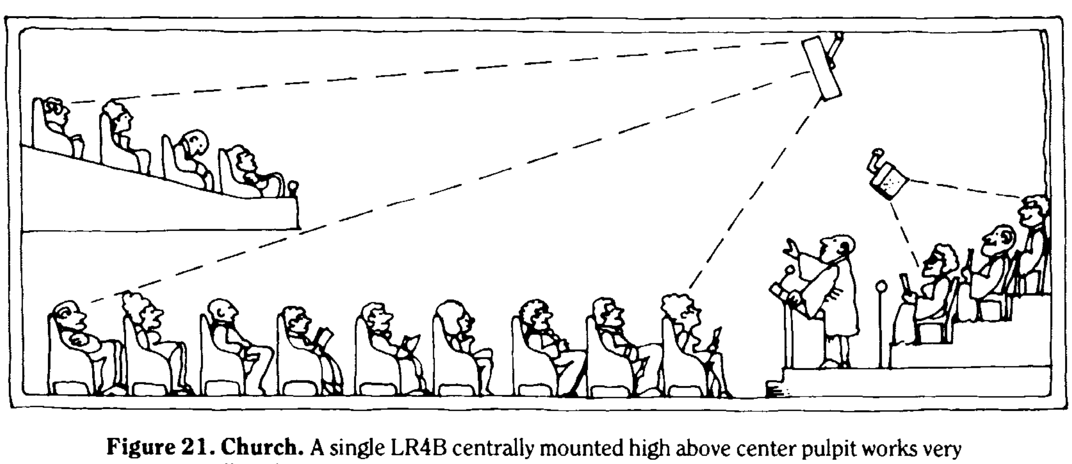


Figure 21. Church. A single LR4B centrally mounted high above center pulpit works very well in churches. One Musicaster™ 100 may be used for choir coverage.

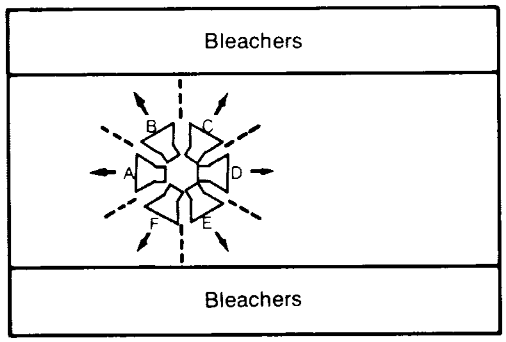


Figure 22. Small Gym

The central cluster, mounted slightly off center, would consist of either six FC100 horns with 1829T drivers or six Cobreflex IIB horns with ID60CT drivers. A good choice for an announcement microphone would be the 664A.

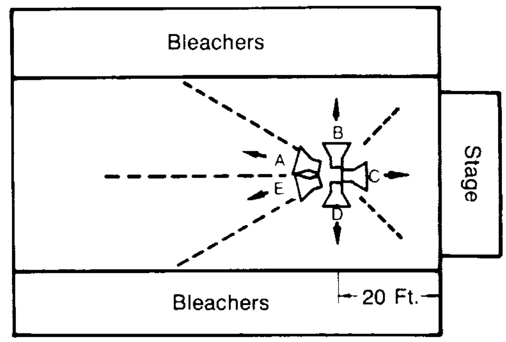


Figure 23. Small Gym with Stage

The central cluster would be mounted about 20 feet from the stage end of the gym, high under the girders. The cluster would be made up of the following: two Cobreflex III horns with ID60CT drivers (A&E); two Cobreflex IIB horns with ID60CT drivers (B&D); and a single Cobreflex IIB horn with an ID30CT driver for stage foldback (C).

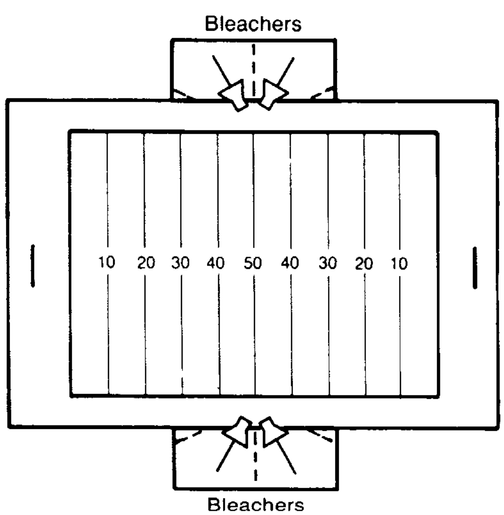


Figure 24. Small Football Stadium

There would be a small cluster on each side of the field mounted in front of the bleachers on a pole about 25 feet high. Each side would have either two FC100 horns with 1829T drivers or two Cobreflex IIB horns with ID60CT drivers. A good choice for an announcement microphone would be the 664A with the 360 windscreen.

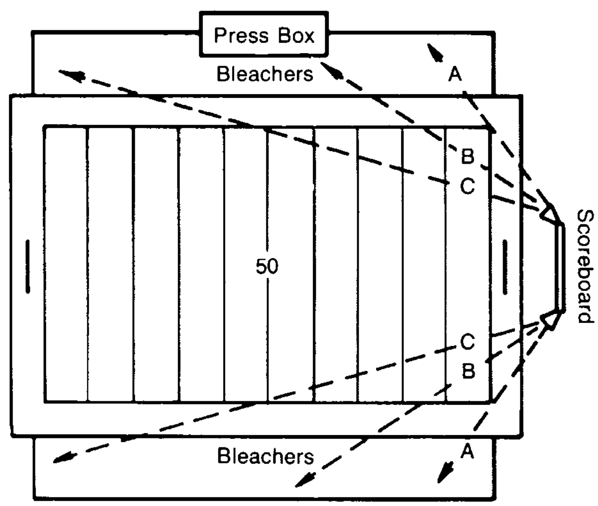


Figure 25. Medium-Sized Football Stadium

A central cluster would be mounted at one end of the field on either side of the scoreboard or on poles. The cluster would consist of the following: two Cobreflex IIB horns (one facing each side of the field) with ID60CT drivers for short throw (A); two Cobreflex III horns (one facing each side of the field) with ID60CT drivers for medium throw (B); and four Cobreflex III horns (two facing each side of the field) with ID60CT drivers for long throw (C). A good choice for an announcement microphone would be the 664A with the 360 windscreen.

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