Sound Reinforcement for Christian Conventions

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Sound Reinforcement for Christian Conventions
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INTRODUCTION

The Purpose of the *Sound Handbook*

1. The success of modern-day Christian conventions of any size requires the application of some type of sound-reinforcement system. God's Word reminds us of the principle that "unless you with the tongue use speech that is easily understood, how will anyone know what is being said? You will, in fact, be speaking into the air." Indeed, to the extent that those in the audience "do not understand the sense of the speech," it will be as though we are "speaking into the air" and much of the spiritual benefit will be lost. Thus, the basic objective of those entrusted with the responsibility for good sound is to communicate clear and intelligible speech that can be "easily understood" by each listener at our Christian conventions. Those who have access to the *Sound Handbook* may distribute it to those who are involved in sound reinforcement at theocratic events.—1 Cor. 14: 8-11.

2. Over the years, many brothers have recognized the vital part that good sound plays in the successful conventions of God's people and have lovingly and willingly volunteered their time, effort, and skills—frequently working with very limited equipment and time schedules—to provide the best sound possible. They are to be commended for their 'faithful work and loving labor.' (1 Thess. 1:3) With the increasing number and size of Christian gatherings throughout the world, however, there is a need to equip more brothers with an expanded knowledge and understanding of basic acoustic and electrical principles, as well as to familiarize them with modern audio equipment and its application in successful sound reinforcement.

3. Since many brothers have busy schedules and limited time, it is not our intent to burden them with complex technical material and engineering data that require heavy investments of study time to assimilate. Rather, our objective is to provide simple explanations of the basic acoustic and audio principles that affect sound reinforcement. An understanding of how physical laws impact specific acoustic environments will be of assistance in designing and planning satisfactory, predictable sound systems. Adherence to basic electrical principles will help to ensure good audio quality and a stable, reliable system. Large sound systems are not designed by a simple engineering formula or by a calculator or a computer. Rather, good sound is often the result of empirical judgment—in other words, good sense coupled with experience and accurate knowledge.

4. The recommendations offered in this handbook are not to be construed as rigid rules, nor are they intended to force the abandonment of existing successful designs and procedures in favor of new ones that may be described herein. Situations vary from place to place and from country to country. While electrical and acoustic principles do not change, there are sometimes other variables that require a measure of flexibility. Factors such as costs, legal restrictions, or limited availability of equipment may call for the use of a different approach. Each specific question or situation will have to be considered on its own merit by those having responsibility for the convention operation.

How to Use the *Sound Handbook*

5. What type of material is contained in the *Sound Handbook?* Its main section covers in logical sequence the steps necessary for planning and achieving a successful convention sound system: selection, design, installation, optimization, operation, troubleshooting, and dismantling. Efforts have been made to provide essential information that outlines efficient and accepted procedures unique to each phase of the work, including pitfalls to avoid.

6. For those who need or desire additional technical information, eight appendixes, including a helpful glossary, are included. Among the topics discussed are basic electrical and acoustic fundamentals, interpretation of component specifications, and proper maintenance of equipment. With its comprehensive index, the *Sound Handbook* should serve as a useful reference for answers and solutions to many problems unique to convention sound systems. Although not every soundman will find an immediate need for all the material presented, it is comforting to know that helpful information is available when needed.

7. As in any technical field, new concepts and designs are continually being researched and developed, and these may in time modify some of the information presented. Also, in certain aspects of sound reinforcement, there are differing schools of thought as to how things should be done. In addition, a variety of complex mathematical formulas have been developed in the sound industry over the years, requiring an in-depth engineering background in order to utilize them effectively. Many sophisticated instruments and costly computer programs have also been developed. Some of these require the gathering of large amounts of data before they can perform their functions. Others may be able to measure and evaluate the performance of a sound system accurately—but only after the system is installed and operating.

8. This handbook, however, emphasizes practices and procedures that have proved practical and that work well when implemented at our Christian conventions. If additional technical data on a specific subject is desired, references to various publications on sound engineering (in English) are listed at the end of Appendix 7.

9. The *Sound Handbook,* based on current knowledge and years of experience, is intended to help the brothers to avoid technical mistakes and to unify their efforts in the field of sound. No doubt it will be of value in the future design, installation, and operation of successful sound systems and will aid in producing sufficient numbers of willing and qualified men to assist in this important feature of Christian gatherings.

CHAPTER ONE

Sound Reinforcement at Christian Conventions

1. Centuries ago, Jehovah God lovingly commanded his worshippers in Israel to assemble in large and small groups for their spiritual instruction. These 'assemblies' or 'holy conventions' served to unify the worship and activity of God's people by communicating the same instruction and teaching to everyone at the same time. (Ex. 12:3, 16; Deut. 16:13-17) Loyal family heads saw to it that their entire family attended, in obedience to Jehovah's specific instruction to "gather the people together, the men, the women, the children, and your foreign resident who is within your cities, in order that they may *listen* and learn about and fear Jehovah your God and take care to carry out all the words of this Law."—Deut. 31:12; Neh. 8:1-8; Luke 2:41-45; *be* p. 139.

2. Similarly, in modern times large and small conventions are arranged under the direction of the Governing Body of Jehovah's Witnesses to communicate spiritual "food at the proper time" and to unify the worship and activity of God's people on earth today. (Matt. 24:45) Indeed, many of these past conventions are viewed as milestones in the forward progress of Jehovah's earthly organization. As the worldwide brotherhood has grown in numbers, so have the number and size of our Christian gatherings, embracing hundreds of languages and nationalities.—Heb. 10:24, 25.

3. Currently, thousands of 'assemblies' and 'holy conventions' are held annually by Jehovah's Witnesses around the earth in a great variety of environments where modern *sound-reinforcement systems* enable literally millions of persons to hear the live spiritual program and benefit from its presentation. However, when a sound system performs poorly or fails to provide adequate volume or clarity to all or part of the audience, the overall objective of the program is compromised. Although a talk or demonstration may be well prepared and well presented, it will be of little value to those who cannot hear it clearly. We appreciate that our brothers and their families—often at considerable personal cost in time, effort, and funds—have come to *hear* the program in expectation of receiving its spiritual encouragement and benefit. Our objective is to provide them with an effective communication link, "in order that they may *listen* and learn," so that at the conclusion of the program, each person in the audience may leave spiritually refreshed and satisfied that his expectations were fulfilled and his efforts rewarded.—Matt. 5:3.

The Goals of Sound Reinforcement at Christian Conventions

4. In this handbook we will address many factors that contribute to the reproduction of good sound, on the premise that adequate and intelligible sound will provide an effective communication link, promote greater attentiveness and concentration, and contribute significantly to the success and spiritual benefit of the program. It follows that those entrusted with this responsibility not only should be conscientious in reaching these objectives but must also be adequately trained and equipped to accomplish their assignments successfully.

5. There are three basic goals that should be considered in the planning and operation of sound-reinforcement systems at our Christian conventions:

(1) ***Adequate sound level*****—**Uniform sound coverage of the seating areas to be used. This does not necessarily mean *equal* sound level in all areas, as sound requirements can vary greatly, especially in large areas or multiple rooms.

(2) ***Intelligibility*****—**Good speech articulation and clarity without distortion, acoustic feedback, extraneous noise, or excessive reverberation—all of which can degrade articulation and impair attentiveness.

(3) ***Naturalness*****—**Comfortable and natural sound quality will help the listener to remain focused on the program without distraction.

6. When these goals are successfully achieved, our brothers and sisters, young and old, will derive the greatest benefit from the spiritual programs prepared by ‘the faithful slave.’ Thus, our aim is to have the sound system reliably installed, thoroughly tested, and ready for operation before the convention opens. When a well-planned, properly managed sound system successfully attains the goals mentioned above, the average listener is seldom aware of its presence. Experience has taught us that when our brothers listen attentively throughout the program—when virtually no one is seen walking in the aisles or milling in the corridors and when, at the conclusion of the program, our brothers are bubbling with enthusiasm over what they have heard and learned—we can feel confident that the sound system has performed satisfactorily.

7. Certain acoustic deficiencies, including *weak sound, loud ambient noise,* or *excessive reverberation,* can result in many listeners missing some of the spoken words or syllables. This often calls for intense concentration to follow the thought content of the program. Such concentrated effort can be quite tiring, and in time, minds may begin to wander because of *listener fatigue*. This is especially true among those with less than average hearing ability. Conversely, *excessive loudness* can be unpleasant, distracting, and even oppressive—and we should not overlook the fact that extended exposure to excessively loud sound can be damaging to our delicate hearing faculties.

8. At times, we may face challenging situations where the desired goals are not easily achieved, perhaps due to excessive noise, reverberation, echoes, and so forth, because of environmental or architectural anomalies. Special attention and additional time may be required to address these problems and implement a sound system that is reasonably acceptable—a small sacrifice compared to the spiritual rewards that we and our brothers derive from attending a Christian convention of Jehovah's people. Many of these factors will be considered in Chapter 4 and Chapter 7.

9. The person speaking into the microphone is the beginning of the communication link. This is a critical area, and to the extent possible, it is our desire to make each program participant comfortable as he makes his presentation. This will be considered in greater detail in Chapter 8. (To avoid confusion with components of the sound system, in later chapters we will refer to the person speaking into the microphone as the *talker,* rather than the *speaker.*)

10. These are just a few of the factors that concern us in the intelligent planning and smooth operation of a successful sound system. Along with a knowledgeable and professional approach, we must never overlook the spiritual aspects of our work—prayer and our dependence on Jehovah's holy spirit. Successful sound performance includes more than merely operating a sound system correctly. Good coordination among everyone involved with the program is essential. Rehearsals or walk-throughs may be needed for some of the parts that will be presented so that participants will be confident and prepared to use the microphones in the most effective manner. Both prior to and during the convention, we will want to maintain good communication and close cooperation with the convention committee, the chairman, and the platform personnel, as well as with the participants themselves.

11. Achieving these goals begins with good planning, careful design, and a degree of professionalism. Yet, even the best-engineered system could fail or function poorly if it is not installed or operated correctly. This requires volunteers—our brothers—who possess sufficient knowledge to install and operate the components properly. When a venue is scheduled for multiple conventions, very likely a different team of brothers will install and manage the sound system at each convention. Each team needs to be knowledgeable and proficient so as to maintain consistent performance at each convention. Here is where our Christian training and experience come to the fore. Our obedience to direction from Jehovah's organization should be exemplary. (1 Sam. 15:22) Training our brothers and directing them in a kind and loving manner calls for displaying "the fruitage of the spirit" at all times. (Gal. 5:22, 23) We must never forget that we depend on these willing and loyal brothers to assist with establishing and operating this essential communication link successfully. Indeed, regardless of how extensive our personal technical skills may be, our most powerful tool is Jehovah's holy spirit on which we rely in all our efforts.

CHAPTER TWO

Basics of Sound

1. It is a tremendous privilege to be used by Jehovah's earthly organization to support Christ's brothers in dispensing spiritual truths at Christian conventions. Such truths fortify our faith and glorify our God, Jehovah.

2. To make sure every spoken word of truth can be heard by those in attendance requires much work and many willing workers. All of us recognize that there are various degrees of experience, knowledge, skills, and responsibilities among those working with sound. With this in mind, the aim of this section is to provide a basic understanding of what sound is and the terms that quantify it. You will find these terms succinctly defined below. For a more thorough explanation of these physical properties and their formulas, please see Appendix 2.

What Is Sound?

3. To the average person, sound is what we hear, whether it is speech, music, or some sort of noise. Simply defined, sound is a type of kinetic (involving motion) energy that is called acoustic energy. It is made up of pressure waves that vary in size, rate of motion, and complexity. These pressure waves decay, or diminish, at a predictable rate as they travel through the air. Now that we understand what sound is, we will quantify the physical properties of sound in simple terms that will be built upon as you progress through this handbook.

4. *Intensity:* This refers to the measure of the amplitude of acoustic energy passing through a given area. Our ears interpret intensity in terms of sound level, leading many to confuse it with volume, or loudness. However, volume is defined as a power level. For example, turning up the volume on an amplifier merely increases its power. Loudness is a purely subjective measure of sound that varies from person to person, depending on their hearing acuity, the amount of surrounding noise, and their preference. Intensity, however, can be precisely quantified, the most common way being to measure sound pressure level (SPL).

5. *Velocity:* Sound does not travel through the air instantaneously, but it does travel through the air at a fairly constant rate of motion. "Velocity" is a term that defines the motion of sound, allowing us to predict how long it will take for sound to get from point *A* to point *B*. The nominal velocity of sound is 1,130 feet per second (344 mps).

6. *Frequency:* What we interpret as sound is acoustic energy consisting of fluctuating waves of pressure traveling through the air. The rate at which a wave fluctuates each second is called frequency. In order for the sound to be audible to humans, the waves of pressure in the air must fluctuate at a rate between 20 and 20,000 cycles per second, commonly known as hertz (Hz).

7. *Wavelength:* A wavelength is the physical distance between two successive peaks of pressure of a sound wave. The wavelength varies with frequency. The lower the frequency, the longer the wavelength and the lower the tone will sound to us. Conversely, the higher the frequency, the shorter the wavelength and the higher the tone will sound to us.

8. *Phase:* Phase is the time relationship between two or more sound waves. If two identical waves are in phase in every respect, they combine to produce twice the level of sound. However, if two identical waves are perfectly out of phase, they completely cancel each other out. When two identical waves are partially out of phase, either a cancellation or a reinforcement takes place and the resultant wave is somewhere between zero and twice the original level.

9. *Inverse Square Law:* The inverse square law defines the relationship between sound pressure level and distance. For example, as we walk away from a person who is speaking, the sound of his voice continues to decrease in intensity until it eventually disappears into the surrounding noise or drops below the threshold of hearing. This law enables us to predict how much the sound level will drop, or decrease, based on distance.

10. *Reverberation:* Reverberation is the buildup of acoustic energy within an enclosed room. Ideally, the majority of the energy from a loudspeaker should be absorbed by the audience or other materials in the room. Realistically, though, much of the energy from the loudspeakers is reflected back into the room. These sound waves ricochet from surface to surface, losing energy with each reflection, until the energy finally decays, or diminishes. Each loudspeaker or cluster of loudspeakers contributes to the buildup of the reverberation.

11. *Critical Distance:* The critical distance is the point where the direct sound from the loudspeaker and the reverberant sound are of equal sound pressure. As a listener moves away from the loudspeaker and further into the reverberant field, he will notice a gradual loss of articulation. As he moves past the critical distance, the reverberant sound is louder than the direct sound and the program will gradually become less intelligible.

12. A basic understanding of what sound is and the terms that define it will aid us in selecting a successful sound system, which will be discussed in Chapter 3.

Chapter Three

Selecting a Successful Sound System

1. The process of selecting a successful sound system requires taking various factors into consideration. As we discussed in Chapter 1, quality sound involves having (1) adequate sound level, (2) intelligibility, and (3) naturalness. By attaining these three key goals, our brothers will be able to derive the greatest spiritual benefit from the convention program. In this section we will identify the basic elements that make up a sound system and discuss how to test and select the individual components that will allow us to deliver high-quality sound.

Basic Elements of a Sound System

2. Understanding and evaluating a sound-reinforcement system does not need to be complicated. The sound system consists of the following four basic elements: (1) the source equipment, (2) the front-end equipment, (3) the amplification equipment, and (4) the loudspeakers. (See Figure 3.1.) The type and complexity of equipment within each element can vary greatly, depending on locally available equipment and the acoustic conditions of the convention site.

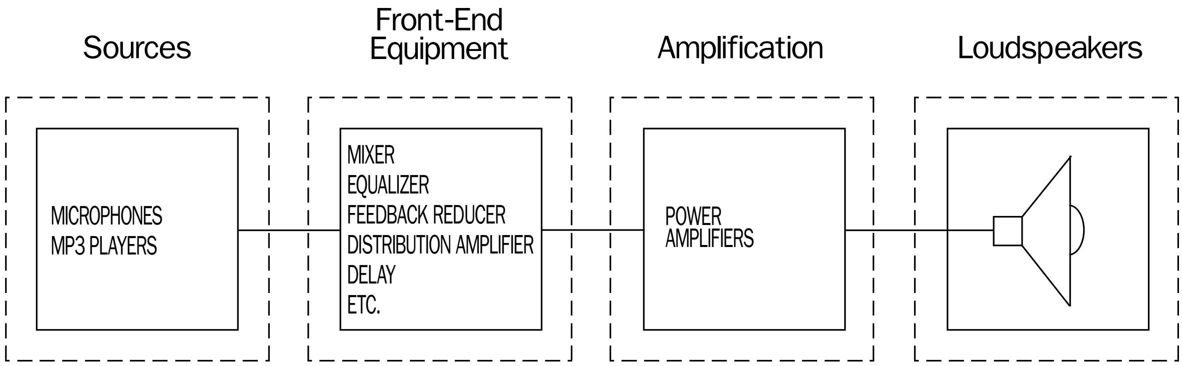


Figure 3.1 Elements of a Sound System

Selecting a Sound System

3. Many modern convention facilities are equipped with a permanent sound system. Some are well engineered, providing sound that is adequate both in quality and in level. Naturally, where arrangements can be made to utilize such a sound system, much time and effort can be saved. However, the majority of these existing sound systems are not suitable for our Christian conventions, where the audience will be listening to the spoken word for sustained periods of time. When evaluating a convention facility, we must first determine if the facility already has a satisfactory sound system. If so, we must decide whether to use all or part of it or to design and install a complete system. This decision basically comes down to four options:

(1) ***Use the existing house system.*** If we are using a convention site, such as an Assembly Hall, that has a sound system that is complete from microphone to loudspeakers and produces good-quality sound, much of our work is done for us. However, we must be thoroughly familiar with how to operate the sound system.—See Chapter 8.

(2) ***Use the existing house system, but supplement areas that have weak speaker coverage.*** Sometimes there are gaps in the sound coverage in the corners of an arena where the direct sound from perpendicular speaker clusters meets. We can save considerable time and effort by supplementing the house speakers with additional loudspeakers only in such areas.

(3) ***Use part of the existing house system.*** If the house system has good speaker coverage but the processing equipment is not adequate, perhaps because of poor maintenance or setup, we can provide the front-end equipment only and still save considerable time and effort.

(4) ***Provide a complete stand-alone sound system.*** For further information, see Chapter 4.

If we choose either option 2 or option 3, we will need to evaluate the existing house system to decide which parts we can use.

Evaluating an Existing Sound-Reinforcement System

4. Many months prior to a convention, it is necessary to evaluate the convention site to see what is required to provide good sound for our brothers. (Luke 14:28-31) When evaluating a site that has an existing sound system, it is best to do so under the actual operating conditions that we would expect during our convention. For example, the house mechanical equipment used to provide air-conditioning or heating can greatly increase ambient noise, which can be difficult to overcome.

5. When evaluating a house system, the following procedures and tools can be used to help you determine what is necessary to provide high-quality sound so that all in attendance will easily hear the spiritual program. The points outlined below are intended as guidelines, not as rules. We recognize that conditions, available test equipment, and the skill level of volunteers may vary in your area.

6. *Tools:* You will need a good-quality cardioid microphone, microphone cables, an audio mixer, various patch cables to tie into the house system, and suitable reading material. We also recommend having a multimeter, a sound pressure level (SPL) meter, a spectrum analyzer, and an audio signal generator. If an audio signal generator is not available, an MP3 player and audio test signals can be used. The available audio test signals should include pink noise, a one-kilohertz tone, and a click track.—See Appendix 6.

7. *Sound System Evaluation Test:* First, get a quick overview of the facility. Is the room reverberant? A sharp handclap will often reveal significant reverberation or echoes. Are there large reflective surfaces, such as glass, concrete walls, or unused sections? Are there curtains that can be hung behind a stage or in front of such surfaces to reduce reverberation?

8. Next, try to discern how much ambient noise you will need to overcome during the program. Check to see if the ventilation or air-conditioning is on. If not, can it be turned on? Also, make sure all the lights that will be used during the program are on, since some lighting systems generate considerable buzzing and vibration. How much ambient noise do you hear?

9. Locate the house sound equipment—the audio mixer, loudspeakers, microphone and line-level patch panels to connect to the house system, and so on. If an on-site sound technician is available, perhaps he can give you his observations about the system and show you where the sound equipment is located, as well as be on hand for your evaluation of the sound system.

10. Start with the house mixer. Can it be used? How old is it? Is it simple enough for the brothers to use? Has it been well maintained? Check the control knobs or sliders. Do they operate smoothly, or are they noisy? Is the mixer in the direct field of hearing, and does the operator have a clear view of the stage? It may be determined that the house mixer can be utilized if it is simple and easy to use, of high quality, and well maintained and if the operator is in the direct field of hearing and has a clear view of the stage. If the house mixer is going to be used, connect the MP3 player, audio generator, and microphone to continue the test.

11. If the house mixer is not going to be used, locate the house patch panel. Using the audio mixer you brought, connect to the house system via the patch panel. Also, connect the MP3 player, audio generator, and microphone to the mixer. The microphone should be set up so that a brother can stand where the stage will be located.

12. To proceed with the test, ask various ones to read aloud over the system at a comfortable level and consistent pace. The use of live readers is better than prerecorded material because it allows you to check if the system tends to produce feedback. During the test, it is good to use more than one reader, since a variety of voices will give a more realistic picture of the sound system's performance. Although we are interested in music quality, our primary concern is clarity of speech. Therefore, we cannot make a judgment by merely listening to a music program. In a reverberant room, music may sound fine over a system that is unacceptable for speech.

13. Walk through the entire seating area, *sitting down* and listening carefully in different locations. Stay seated long enough in each section to analyze the quality of the sound thoroughly and make notes on it. Is the sound clear and articulate throughout the seating area? If so, intelligibility will most likely be satisfactory with an audience present. Sound absorption from the audience will typically help with sound quality, but it will not correct serious acoustic difficulties such as very long reverberation time, strange echoes, or exaggerated high or low frequencies.

14. Take note of any conditions that could affect the quality of the sound. Are there interfering noises such as low-flying aircraft, heavy vehicular traffic, or passing trains? If so, can the house system overcome the interference? Would our own system likely do better? Note any areas where the sound is low or distorted. This could indicate inoperative or faulty house amplifiers or loudspeakers. Also, note areas where ambient noise seems excessively loud, perhaps at the top of seating areas close to air handlers. If an SPL meter is available, record the ambient noise levels in these locations in case we have to design and provide a complete sound system.

15. If the house system provides an adequate sound level throughout the seating areas, check how much level you can get before feedback. Have the brother reading gradually increase his distance from the microphone while you raise the gain to maintain a good listening level in the audience. What is the greatest possible working distance from the microphone while still maintaining an acceptable level without ringing or feedback?

16. If you must be very close to the microphone to be heard without feedback, it would be good to investigate the reason for this. Consider the following questions: Is the system correctly equalized? Improper equalization—or the absence of equalization—can reduce working distance significantly. If the system is not properly equalized, can it be improved? Can the house system be equalized, using either their equipment or ours? Play pink noise over the system and use a spectrum analyzer to see how the system is equalized. Look especially for frequencies that are substantially higher than their adjacent frequencies. This, along with what you have heard during your test, can help you determine if you can use the house system.

17. After checking the house system in this way, it may be determined that the system cannot provide adequate sound level, intelligibility, and naturalness. Perhaps the house processing equipment could be bypassed and our front-end equipment patched into their amplifiers and loudspeakers. If so, oversight may decide to provide the processing equipment and patch into the house amplifiers and loudspeakers to save time and expense. If none of this can be done, you will have to design and install a complete sound system. In the next chapter, we will discuss how to design such a system.

Chapter Four

Designing a System for Clarity and Coverage

1. Many factors must be taken into consideration when designing a quality convention sound system. Design options are largely dependent on available sound equipment and the convention site. With this in mind, the following examples and principles can serve as a guide when designing a sound system as well as help you understand what has already been designed or correct problems related to design. The sound systems often found in the facilities that we use are not designed for audiences that must pay close attention to the spoken word for many hours, so at least some audio design on our part is typically required. In this chapter, we will go through the step-by-step process of designing a sound system for an indoor venue, although these same steps and principles would also apply to an outdoor facility. The bold subheadings will isolate the specific steps and provide other important information.

Gather Site Information

2. We must remember that our design and calculations will only be as good and as accurate as the dimensions that we use to design. Therefore, once the facility that will be used for our convention has been determined, it is important to gather as much information about the site as possible. The convention representative should contact the building management to inquire if they have detailed prints of the facility. This is a common request, and the building management will generally be very helpful in providing electronic or hard-copy scale drawings. Although a seating chart may give a good overview of the entire facility and prove useful in planning where departments may be located, scale drawings of the floor and section plans should be obtained, preferably in an electronic format such as *AutoCAD*. When this is not possible, the following procedure will help you to obtain appropriate measurements and dimensions.

3. First, we need to consider the *section* view, as noted in Figure 4.1. This view allows us to look at the seating area from the side. Both the horizontal and vertical dimensions are necessary in order to establish the correct angle of inclination of the seating area. The section view is useful in determining how to aim the loudspeakers so that as much energy as possible is focused on the audience. This view also enables us to determine how far we must project the sound to reach the most distant seats.

4. If we are unable to measure long dimensions or if we are finding it difficult to get accurate dimensions, we can use the alternative method shown in Figure 4.2 to draft a section view. This method allows us to obtain the complete vertical and horizontal dimensions by building upon small measurements.

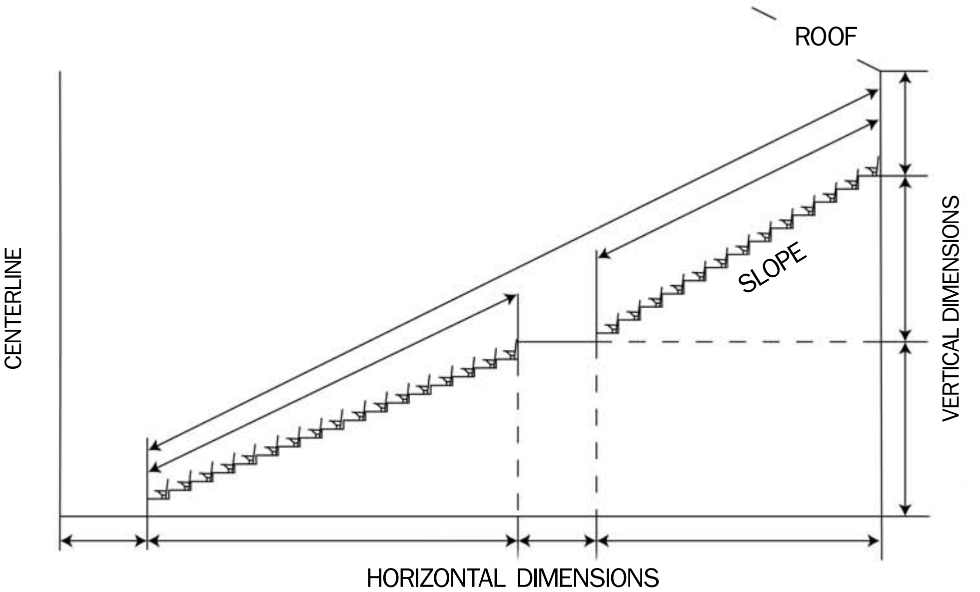
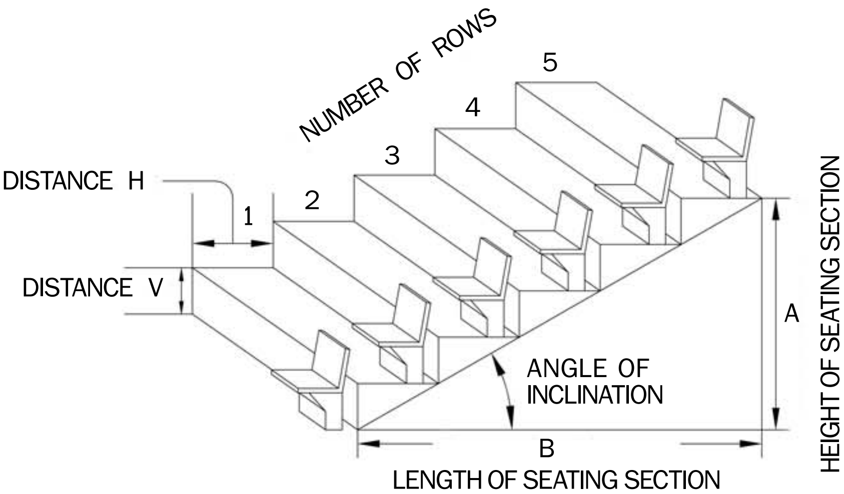


Figure 4.1 Typical Section View



CALCULATING LENGTH AND HEIGHT OF EACH SEATING SECTION:

1. MEASURE THE HORIZONTAL DISTANCE (H) OF A NUMBER OF ROWS IN THE SECTION TO CONFIRM THAT THEY ARE EQUAL.
2. MEASURE THE VERTICAL DISTANCE (V) OF A NUMBER OF ROWS IN THE SECTION TO CONFIRM THAT THEY ARE EQUAL.
3. COUNT THE NUMBER OF ROWS IN THE SECTION.
4. LENGTH OF SEATING SECTION = DISTANCE H · NUMBER OF ROWS.
5. HEIGHT OF SEATING SECTION = DISTANCE V · NUMBER OF ROWS.

OR

1. SLOPE = DISTANCE A/DISTANCE B
2. HEIGHT OF SEATING SECTION = SLOPE · DISTANCE C, E, H, OR J (IF KNOWN FROM FLOOR PLAN VIEW—FIGURE 4.3)
3. ANGLE OF INCLINATION = INVERSE TANGENT (TAN^{-1}) OF $\left(\frac{\text{DISTANCE A}}{\text{DISTANCE B}}\right)$

Figure 4.2 Alternative Method for Determining Dimensions

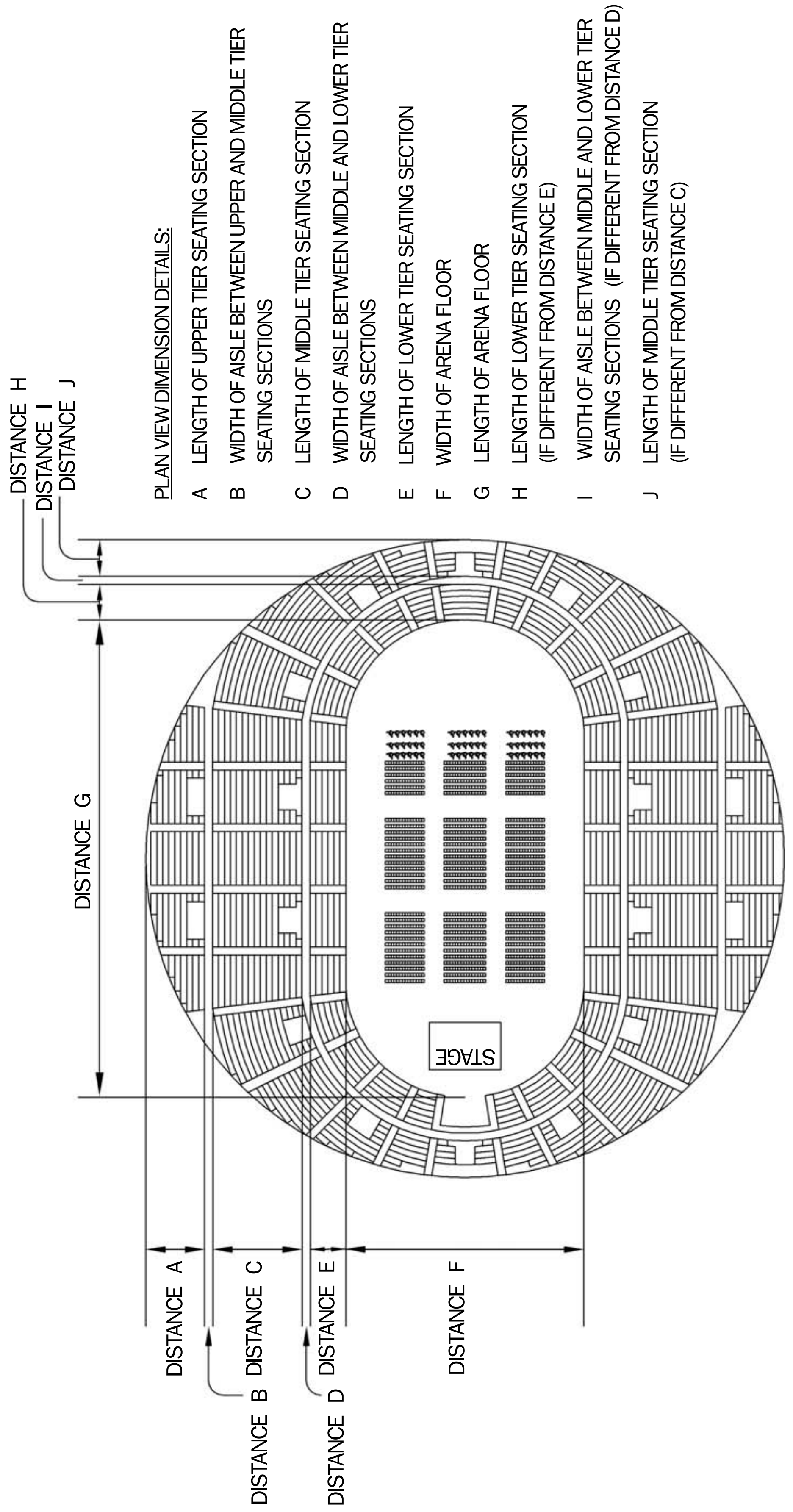


Figure 4.3 Floor Plan View

5. Next, we will consider the *floor plan* view, as shown in Figure 4.3. The floor plan will help us to lay out the speakers in a uniform manner and establish our speaker zones. This view, looking down on the floor, can prove very helpful in noting areas of concern, such as areas where high ambient noise might be a problem.

Determine Stage Location and Construction

6. Before designing the sound system, the location of the stage must be determined. The stage should be located so that program participants have a clear view of the majority of the audience, thus enabling the participants to have good audience contact. If possible, avoid locating the stage where those in attendance may be unduly distracted by pedestrian traffic, such as directly in front of restroom access. Since the stage is where the microphones are used, both the design and the location of the stage are very important. It should not be located in the direct coverage of the main loudspeakers, where it will receive a lot of sound energy, as this will make it difficult to control ringing and feedback. In addition, the stage should be designed and constructed in such a way that reflections of sound toward the microphones are minimized. For example, a concave wall at the back of the stage should be avoided because its shape would function as a parabolic reflector, reflecting sound and ambient noise to the center of the stage, raising noise levels, and narrowing our feedback margin. Irregular surfaces, however, are acoustically advantageous, since reflected sound is dispersed in various directions rather than focused on an area where it is not desired. Stage surfaces, such as the backdrop, should be constructed with either absorptive or acoustically transparent materials. Carpeting the stage floor will not only assist in reducing reflected sound but also serve to muffle footsteps and other noises. Unquestionably, the location, design, and construction of the stage, as well as the materials used, will greatly influence the overall acoustic quality and performance of the sound system.

Determine Seating Layout

7. Every effort should be made to plan a general seating layout that will minimize distractions and enable our brothers to focus on the spiritual food being presented. As much as possible, our design should allow for a clear view of the stage. Additionally, safety must be kept in mind at all times. Aisles need to be wide enough to allow for easy egress in case of an emergency, such as a fire. We should also be sure to cooperate with any officials, such as a fire marshal, or agencies that regulate and govern large public gatherings in the local area. (Rom. 13:1) Consideration should be given to the elderly and disabled, so that they are able to have seats reserved for them that provide adequate sound and easy access to restroom facilities. —Lev. 19:32.

Determine Mixer Location

8. The location of the mixer is critical, since it has a direct bearing on the overall quality of the program. Therefore, the mixer should be located where the operator will have a clear, close view of the stage, allowing him to see easily which microphones are being used, but it should not be located at the edge of the stage. A location within the direct coverage of the main loudspeaker system will allow the operator to hear the program at approximately the same sound pressure level (SPL) as the majority of the audience. This location should have adequate power, preferably with dedicated AC circuits. If amplifiers are used, they should be installed in the same area as the mixer so that they can be easily monitored and adjusted if needed. It is best if the amplifiers also have their own electrical circuit. Consideration should also be given to the distance needed to route and install the microphone snake. If the distance is excessively long, our system will be more susceptible to noise and radio-frequency interference.

Loudspeaker Selection

9. "Loudspeaker" is a generic term that refers to a wide variety of transducers that convert electrical energy into acoustic energy, or sound. In a sound-reinforcement system, loudspeakers play a very important role, as they are the final link between the sound source and the brothers and sisters listening to the program. In our discussion, basic information will be provided to help you understand the characteristics of a loudspeaker so that you can make a proper selection and have natural-sounding, uniform coverage reaching the most distant seats. The main attributes that are normally given in loudspeaker specifications are *frequency response, power handling, sensitivity, impedance,* and *directional characteristics.*

10. *Frequency Response:* This specification tells us the range of frequencies that the loudspeaker is designed to reproduce accurately within a given tolerance. Ideally, we desire a loudspeaker that can accurately reproduce all frequencies within the audible spectrum of sound.

11. *Power Handling:* This specifies the maximum amount of power the loudspeaker can safely handle.

12. *Sensitivity:* This specification tells us the SPL produced by one watt of power applied to the loudspeaker input and measured at a distance of one meter* directly in front of the loudspeaker. For example, if the sensitivity of a loudspeaker is listed as 98 decibels (dB), we know that one watt feeding into the loudspeaker's input will

* Most manufacturers reference sensitivity to one meter. However, from here on we will reference U.S. measurements first, followed by the metric equivalent.

result in 98 dB-SPL at 3.28 feet (1 m) in front of the loudspeaker. A highly sensitive loudspeaker is desired because it requires less power from the amplifier to produce more sound. An increase of 3 dB in sensitivity creates an increase in SPL in front of the loudspeaker that is effectively the same as *doubling* the amplifier power.

13. *Impedance:* This is the total opposition, or resistance, to the flow of alternating current in an electric circuit. The maximum power is transferred when the load (loudspeaker) impedance matches the source (amplifier) impedance. For example, if a power amplifier is rated at 100 watts into an 8-ohm load, an 8-ohm loudspeaker rated at 100 watts has the potential to transfer the full 100 watts from the amplifier.

14. *Directional Characteristics:* The most common method used to specify a loudspeaker's directional characteristics is by directly stating its horizontal and vertical dispersions, which are usually listed separately in degrees. These figures normally represent the points at which we will lose approximately 6 dB compared to the level found directly in front of the loudspeaker.

15. In order to specify a loudspeaker's directional characteristics more fully, polar plots are sometimes provided. Polar plots can specify exactly how much loss we can expect at any point around a speaker, both horizontally and vertically. Because a loudspeaker's dispersion will change significantly depending upon the frequency it is reproducing, manufacturers will commonly offer polar plots at several different frequencies.

16. In Figure 4.4, sample polar plots are provided for a full-range loudspeaker that consists of a high-frequency horn and a 12-inch (300 mm) woofer. Each polar plot specifies the horizontal and vertical dispersion at a particular frequency. A solid line represents the horizontal dispersion, while a dashed line represents the vertical dispersion. Each plot is drawn with 5 dB divisions.

17. First, look at the plot for 50 hertz (Hz). You will note that the energy radiates from the loudspeaker in an omnidirectional manner, which is typical of almost every loudspeaker that produces lower frequencies. If you look at both the horizontal and the vertical dispersions on the polar plot, you can see that almost as much energy is radiating behind the loudspeaker as is being projected from the front, with only about 5 dB difference. As you study the polar plots, note that the higher the frequency, the more directional the loudspeaker becomes. Look at the 2 kilohertz (kHz) polar plot and note the 6 dB down points. We have indicated the 6 dB down points on this polar plot in red. They are between 32 and 35 degrees, making a total dispersion of between 64 and 70 degrees.

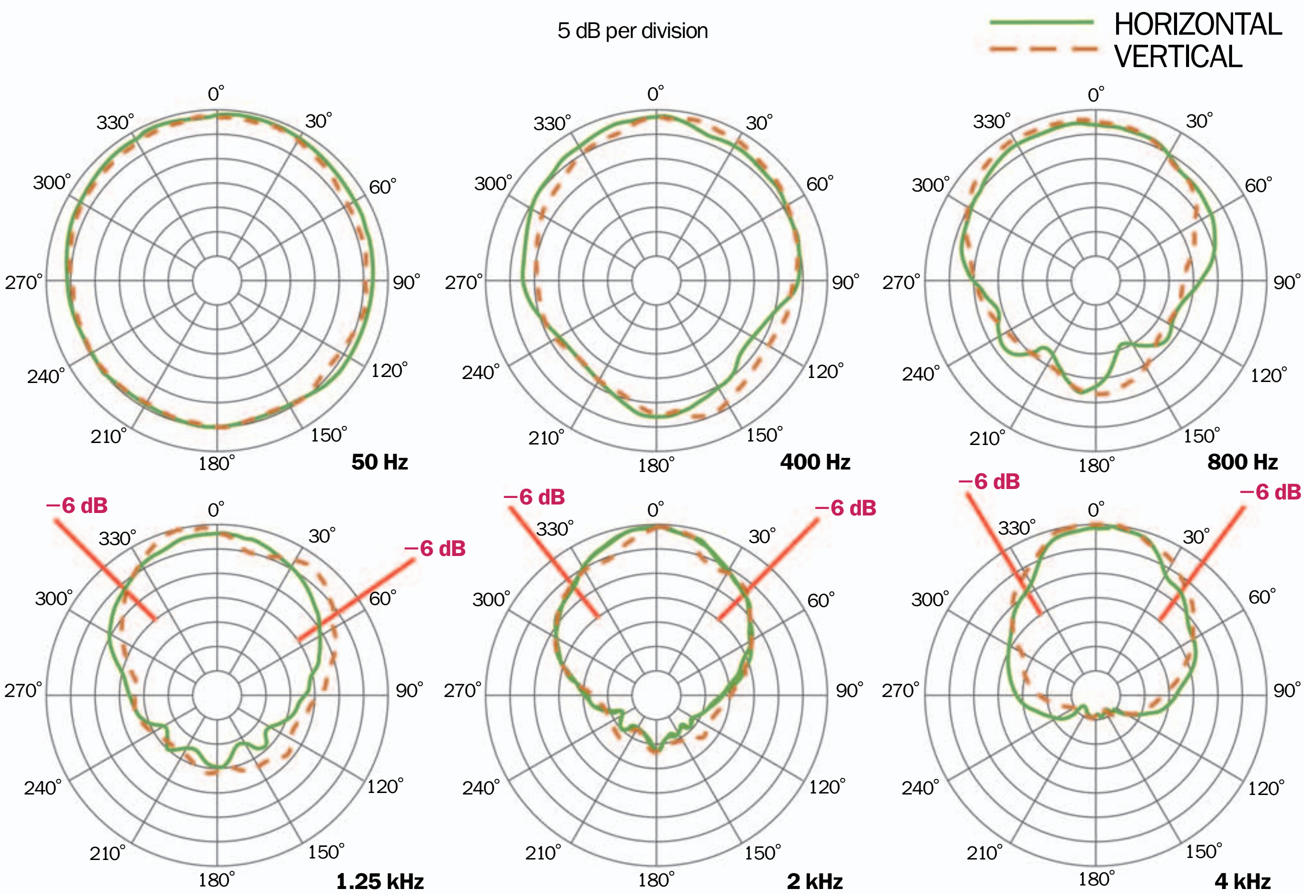


Figure 4.4 Example of Typical Polar Plots

Also, note that there is far less energy—nearly 25 dB less—radiating behind the loudspeaker at this frequency.

18. If the specification gives an average dispersion and does not provide frequency-specific polar plots, the dispersion angles represent an average that applies only to the midrange and high-range frequencies. This is because all loudspeakers tend to become omnidirectional at lower frequencies.

19. A simple guideline to help determine if the loudspeaker being used is capable of projecting sound the desired distance is to remember that the narrower the dispersion, the greater the distance the loudspeaker can extend the sound coverage. Conversely, the wider the dispersion, the shorter the distance the loudspeaker can extend the sound coverage.

20. It is helpful to know that the size of a speaker cone also has a bearing on its dispersion characteristics, since the directionality of a speaker is dependent on the relationship between the size of the speaker and the wavelength of the frequency that it is reproducing. As an example, a 12-inch (300 mm) loudspeaker has a narrower dispersion than a 4-inch (100 mm) loudspeaker, especially in the primary speech ranges. Similarly, the longer the horn's waveguide, the more capable it will be in controlling lower frequencies. This knowledge proves helpful when trying to project sound a greater distance.

Types of Loudspeakers

21. *Speaker Columns:* These are constructed with multiple cone speakers designed to cover most of the audio spectrum; usually, there are three or four speakers stacked vertically. Stacking the cone speakers vertically narrows the directional characteristics of the speaker vertically but does not help horizontally. This can be seen in the dispersion characteristics of many column-type loudspeakers. For an example of how to build a speaker column, see Appendix 7, Figure A7.17. This 8-inch (200 mm) speaker column's horizontal dispersion between 2000 and 4000 Hz is 90 degrees, and the vertical dispersion is 60 degrees. Since these 8-inch speaker columns are not highly directional, they are usually used only in a low-level distributed system where they would only have to project 40 to 50 feet (12.19 to 15.24 m). It should be noted that most column-type loudspeakers tend to project a strong lobe of sound directly behind them. Therefore, this is another reason they should only be used in low-level distribution systems.

22. *Full-Range Loudspeakers:* These loudspeakers are usually constructed with a small-format horn and a cone speaker that combine to cover most of the audio frequency range. The small-format horn handles the mid-high frequencies, and the cone speaker handles the low frequencies. Most full-range loudspeakers have a built-in crossover that sends only mid-high frequencies to the horn and lower frequencies to the cone speaker. Some full-range loudspeakers require a biamplified configuration, with the low-frequency transducer(s) and the high-frequency transducer(s) each having its own amplifier feed that can be processed individually. Typically, full-range loudspeakers provide useful directional characteristics because of the dispersion-controlling horn. The speaker will be most suitable for convention use if the crossover frequency is below 2 kHz, so that the horn is handling the articulation range of speech.—See Appendix 5:14.

23. *Horns:* These are very directional and come in small, medium, and large formats. A medium-format horn, such as one with a 40-degree horizontal and 20-degree vertical dispersion, is often used for larger conventions. These highly directional horns are capable of projecting the sound 200 feet (61 m) or more and work especially well for outdoor applications.

24. As a word of caution, these horns (high-fidelity, not paging horns) are designed to reproduce sound in the mid-high frequencies. The compression driver that is used with this type of horn is very sensitive and will be damaged if fed low-frequency signals. Using such a horn requires that another loudspeaker, such as a woofer, be used in conjunction with it. A crossover should be used to send mid-high frequencies to the horn and low frequencies to the woofer.

25. *Reentrant Horns:* Sometimes referred to as paging horns, these are constructed so that the driver discharges energy into the horn in such a way that it doubles back, or is folded, thereby enabling the horn to produce and control lower frequencies despite its relatively small size. They are constructed in a robust manner that makes them ideal for outdoor use. With proper equalization, they can produce intelligible and natural sound.

26. *Low-Frequency Speakers:* For our convention purposes, low-frequency speakers are typically used with a biamplified system to reproduce the low frequencies lacking in a high-frequency horn. A low-frequency speaker generally has a larger cone area to reproduce low frequencies efficiently.

27. *Line Arrays:* For convention purposes, line arrays are not typically used because of cost and rigging issues. Line arrays are made up of a number of loudspeakers coupled together in a line, thus narrowing the vertical dispersion, and because of phase coupling, they may lose only 3 dB per doubling of distance.

Loudspeaker Placement

28. When designing a sound system, we must remember that it is impossible to give everyone in the audience the exact same sound level and coverage or to have all the energy focused only on the audience so as to avoid all reflective surfaces. This is because each seat varies in its distance from and angle to the loudspeaker, noise sources such as air handlers, and reflective surfaces. Since this is the case, we must decide on the best *overall compromises* within various parameters. For example, we realize that the sound level at the seats nearest the loudspeakers will naturally be louder than at the most distant seats. The compromise is making sure that the sound level for the nearest seats is not too loud and that the level for the most distant seats is not too low. If we meet those requirements, we know that the sound level for the seats that lie in between will generally be acceptable.

29. Acoustics are often very difficult to contend with in an indoor facility, since such facilities are usually constructed with hard reflective surfaces, such as concrete, steel, and glass, and have very few absorptive surfaces. Thus, a simple principle can guide us through the design process for *any venue* and should always be kept in mind when placing speakers: **We should always endeavor to focus as much sound energy as possible on people and as little as possible on reflective surfaces.**

30. Before placing the first speaker, we must have an idea of how much SPL is needed in the most distant row of seats. To establish the SPL, we need to give attention to the following three pieces of information:

(1) The average person's speaking voice is about 65 dB-SPL. This level gives us a starting point, since most listeners consider this a comfortable level in a quiet environment. Logically, this is the desired minimum SPL only in a *quiet environment;* however, the sound level at our conventions is usually between 70 and 75 dB-SPL or a little higher, depending on the amount of ambient noise.

(2) We need to know the ambient noise level for the area being covered. If possible, the program audio should be at least 15 dB greater than the ambient noise in order to provide adequate volume and *intelligible* sound. We must also consider things that might create increased noise during a program, such as large amounts of pedestrian traffic in a main aisle.

(3) The SPL should not be above 85 dB for any sustained period of time, as this can damage hearing. Most in the audience will start feeling uncomfortable when the SPL approaches 80-82 dB-SPL.

With these three factors in mind, we are able to determine that our SPL will need to range between 65 and 85 dB-SPL, depending on the ambient noise.

Calculating SPL Loss

31. Because sound dissipates at a rate that we can calculate, we can determine how much SPL will be lost at any given distance. Keeping this factor in mind, a few simple calculations will give us a very good idea of whether a particular speaker layout will adequately cover our audience.

32. When dealing with loudspeaker placement, there are two types of loss that must always be accounted for: (1) distance loss and (2) off-axis loss. In the next few paragraphs, we will define exactly what these are and how to calculate them.

33. *Distance Loss:* If we were to stand in front of a loudspeaker and slowly begin walking directly away from it, we would perceive that the farther away we walk, the more the sound level diminishes. This decrease is due to distance loss, and since sound dissipates at a consistent rate, it can be precisely calculated. Figure 4.5 illustrates how sound diminishes with distance. In this illustration, the person is always directly in front of the loudspeaker so that he is in line with the greatest amount of energy discharged by the speaker. We refer to this as being on-axis with the loudspeaker. You will note that a person standing 100 feet (30.48 m) away from the loudspeaker in this example is able to hear 76 dB-SPL. As the distance is increased to 200 feet (60.96 m)—a doubling of the distance—the sound will diminish by 6 dB-SPL and our listener will be able to hear 70 dB-SPL. At 400 feet (121.92 m) away, doubling the distance yet again, the sound will diminish by another 6 dB-SPL and the person will hear about 64 dB-SPL. Even though in one instance the change in distance was 100 feet and the other was 200 feet, the sound level decreased by 6 dB-SPL in both cases. This is because sound loss follows a logarithmic curve rather than a linear one.

34. There are two ways to determine distance loss: (1) Calculate it mathematically using inverse square law (dB-SPL = 20 log D_f/D_n) or (2) use a nomogram to determine inverse square law. For a detailed discussion of these two methods for calculating distance loss, see Appendix 1, paragraphs 57-60.

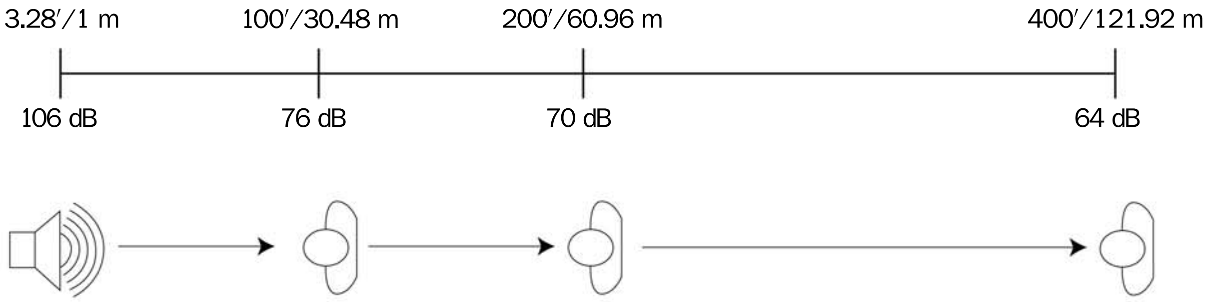


Figure 4.5 Distance Loss

35. *Off-Axis Loss:* If we were to stand directly in front of the loudspeaker and slowly begin walking off to the side so that the loudspeaker was no longer directly facing us, our level of perceived sound would similarly diminish. This would be true even if we took care to walk in such a way that we always maintained the same distance from the loudspeaker. This decrease in level is not due to distance but due to being off-axis from the loudspeaker. The farther the speaker is from facing us, the more it can be said that we are off-axis. To define how far a point is from being on-axis, we can measure in degrees, with 0 degrees describing a point directly in front of a loudspeaker and 180 degrees being a point directly behind. It should be noted that a point can be considered off-axis not only horizontally but also vertically. As previously discussed in paragraphs 14-20 concerning the directional characteristics of a loudspeaker, polar plots help us to specify how directional a loudspeaker is by identifying how many degrees off-axis the 6 dB down points are. Polar plots also allow us to quantify the off-axis loss for *any* angle at a given frequency. As an example, we have taken the 1.25 kHz polar plot in Figure 4.4 and illustrated its off-axis loss in Figure 4.6.

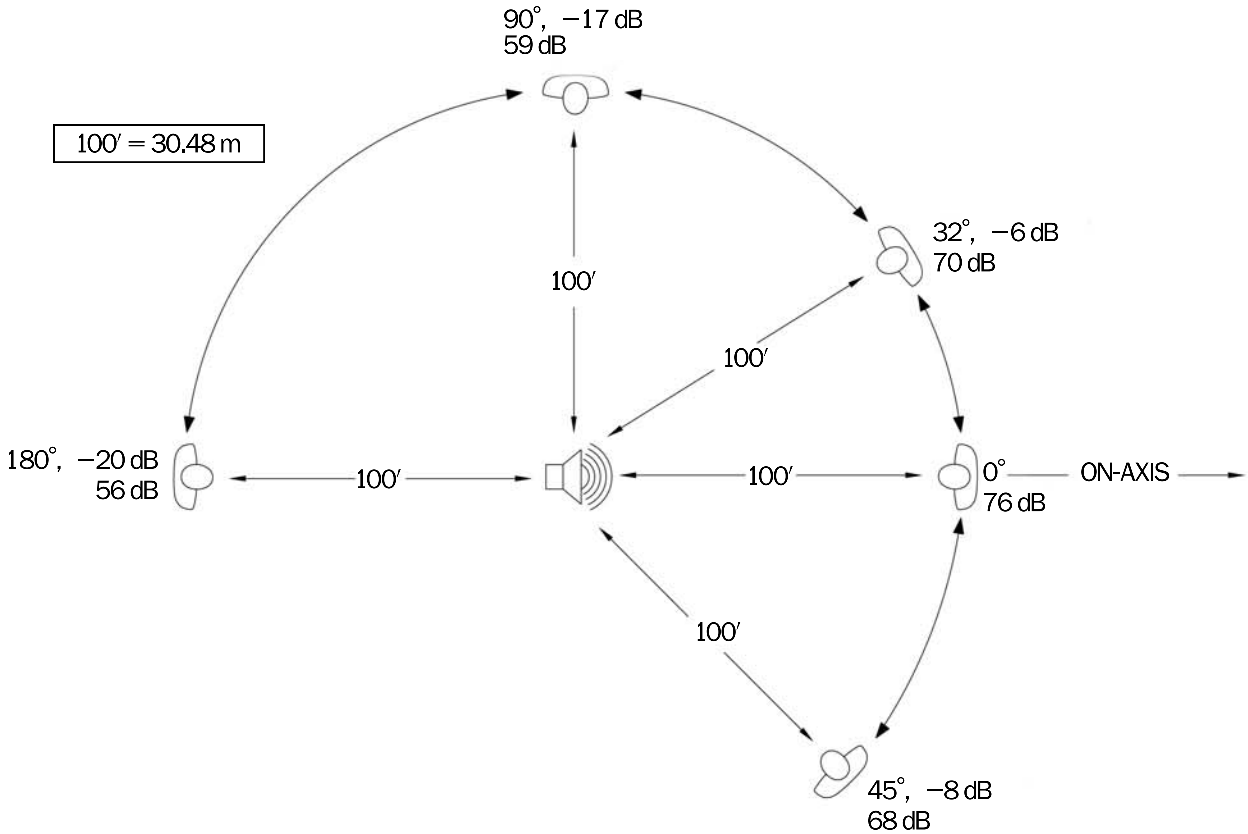


Figure 4.6 Off-Axis Loss at 1.25 kHz

Timing Between Loudspeakers

36. It is a well-known fact that sound does not travel through the air instantaneously but, rather, at a fairly constant speed, or *velocity.* This velocity is defined as approximately 1,130 feet (344 m) per second. Although the velocity, or speed, of sound varies slightly with changes in temperature, humidity, and atmospheric pressure, the figures given here for a temperature of 70°F. (21°C.) are adequate for our purposes. Here are simple equations for determining the delay for any given distance:

Delay (seconds) = Distance (feet) ÷ 1,130 or Distance (meters) ÷ 344

If we prefer to calculate delay time in *milliseconds* (ms), the following equations are useful:

Delay (milliseconds) = Distance (feet) • .885 or Distance (meters) • 2.9

37. In most simple, well-designed installations, this delay may be of little concern. Even when seated at a considerable distance, listeners will be content as long as sound quality is adequate and intelligible. The situation becomes quite different where the dispersion patterns of two or more loudspeakers overlap. If the arrival of the more distant sound is sufficiently late, listeners will hear two distinct signals, severely degrading the intelligibility. Similarly, a strong echo in certain areas of seating will have the same negative effect on clear articulation. Acoustic delay is a fact of life, an important factor that must be taken into consideration when designing a practical layout for the loudspeaker system.

38. While the loudspeaker layout should avoid large differentials between arrival times at any one listening position, it is good to know that the marvelous ear-brain faculty of each listener works to our advantage. A coherent sound arriving up to 35 milliseconds later than the original sound is still accepted by the auditory cortex of our brains as a single sound. This suggests that loudspeakers can be separated by up to 40 feet (12.19 m) before sound delay becomes a serious threat to intelligibility. Differentials in arrival times greater than 35 milliseconds may call for some type of signal delay.

39. Also, we must be aware that the amount of interference varies significantly with the difference in level between direct and delayed sounds. As the sound level of the delayed signal decreases, it gradually becomes imperceptible. It is generally agreed that a delayed signal that is 25 dB-SPL or more below the main signal will have a limited effect on intelligibility.

Constant-Voltage Systems

40. With a constant-voltage system, the output voltage remains the same regardless of the amplifier's power rating or the load it is feeding, making it possible to add or remove any number of loudspeakers. Constant-voltage systems can be designed for 25, 70, 100, 140, or 200 volts (V), depending on the country where the system will be used. Constant voltage does not mean that the voltage designated is always present. The system voltage is constant in that it does not vary as a function of the number of loudspeakers driven. With a constant-voltage system, the audio transformers feeding the loudspeakers are wired in parallel across the distribution line (see Figure 4.7) and provide a relatively high-impedance circuit to the amplifier.

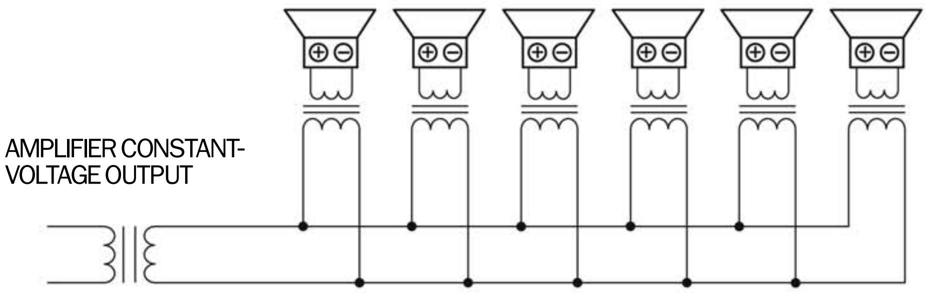


Figure 4.7 Typical Constant-Voltage Circuit

41. Each transformer has various taps (labeled in watts) that allow us to vary the power delivered to each loudspeaker, giving us the flexibility to increase the SPL for small areas if necessary. To calculate how much power is required in a constant-voltage system, we simply sum the tapped values of all the transformers in the circuit. For example, if each loudspeaker's transformer in Figure 4.7 is tapped at 30 watts, we know the amplifier must be rated for at least 180 watts (6 · 30 watts = 180 watts). There are some additional factors that will need consideration when determining power requirements in a constant-voltage system.—See paragraph 55.

42. A constant-voltage system is preferred for most convention loudspeaker systems because (1) it simplifies the design process and the power calculations, (2) it allows us to add or remove loudspeakers without having to redesign the system, (3) the failure of one loudspeaker does not affect the rest of the loudspeakers in the circuit, and (4) it is much simpler to troubleshoot.

Low-Impedance Systems

43. A low-impedance system allows the maximum transfer of power from the amplifier to the loudspeaker system without the use of transformers at every speaker. Many audio amplifiers are equipped with an internal transformer that provides an output impedance of 4, 8, and 16 ohms. Higher-powered amplifiers have a constant voltage output that can drive a variety of low-impedance loads.—See Appendix 5, paragraphs 25-31, for more details on amplifier outputs.

44. A low-impedance system can be wired in series, parallel, or a combination of the two. With a large low-impedance sound system, the series-parallel method must be used for wiring. Series-parallel allows for a number of loudspeakers to share the amplifier's power equally. For example, in Figure 4.8 you have eight 8-ohm loudspeakers that make up a circuit having a 16-ohm load. If hooked up to a 200-watt amplifier, each loudspeaker could receive up to 25 watts feeding it.

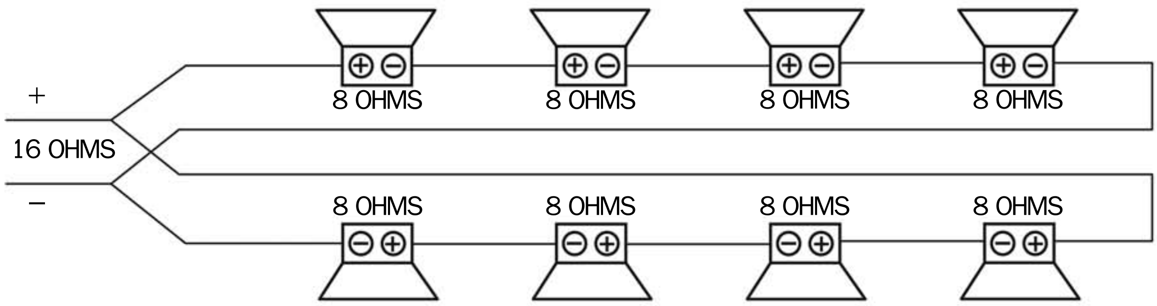


Figure 4.8 Typical Series-Parallel Circuit

45. The disadvantages of a series-parallel system are that (1) it is a more complicated system to design, install, and troubleshoot; (2) a loudspeaker cannot be added or removed from the circuit without redesigning the entire system; and (3) when one of the loudspeakers fails or a wire connection is severed (creating an open circuit), the series is broken and all the loudspeakers in that series will stop working. For more information on calculating series-parallel, see Appendix 1, paragraph 21.

Designing the Sound System

46. Thus far, we have reviewed how to gather information so that critical details for designing the sound system are available and how to decide on the stage location, seating layout, mixer location, and loudspeaker selection. Further, information has been provided to help you understand how to determine distance and off-axis loss, timing between speakers, and the choice between a constant-voltage or a low-impedance system. Now we are ready to design a sound system. The following steps will guide you through the design process.

47. (1) *Determine if you will design a constant-voltage* (*70- or 100-volt*) *or a low-impedance* (*4-, 8-, or 16-ohm*) *system.*

- Check the amplifiers that you have at your disposal. Do they have the power to drive the loudspeakers you are going to use? Do the amplifiers have a constant-voltage or a low-impedance output? The equipment that you have available will dictate, for the most part, the type of system you will design. If good transformers and high-output power amplifiers are easy to obtain at a reasonable cost, constant-voltage systems are our first choice.

48. (2) *Use the section plan to make an educated decision on the placement of the first loudspeaker.*

- The greater the distance that sound must be projected, the more energy you will need from the loudspeaker.
- The greater the loudspeaker's energy, the farther back and/or higher the loudspeaker will need to be placed from the nearest seats to ensure that the SPL is not too great for these listeners.
- Check the loudspeakers that can be used. Are they capable of covering the distances required? Projecting sound over long distances requires that loudspeakers have good power handling, high sensitivity, and relatively narrow directional characteristics. If you do not have such loudspeakers, you will be required to split the distance, using a low-level distributed system.
- Check the mounting towers that are available. Can they safely lift the loudspeaker to the desired height?

49. (3) *Calculate SPL loss at the farthest seat and determine how much energy is required at the loudspeaker.*

- First, draw the loudspeaker's average vertical dispersion angle (–6 dB down points) with the centerline being the axis used for aiming. If you are not able to do this with the aid of a computer, you can cut the dispersion angle out of clear Mylar or a similar transparent material and use it in conjunction with a scaled drawing.

- Aim the loudspeaker so that the centerline is pointing approximately three quarters of the way up the audience. Look at each –6 dB down point and try to make sure the seating areas will be somewhere between the two down points to get as much energy on the audience as possible.
- Use the inverse square law (dB-SPL = 20 log D_f/D_n) to calculate the distance loss to the farthest seat. Using the loudspeaker 1 kHz polar plot (or as close to that value as possible), determine the off-axis SPL loss. (In this instance, we will use the 1.25 kHz polar plot in Figure 4.4, which will be referenced throughout this exercise.) Add the distance loss and the off-axis loss together to get the *overall SPL loss*.
- Add the overall SPL loss to the desired audio program level. This will provide us with the amount of energy we will need 3.28 feet (1 m) in front of the loudspeaker. For example, if we desire the farthest seats to have 76 dB-SPL and our overall SPL loss is 26 dB, we simply calculate that 76 + 26 = 102. Therefore, we must have 102 dB at 3.28 feet in front of the loudspeaker to have 76 dB at the farthest seats.
- When determining the desired SPL for the farthest seats, we must remember that we desire the program audio to be 15 dB-SPL or more above ambient noise if possible. Realistically, there may be locations in the venue where this is impossible, such as seats directly under an air handler. Overcoming this problem will be considered in Chapter 7.

50. (4) *Calculate the SPL loss at the nearest seats.*

- Use the inverse square law (dB-SPL = 20 log D_f/D_n) to calculate your distance loss to the nearest seats. Using the loudspeaker 1 kHz polar plot (or as close to that value as possible), determine the off-axis SPL loss. Add the distance loss and the off-axis loss together to get the *overall SPL loss*.
- Subtract the overall SPL loss at the nearest seats from the SPL at the loudspeaker calculated in the previous step. For example, let us say that our overall SPL loss was 20 dB at the nearest seats. Since we need 102 dB-SPL to have 76 dB at our farthest seats, we would simply take 102 – 20 = 82. Therefore, we would have approximately 82 dB-SPL at the seats nearest the loudspeaker.
- Always double-check the SPL at low frequencies, since these radiate in an omnidirectional manner. Having less off-axis loss at these lower frequencies can result in having SPL that is too loud for the near seats. Keep in mind, however, that this can be managed to some extent with careful equalization.—See Chapter 7, paragraphs 27-50.

51. (5) *Balance levels between far and near seats.*

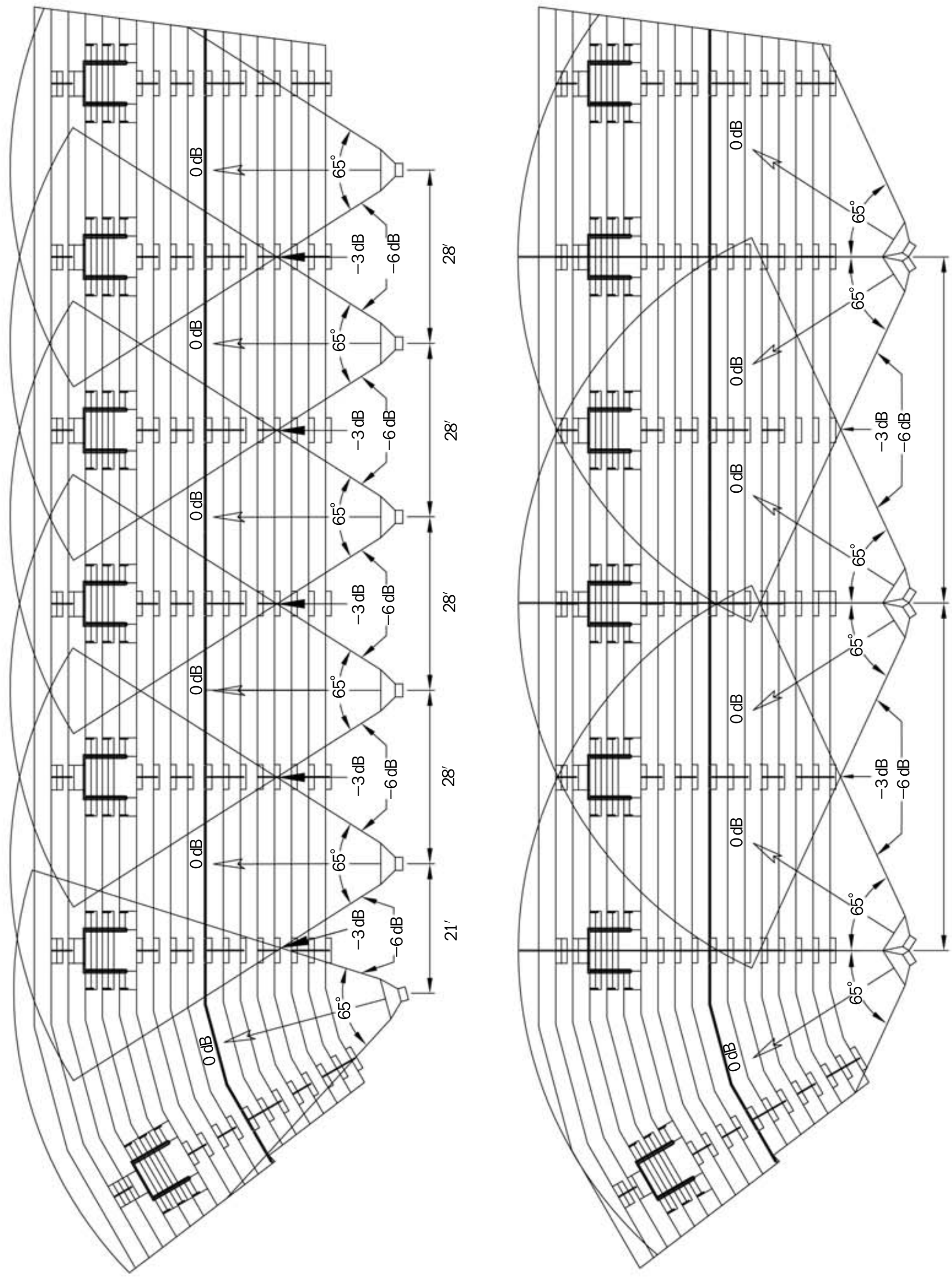
- If the SPL is too loud in the nearest seats, there are basically four options: (1) Aim the loudspeaker higher in the audience so that its centerline is aimed more toward the farthest seats, (2) move the loudspeaker farther away from the nearest seats, (3) increase the height of the loudspeaker, or (4) use a combination of all of the above. Each option requires some compromise, which may include not having as much energy placed on the audience.
- Each time we move or adjust the aim of the loudspeaker, we must recalculate all levels.

52. (6) *Use the floor plan to lay out loudspeakers.*

- The floor plan shows the exact location of each loudspeaker and its coverage. If the layout is not too congested, loudspeaker zones or other additional information can also be shown. If there is not enough room on the floor plan to show measurements, wiring paths, and loudspeaker zones for installation, a separate layout noting this information should be provided. In Figure 4.9, there are two different loudspeaker layouts. Both layouts make use of the same type of loudspeaker with a 65-degree by 65-degree dispersion and placed the same distance from the nearest seats.
- The first layout places each loudspeaker directly in front of a seating section with the speaker aimed at the middle to keep as much energy as possible on the audience. You will note that the −6 dB down points cross at the aisles and therefore sum so that there is only a 3 dB loss. (Loudspeaker interaction will be discussed in step 7.) This is a preferred layout that provides uniform coverage.
- The second layout places two loudspeakers together on one mount. Each set of loudspeakers is positioned in line with the aisles. The speakers are splayed so that each loudspeaker covers a seating section on each side of the aisle. Since the loudspeaker is aimed across the seating section, the design keeps as much energy as possible on the audience. Once again, you will note that −6 dB down points cross at each aisle and sum so that there is only a 3 dB loss. This layout also provides uniform coverage. Caution needs to be exercised that the loudspeakers are not placed too far apart horizontally or do not produce such a high SPL that they create timing issues. This becomes a possibility because the axis of the loudspeaker is aimed across the audience, and without additional off-axis loss, the sound can extend a good distance into another loudspeaker's coverage.

53. (7) *Verify loudspeaker interaction.*

- When calculating levels, we must take into consideration how other loudspeakers will affect the overall level. This can be especially important for the near and far seats. For example, if



21′ = 6.40 m
28′ = 8.53 m
56′ = 17.07 m

Figure 4.9 Typical Loudspeaker Options

the SPL on the near seats is close to being too loud, the energy from two loudspeakers can sum, making it unbearably loud. On the other hand, in some instances two loudspeakers can sum so that it may be just loud enough for the far seats. The chart in Figure 4.10 gives us an understanding of how loudspeaker interaction can affect SPL.

When two levels differ by:	Add this dB-SPL to the higher value:
0 to 1 dB-SPL	+ 3 dB-SPL
2 to 3 dB-SPL	+ 2 dB-SPL
4 to 8 dB-SPL	+ 1 dB-SPL
9 or more dB-SPL	0 dB-SPL

Figure 4.10 Effects of Loudspeaker Interaction

- As explained earlier, the timing between loudspeakers has a direct bearing on the program's intelligibility. To calculate this relationship, first look at the section plan. If more than one loudspeaker is used to extend the sound coverage, measure the distance between each zone of loudspeakers. Calculate the delay between each zone, and note the delay required for each zone on the print. The loudspeakers should be timed so that they release their sound to coincide with the direction the sound is traveling.
- Next, look at the floor plan and make sure the loudspeakers are not placed too far apart horizontally. This is especially important when the loudspeakers are splayed and aimed across the audience. When the loudspeakers are aimed at an angle, the on-axis sound can easily extend into another loudspeaker's coverage with minimal SPL loss. Experience has shown that timing problems start to occur when loudspeakers are placed more than 60 feet (18.29 m) apart horizontally. The extent to which the intelligibility will deteriorate is dependent on both the angle of the loudspeaker and its SPL output.
- At times, you will need to determine the delay for a loudspeaker that is within the direct coverage of multiple loudspeakers. First, calculate how long it will take the sound to travel from each loudspeaker to the loudspeaker that you are trying to calculate the delay for. If the sound arrives from all the other loudspeakers within a 35-millisecond (ms) window, adjust the delay to match the first arrival. If sound arrives from one or two loudspeakers *outside* the 35-ms window, average the time for all sound arrivals and use this number to set the delay. This will mean that sound may arrive a few milliseconds early from one or more loudspeakers; however, very few listeners will be able to discern this and those who do will quickly become accustomed to the sound's early arrival.

54. (8) *Place stage monitors.*

- Stage monitors should be placed so that they face the participants. This allows the participants to hear themselves and, in the case of a demonstration or an interview, hear each other. This configuration also aligns the microphone so that its greatest rejection is almost always directly in line with the stage monitors. (See Figure 4.11.) Properly set-up stage monitors should be used for the entire program, not just the drama. Setting up and equalizing stage monitors will be discussed in greater detail in Chapter 7, paragraphs 57-60.

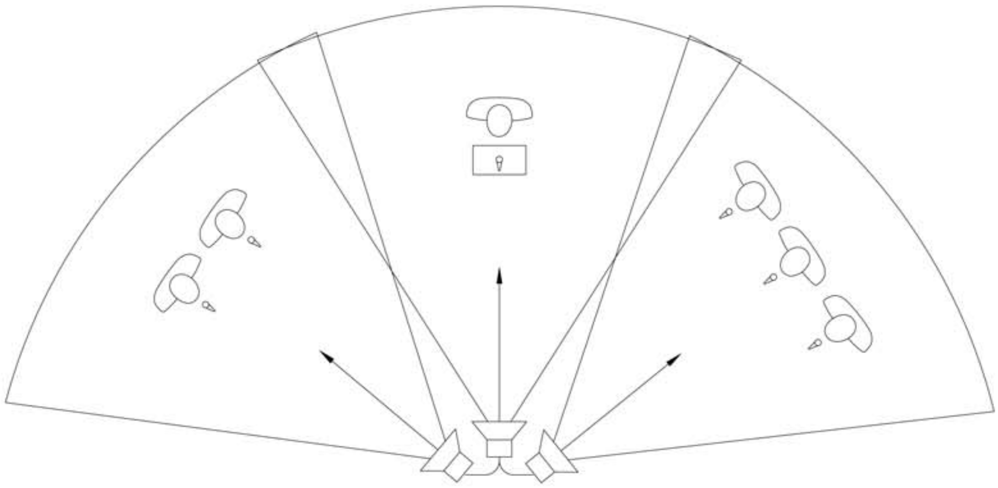


Figure 4.11 Recommended Stage Monitor Layout

55. (9) *Determine speaker zones and required amplification.*

- There are four primary reasons for using loudspeaker zones: (1) delaying loudspeakers for proper timing, (2) managing amplifier power, (3) special processing or level control for certain areas, and (4) simplifying troubleshooting.
- Looking at the floor plan, identify groups of loudspeakers that will require a delay. When more than one row of loudspeakers is needed to extend sound coverage and the distances require it, each zone of loudspeakers after the first zone needs to be on its own delay and timed to coincide with the direction the sound is traveling. If a house system is being used, each zone of loudspeakers that supplements the house system needs to be on its own delay.
- A zone can be used in areas where greater sound-level control or processing may be required. For example, stage monitors will always be on a separate zone so that they can have their own equalization. There may be zones where we desire to roll off low frequencies more than normal to provide greater articulation, such as floor seating where older ones tend to sit.

- Well-thought-out zones can prove advantageous if an unforeseen problem occurs. Having multiple zones not only minimizes the number affected by the problem but divides the system into manageable sections for easier troubleshooting.
- Once you have identified zones requiring a delay, calculate how much power each loudspeaker needs to project the sound as designed. To do this, look at the loudspeaker's sensitivity specification. Each time the SPL is increased by 3 dB, the power feeding the loudspeaker must be doubled. Therefore, if a minimum of 100 dB is needed and the loudspeaker's sensitivity at 3.28 feet (1 m) is 92 dB at 1 watt, 95 dB will require 2 watts, 98 dB will require 4 watts, 101 dB will require 8 watts, and so on. So the loudspeaker would need a continuous 8 watts of power to provide the required 100 dB-SPL. It is a good practice to design the system to allow for a minimum of 6 dB for headroom. Therefore, in this instance, 32 watts of power should be reserved for each loudspeaker. This emphasizes the importance of using highly sensitive loudspeakers.
- Determine the wire size needed to feed the loudspeakers. Wire resistance can be a source of significant power loss, especially over long distances. Therefore, due consideration must be given to sizing the wire properly for the electrical load that it must handle. A power loss of 0.5 dB represents approximately 11 percent of the total power output of the amplifier. Long lengths of wire, especially the initial feed from the amplifier to the first loudspeaker in a zone, should be given consideration because of the wire length and the need to carry the entire electrical load of the speaker zone. The chart in Figure 4.12 lists the resistance in ohms for various common wire sizes and also indicates the maximum feeder length that can be used in order to limit wire losses to 0.5 dB. Also, a column for 70-volt systems notes the gauge of wire to use based on maximum power needed. Keep in mind that a drop of 3 dB represents a waste of one half of the amplifier's available power.
- It is a good practice not to load the amplifier more than 80 percent. This will help with power loss from wire resistance and allow the amplifier to run cooler. For example, if you have a 100-watt amplifier, you would only put an 80-watt load on it, such as five loudspeakers tapped at 16 watts ($5 \cdot 16W = 80W$).
- After determining the power needed for a loudspeaker zone, verify that the amplifier to be used is capable of producing enough power. For example, imagine that a system has four loudspeakers that require 32 watts of power to produce the required SPL. It is a 70-volt system using transformers that have the following tap values to choose from: 7.5W, 15W, 30W, and 60W. Ideally, the 60W tap would be used to give a little more headroom. Therefore, an amplifier capable of producing

POWER LOSS DUE TO LOUDSPEAKER LINES														
					FEEDER-PAIR LENGTH (FT) FROM AMPLIFIER TO LOAD FOR APPROXIMATELY 0.5 dB LOSS IN POWER (11%)									
								POWER AND LOAD IMPEDANCE AT 70V						
WIRE SIZE AWG	NEAREST BRITISH SWG	DIAMETER (mm)	OHMS PER 1000′ PAIR (304.80 m)	MAX SAFE POWER AT 70V	4Ω	8Ω	16Ω	30W 167Ω	40W 125Ω	60W 83Ω	100W 50Ω	250W 20Ω	400W 12.5Ω
10	12	2.59	2.0	1750W	120	240	475	4930	3690	2450	1475	590	370
12	14	2.05	3.2	1400W	75	150	295	3080	2300	1530	920	370	230
14	16	1.63	5.2	1000W	45	90	180	1895	1420	945	570	225	140
16	17-18	1.29	8.0	420W	30	60	120	1230	920	610	370	150	90
18	19	1.02	13.0	210W	15	35	70	760	565	380	230	—	—
20	21	0.81	20.6	70W	10	25	45	480	360	240	—	—	—
22	23	0.65	32.6	35W	7	15	30	300	—	—	—	—	—

Figure 4.12 Chart for Determining Power Loss

300 watts would be needed for four loudspeakers tapped at 60W plus 20 percent, or (4 · 60) · 1.2 = 288. If the additional SPL is not needed, an amplifier capable of producing 150 watts could be used by tapping the loudspeakers at 30W, which would result in (4 · 30) · 1.2 = 144. Alternatively, a series-parallel system could be used. Remember, the power is divided equally between the loudspeakers. Therefore, an amplifier capable of producing 150 watts would be needed for four loudspeakers drawing 32 watts plus 20 percent, or (4 · 32) · 1.2 = 154.

56. (10) *Determine electrical power requirements.*

- Good reliable power is necessary for a sound system to function to its full capacity. The AC circuits supplied should be dedicated circuits feeding only the sound equipment. All AC circuits for the sound equipment should be fed from the same electrical panel to avoid differences in ground potential. Making sure the necessary power is available is a part of our design process; overlooking this can have dire consequences. The sound overseer and his assistant should know the location of the electrical panel in case a circuit breaker trips. Arrangements should be made so that these panels can be accessed at any time during the convention program, should that be necessary. After designing the sound system, you should total up how much electric current the sound equipment draws. It is a good practice not to load the AC circuit more than 80 percent.
- The front-end equipment should be on its own circuit. Clean power is very important for front-end equipment, especially for digital-signal processing and computers. If the power is unbalanced because of grounding problems, a portable isolation transformer is recommended. The transformer should be specified to match the voltage, cycles, and circuit load.

- Amplifiers should likewise be on their own circuit. Although amplifiers are not as susceptible to power issues as the front-end equipment, problems can still occur. If this happens because of poor grounding, an isolation transformer is recommended. Depending on the number of amplifiers and how much current they are drawing, multiple AC circuits may be required. To determine this, refer to the power information on the amplifier. It is generally noted near the power cord on the back of the amplifier. In Figure 4.13, note that the maximum output of the amplifier is 600 watts at 70.7 volts and that it will draw 8.5 amperes (amps) at full power with a 120-volt feed.

Figure 4.13 Amplifier Power Information Label

- An uninterruptible power supply (UPS) is typically not necessary. In the event of a major power outage, the building system, if so equipped, will cut over to emergency power after a short period of time. The brothers will understand the interruption and wait for the program to restart.
- If a portable generator is necessary to run the sound system, it should be purchased from a reputable manufacturer and should provide clean power that matches the frequency (50 or 60 Hz) of the country. It should be able to provide enough power to supply the electrical needs of the sound system. A seven-horsepower unit can provide two 20-amp circuits and will generally suffice for a small sound system. There should be circuit breakers to monitor overcurrent and circuit problems. The generator should also have an engine shutdown switch and an automatic engine shutdown to protect the engine if the oil level is low. A good-size fuel tank will allow extended operation, and a fuel gauge is helpful to avoid running out of fuel during the program. An onboard voltmeter will indicate how much voltage is being produced. The generator should have a good muffler so that it operates quietly and should be installed outside the seating area in a secured and well-ventilated location.

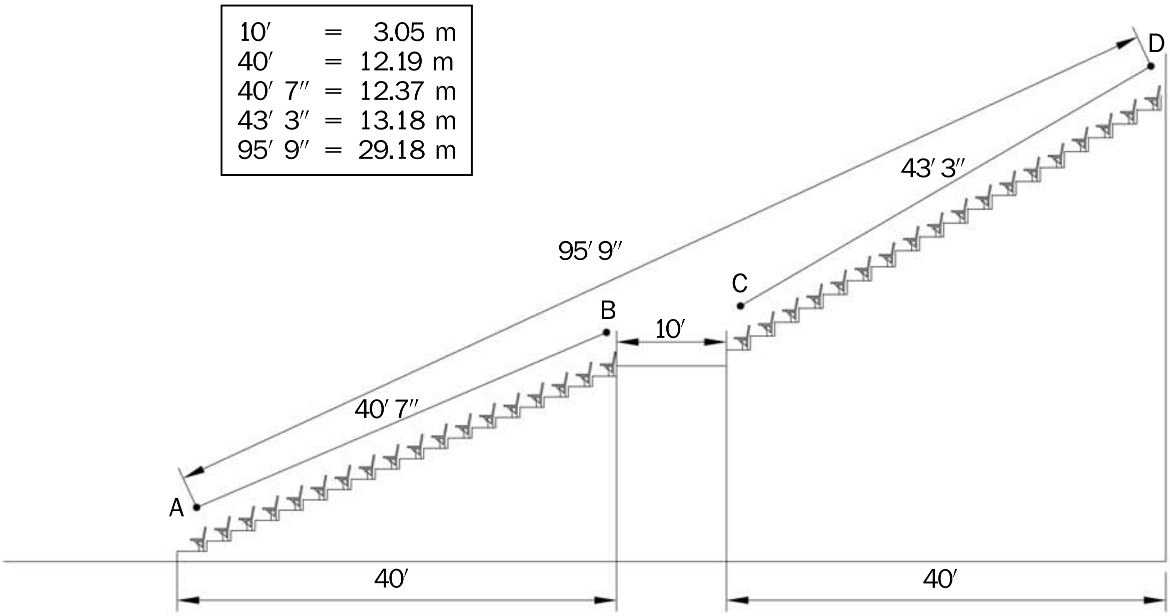


Figure 4.14 Section View Dimensions

Loudspeaker Aiming and SPL Calculation Exercise

57. This exercise will be based on Figure 4.14, which depicts the section view of a typical sports venue. You will note that we have placed solid circles above the nearest and farthest seats in each seating tier. The circles are placed four feet (1.22 m) above the floor of each seat location, as this is the typical ear height of a seated person. These points have been labeled *A, B, C,* and *D,* with points *A* and *C* being the nearest seats in their seating tier and points *B* and *D* being the most distant seats in their seating tier. We will refer to these points as we go through the design process.

58. (1) *Use the section plan to make an educated decision on a starting point.*

- As discussed in paragraph 48, first make an educated decision on a starting point. Will the sound have to cover more than one seating tier? In our example, we have two tiers of seating, each over 40 feet (12.19 m) deep, with a 10-foot (3.05 m) aisle between them. Ideally, we would prefer to use one speaker on the floor to cover both tiers of seats, since this would require less labor and equipment to install. However, using only one speaker on the floor will require the sound to be projected more than 95 feet (29 m), which means we must have a highly directional loudspeaker capable of projecting sound that distance.
- However, let us say the choices of loudspeakers are limited to speaker columns that are capable of projecting the sound between 40 and 60 feet (12.19 and 18.29 m). Such a speaker column cannot cover the entire distance; therefore, we are forced to split the distance and use two speakers to cover the distance

needed. Since we are projecting the sound approximately 50 feet (15.24 m), we will set the loudspeaker on the floor 10 feet (3.05 m) in front of the nearest seats and 10 feet off the floor. Each loudspeaker will be aimed so that the audience is within the –6 dB down point, in an effort to keep as much energy on the audience as possible. The upper speaker should be secured using the best mounting options.—See Figure 4.15.

- Let us say that the notes from the evaluation walk-through state that the ambient noise is 60 dB-SPL in the lower seating section. So we will strive to have 75 dB-SPL at point *B*. This will be 15 dB above ambient noise and at a comfortable level for our brothers to hear the program.
- In order to simplify this exercise, we will give you the distance and the number of degrees off-axis points *A* and *B* are from the loudspeaker's axis so you can make your calculations. Use the vertical dispersion on the 1.25 kHz polar plot (Figure 4.4) to determine your off-axis loss. You will note that there is a slight crater in the vertical dispersion on the right side of the 1.25 kHz polar plot and that it is slightly raised on the left side. We will take an average between the two sides to determine our off-axis loss. All calculation results will be rounded to the nearest whole number.

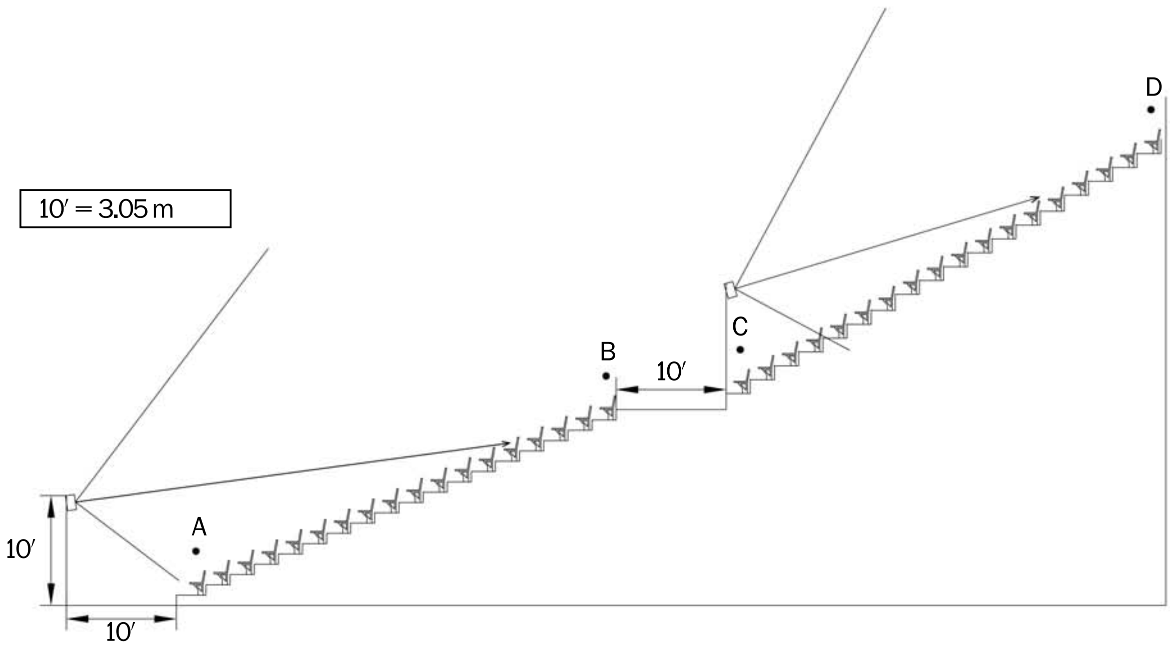


Figure 4.15 Example for Calculating SPL

59. (2) *Calculate SPL loss for the farthest seat.*

- The distance from the loudspeaker to point *B* is 47 feet (14.33 m) and is 6 degrees off-axis. Calculate the distance loss using the inverse square law (dB-SPL = 20 log D_f/D_n). We will take this first calculation step-by-step. First, we have to figure out the ratio between point *B* (D_f [distance far]) and the 3.28-foot (1 m) point in front of the loudspeaker (D_n [distance near]), or $47 \div 3.28 = 14.33$. To find the distance loss, take the log of 14.33, multiply it by 20 ($\log 14.33 \cdot 20 = 23.12$), and round it to 23 dB-SPL. Look at the 1.25 kHz polar plot (Figure 4.4). At 6 degrees off-axis, 2 dB are lost. Adding the distance and the off-axis losses gives you a total loss of 25 dB.

- Calculate how much SPL is needed at 3.28 feet (1 m) in front of the loudspeaker to provide 75 dB at point *B*. The total should be 100 dB. This is the sum of the desired SPL at point *B* and the overall SPL loss: 75 + 25 = 100 dB-SPL.

60. (3) *Calculate SPL loss for the nearest seat.*

- The distance from the loudspeaker to point *A* is 9 feet (2.74 m) and the angle is 30 degrees off-axis. Calculate the distance loss; it should result in 8.77 dB. Look at the 1.25 kHz polar plot and find the off-axis loss. You should have determined that there will be a 3 dB loss. What is the overall SPL loss? This is the sum of the distance loss and the off-axis loss (8.77 + 3), which is 11.77, or rounded up, 12 dB.
- Calculate the SPL at the nearest seat. You should have come up with 88 dB by subtracting the 12 dB loss from the 100 dB at 3.28 feet (1 m) in front of the loudspeaker.
- Remember to double-check the SPL at a lower frequency, such as 400 Hz, to make sure it is not too loud at the nearest seat. In this instance, with only 1 dB off-axis loss at this frequency, the result is 90 dB.

61. (4) *Balance levels between far and near seats.*

- With 88 dB at 1.25 kHz and 90 dB at 400 Hz, we need to make adjustments to lower the SPL at the nearest seats. As mentioned earlier, loudspeaker placement and aiming involves choosing the best overall compromises within various parameters to get the best overall results. By looking at Figure 4.15, we can determine our best options before having to re-aim and/or move the loudspeaker, which would require calculating all the levels again. Notice that both loudspeakers have similar parameters; both have to project the sound about the same distance. So we can conclude that we will have similar problems with the sound being too loud in the front rows of the second seating section as well. Logically, our first thought is whether we can drop the SPL at the loudspeaker. If we do, it will reduce the ratio between direct sound and ambient noise, thus affecting intelligibility. Since the SPL from the loudspeaker is 15 dB more than the ambient noise, we can afford to drop 2 or 3 dB. However, we cannot drop the loudspeaker's output level enough to solve the problem entirely.
- We can aim the loudspeaker so that it is on-axis with point *B*, the last row of the first tier. This will eliminate the off-axis

loss so that we can gain 2 dB. This will send more energy toward the ceiling. However, gaining 2 dB is worth the compromise. There will be some SPL gain from the second loudspeaker at point *B* because the second loudspeaker is also discharging about 100 dB. However, if we look at the polar patterns in Figure 4.4, we can quickly discern that we will have no gain from summing at higher frequencies not only because of the 13 dB distance loss from the second loudspeaker to point *B* but especially because of being 163 degrees off-axis. The off-axis loss from the second loudspeaker to point *B* at 1.25 kHz is approximately 18 dB. Adding the distance and the off-axis losses (13 + 18 = 31) gives us a total SPL loss of 31 dB. If we subtract 31 dB from the 100 dB at the second loudspeaker, we will have 69 dB at 1.25 kHz at point *B*. If we have 75 dB at point *B* from the first loudspeaker and 69 dB from the second loudspeaker, we will have +1 dB of gain added to the first loudspeaker level of 75 dB, giving us a total of 76 dB at 1.25 kHz at point *B*. This is because there is a 6 dB-SPL difference between the two loudspeakers at point *B*. According to the chart in Figure 4.10, we can thus conclude that there is a gain of 1 dB. As we drop in frequency, there is less off-axis loss, so we will start to gain more level from summing, between 2 and 3 dB at 800 Hz. Interestingly, at 400 Hz there is 81 dB-SPL from the second loudspeaker at point *B,* which is more than the amount being received from the first loudspeaker. (See the chart in Figure 4.16 for the interaction of both loudspeakers at point *B*.) The SPL increase at these lower frequencies will not help with articulation. However, it does tell us that we can reduce the SPL that is being discharged from the first loudspeaker.

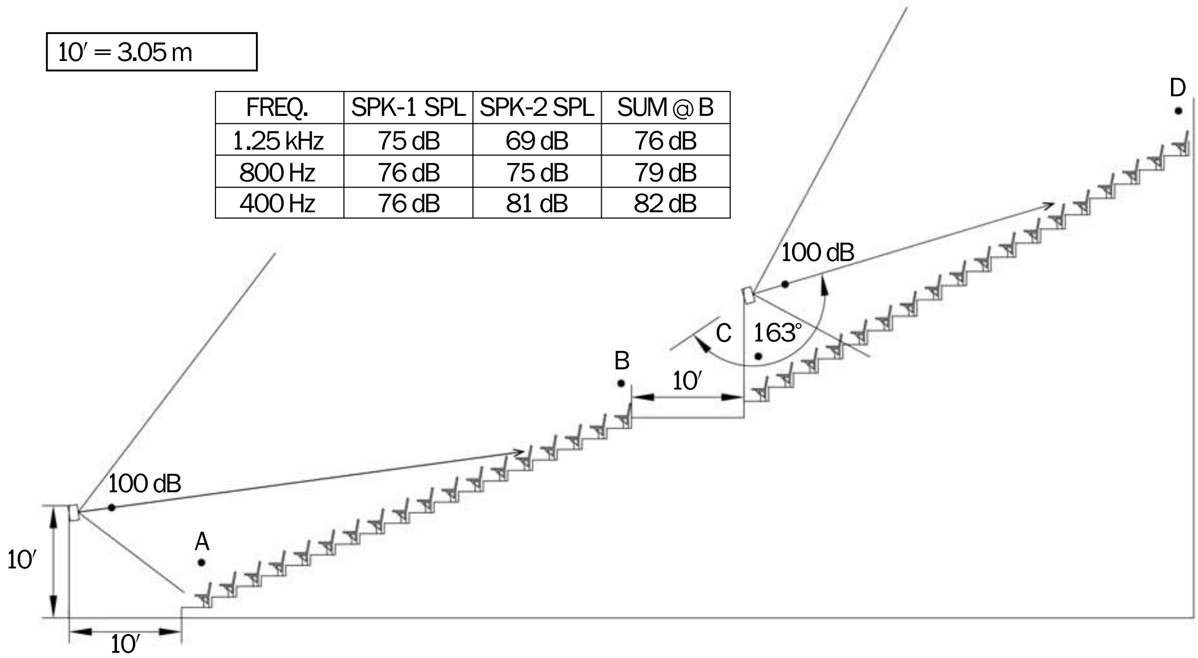


Figure 4.16 Example of Loudspeaker Interaction

- If we use these adjustments, we could drop the SPL level of both loudspeakers to reduce the SPL in the front rows by 5 or 6 dB without having to move the loudspeaker farther away or higher, which would keep it within our parameters. However, the SPL will be a little lower than planned at point *D,* where there might be more ambient noise. The best solution is to raise the height, re-aim the second loudspeaker so that it is on-axis with the last row, and decrease the output level slightly. After making any adjustments, you must recalculate your work.
- If you are not able to get the desired SPL at the near and far seats after making such adjustments, two loudspeakers can be used for each seating section as illustrated in Figure 4.17. The two loudspeakers in the illustration are aimed toward the middle of the last row. This reduces the SPL level in the front row by 2 or 3 dB and increases the SPL in the back rows by means of summing. The drawback of this layout is that it requires more equipment. Also, aiming the sound at an angle across the audience can create timing issues, depending on levels and the distance between loudspeakers. The short distance between loudspeakers as illustrated in Figure 4.17 poses no problems with timing.

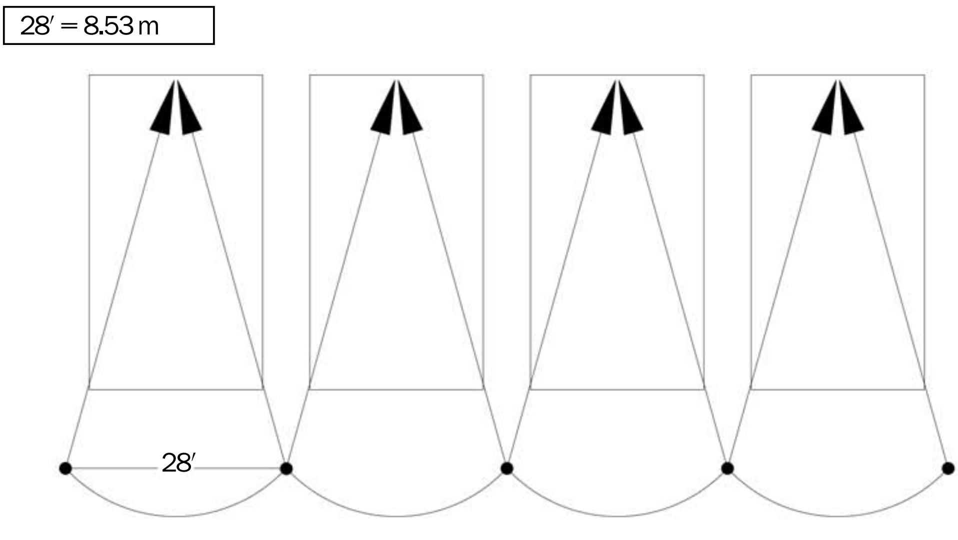


Figure 4.17 Summing Loudspeaker Layout Example

62. (5) *Check the loudspeaker timing.*

- Since two loudspeakers are needed to extend the sound coverage, we must delay the second loudspeaker so that it is in time with the first loudspeaker. There is a distance of 63 feet (19.20 m) between the two loudspeakers, as noted in Figure 4.18. Calculate how much delay is needed to the second loudspeaker. To do this, we simply multiply the distance between the two loudspeakers by .885 (if in feet) or 2.9 (if in meters), which gives us a delay of 56 ms.

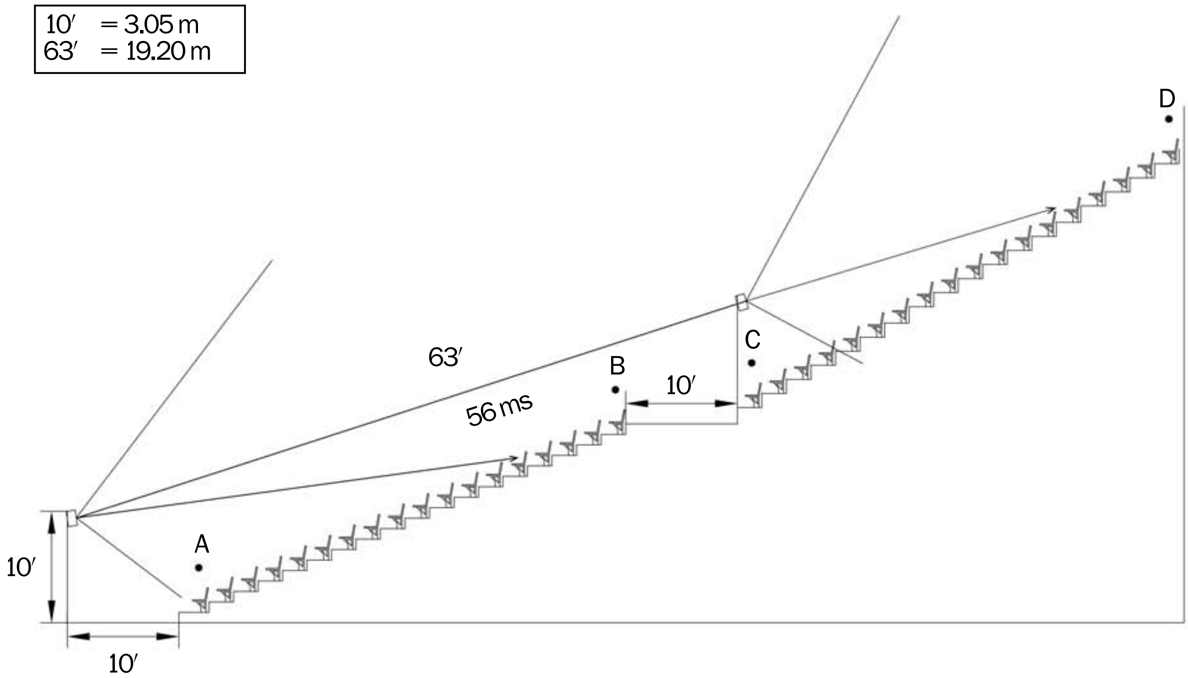


Figure 4.18 Example for Calculating Loudspeaker Delay

Designing the System Front End

63. Now that we have discussed how to design a loudspeaker layout, let us review the elements in the signal chain that feeds the loudspeakers, noting their function and where they fall in the signal chain. The physical connections will be discussed in Chapter 6, and how the front-end equipment is adjusted will be discussed in Chapter 7.

64. *Source Equipment:*

- *Microphones:* A microphone is a transducer that takes acoustic energy and changes it to an electrical signal that can be processed and amplified. Microphones have various pickup patterns, with the cardioid pickup pattern being the most common for our application.
- *MP3 players:* These are used for playback of prerecorded audio supplied by the branch.

65. *Front-End Equipment:* The front-end equipment is composed of different components designed to control, shape, or solve an acoustic or signal problem. With analog front-end equipment, each piece of equipment typically performs one function, whereas a digital signal processor (DSP) usually provides multiple functions. A DSP is also more cost-effective and gives us greater control over the audio signal chain. Note the following brief descriptions of the front-end equipment we commonly use for our conventions:

- *Audio mixer:* In the simplest of terms, an audio mixer allows us to control the audio signal, adjusting the volume and turning the signal off and on. Some mixing consoles have equalization built in. The audio mixer is the first piece of equipment after the source equipment.

- *Audio meter:* A meter is often used to monitor the signal level from the mixer into the system. The most common are the VU, peak, and loudness meters. Each meter tracks the signal differently. A VU meter measures and tracks the average signal level, while the peak meter measures and tracks the peak signal level and the loudness meter measures and tracks a combination of average and peak signals.

- *Limiter:* This is used to protect against sudden bursts of volume that could possibly damage loudspeakers. It typically has a faster reaction time than a compressor and is generally used within the DSP. For our convention use, the threshold should be set high enough so that it is not reached during normal operation.

- *Equalizer:* This is used to adjust the frequency response of audio signals. Speech and music should be equalized separately for conventions. There are two types of equalizers, graphic and parametric. Depending on the graphic equalizer purchased, the audio spectrum is typically divided into bands of either 1 octave, 2/3 octave, or 1/3 octave and the equalizer gives a graphic approximation of the spectral response of the unit. A parametric equalizer allows a number of filters, varying in width and depth, to be placed at any frequency. Parametric equalizers are preferred because of their precision.

- *Feedback reducer:* This piece of equipment detects and reduces feedback. It does so, first, by deploying narrow-band notch filters that are generally increased in 3 dB increments and, second, by widening the notch filter until the feedback stops. While a feedback reducer helps reduce feedback, it cannot entirely eliminate it. The feedback reducer should be installed after the speech equalizer and should not be used with music.

- *Cut shelf:* This cuts frequencies above or below a selected frequency known as the cutoff frequency. A high-pass filter will allow only frequencies above the cutoff frequency to pass unaffected. The low frequencies below this cutoff are attenuated at a constant rate, usually 6, 12, or 18 dB per octave.

- *Crossover:* This piece of equipment divides the full-range audio signal into low- and high-frequency components, directing each only to the appropriate output. Sometimes this is called a frequency-dividing network. The crossover is placed early in the output chain, before the amplifiers feeding the high- and low-frequency drivers.

- *Compressor:* A compressor reduces the dynamic level of an input audio signal. The ratio setting determines how much the output level is changed. For example, a gentle 2:1 compression ratio simply means that for every 2 dB of input level, the output level will be changed by 1 dB, or half the level. Typically, a compressor has a slower reaction time than a limiter. For our application, a compressor is used in the signal chain feeding the FM transmitter to ensure a consistent signal level, thus enabling us to maximize its transmitting power. Generally, we do not use compressors on other outputs where they can work against us. For instance, at the beginning of each session the chairman must overcome significant noise in an effort to get the attention of the audience. If a compressor is used on the program audio, it will likely prevent him from reaching the necessary level.
- *Distribution amplifier:* This piece of equipment accepts at least one input signal and provides the same signal to multiple isolated outputs, which allow the signal to be distributed without ground loops and signal degradation. Generally, the power for each output can be adjusted to provide the optimum input for the power amplifier. A distribution amplifier comes after the majority of front-end equipment and feeds the zones of amplifiers. However, if a signal delay is required, the delay is typically installed after the distribution amplifier so that zones can be individually delayed.
- *Delay:* A modern digital delay stores an audio signal temporarily and then releases it after a preset time.
- *Power amplifiers:* These take a nominal audio signal and increase the power of the signal. They are the last piece of electronics before the loudspeakers. Based on our design process, these will be either constant-voltage or low-impedance amplifiers.

How and Why Equipment Is Connected

66. Figure 4.19 shows the plan view of a sound system that utilizes the house system.

67. The house system's loudspeakers are generally rigged from the ceiling. Their height above the floor evens out the distance between the far and near seats, allowing for a more even SPL. In Figure 4.19, you will note that there are two areas in the corners where the house sound system coverage is weak and requires supplemental loudspeakers. Floor loudspeakers are also needed to cover the floor seating. They are placed within 40 feet (12.19 m) of each other so that there are no timing issues between them. All the loudspeakers that we install to supplement the house system must be timed to match the house loudspeakers. Stage monitors, which are set up for the participants and are on their own zone, are the exception to this rule. The stage monitors are always in real time.

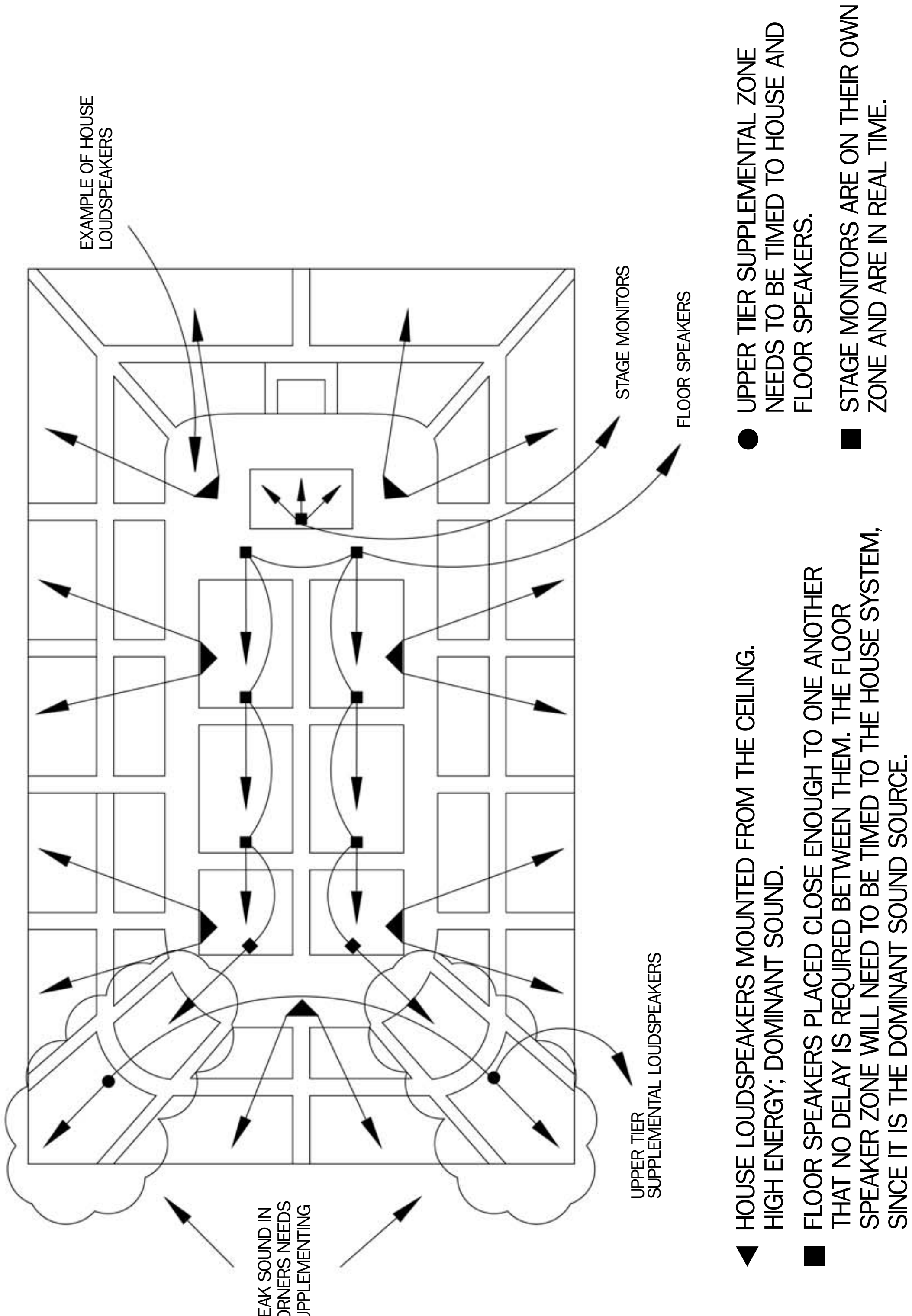


Figure 4.19 Typical Plan View Including House System

68. We will look at this sound system by means of two block diagrams: a simple analog system using individual components in Figure 4.20 and a digital signal processor that has more processing capabilities in Figure 4.21. In comparing them, you will note that the signal chain is the same. However, the DSP allows us to add processing to individual zones, giving us more control. The DSP also eliminates all the cabling between the processing equipment in an analog system. You can use the analog block diagram to visualize the installation order of the equipment. Detailed explanations pertaining to its setup will be given in Chapter 7.

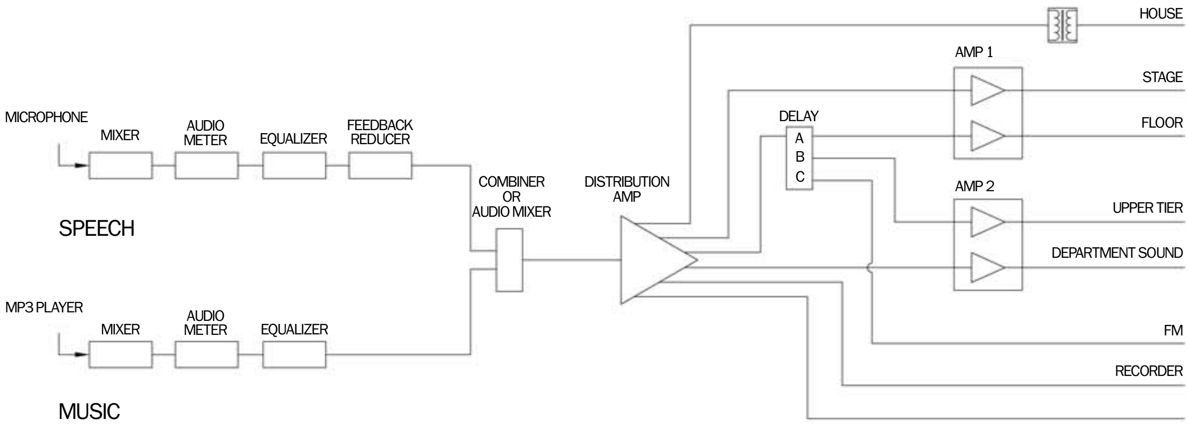


Figure 4.20 Component Block Diagram (Analog System)

Input Side of Block Diagram:

69. First, let us look at the speech chain. At the start of the chain, microphones feed the audio mixer, which gives basic control over the audio signal. Next, we have a limiter to protect the loudspeakers from a sudden burst of sound. A limiter may not be available if you are installing analog equipment. Following that is an equalizer, which provides a low- and high-cut shelf for the speech spectrum and reduces unwanted reflections on the stage. Next, we have a feedback reducer to notch any unwanted frequencies that enter the microphone.

70. Now, let us examine the music signal chain. At the start of the chain, the MP3 player feeds the audio mixer, which gives us basic control over the audio signal. Next, a limiter protects the system from any high-level spurious signals that could damage our loudspeakers. Having an equalizer in place is good, since at times some modest equalization may be needed.

71. The speech and music strings are now combined; a fairly inexpensive combiner can be used. If such a combiner is not available, an audio mixer can be used. There are also distribution amplifiers available with multiple inputs that can be switched to any output.

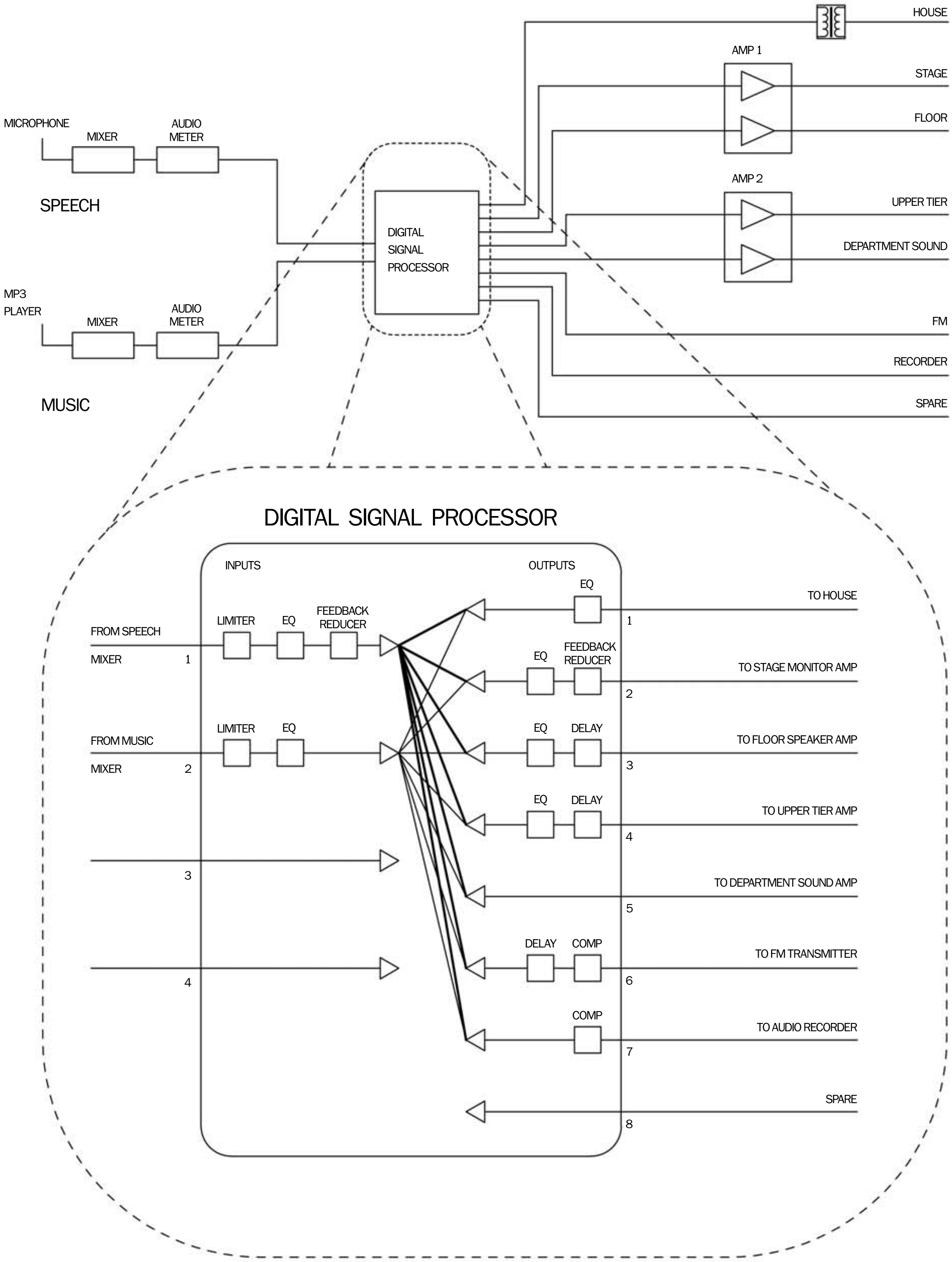


Figure 4.21 Component Block Diagram (With Digital Signal Processor)

The distribution amplifier feeds the various output zones. With a DSP, the signal distribution to the various outputs takes place internally. **Caution:** A Y-adapter should *never* be used to combine two outputs. Outputs are low impedance and must *only* be connected to high-impedance inputs. If a Y-adapter is used, one output will try to drive the other output, forcing both outputs into current-limit and possible damage. It will also result in severe signal loss. If proper combining equipment is not available, see Appendix 7, Figure A7.12, for information on constructing an unbalanced summing box.

Output Side of Block Diagram:

72. *House feed:* Because of the difference in ground potential between the house system and our sound system, there is a likelihood of creating ground loops, which can introduce an unacceptable amount of hum in the audio program. Noise-free operation is best ensured by the use of transformer coupling between the systems. When suitable transformer isolation does not exist or is in question, then a suitable isolation transformer should be used. For details on the installation of isolation transformers, see Chapter 6, paragraphs 42 and 45 and Figure 6.3.

73. At times, there may be a need to adjust the house equalization. With a DSP, we can easily install an equalizer on the house feed to roll off the low end or make other adjustments if necessary. If you are installing an analog system, you may not have an extra equalizer. If this is true, you may need to see if the building management will let you make some adjustment to the house equalization.

74. Stage monitors usually require additional equalization to provide adequate SPL without feedback. Since we are not as concerned with naturalness, we can adjust the equalization with emphasis solely on the articulation range. With a DSP, an equalizer and even a feedback reducer can easily be installed for the stage monitors only. The same procedure would apply with analog equipment, using an extra equalizer and feedback reducer. If this equipment is not available, the SPL will have to be lowered.

75. *Floor seating feed:* In our earlier example, the loudspeakers are set up less than 40 feet (12 m) apart so that they do not require a delay between them. However, the zone will require a delay because everyone within the floor seating will hear the house system to some extent. It is a good practice to install an equalizer on the floor seating to increase articulation.

76. *Upper tier feed:* It is a good practice to install an equalizer on the supplemental loudspeakers if available. This zone will also require a delay. Since all those sitting in this area will hear the house system, the timing will need to be matched.

77. *Department feed:* With an analog system, a straight connection can be made from the distribution amplifier to an amplifier feeding loudspeakers in various departments, such as Administration, Cleaning, and Attendants. If a department desk is situated where sound from the house system may also be heard or if the audience will hear the department loudspeaker, the feed will need to be delayed.

78. *FM transmitter feed:* An FM transmitter is used primarily to assist those with impaired hearing. If available, a compressor should be used to feed the FM transmitter to provide the highest possible level without clipping. Details on the installation and optimization of this feed will be discussed in succeeding chapters.

79. *Recorder feed:* The program will be recorded for the benefit of the elderly and infirm who are unable to attend. If available, a compressor should be used on this output to prevent clipping. Special care should be used so that the entire program is recorded with the highest possible quality. It is beneficial to have a brother monitor the recorder for the duration of the program.

Transmitting to Multiple Sites

80. When special events such as zone talks or international conventions are held, at times multiple sites must be tied in. Clearly, much advance planning will be necessary. There are a number of methods available to tie in multiple sites, and we will now discuss some basic guidelines.

81. Each of the following methods has positive and negative aspects that should be given consideration. How often will such programs take place, once a year or every four years? How many sites will be tied in? Will the event require one-way transmission, or will there be program participants at multiple sites? Knowing these answers may determine whether we purchase or rent equipment and whether it is set up permanently or temporarily. Consideration should also be given to transmission cost and the audio quality that the provider is able to deliver.

82. Regardless of the method chosen, the output from our sound system should have its own level control to provide the optimum audio level for the patch or transmission equipment. Additionally, such equipment must be fed from an isolated output. If you are using an audio mixer or a distribution amplifier, an internal isolation transformer should be used on its output. If the facility does not have such a transformer, an external isolation transformer should be installed at the input of the patch or transmission equipment.

83. *Telephone Line Patches:* If the telephone system is going to be used to distribute the spiritual program to multiple sites, the telephone company should be contacted well in advance of the special event to ascertain what they require, such as jack locations and audio level. We must remember, though, that telephone lines have a limited bandwidth. A typical telephone line passes 300 to 3,000 Hz, which will provide intelligible sound for speech. However, it will not sound natural and will severely affect the quality of the music. If a telephone line will be used to tie in another site, equipment designed for this purpose is recommended. The manufacturer's specifications for the telephone patch will list the required audio level for its input. Generally, the audio input is 0 dBm (0.775 volts/600 ohm circuit). (See Appendix 1 and Appendix 7, Figure A7.7.) The telephone patch will provide a circuit to connect and transmit the audio signal at a standard telephone operating level, as well as a standard telephone jack for its output. Using equipment made for this purpose will minimize hum or noise that might be generated from improper levels or connections. For this type of telephone link, *each* site will require a telephone patch for either sending or receiving and each unit must be associated with an assigned telephone number. Each time a site originates a portion of the program, the number of telephone patches required is doubled.

84. *Telephone Conference Bridge:* The telephone company may be able to provide a conference bridge that can be rented for our event. A conference bridge simplifies feeding multiple sites. The same telephone patch equipment is used, but a conference bridge, generally located at the telephone company facility, requires only the originating site to send the audio program to the bridge. All the other sites receiving the program will use their telephone patch to dial into the conference bridge to receive the program. A telephone number is assigned to the conference, and each site is required to dial into the conference number and use a secure access code. This method will reduce the number of telephone patches required at the originating site.

85. *Audio Streaming Over Internet:* The audio quality is greatly enhanced compared to telephone audio because of having a much greater bandwidth, although a word or two may be dropped or skewed by Internet traffic. Using such technology requires an outside service provider to gain access to the Internet. A computer with a good sound card can be used to stream high-quality audio by means of the Internet. A number of computer programs can be used to accomplish this task. Consideration should be given to the computer's specifications, making sure it has the processing power and memory required to stream an audio program for an extended period of time. Audio streaming systems should always be thoroughly tested for reliability, quality, and security. The Internet also provides the means to transmit video. Video transmission requires more robust computer equipment, bandwidth, video cameras, and/or display equipment for each site. Because of the additional expense and complexity associated with video, permission for its use needs to be obtained from the Teaching Committee of the Governing Body.

86. *Satellite Links*: Using satellite technology requires an outside service provider to gain access to satellites. This is a service that is very reliable and secure, and it delivers a high-quality audio/video program. The equipment used to transport the audio/video program from site to site is rented from a satellite service provider and requires assistance from their personnel. Using a satellite system to feed multiple sites requires permission from the Teaching Committee of the Governing Body.

Multitranslation System

87. Our international conventions require the spiritual program to be transmitted to our multinational brotherhood in their native tongues. The complexity of the sound system can vary, depending on the number of languages the program is translated into, the number of brothers attending within each language group, and whether the convention is indoors or outdoors. To start with, we will discuss how this can be handled with an indoor facility.

88. If all attending a convention are seated within the same room and a small number of our brothers require the program to be translated, an FM transmitter can be used to provide the translated program. A room should be set up for the translators to enable them to hear the spiritual program without being distracted. The translators need to hear the originating program on headphones (not from loudspeakers) so that sound from the originating program does not mix with the translated program. If it is not possible to provide a hardwired feed for the translators' headphones, an FM receiver tuned to the frequency transmitting the originating program can be used. The microphone(s) that the translator(s) uses should feed into an audio mixer that feeds an FM transmitter. The brothers desiring to hear the translated program can simply tune their FM receivers to the frequency used to transmit the translated program. Of course, if this method is used, advance notice needs to be given to all those needing to hear the translated program so that they can bring their FM receivers.

89. Large international conventions require much advance planning and good communication, especially when the program calls for a talk to be delivered to all in attendance and translated simultaneously by each language group's translator. We will first discuss what is necessary at an indoor facility where multiple language groups will be in attendance, each having its own venue and all needing to be interconnected. The same principles for designing a sound system that we have discussed thus far also apply to the sound systems used at each venue. The challenge is connecting the different venues together. The amount of wire and the type of switching required depend on where talks originate and which venues receive them. Using an A/V router that is capable of switching multiple inputs/outputs and that can be programmed for different configurations greatly simplifies the technical operation of the program. Each venue's sound system should send a feed to the A/V router. The translator's microphone in the originating venue should be on its own mixer and sent to the A/V router. In this way, the translator in the originating venue can be heard by selected audiences and not by others, thus allowing simultaneous translation. Each venue should receive a feed from the A/V router's outputs. When switching between venues, good communication needs to take place so that none of our brothers miss any part of the program.

90. Large outdoor stadiums are often used for international conventions. As mentioned earlier, the design principles discussed in this chapter also apply to sound systems used in an outdoor facility. Each language group will need to have a separate sound system, and similar switching and wiring will be required as discussed in the previous paragraph. However, with a large outdoor facility, each language group will generally be assigned to sit in a certain location in the stadium. If extra seats are available, a seating section will be left empty between language groups to minimize overlapping coverage by the loudspeakers. Reentrant horns are often used for this application, since they are capable of projecting sound over a long distance and allow for tighter control of frequencies as low as 300 Hz.

Final Step

91. Once you have double-checked all your work, draft a final copy of all pertinent information on your floor plan and section view for a permanent record. Make sure that all the information needed to install and set up the equipment properly is noted. Copies should be made for the brothers to use at the convention site, and the original print should be filed for reference or future needs. A detailed equipment list should also be compiled to make sure all the equipment will be shipped and on-site for the installation as well as accounted for when dismantling the system.

CHAPTER FIVE

Safety and Personnel

Safety a Primary Concern

1. As the "Grand Creator," Jehovah God considers life sacred and highly valuable. (Eccl. 12:1; Ps. 36:9) As his servants, we are required to adopt that same view. (Eph. 5:1) We do not take unnecessary risks with our life. We understand the need to be safety conscious at all times. (Deut. 22:8; Eccl. 10:9) We sincerely endeavor to apply Scriptural principles so as to protect ourselves and others from physical harm.—Prov. 3:21, 22; Phil. 2:4.

2. One basic Bible principle states: "The shrewd one sees the danger and conceals himself, but the inexperienced keep right on going and suffer the consequences." (Prov. 22:3) What serious consequences there can be if such advice goes unheeded! When a dangerous situation is not avoided, serious injury or death can occur. Words cannot express the physical and emotional impact this can have—it is devastating. All of us, therefore, have a very serious responsibility to work safely.

3. Those volunteering to assist with the installation and dismantling of the sound system should be familiar with the principles outlined in the booklet *Working Together Safely—Safety Rules and Standards for Volunteer Projects* (S-109). While this booklet deals primarily with the safety of volunteers on *construction* projects, the principles contained therein provide many excellent reminders that apply to the heavy work that is done in conjunction with a convention. Particular attention should be given to the following sections:

- Attire and Personal Health
- Work Habits and Conduct
- Injuries
- Housekeeping
- Personal Protective Equipment (PPE)
- General Tools and Ladders
- Higher-Risk Tools and Equipment (for those who will use lasers or motorized vehicles)

4. In addition, the chapters in this sound handbook dealing with installing and dismantling the sound system each begin with a discussion of specific safety concerns. This information along with relevant safety points from the booklet *Working Together Safely—Safety Rules and Standards for Volunteer Projects* may be used as a basis for any brief safety meetings held prior to installing or dismantling the sound system. In addition to safety points, reminders about care and concern for the building and equipment will also prove helpful.

Personnel

5. The success of a department is directly related to its personnel and the work that they accomplish. The Sound Department overseer and his assistant will play a large role in selecting willing volunteers to assist with the installation, operation, and dismantling of the equipment. Therefore, these keymen must be very carefully chosen.

6. When selecting a suitable overseer and assistant for the Sound Department, it is vital that only *spiritually* qualified men be selected and trained. Since the work to be done is of a technical nature, it may be tempting to select an overseer based solely on his technical aptitude. Interestingly, when a problem arose regarding food distribution to needy widows in the first century, the apostles knew exactly what to look for: "reputable men . . . full of spirit and wisdom." (Acts 6:3) Among those selected was Stephen, who was described as "a man full of faith and holy spirit." (Acts 6:5) Even though the work to be done was secular in nature, the key qualifications were not secular but spiritual. Jehovah's organization today similarly seeks to use men who are *spiritually* qualified. Many excellent Sound Department overseers had little or no previous experience with sound equipment, but they were mature, spiritual men who took the time to learn a few necessary technical details so they could be used in this capacity. Spirituality, discernment, and solid oversight skills are essential—technical skill is not.

7. Those who will serve in an oversight position must communicate well and cooperate closely with the direction of the Convention Committee. When a sound system design is provided by the branch, the overseer will work within the bounds of that design. He should not feel compelled to experiment unnecessarily with either the equipment or the design. If changes are deemed necessary, he will report to the committee and they will then determine if the changes are truly required. If any departure from the branch design is needed, the committee will arrange to contact the branch. Brothers used in oversight should have an earnest desire to share their knowledge and train others, not feeling obligated to do everything themselves.

8. The overseer will work with his assistant to identify individuals who can be used for installing, operating, and dismantling the sound equipment. Before making any assignments, they will carefully consider the age, experience, and qualifications of the personnel involved. Any volunteers selected must be known as spiritual individuals. Prior to the convention, the department overseer, his assistant, and other keymen will remind all volunteers to be exemplary in conduct, dress, and grooming, so as not to bring reproach on Jehovah or his organization.

Chapter Six

Installing for Quality and Reliability

1. For any sound-reinforcement system to perform reliably, special attention must be given to the installation. The key to success lies in good preparation along with care and concern for the quality of the work. When the installation has been handled properly, we can be more confident that we will have a safe, trouble-free system for the duration of the convention.

Specific Safety Concerns

2. Injuries are more likely to occur during the installation and dismantling of the sound system. Therefore, before beginning the installation process, it would be wise to review Chapter 5 for general principles of safety. No doubt this material would also provide a good basis for a brief meeting that could be held with the volunteers on the day of the installation. Here are a few areas that should be given special attention.

3. During the installation of a large sound system, many cables will need to be run—some for considerable distances. Special care must be taken that these cables do not become a trip hazard for our brothers either before or during the convention.

4. Any object protruding at head or face level could seriously injure the unsuspecting. Therefore, mounting bolts and brackets should be sized, positioned, or covered in such a way that sharp corners or other protrusions cannot injure passersby.

5. At outdoor conventions, care must be taken to protect operators from electrical shock. A raised covered platform will usually suffice to keep operators and equipment safe and dry in case of rain. It is also the course of wisdom to use a ground fault circuit interrupter, also known as a residual current device, to provide additional protection on all electrical equipment used outdoors.

6. Another area of concern is the mounting of equipment overhead. In some lands speakers will be "flown," or rigged, above seating sections or walkways high above the audience. Local laws may govern who is qualified to hang such equipment. Clearly, this is one aspect of the installation work that should be checked and rechecked and performed only by someone well qualified. Have the mounts been designed and installed with a sufficient safety factor to ensure that they can support the necessary weight and not come loose? Is there a way that they can be tethered so that, even if they did come loose, they cannot fall to the crowd below? If the mounts are to be used outdoors, are they capable of dealing with additional stresses, such as rain, wind, or extreme temperatures? The safety and well-being of our brothers during installation and throughout the convention should be one of our foremost concerns.

Preparation

7. Careful advance preparation can prevent much extra work and frustration on the day of installation. It should be noted that even sound systems for very large venues can be installed surprisingly quickly when time has been taken to coordinate the effort properly. What can be done to prepare?

8. In rare instances, the sound design may have been handled locally. However, most conventions will receive detailed sound design documentation that includes complete drawings of exactly how the sound equipment should be installed. When such documentation is provided, it typically comes from the branch office. Regardless of where the design originated, the sound overseer, his assistant, and other keymen will need to make sure that they have a firm grasp of the design concept. Only after they fully understand the design will they be able to instruct others clearly on how the installation should be done. They will also want to note carefully what they will need to bring with them to install the equipment. How much speaker wire will be used, and what gauge should it be? Are there interconnecting audio cables that can be purchased or made ahead of time? What tools are needed? Do any additional pieces of audio equipment need to be picked up from a neighboring Assembly Hall or Kingdom Hall? What test equipment will be needed? How many copies of the design drawings will be needed for those who will be working with the installation? These and similar questions must be considered beforehand in order to be successful.

9. From the time he receives his assignment, the sound overseer and his assistant should begin to search out willing volunteers, both skilled and unskilled, to assist with setup. From this list of volunteers, specific crews can be carefully arranged ahead of time to make the best use of the qualifications of each person. The types of crews that will be needed will largely depend on the type of work to be done and the skill level of the volunteers available. For example, if a large distributed system will be installed, several crews may be needed. Perhaps one crew could be assigned to unload and inventory the equipment and, once finished, to install the front-end equipment. Another crew could measure and mark the designated locations where speakers will need to be placed and, when finished, install the cables for the microphones and stage monitors. A third crew could be assigned to set up tripods, another just to run the speaker cabling, and another to mount and aim the speakers according to the design. A sound overseer can display keen discernment in arranging the crews in a way that will prove most effective.

10. It may be possible—long before the setup day—to provide these crews with basic information about the type of work that they will be doing. For instance, if a crew will be used to install speakers, even a simple hand-drawn sketch will help them approach the job with greater understanding. If drawings of the sound design were initially provided, enough copies should be made so that all crews will have the set of drawings needed for their specific assignment. It will also be helpful if the volunteers know what tools they should bring to accomplish their work. When small details such as these are handled beforehand, it greatly speeds up the pace of the work on setup day.

11. There are occasions when some of the sound equipment must be borrowed or rented. If any of the equipment is unfamiliar or if it has not been used for some time, it would be prudent to have those components tested beforehand. When equipment is critical to the function of the sound system, it may be advantageous to have quality spares added to the inventory.

12. In some cases, there will be a room where the sound equipment can be securely stored on-site. In many locations, however, the sound equipment will have to be brought in on the day that setup is to begin. It is beneficial, therefore, for the sound overseer to verify arrangements for the transport of the equipment to the facility. When is the equipment scheduled to arrive? Can it be packed last on the truck so that it can be unloaded first? Clearly, this is a case where a few advance phone calls to responsible brothers can save significant time later.

Installation Day

13. Many find it beneficial to begin the setup day with a brief meeting. As mentioned at the beginning of this chapter, relevant safety points from Chapter 5 may be used as a basis for this meeting. In addition to safety points, reminders about care of and concern for the building and equipment would be helpful. In most cases, this meeting should take no longer than a few minutes. If crews have been previously assigned, it should prove quite easy to dismiss the meeting and have everyone immediately know what they will be doing and with whom they will be working. When the work has been delegated properly, the sound overseer will not feel that he must do everything himself. A good sound overseer should be free to step back and truly *oversee* the work, assisting in areas where he is needed most and periodically checking the overall quality of the work. His ability to oversee and keep everyone busy will play a large part in getting the job completed on time.

Receiving Equipment

14. All sound equipment should be carefully inventoried and never be left unattended in an unsecured location. Complete and accurate records of all transfers of sound equipment are important in order to prevent confusion as well as loss of time or equipment. A responsible brother should be assigned to maintain a complete inventory of the various components received and used throughout the convention.

15. Any who handle the equipment should also realize that *all sound equipment is fragile*. The components that make up a sound system are not made to be dropped, tossed, or stood on. Sound equipment is expensive, and each piece of equipment should be treated as if we had to pay for it ourselves.

16. In many cases, sound equipment may be provided from several different sources. Perhaps the branch supplied the bulk of the equipment but a few pieces came from an Assembly Hall and other pieces from a Kingdom Hall. All equipment should be carefully returned to its owner in good condition. Identification tags and other methods of labeling can and should be secured to each component without damaging it.

17. All special packing crates, cartons, packing materials, and fillers should be stored in a secure location during the convention. This will prevent needed packing material from being discarded or misused and will save considerable time when dismantling the system.

Installing Primary Audio Cables

18. When running audio cable, attention should be given to a number of factors. For instance, some may feel that running any cabling on the floor will prove to be a trip hazard. They may wish to suspend all speaker wires overhead from speaker to speaker. However, this will result in an unsightly appearance, and if one speaker is knocked over, others will likely be pulled down. Securing speaker wires neatly on the floor, under seating wherever possible, is usually the safest option. It should be noted that in some areas local governmental agencies may dictate how cabling may be installed in public places and we would always want to abide by any such regulations.

19. In another instance, a microphone snake (multipair cables that carry audio from the stage to the mixer) may need to be placed in an area that must be accessible to forklifts or other large equipment. In this case, it may be necessary to run this cable overhead. Obviously, there is need for good judgment and good communication with the Convention Committee.

20. As when mounting equipment, proper respect should be shown for the convention facilities when running cables. If the choice is made to tape cables to the floor, we should be sure that the tape can be removed without causing damage or leaving a sticky residue. For example, duct tape may be relatively inexpensive and readily available in most areas, but it is very difficult to remove. For this reason, it is not recommended. Painter's tape tears very easily and does not have the adhesive strength required for installation use. Gaffer tape is an excellent choice for our needs even though it is more expensive. It is very strong and can be cleanly and easily removed. Cables that are run in high-traffic areas or cables that might be more sensitive, such as microphone cables, may need additional protection or guards over them. Discretion must be used in the purchase or construction of such cable guards to ensure that they will fully protect the cables, keep different signal types separate, and not pose a trip hazard to our brothers.

21. Those who will be responsible for running all the cables should be sure to label both ends clearly and carefully. This basic yet often overlooked step will allow equipment to be connected quickly and accurately later. The cables should also be run neatly. Whenever they can be placed under seating or in other locations that afford a measure of protection, we should do so. When connecting speakers and speaker zones, proper observation of polarity is essential. Systems that have complicated wiring, such as series-parallel systems, will especially need to have the speaker cables checked very carefully.

22. Consideration must also be given to the types of cables that will be run in close proximity to one another. In some cases, several cables are used for similar purposes, as in the case of speaker wiring that feeds a large distributed system. In this instance, the cables are all of the same signal type and there is a negligible effect if they are all run close together. However, there may be several different signal types in other cases. For instance, the cabling to the stage may include a microphone snake, a 70-volt speaker cable for stage monitors, and an AC power cable for a clock. If all these cables are placed close together and run for even a short distance, there will almost certainly be problems. In this example, the 50/60 cycle hum from the AC power will likely be picked up by the microphone snake, causing a significant hum in the system whenever the level on one of the microphone inputs is raised. Loudspeaker cabling should also be separated from microphone or other low-level circuits in order to prevent similar problems. As a general rule, *cables carrying different signal types should always be separated by a minimum of six inches* (*15.24 cm*). If they must intersect, this should be done at a right angle.

23. In a large installation, it may be necessary to run speaker cables for hundreds of feet. These long runs of cable can cause significant power loss unless the time is taken to size them correctly. Before specifying the size of speaker cable to be used, please review Figure 4.12 and the accompanying section in Chapter 4.

Installing Speakers

24. Various ways of mounting speakers are used throughout the world. One thing that should always remain constant, however, is the care that we show for convention-site property. In most cases, conventions are held in rented facilities. We want to make sure that we show consideration for building management personnel and respect their wishes when it comes to mounting and installing equipment. If speakers are to be mounted to the building structure, we should be careful to do it in such a way that we will not damage any paint or other finishes. If we are outside on a playing field, we should be careful to protect the turf. Regardless of how well maintained the building may look when we arrive, it should always look better for the final walk-through following the convention.

25. If it is necessary to manufacture our own speaker mounts, great care should be taken. Not only will the mounts need to have sufficient strength to carry the weight of the speakers but they must also allow for a safety factor of around three to four times that amount. For example, a loudspeaker that weighs 20 pounds (9 kg) would require a mount rated to carry a load of 60 to 80 pounds (27 to 36 kg). Any mounting bolts must similarly be appropriately sized for the loads they will carry. Washers should be used for improved stability, particularly when equipment will be bolted to wooden supports. If setting up outside, consideration must be given to any adverse conditions, such as rain or wind, that could threaten the stability of our mounts or damage our equipment. As mentioned at the outset of this chapter, even greater concern would be in order when mounting speakers overhead. We should always be careful to abide by all local laws governing the rigging of overhead components, and only well-qualified personnel should be used for this work. As an additional precaution, safety tethers may be installed on overhead equipment.

26. Tripods and towers for the mounting of speakers must be set up properly and never overloaded. The load must always be balanced on the tripod. Large speaker towers are usually designed in such a way that they can be precisely leveled. As shown in Figure 6.1, by using either a level or a plumb bob on two separate vertical axes, verify that the tower is exactly level before raising it to full height. Even when using smaller tripods and stands, always be careful to verify that they are on level, stable ground. For added stability, sandbags or other weights can be placed on each leg. In some areas, seismic or other factors will dictate that additional measures must be used to ensure stability. In all cases, local laws and the manufacturer's specifications on installation should be followed.

27. If possible, connect all the speakers and verify their operation *before* aiming them precisely and raising them to their

Figure 6.1 Always Verify That Tripods and Towers Are Level

full and final height. Speakers often have the simplest connections of any of our equipment, but there is still a need to connect them carefully and test them thoroughly. Always verify that the speaker is being connected to the proper zone. When installing constant-voltage systems, we may need to install a separate transformer for each individual speaker. A transformer connected backwards will reflect a virtual short circuit back to the amplifier, an error that must be corrected before the system can operate, so great care must be exercised in this regard. Wiring for correct polarity, although quite simple to achieve, is absolutely vital. As each loudspeaker circuit is completed, it is wise to measure the actual impedance before connecting it to the amplifier. Any significant departure from our calculated impedance may indicate a wiring error, which will obviously need to be corrected.

Racking Equipment

28. Likely, the system has been designed so that all the front-end equipment is together in one location near the stage. The mixer, MP3 player, and any other necessary pieces of equipment should be placed on sturdy tables so that they can be used comfortably.

29. Where much equipment is required, it is a good practice to rack-mount amplifiers and any other front-end processing equipment that does not require constant adjustment during the program. Since heat is a primary enemy of electronic equipment, any equipment racks should be well ventilated and no equipment should be left out in direct sunlight. Another precaution that should be observed when racking equipment is continued separation of different signal types. Many have found it helpful to run all the power cords on one side of the rack, while keeping all the audio cables on the other. Line-level cables and microphone-level cables should also be kept separate from speaker cables.

Connecting the Front-End Components

30. The connection of the front-end components is another area that should be handled by experienced personnel. The components must be carefully wired as designed and specified in the documentation. Carelessness in this area is frequently the cause of problems with the sound system.

31. It will be necessary to use a number of audio cables to connect all the system components. At some conventions, it has been observed that well-meaning brothers bring a number of miscellaneous audio cables of dubious quality. A closer examination of these sometimes reveals incorrect wiring, corrosion, bare wires, and a host of other problems. We should be sure that the cables used to connect the components are always of good quality. Additional information will be provided in Appendix 6, paragraph 10.

32. At times, it may become necessary to fabricate some cables. Will any balanced-to-unbalanced (or vice versa) cables need to be made? To assist in making these, Figure A7.15 in Appendix 7 provides a chart that shows some typical interconnecting cables and adapters. It is important that the person making these cables be familiar with proper soldering techniques, which are summarized below.

33. Solder is an alloy primarily composed of tin and lead, and it inherently has a low melting point. For electronics work, a ratio of approximately 60/40 (meaning approximately 60 percent tin and 40 percent lead) typically works best. There has been a trend toward using far less lead in solder, but this significantly raises the melting point and makes the solder more difficult to use for elec-

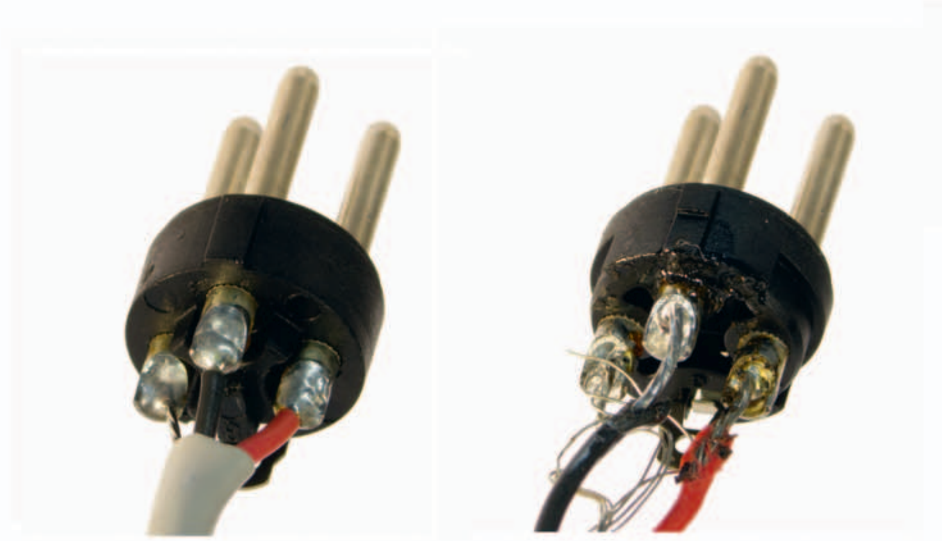


Figure 6.2 Proper and Improper Connections

tronics work. The solder we use generally has a rosin core (acid-core solder is not meant for electronics work and should never be used). The rosin acts as a flux, which works to clean the surface and helps the solder to adhere to the base metal. In order for solder to flow onto a material, the temperature of both the base metal and the solder must be raised above the fusing point of the solder. A point of caution is needed, however, because raising the temperature too high can result in melted insulation on a wire or damage to a connector. All surfaces to be soldered should be clean and relatively free of corrosion. After tinning the tip of the iron with solder, press the tip against the work until the point to be soldered reaches a temperature high enough to melt solder. Solder should then be applied to the work, rather than to the tip of the iron. Both the base and the wire should be sufficiently hot so that solder flows smoothly onto both of them. When both points to be soldered are the correct temperature, the applied solder will appear to flow *into* the joint, filling indentations and coating the wire evenly. If either point is not at the correct temperature, solder will appear to fall *onto* the joint and will not adhere fully. The correct amount of solder will flow into the joint and create a smooth and shiny surface. If too much solder is applied, it will blob or drip. Solder that has been overheated or cooled too quickly will appear gray and porous and will not be nearly as strong.—See Figure 6.2.

34. When making connections between front-end equipment, preference should be given to the use of balanced connectors whenever possible. Balanced wiring helps to reduce some external noise (such as radio frequency interference) and is generally used on professional audio equipment. The XLR cables used for good-quality microphones provide an example of *balanced* wiring. Each cable includes three conductors, which could be identified as positive, negative, and shield. On a balanced line, the shield is connected to ground and carries no signal. MP3 players normally use *unbalanced* wiring with RCA, or phono, connectors. Only two wires are needed in this unbalanced configuration, since the shield performs double duty by carrying the return signal in addition to being connected to ground. Interestingly, so-called constant-voltage speaker lines are *balanced,* despite having only two conductors. However, in most instances of interconnecting equipment, we can be confident that we are dealing with an unbalanced line when only two conductors are seen.—Please see Appendix 7, Figures A7.15 and A7.16, for more information on balanced and unbalanced connections.

35. Before connecting speaker lines to amplifiers, a quick check should be performed with an impedance measurement device. Any significant difference between the actual impedance and our calculated impedance may indicate a wiring error that must be corrected.

36. Many amplifier outputs make use of screw terminals for connecting to the speaker lines. Even with this very simple type of connection, correct installation is important. When attaching a conductor under a terminal screw, make certain that the wire is neatly twisted and not frayed. It should be wrapped under the screw just once and in such a way that the wire will wrap more tightly around the screw as the screw is tightened. The wire should not cross over itself, and excess wire can be snipped off. While not required, ring- or spade-type crimp terminals are convenient when using terminal strips and, when properly crimped, contribute to a clean installation.

37. Since it is not possible to give a detailed description of every possible audio connection within the scope of these pages, it is our hope that these general guidelines will be sufficient in most cases. There are, however, certain connections and installations that require particular attention. A few of these items will now be considered.

Installing FM Transmitters

38. FM transmitters are generally connected in the same way as any other piece of line-level equipment, but there are a few points to be particularly aware of. Always refer to the owner's manual regarding any special installation requirements.

39. At times, it is necessary for an FM transmitter to be placed in a remote location to provide better coverage. It is not always possible to run a separate line-level cable to such distant locations. When this situation arises, a transformer (such as the RDL TX-70A) can be used. This small transformer can be installed on a 25-, 70-, or 100-volt speaker line and will provide us with a variable unbalanced line-level output suitable for driving an FM transmitter. Since an FM transmitter connected in this way can no longer be separately equalized or compressed but must use the same processing as the speaker line on which it has been placed, this is not the first choice. However, if the situation dictates that a transmitter must be placed in a remote location and a constant-voltage speaker line is nearby, using a transformer may save us considerable time and effort.

40. When using FM transmitters, care must also be given to selecting a suitable frequency and adjusting the antenna properly. This will be further described in Chapter 7, paragraphs 18-20.

Connecting Computer Recording Equipment

41. Computerized audio-recording equipment can be connected in a number of ways. Some brothers may choose to use the line-in jack available on most computers, while others may make use of special converters that plug in via a USB or FireWire port. Regardless of the method used, it is extremely important to verify the quality of the signal being fed into the computer recording program. Some have also found it beneficial to place a compressor on the output feeding the computer for additional protection from overload during the playback of music.

Connecting to the House System

42. When it is necessary to combine our sound equipment with an existing house system, some precautions will certainly be in order. Always work closely with the management of the facility to be certain that we will not damage any of their equipment. We must also verify exactly what input level the house requires. As an example, serious damage could occur if we were to feed a +4 dBu line-level signal into a microphone input. Also, if the input into the house system is not transformer-isolated, a separate isolation transformer may be needed as shown in Figure 6.3.

Grounding to Prevent Noise Issues

43. Issues with improper grounding can wreak havoc on a sound system, and their cause can be somewhat elusive. Solid design and installation should do much to prevent these problems. For instance, as mentioned in Chapter 4, paragraph 56, all circuits used to power the front-end equipment should be taken from the same panel. Similarly, although rack-mounting the equipment does not guarantee success, it does help to ensure that all components have a common ground. Using careful wiring techniques, providing adequate shielding, and avoiding ground loops are also among the precautions that can be taken to avoid needless issues.

44. Ground loops are commonly introduced when tying into another system. For example, this problem often presents itself when convention sound equipment is connected to an existing house sound system. What can be done to prevent such issues?

45. Lifting the shield ground at one end of the transmission line may help to reduce hum and noise, but it does not ensure immunity from such problems. Noise-free operation is best ensured by the use of transformer coupling between the systems. In some cases, this isolation may already exist. For instance, a venue may have a professional mixer in which each individual input is transformer-isolated. When this is the case, no additional isolation should be needed. If this isolation does not exist or is in question, then a suitable isolation transformer should be used. It is preferred that the transformer be installed at the input point of the house system and connected as shown below in Figure 6.3. Please note that the *shields are not connected* at the transformer. As a reminder, because isolation transformers are sensitive to electromagnetic interference, they must not be placed near other transformers, power supplies, motors, or lighting fixtures.

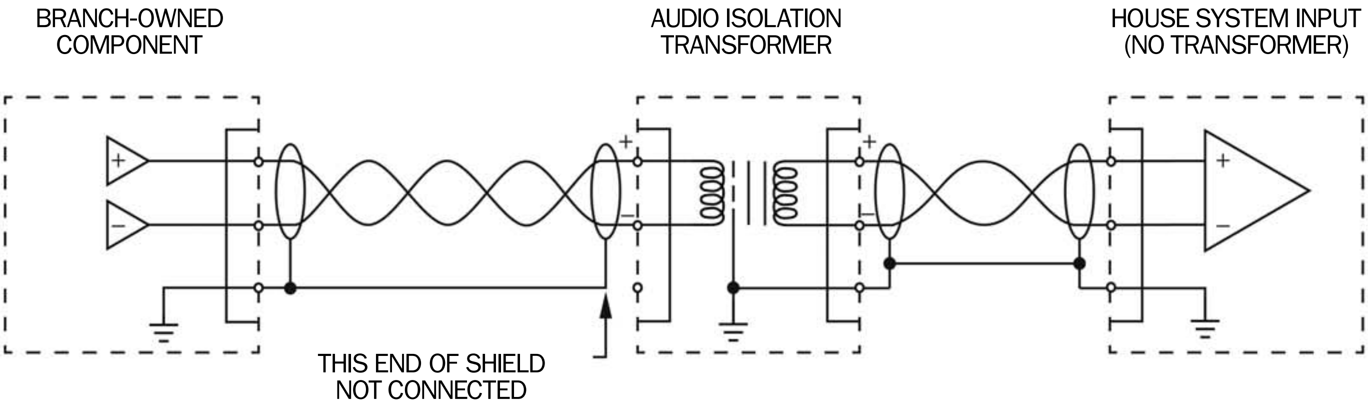


Figure 6.3 Connecting to an Existing House Sound System

Grounding for Safety

46. Not only is proper grounding essential for keeping noise out of the system, it is also essential for safety. At times, well-meaning brothers will use a three-prong-to-two-prong AC ground lift adapter in an effort to minimize hum in the sound system. In some instances, the grounding pin of the AC cord has even been cut off completely. Ground lift adapters, if they must be used at all, should only be used to assist in identifying the source of ground loop problems. They should *immediately* be removed once testing is completed. Out of respect for life and safety, ***never*** permanently remove an AC safety ground.

47. While most soundmen would consider themselves quite knowledgeable about power-related issues, a qualified electrician should always be consulted whenever significant power problems surface.

Testing the System

48. Once everything has been properly connected, the system can be tested. In order to prevent unnecessary and possibly damaging pops, the sound equipment should always be powered on in the correct sequence, with low-level equipment turned on first and amplifiers last. When shutting down, this process is reversed, with the amplifiers being turned off first, followed by the low-level equipment.

49. Initially, all the low-level sources such as microphones and the MP3 player should be tested through the mixer for operation. Be sure to make a *full* test of each microphone, cable, mixer input, MP3 player, and so on. Everything must be thoroughly checked and properly identified with an appropriate label.

50. Once we are satisfied that the signal-processing chain is functioning correctly, we can verify the speaker zones and individual speaker operation. An audio signal generator set to output pink noise can be used for this test. Turn up the level for one zone at a time and then walk past every speaker in that zone. If some of the speakers in the zone are *not* working or if any speakers outside the zone under test *are* working, it may be an indication of a wiring error. When it has been confirmed that all the speakers in the zone under test—and only those speakers—are working, we can then check the sound of individual speakers. When walking through the zone, there should be very little difference in sound from speaker to speaker. If there is a significant difference in level or equalization between speakers, further testing should be done to identify and correct the problem.

Aiming the Speakers

51. It is critical for our speakers to be carefully located and aimed according to the design. What a waste it would be to spend much time and effort in preparing a quality sound design and then fail to install it according to plan! When a design stipulating mounting locations and aiming angles for the speakers has been provided, we should always carefully follow such direction. To do that, it will likely prove helpful to have on hand a sufficient number of tape measures and inclinometers, or “angle-finders,” to make the necessary adjustments accurately. (See Figure 6.4.) In some large facilities, a small laser can be used to project a beam from the speaker to a specific location, thus allowing for precise aiming of horns that will be located a considerable distance from the audience. Remember that all lasers, even low-power models, are dangerous to the eyes if used improperly. Carefully adhere to all the manufacturer’s safety guidelines for their use.

52. Once the speakers have been properly raised and aimed, they should be locked down in their final position. It should also be verified that all speaker wires are neatly and carefully secured and that all areas in which work has been done are clean and orderly.

Figure 6.4 Using an Inclinometer to Adjust Speakers

In Conclusion

53. As we have seen, the installation of a sound-reinforcement system truly requires a group effort. From the overseer to the unskilled volunteer, any who will be used for this task must be able to follow direction. The ability of these ones to communicate effectively and work together harmoniously will play a large role in making the installation a success. However, the installation is not truly complete until the system has been fully optimized. This will be discussed in the following chapter.

CHAPTER SEVEN

Optimizing Sound System Performance

1. Without a doubt, the responsibility of providing quality sound to thousands of listeners is a weighty one. It should also be acknowledged that a number of factors beyond our control may limit the functionality of a particular sound system. For instance, certain desired pieces of equipment may prove prohibitively expensive or we may not be able to place the loudspeakers exactly where we would prefer. Happily, one key factor is always under our control—we have the ability to make the available equipment and design sound as good as they possibly can.

2. At times, reports have been received that a convention experienced sound problems on the first day but that things were better by the last day. This is not acceptable. The sound system should be completely set up, balanced, and ready well before the opening session. Some might feel that it is not possible to optimize the sound system without first having the location filled with people. While this does present a challenge, the arrival of the crowd minimizes reflections in the room, which will generally only improve the sound quality. It has been proved time and again that a properly optimized sound system will usually require only minor adjustments once the crowd arrives. We can follow this guiding principle: If we are able to make the system sound *good* with an empty room, the quality should be *excellent* with a crowd. In this chapter we will discuss steps that can be taken to optimize the sound system's performance fully.

Gain Structure

3. To put it simply, every piece of audio equipment has a certain level at which it wants to "hear" an incoming signal. Also, each piece of equipment outputs sound at a certain level. Sometimes it is a "whisper," like the scant signal provided by a microphone output. At other times it is more of a "shout," like the strong line-level signal of professional broadcasting equipment. Problems will occur when a piece of equipment expecting a whisper receives a shout or when equipment needing a shout hears only a whisper. With audio equipment, too much level will likely cause clipping and distortion, while too little will likely result in hiss and excessive noise. To get a better idea of the types of issues that might be encountered, please note three examples of gain-structure problems.

- **Problem:** A sound system with a gated mixer works fine when all the inputs are tested individually. During the first interview of the convention, with several inputs turned up at once, the mixer fails to gate the correct microphones.
  The reason: The mixer's microphone inputs were set to line level instead of microphone level. The incoming signal was so

low that the mixer considered it negligible and therefore did not switch over when a microphone input was used.

- **Problem:** An amplifier with plenty of power for the desired speaker load has been specified, and yet, with the gain controls set to maximum, it is still unable to drive the loudspeakers at the desired level.
  The reason: The amplifier is being fed consumer line level (-10 dBu) instead of the required professional line level (+4 dBu).
- **Problem:** An FM transmitter sounds fine during most of the program but distorts whenever music is played.
  The reason: The increase in level during the musical portions of the program overdrives the input and causes distortion.

4. Every piece of equipment should receive the appropriate level throughout the entire signal path. How can the sound system be adjusted correctly to meet this goal?

5. It would be a mistake to think that we should simply adjust all processing equipment to "unity gain," that is, the point where the level of what comes in is equal to the level of what goes out. Granted, this may work on occasion. However, since audio equipment has many different input- and output-level requirements, consideration should always be given to proper system gain structure.

6. To begin, we should check the input and output specifications for each piece of equipment. This check may quickly identify where there will be a problem. If one piece of equipment outputs an extremely high level and the device after it requires a low level, something must be done. Is there some way the sound level can be adjusted on the equipment? If not, an inline pad may be needed to reduce the level. Perhaps the level is extremely low and there is no way it can be increased by the previous device. In that case, a line amplifier of some sort may be necessary. Is the output balanced and the input unbalanced? This, too, could cause problems and was discussed in Chapter 6, paragraph 42.

7. In most sound systems using modern equipment, gain structure can be adjusted quite simply. For instance, if all our processing is handled digitally, there may be just one device between the mixer and the amplifiers. Additionally, many signal processors allow levels to be adjusted and include meters that allow easy verification of signal levels.

8. There are many possible ways to optimize gain structure. Some methods are extremely complex and require a number of specialized tools. The following information, however, provides a very basic but reasonably accurate method of adjusting system gain structure and is primarily meant to demonstrate the principles involved. Once the key principles are understood, it should become apparent what needs to be done when we are confronted with any number of different system configurations. To perform these adjustments, we will need a multimeter and a 1 kilohertz (kHz) tone from either a test CD or an audio signal generator.

9. Start with all amplifier volume controls turned completely down and all equalization and other processing equipment bypassed. Beginning with the mixer, the goal is to optimize each processing component so that we can keep the signal level as high as possible while maintaining the desired headroom.

10. If a console-style mixer is being used, there will be an input gain/trim control for each channel. Please note that this is not to be confused with the main fader for each channel. The gain/trim control is a knob used for input gain adjustment only. One by one, adjust each channel's main fader to "0" and have someone speak into the corresponding microphone at the *absolute maximum* level that will be used on the program. The input gain/trim should then be adjusted until the overload (or clip) light only flickers occasionally. These steps should be repeated with all source equipment on the mixer to ensure the best possible signal-to-noise ratio.

11. When we work with all the individual pieces of processing equipment, we must be sure to keep in mind the input and output specifications that were mentioned previously. The 1 kHz tone should be injected into the mixer and the mixer turned up so that any meters on it read "0." Test the output of the mixer by checking the AC voltage on pins 2 and 3 of the XLR connector. (On an unbalanced connector, the tip and sleeve can be used for testing.) Does the output voltage equal the proper input voltage for the next piece of equipment? If the required level is being supplied, any meters on the next component should also read "0" once the component has been connected. Now test the output level for this piece of equipment. As was done previously, verify that the output voltage is equal to the proper input voltage for the next item in the signal chain. This process should continue through all pieces of equipment until it has been verified that each piece is receiving the required input voltage. Additionally, whenever an external meter (such as a VU meter) is used, it must be verified that it also reads "0." Once this has all been completed, the overall system sound pressure level (SPL) can be adjusted using the amplifiers.

12. As a final check, listen carefully for any unusual amount of hiss or noise coming from the system. If everything seems acceptable, music can be played at a level simulating the loudness needed to lead the singing during the convention. Is any distortion heard? Verify that none of the equipment indicates clipping. Next, have someone speak into the main microphone at a high level similar to what the session chairman would use when gathering the attention of the audience. Again, is any distortion heard? If none of the equipment indicates clipping, we will likely have sufficient headroom for the program.

13. The key to adjusting system gain structure lies in confirming that there is neither *too much* nor *too little* signal appearing at the input of each piece of equipment.

Adjusting Dynamic Processors—Limiters and Compressors

14. In order to understand the role dynamic processors play in a sound system, it may be helpful to think of it in the following way: In a sound system, we are concerned with both frequency and level. Frequency is adjusted by the use of equalizers, while constantly changing levels can be tamed by dynamic processors, primarily limiters and compressors. Because these devices influence level, they must be carefully adjusted so as not to cause unexpected problems. When properly set, they can protect equipment and reduce clipping, such as when a compressor is used ahead of an FM transmitter. Proper gain-structure optimization requires the adjustment of any limiters or compressors in the signal path.

15. Compressors and limiters are primarily differentiated by the way they are used and typically have a few controls in common. The four main adjustments include threshold, ratio, attack, and release.

(1) ***Threshold*****—**This describes the level at which the unit will begin to apply processing. This term can be used to refer to either a maximum or a minimum level.

(2) ***Ratio*****—**This describes how much processing will be applied once the threshold is reached.

(3) ***Attack*****—**This describes how quickly the unit will react once the threshold is reached.

(4) ***Release*****—**This describes the amount of time before the unit ceases processing once the levels fall below the threshold again.

16. Limiters are used to provide a ceiling, or maximum, for the level. They are helpful both in protecting equipment and in preventing distortion. When used in convention systems as shown in Figure 4.21 of Chapter 4, extreme care must be used in their adjustment. For example, if set up improperly, a limiter could prevent a chairman from being able to speak with enough level to catch the attention of the audience. To avoid such a problem, the threshold must be set very high. During testing, always verify that the threshold is set high enough that it will not be exceeded during normal operation. Since the limiter is in place to act as a ceiling, the ratio will typically also be set high and the attack and release times will be set to react quickly.

17. Compressors are used to reduce the overall dynamic range of the program. For our purposes, they are only to be used as a means of scaling the wide dynamic range of the program so as to match audio equipment with a smaller dynamic range (such as FM transmitters) and are not recommended for use on the primary inputs. Improperly set compressors can increase breath sounds from the person at the microphone and create a "pumping" effect during the program. Compared to limiters, they will be adjusted with lower threshold and ratio settings and the attack and release times will be moderate. In the hands of someone skilled in their operation, compressors can be a useful tool. In the hands of someone who *thinks* he is skilled in their operation, they can cause many problems.

Optimizing FM Transmitters

18. A portable radio should be used to find a suitable open frequency. Not only must the specific frequency be clear but adjacent frequencies should also be unused if at all possible. Once the selected frequency has been set on the transmitter, some units require adjustments to the antenna. Always refer to the manufacturer's instructions regarding special requirements for installation. Please note that each transmitter requires its own specific frequency when using multiple transmitters. Never attempt to run separate transmitters on the same frequency.

19. Using the procedures outlined previously regarding optimizing system gain structure, confirm that the levels are properly adjusted for the transmission of both speech and music. If a compressor is used ahead of the transmitter, this must also be verified and tested before use.

20. Some in the audience may use an FM receiver even though they are not hard of hearing. A 30-millisecond delay placed before the FM transmitter will soften the effect of any delayed sound.

Adjusting Delays

21. A well-designed system will list the approximate delay times for different zones, and this provides an excellent starting point for optimizing. In some cases, though, there may be a need to match the timing of the existing house system with sound equipment that was installed specifically for the convention. In such a case, it likely will not be possible for the designer to provide an approximate delay time. In both cases, how can delays best be optimized?

22. At times, it may be possible to measure the distance between the house loudspeakers and the newly installed speakers. Once careful measurements have been made, the formulas found in Chapter 4, paragraph 36, of this handbook will prove helpful in determining an initial delay time.

23. Determining whether the delays have been properly adjusted can be difficult with voice or music programs. A click track can be used as an aid in adjusting delays. It consists of a click, or pop, approximately every second and provides us with an easily distinguishable sound for comparing arrival times at a specific location. While a listener stands in the coverage of the two zones to be synchronized, the click track can be played and the delay times slowly adjusted until the clicks from both zones seemingly arrive in unison.

Balancing Initial Loudspeaker Zone Levels

24. A perfect sound system would cover every seat with direct sound at exactly the same level. This will not happen in the real world. Some have claimed that they were able to walk through an arena and notice a difference of only 3 dB-SPL between any two seats in the house. It should be noted, however, that an empty building will provide an enormous amount of reflected sound, which an SPL meter cannot distinguish from direct sound. An SPL meter also cannot account for noisy ventilation or a host of other ambient noises. Knowing this helps us to understand why an SPL meter may display a reasonable 75 dB-SPL and yet some will still complain that they cannot hear the program well. Be assured that there will *always* be differences in the level of direct sound throughout a convention site. Our goal is to minimize those differences as much as possible. How can this be done?

25. For this test, a pink-noise generator and an SPL meter will be needed. It will also be helpful to have two-way radios on hand so that the person checking levels can quickly relay information to the one making the adjustments at the amplifiers.

26. There is no way to shortcut the process of adjusting the levels for each speaker zone. The only truly accurate way to determine levels is to walk to each area and listen until the proper level is achieved. This does not mean that an SPL meter should not be used but, rather, that the meter alone will not tell us if we are in the direct field of sound. Additionally, the interaction between all the loudspeakers in the house cannot be overlooked. As the volume is increased in a zone, areas outside that zone will also need to be rechecked. One check should be performed when the initial zone levels are set, another check will be needed once the entire loudspeaker system is operating, and another check once the system has been properly equalized.

Equalization

27. Equalizing a sound system involves selectively increasing or decreasing certain frequency bands in order to improve the overall quality of the reproduced sound. Our primary goals are to optimize intelligibility and naturalness while still providing sufficient headroom before feedback. Let us briefly discuss those three goals before we see how equalization can help us reach them.

(1) ***Intelligibility***—When we say that a sound system is intelligible, we mean that it reproduces speech in such a way that it can be easily understood. This means that not only must there be enough level for the sound to be easily heard but also the sound must be sufficiently articulate. For theocratic events, our primary concern is intelligibility.

(2) ***Naturalness***—Second to intelligibility, naturalness is our goal. Preferably, a sound system would not color, or alter, the sound it is intended to reproduce. When perfect naturalness is achieved, what goes out of the system sounds exactly like what goes in, only louder. Equalizing for naturalness cannot be our main concern, however, because of the size and construction of the facilities used for conventions and the limitations of sound reinforcement equipment. As discussed further in Appendix 3, the acoustic environment can exaggerate certain frequencies, greatly impeding intelligibility. We will gladly compromise a degree of naturalness to ensure that the program is intelligible.

(3) ***Headroom Before Feedback***—Acoustic feedback occurs when sound from the loudspeakers arrives at the microphone both in phase and with amplitude at least equal to that of the originating voice. It may be helpful simply to think of it as occurring when the microphone "hears" approximately the same level from both its input and the loudspeaker system. (See Figure 7.1.) Clearly, feedback directly relates to level, but interestingly, it is primarily the result of individual frequencies at a certain level. Any frequency exaggerated by either the system or the room will be a frequency at which the system will have a tendency to produce feedback. This helps us to understand the part proper equalization plays in providing sufficient headroom before feedback.

28. It should be noted that equalization is neither the only nor even the *preferred* way to reduce feedback. In Chapter 4, we discussed how locating the stage outside the zone of direct coverage by the main loudspeaker system will help to prevent feedback. By also designing the system with the directional characteristics of the loudspeakers and microphones in mind, we can limit the interaction between them. In Chapter 8, we will discuss how keeping the microphone close to the program participants will provide us with yet another way to reduce feedback. All these methods of reducing feedback are superior to equalization adjustments. The situation illustrated in Figure 7.1, therefore, poses both a design problem and an operational problem that cannot be fixed solely with equalization. However, even once these issues are corrected, the fact remains that our conventions are typically held in highly reverberant environments and there will *always* be considerable *reflected* sound on the stage that must be accounted for. For our purposes, we should consider that proper equalization, so as to provide adequate headroom, is essential.

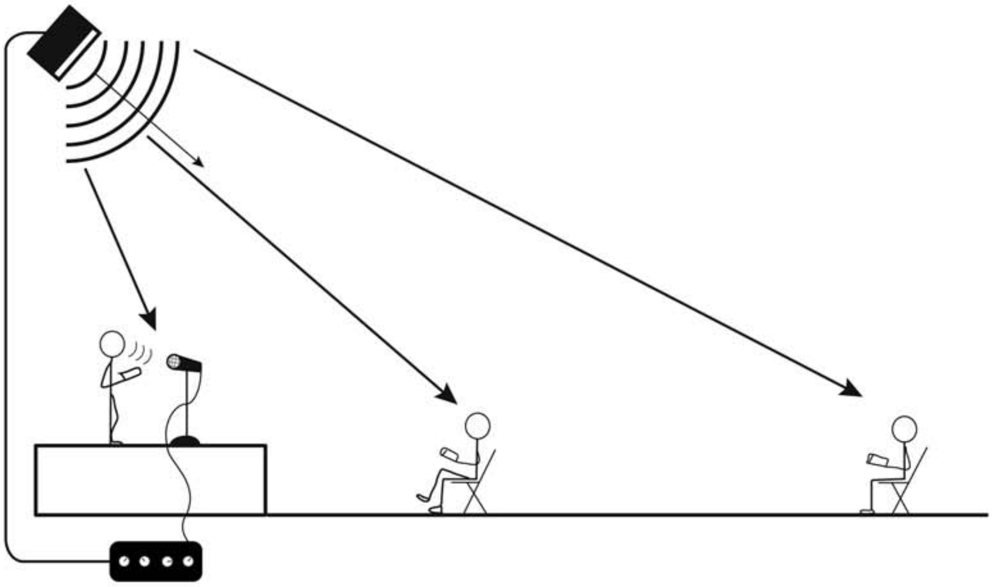


Figure 7.1 Acoustic Feedback Path Caused by Poor Speaker Placement

29. To operate a system even on the *verge* of feedback is highly undesirable, as quality and intelligibility will be greatly reduced. A system operated close to feedback will sound hollow, and words will trail with a slight ringing sound. A reasonable and safe goal would be to provide sufficient level for the audience while still operating the system at least 6 dB-SPL below the onset of feedback. Before discussing how that goal can be achieved, we must first learn more about the devices used to adjust equalization.

30. *Equalizers—Graphic or Parametric?* Equalizers were briefly discussed in Chapter 4. If the system has already been designed, the choice has likely already been made as to whether graphic or parametric equalizers will be used. However, it is still important to understand the differences between the two and what they offer in terms of reaching our equalization goals.

31. Graphic equalizers are very simple to use. Each slider is set to a frequency and can either boost or cut that frequency. The problem is that graphic equalizers adjust the sound in very broad strokes. For example, a 1/3-octave graphic equalizer (sometimes called a 31-band equalizer) is typically the narrowest-band graphic equalizer available. That 1/3-octave designation refers only to the spacing of the filter centers (each separated by one third of an octave) and not to the width of the filter. The width of a typical filter on a 1/3-octave equalizer is about one full octave! Therefore, graphic equalizers are useful, but only to the extent that it is acceptable to change a broad range of frequencies with each adjustment.

32. On the other hand, a parametric equalizer allows the operator not only to decide the amount to boost or cut but also to choose the exact frequency that will be adjusted and the number of surrounding frequencies that will be affected. Although this makes the parametric equalizer a bit more difficult to use, it affords an enormous amount of precision when making adjustments.

33. In summary, though graphic equalizers are easy to use, they impose certain limitations that make them impractical for removing feedback. A parametric equalizer can make adjustments that a graphic equalizer is incapable of, and it would be difficult to name something a graphic equalizer can do that a parametric equalizer in the hands of a skilled operator cannot. In the end, the type of equalizer used may be decided for us, based on equipment availability. Regardless of which type is chosen, careful optimization of the system can result in an acceptable quality of sound.

34. *Necessary Tools and Preparation for Testing:* Admittedly, there may be rare instances where all equalization adjustments must be done by ear because of a lack of test equipment. In recent years, however, improvements in consumer electronics have made simple audio-test equipment significantly more affordable. For the tests that we will describe below, a real-time audio spectrum analyzer and a pink-noise generator or test CD are recommended.

35. Because of the sensitivity of the test equipment, not to mention the noise created during testing, it is recommended that all equalization be performed when the test areas are reasonably quiet and free of crowds. This description of testing procedures also presumes that the sound-system design principles provided within this handbook have been followed and that equalizers have been placed in the system to allow speech, music, and different types of speakers to be equalized individually.

36. *Adjusting Equalization for Loudspeakers:* All loudspeakers have specific frequencies that they tend to exaggerate. As we have learned, any frequency exaggerated by the system tends to produce feedback. As an example, note Figure 7.2, which shows the frequency response for a two-way loudspeaker.

37. As is true of all loudspeakers, the speaker in our example does not have a perfectly flat frequency response. For instance, a peak in the response can be observed at 5 kHz. A smaller peak can be seen at around 600 hertz (Hz). Issues with the acoustic environment notwithstanding, we can be sure that these two frequencies will be among the first to experience feedback.

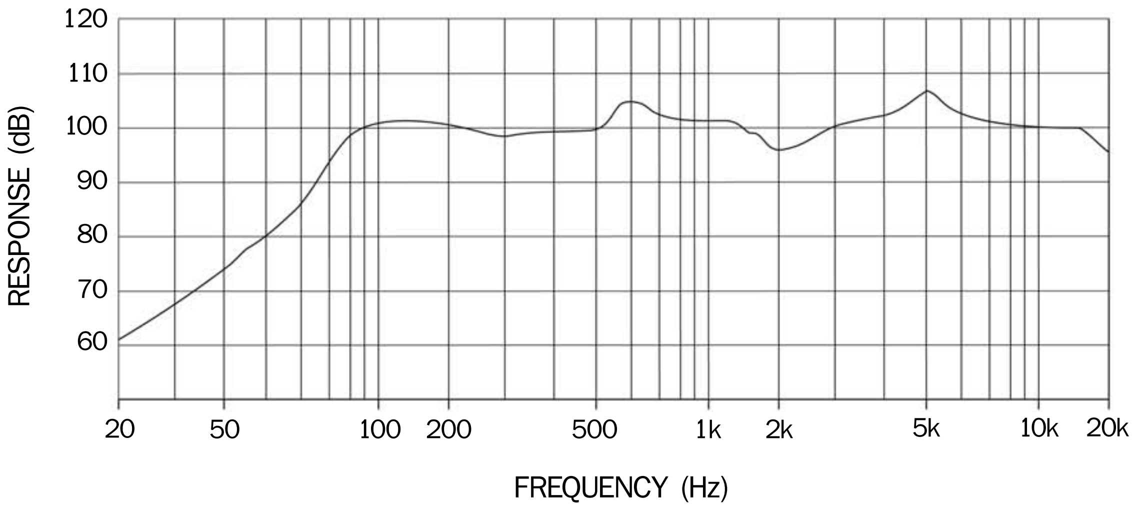


Figure 7.2 Example of Two-Way Loudspeaker Frequency Response

38. We may also note that the frequency response of this speaker drops off abruptly below 100 Hz. This is not a problem but, rather, should be expected. In this instance, the speaker only has an eight-inch (20 cm) transducer for the low frequencies and therefore becomes less and less capable at lower frequencies. No amount of boosting the lower frequencies during equalization will ever make up for this lack. There is also a dip in the response at around 2.2 kHz. In this case, that is where the crossover frequency is located. Everything above 2.2 kHz is sent to the horn, and everything below is sent to the woofer. Try as we may, we will never be able to boost this particular frequency, and it should also be left alone.

39. The goal of equalization is to flatten the frequency response of the speaker to a reasonable degree, keeping in mind that it will never be perfectly flat. As we have seen, there will be some frequency-response issues that can be corrected, while others cannot. As a rule, all equalization adjustments for loudspeakers should be minor. With loudspeakers of reasonable quality, excessive adjustment of equalization will not be necessary.

40. Before testing, the area should be reasonably quiet and all equalizers should be flat. The test signal (in this case, pink noise) must be at least 10 dB-SPL above the ambient noise to ensure that our measurements will be valid. With pink noise being played only by the speaker under test (or speakers of the same type in the same zone), take frequency readings directly in front of the loudspeaker at a distance of about 6.5 feet (2 m). This distance is sufficient to allow the sound from both transducers to combine for two-way systems and yet is close enough to ensure that we are equalizing the speaker and not the acoustic environment. Make a

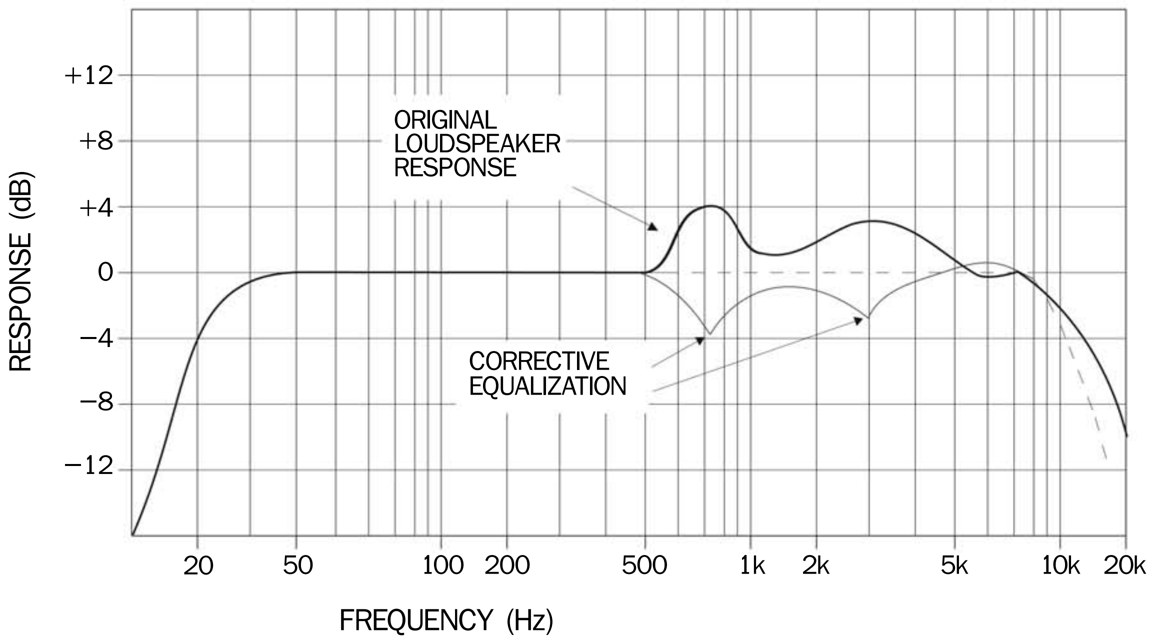


Figure 7.3 Example of Frequency Adjustments to Correct Loudspeaker Response

note of any frequencies that seem to be unusually high compared to others. It is also beneficial to take readings from multiple speakers so as to verify that any anomalies are not limited to a single speaker but are common to all speakers of that type.

41. On the equalizer for that zone only, make *minor* adjustments (typically not more than 6 dB boost or cut) to correct the anomalies. So as not to alter the sound unnecessarily, these frequency adjustments must also be as narrow as possible, affecting only specific frequencies. (See Figure 7.3.) Parametric equalizers are preferred, as they afford the most control over the width of the frequencies adjusted. With quality equipment, typically only a few small adjustments will be needed in order to flatten the response sufficiently. Once the adjustments have been made, additional readings should be taken and additional corrections made as needed. It is not uncommon for this cycle to be repeated three or four times in order to confirm that the adjustments are correct.

42. Whenever multiple equalizers are found in the same signal chain, care must be taken not to overequalize. With that in mind, on *output* equalizers there will typically not be a need to roll off the high or low frequencies, as this will be done separately for speech and music earlier in the signal chain. The exception to this would be outputs with specific high- or low-frequency drivers, as well as zones where additional roll-off is necessary. In that case, appropriate roll-off or crossover filters should always be used.

43. When the proper equalizer settings for a certain speaker have been determined, adjust the equalizers for all zones (aside from stage monitors) that use the *same type* of speaker the exact same way. However, for each *different* type of speaker used, we will need to perform frequency readings using the spectrum analyzer and pink noise to make adjustments as previously outlined.

44. Once all loudspeaker output equalizers have been adjusted, the system should sound natural. However, intelligibility will be lacking until the system has been properly equalized for speech.

45. *Adjusting Equalization for an Existing House System:* At times, the existing house system will be used either by itself or in conjunction with supplemental sound equipment. To test the system, pink noise can be played over the house system only. Using a real-time audio spectrum analyzer, readings can be taken in a few sections covered by the house loudspeakers. Usually, an equalizer feeding the house will need nothing more than a few minor adjustments. Excessive adjustment to an external equalizer feeding the house should always be avoided. Likely, the house system already has a number of equalizers in place, and excessive adjustment to an additional equalizer will not help. If major adjustments seem to be required, it may be that some adjustments to the actual house equipment are necessary. Unless previous permission has been granted, we should never take it upon ourselves to make adjustments to equipment that belongs to the convention facility. The convention representative can contact the building's management to see if any adjustments are permissible. Only after receiving such permission may any adjustments be made to a facility's existing sound equipment.

46. *Adjusting Equalization for Speech:* There are two goals when equalizing for speech:

(1) Passing only the frequencies necessary for intelligible speech.

(2) Compensating for any harmful room acoustics. (See also Appendix 3.)

47. If the system were being optimized solely for the playback of prerecorded material, then we would likely be quite happy with a flat response across the entire frequency range. However, aside from music (which will be discussed later), the frequencies that we are primarily concerned about reproducing comprise only a relatively narrow swath across the middle of the audible sound spectrum. For the majority of our speech programs, any sound that passes through the system beyond the spectrum needed for speech is unnecessary and possibly detrimental to intelligibility. How can speech equalization be properly adjusted?

48. To begin, adjust the equalizer for speech as shown in Figure 7.4, as this will provide a basis for all future adjustments. The high frequencies can be rolled off at about 8 kHz. This is well outside the range of normal speech and will not affect the intelligibility of speech at all. To improve intelligibility, we will be far more interested in the low-frequency roll-off. Initially, the low-frequency roll-off should be set at 200 Hz with a slope of 12 dB per octave. This is only a starting point, as even a moderately reverberant room will need to be adjusted to roll off at a higher frequency. With someone reading into the main microphone, slowly adjust the roll-off to higher frequencies. There will be a point when intelligibility greatly improves because of the reduced low-frequency response. Though we begin at 200 Hz, there may be a need to go as high as 500 Hz or so. As a rule, the more reverberant the room, the higher the roll-off frequency will need to be. When making these adjustments, real-time analyzers can do little to help. We must use our ears to establish when the roll-off begins to improve intelligibility. Once the appropriate low-frequency roll-off point has been determined, confirm optimal intelligibility by having several different brothers and sisters read over the system.

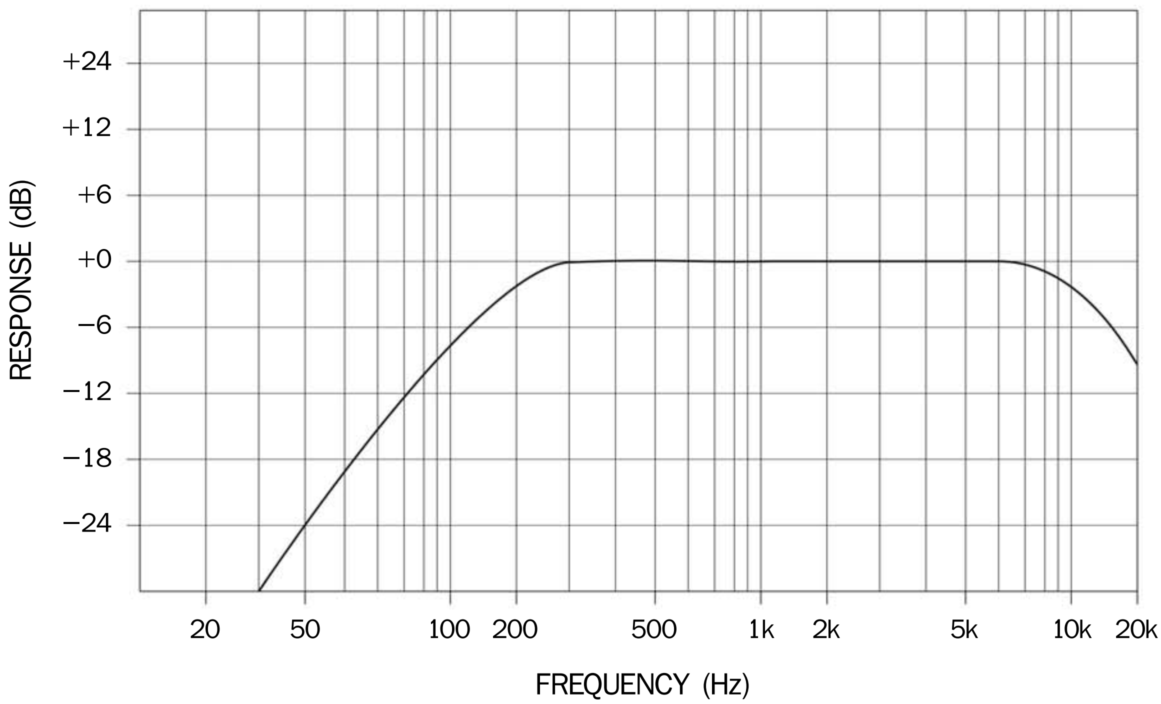


Figure 7.4 Recommended Starting Points for High- and Low-Frequency Adjustments

49. *Adjusting Equalization for Music or Other Prerecorded Material:* Portions of our program make use of prerecorded material. For example, prerecorded audio is used for dramas and whenever music is played to open the session or to lead the singing. There is no need for concern about possible feedback during dramas or music because microphones are not being used. However, a slight equalization issue may arise because any adjustments depend largely on whether the goal is naturalness or intelligibility. With music, intelligibility is not an issue and so the desire typically is to provide the most natural sound possible. However, intelligibility is crucial for an audio drama.

50. At times, it may be necessary to make slight adjustments to the equalization for the dramas or musical portions of the program. For instance, loudspeaker horns have a tendency to exaggerate high frequencies, which can make stringed instruments and cymbals sound strident and harsh. An extremely slight roll-off of the high frequencies should be all that is needed to make the sound considerably more pleasant. Keep in mind that much work has gone into the preparation of the dramas and musical programs and that our desire is to reproduce the sound exactly as it was recorded. Therefore, any adjustments to equalization should only be made to compensate for equipment problems or building acoustics and not because of personal preference. Such adjustments would only be implemented after approval has been given by the sound overseer or someone assigned by him.

51. *Setting Feedback Reduction Devices:* There is no shortage of devices on the market claiming to prevent or reduce feedback, and nearly all digital signal processor devices will offer some sort of feedback-reducing processor. However, once a system has been properly equalized, there is very little that any of these devices can do to prevent feedback. All feedback reducers have the same goal, which is to identify frequency peaks that are feeding back and notch them with a very narrow filter. As we have already learned, any frequency exaggerated by either the system or the room will be a frequency at which the system tends to produce feedback. The goal of equalization is to flatten the frequency response a reasonable amount so that no such peaks exist. In other words, if the system is properly equalized, there will not be many significant frequency peaks for the feedback-reducing device to find.

52. This is not to say that these devices should not be used. One must realize, however, that their use is only of limited benefit. At best, a good feedback-reduction unit on a properly equalized system will capture a few room anomalies and a few peaks on the response of the microphone. Once there are no more peaks to capture, the device has no choice but to start widening the notches and adding more. Basically, the feedback reducer is now trying to turn down the overall gain of the system! There is a point when any additional equalization (feedback reduction) provides no further benefit. Once that point is reached, continuing to equalize will only reduce the quality of the sound.

53. Several things should be determined before using any feedback-reducing device. Most units allow a certain number of frequencies to be set during installation, while a few frequencies are considered dynamic, or adaptive. The dynamic filters are constantly adjusted to different frequencies in an attempt to prevent feedback. Will the unit be allowed to notch feedback during the program automatically, or will it be locked once the initial setup is complete? If the device is allowed to run automatically, how will it react to feedback during the program and how long will it store newly placed filters? Despite some manufacturers' claims, it is not a good practice to run music through a feedback reducer with active filters.

54. Instructions on optimizing a feedback-reducing unit will differ by manufacturer, so please refer to the owner's manual for specific details on installation and setup. The following information is intended only to give a basic idea of how these devices are typically optimized.

55. Most feedback reducers must be set up when the room is completely quiet. Once the device is placed in setup mode, the gain for the primary microphone is slowly raised until feedback starts to occur. If the volume is raised slowly enough, only one feedback tone at a time will be heard and the unit will easily be able to identify and notch it. During setup, many find it beneficial to have someone quietly stand at the microphone to provide a more realistic acoustic environment. Also, in addition to the primary microphone, it is advantageous to capture a few frequencies with other microphones in locations where they will be used during the program. Instructions for some units will encourage the turning up of multiple microphones at the same time in order to stimulate feedback. However, if a gated mixer is being used, the number of open microphones will not pose a problem.

56. Once the setup is complete, the system should be tested by having someone read into one of the microphones. During the test, briefly place the unit in bypass mode to verify that the sound quality has not changed. If the sound has been significantly altered, the setup should be repeated.

57. *Adjusting Equalization for Stage Monitors:* The stage is designed and located in such a way that it will not be in the direct coverage of the main loudspeaker system. Because of this, it typically receives only reflected sound, with some arriving early and some arriving very late. This reflected sound can be extremely disconcerting for a participant trying to give a discourse. On occasion, the sound on the stage is so poor that those being interviewed cannot even understand the questions they are being asked. Stage monitors can help to alleviate such problems.

58. Some have been hesitant to use stage monitors because they are concerned that the monitors will cause the system to produce feedback. However, when stage monitors are properly laid out and equalized, feedback will not be a problem. (For a recommended stage monitor layout, please refer to Figure 4.11 in Chapter 4.) Stage monitors should always be equalized separately, as their primary function is only to provide a moderate level of sound to the participants and naturalness is not a requirement. For our purposes, equalization can begin with a frequency curve similar to the one shown in Figure 7.4. As needed, the low frequencies can be rolled off at an even higher point than shown. Even though naturalness will be reduced, the remaining frequencies will allow for sufficient intelligibility. Since any frequency peaks will cause feedback, the monitors should be checked with pink noise and a real-time audio spectrum analyzer. Any peaks should be removed with careful equalization.

59. In some instances, an additional feedback-reducing device will be used specifically for stage monitors. Where such devices are used, they can be adjusted as previously described in this chapter. If all peaks have been removed by previous equalization, though, the unit may have trouble finding frequencies to notch.

60. During the presentation of dramas, it is usually desired to raise the level of the stage monitors. While this is acceptable, the stage monitors should still be operated at the lowest acceptable level. Loud stage monitors can bleed into the audience seating, interfering with intelligibility. *It is imperative that the stage monitors be restored to their original level as soon as the drama is concluded or significant feedback issues may occur.*

Chapter Eight

Effective Sound System Operation

1. Even the best-designed and most carefully installed system will perform poorly if it is not operated correctly. At times, participants have become very frustrated because they have spent many hours preparing for their parts, only to be confronted with dead microphones, oppressive feedback, or other distracting technical mishaps. Such flaws in operation are not only undesirable but usually unnecessary. This aspect of the sound system involves—not equipment—but people. As we shall see, operating a sound system *effectively* requires that *many* work together in close cooperation. In this section we will discuss the responsibilities of the many different brothers involved in this operation, as well as specific details regarding effective sound system operation.

Sound System Operation Personnel

2. *Sound Department Overseer and Assistant:* The Sound Department overseer will work with his assistant to identify brothers who can be used for operating the equipment during the convention. Since these jobs typically require focused concentration on the task at hand, it is inevitable that operators will miss certain portions of the program. Thus, it is helpful when these assignments can be rotated. When scheduling, consideration should also be shown for the personal responsibilities a brother may have, such as young children in his family. The overseer will arrange for a reasonable schedule to be organized and distributed to those willing to assist.

3. During the convention, the sound overseer will be interested in all facets of operation. Either the sound overseer or someone assigned by him will attend all platform walk-throughs, carefully noting any potential issues. During the walk-through, careful note must be taken of what microphones, audio recordings, and so forth, are necessary for the program parts. The overseer or his assistant will also make sure that sound needs are cared for at any additional convention meetings, such as the meetings held with those considering Bethel service or the School for Kingdom Evangelizers. When initial system checks are performed at the beginning of each session, he should be notified of the results.

4. The sound overseer is primarily concerned with the overall quality of the sound for those in the audience. He should exercise extreme caution when determining who will be allowed to make adjustments to the system equalization. Some mixers have independent equalization for every channel, and it can be tempting for operators to make constant adjustments for every participant that sets foot on the stage. With a properly equalized system, such constant adjustment is unnecessary and should be avoided. That being said, there might be times when there are extraordinary problems with equalization and a subtle change could help significantly. As an example, perhaps prerecorded music is being played over a system that tends to overemphasize the high frequencies. A minor adjustment lowering the high-frequency equalization is likely all that would be needed to remedy the problem. So that such equalization is not overdone, the sound overseer or someone assigned by him should determine whether any adjustments are needed.

5. *Mixer Operator:* The mixer operator is responsible for making sure that the correct microphone and MP3 inputs are turned up at the proper time. In addition, he is responsible for keeping the system operating at a predetermined optimal level, preferably aided by the use of meters. He should have a good ear for feedback or other tonal problems that might arise with certain participants. A mixer operator must be sharp and alert, a person who takes a serious approach to his assignment. He must focus his full attention on what is happening or about to happen on the stage, being able to concentrate despite distractions. Owing to this, he simply will not be able to take notes or look up all the scriptures while operating the mixer. Therefore, qualified brothers should be rotated after every session to allow them to get full enjoyment from the program. Operating a sound system skillfully is an art for which one must develop a feel. Some develop this quickly, while others never will. Therefore, this is hardly a responsibility where *"everyone* should take his turn." The department overseer should always be on the lookout for spiritually-minded brothers who have the necessary qualifications and assist them to develop these skills. Preferably, brothers used in this capacity would already have proved themselves capable at circuit events.

6. *MP3 Player Operator:* It is possible that a mixer operator could make a number of minor mistakes, none of which would be noticed by the audience at large. Any mistake by the MP3 operator, however, will almost always be noticed immediately by all. If the music is stopped too early or an incorrect song is played, it can be difficult to correct. While this assignment is generally less involved than operating the mixer, the MP3 operator must pay close attention to detail and be precise and careful at all times.

7. *Recording Monitor:* A reliable brother should be assigned to monitor the official recording of the program. He should be familiar with whatever recording device is being used and should alert the sound overseer immediately if any problems are noticed.

8. *Sound Checks:* Each day before the doors are opened and during the first talk of each session, all audience seating areas must be thoroughly checked for sound quality. Who will perform these sound checks? Because of the legwork involved, the inclination may be to assign one of the younger—and possibly less experienced—members of the department. However, the final check is best made by an experienced man with the benefit of well-trained ears. At the same time, a measure of discernment must be used. It is a well-established fact that hearing acuity, particularly in the higher frequencies, declines steadily with age. Therefore, balance must be used in regard to those chosen to help in this area.

Departments Closely Assisting the Operation of the Sound System

9. *Platform Department:* The platform overseer works closely with the chairman and the Sound Department to ensure that the program is properly presented. All brothers assigned to adjust the microphones should be trained by the Sound Department in advance.

10. *Attendant Department:* While not technically members of the Sound Department, the brothers who serve as attendants for the program can do much to assist. It has been noticed that at some conventions brothers are allowed to sit in places that are not covered by the sound system or are too close to hallways or corridors where the sound from passersby is too loud to overcome. The attendants are responsible for attempting to limit these outside distractions as much as possible and for getting the brothers to sit only in the places allowed. Unfortunately, at times our elderly or infirm ones have been unwittingly seated in areas not covered by the sound system. If such a situation occurs, the sound overseer should bring the matter to the attention of the Convention Committee.

Initial Sound Checks

11. The sound system should be thoroughly checked before the attendants' meeting on the first day of the convention and every morning before the doors are opened to attendees. This initial sound check should not take much time, but it should be done sufficiently early so that there is enough time to correct any problems that may be found.

12. Typically, this check will involve the following steps:

- Fully power on the system (low-level equipment first, amplifiers last).
- Ensure that both MP3 players are operational.
- Check all microphones and cables to be used that day. (Microphones should be checked by speaking into them. Blowing into, tapping, or snapping fingers in front of a microphone is a poor test.)
- Verify that all loudspeakers are operational. (This would be especially true when either a partial or a full sound system had to be installed.)
- Double-check safety—confirm that no cables or tripods pose a trip hazard.

Platform Walk-Through

13. The convention chairman will schedule a walk-through of the participants before each session begins. The purpose is to make sure that the participants, platform personnel, and Sound Department are all prepared for the needs of each part. Members of the chairman's office as well as brothers working with the Platform and Sound departments must communicate carefully. Stage forms are used to provide detailed information on microphone requirements and locations. Copies should be provided for the mixer operator and any brothers who will need such information for platform setup.

14. During the walk-through, it is important for the participants to receive proper coaching on microphone use and other reminders. All brothers assigned to adjust the microphones should be trained by the Sound Department in advance. As aids in conducting a thorough walk-through, reminders are listed below and in Appendix 7. Concise information outlining microphone usage is also provided in Appendix 7.

- Any cellular phone taken on the stage must be powered off.
- Do not pull the lectern away from the microphone stand.
- The microphone should generally be about four to six inches (10 to 15 cm) from your mouth.
- Project your voice. Use a little more volume and intensity than you would in conversation.
- When addressing other participants on the stage, always turn your head in such a way that you are still speaking into the microphone.
- If speaking softly for effect, move very close to the microphone. If speaking loudly, pull away.
- If you hear popping from breath noise, speak across the microphone rather than directly into it.
- If you hear feedback, move closer to the microphone.
- If you need to clear your throat or have the urge to cough or sneeze, be sure to turn your head away from the microphone.

Microphone Technique and Placement

15. Stationary microphones used for talks are typically best supported by a separate floor stand with a boom. These stands and booms should be in good condition and should not be overly noisy when adjusted. Some have preferred the convenience of a gooseneck microphone attached directly to the lectern. While these can work quite well for Kingdom Halls, they can present insurmountable difficulties at a convention. For example, what will happen if an extremely tall participant prefers the lectern at a lower height? In such an instance, there may be no practical way to get the microphone close enough. The possible pickup of noise

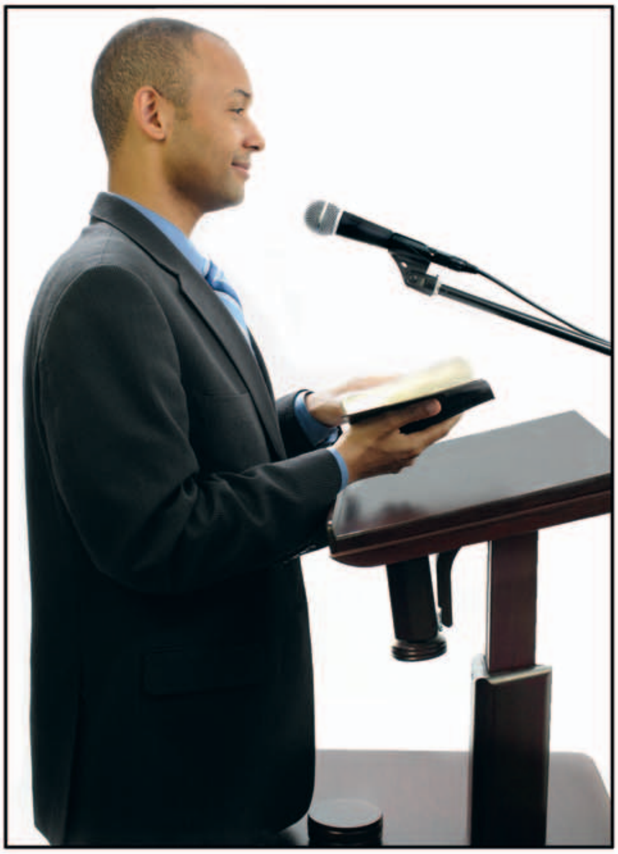


Figure 8.1 Proper Stand-Mounted Microphone Position

from the lectern is a further disadvantage to this approach. For a stationary microphone, a separate boom mounting is generally the most practical for convention applications.

16. The inverse square law applies to microphones the same way it applies to loudspeakers: Every doubling of the distance results in a loss of six decibels (dB). Knowing this helps us to understand why setting a microphone *extremely* close to a participant is to be avoided. For example, if the microphone is one inch (2.54 cm) away from the participant and he moves one additional inch away, there is a loss of 6 dB. However, if the microphone is placed a more appropriate six inches (15.24 cm) away, the talker would have to move six inches farther away to lose that same 6 dB. Thus, for most conventions a distance of about four to six inches (10 to 15 cm) is optimal.

17. In addition to the proper distance, the microphone must be at the proper angle. The microphone should typically be level with the talker's chin and positioned slightly upward toward his mouth at a 30-degree angle. (See Figure 8.1.) At this angle and at a distance of four to six inches (10 to 15 cm), the microphone will not block the participant's face, his view of the audience, or his notes. Signal level will not change significantly with normal head movement, and breath noises will be minimal.—See also the section

Figure 8.2 Proper Handheld Microphone Position

"Proper Microphone Techniques" in Appendix 7 and pages 139-142 in the *Ministry School* book.

18. Handheld microphones work well for multiple participants, especially when some motion is desired. In addition to portability, they usually permit a reduction in the total number of microphones, which is *always* desirable. Fewer live microphones will permit greater headroom before feedback and make the work of the mixer operator and platform personnel simpler and more manageable.

19. These handheld microphones may be of the same type as the stand-mounted microphone used for the lectern, provided they have some shock resistance built in. Some microphones are not intended for handheld use and will transmit unwanted handling noise if misused.

20. The handheld microphone should be positioned similarly to a stand-mounted microphone. The microphone should be held at a distance of four to six inches (10 to 15 cm) from the talker, slightly below the lips at chin level, with the talker speaking directly over the top of the microphone.—See Figure 8.2.

21. Over the years, brothers have experimented with a variety of microphone types and locations and even the use of multiple microphones to cover a single participant. For convention use, simpler is always better. Arrangements using multiple microphones for a single user are unnecessary. A single hardwired cardioid microphone on a stand is sufficient for those giving a talk. For other participants, handheld wired microphones will work quite acceptably, and have for many years.

22. Significant advances have been made in wireless microphones and because of their common use as roving microphones in many Kingdom Halls, some have wondered about their use for conventions. The primary advantage of wireless microphones is the extra mobility that they afford. However, the disadvantages include increased expense, decreased reliability, and the need for batteries. Compared to a wired microphone, even the best wireless models are more vulnerable to harmful radio-frequency interference. Therefore, wireless microphones should *never* be used at the main lectern. They would provide no benefit and would only increase the potential for problems. If the use of wireless microphones is being considered for participants not giving talks, careful thought must be given to whether the slight advantage will outweigh the numerous disadvantages.

23. At times, inquiries have been made regarding the use of wired or wireless headworn microphones on the platform. The rationale is that this style will allow the microphone to be as close to the talker as possible and allow participants in demonstrations to have both hands free. While this would offer some benefits, these would likely be outweighed by the many disadvantages. For example, if headworn microphones were used, many more microphones would be needed. Not only would each participant on the stage need to be outfitted with his own microphone but there would also need to be enough microphones for any participants that directly follow those on the stage. Additionally, wireless headworn microphones are significantly more expensive than good-quality standard wired microphones. Because of hygienic concerns, headworn microphones would need to be thoroughly sterilized before they were used by another participant. Sisters could face more challenges, since certain hairstyles might pose a problem with the headset and most dresses have no place to attach the wireless transmitter belt pack. For these reasons and more, many branches do not approve of their use.

Mixer Operation

24. When operating the mixer, there are a number of things to be considered and diligent focus is required. To avoid distracting the mixer operator and those assisting him, traffic around the control area should be kept to a minimum and access limited to those who are necessary to operation.

25. Primarily, the operator will endeavor to keep the sound levels consistent. There are a number of factors, though, that can make this task a difficult one. For instance, the operator's own perception of sound levels will change the longer he listens to the program. This "listener fatigue" may cause him to allow the levels to wander or he may make unneeded adjustments. Therefore, it is recommended that some type of visual indication be provided, such as a VU meter or an equivalent instrument. The combination of careful listening and visual indication should assist him in keeping the levels constant. Just as there can be "listener fatigue," there can also be what we could term "participant fatigue." Because of being nervous, many participants tend to begin their parts by speaking in a louder fashion. However, as the part progresses, their nerves subside and they begin to speak more softly. Oftentimes, changes in level such as these will be easily recognized and compensated for by an alert operator. As a word of caution, care must be exercised in our efforts to provide consistent sound levels so that we never inadvertently take away an experienced participant's effective use of modulation.

26. At times, there may be a number of participants on the platform. As the number of participants increases, the operator will need to pay close attention to the number of open microphones, as this can have a significant and detrimental impact on effective gain before feedback. With a large number of participants, it can also be difficult to keep track of which microphones are being used. At some conventions it may be possible to use a gated (sometimes called "automatic") mixer. Such mixers have sophisticated electronics to determine which microphones are being used and automatically switch between them. In no way do these mixers replace human operators, but they can offer a measure of assistance in cases involving large numbers of open microphones. It should be noted, however, that some low-cost gated mixers can be more problematic than helpful.

27. One of the most common problems an operator will experience is feedback. It is very important that the operator develop a feel for how the system reacts to adjustment of the controls. As soon as the slightest hint of feedback becomes apparent, he should gently reduce the gain. Operating even on the verge of feedback will have an undesirable effect on intelligibility. Therefore, the operator must decide when a compromise will be made, sacrificing level for the sake of intelligibility.

28. Not all persons speak articulately, nor do all speak with the same intensity. Most convention participants will use precise enunciation, but some, particularly those with strong regional or national accents, may not be as easily understood. Some participants may also be "microphone shy," that is, they speak very quietly when on the stage. Raising the volume is not the solution in every case. For example, a heavy accent will not be made any clearer by an increase in level; such an increase may even make the intelligibility worse. Therefore, discretion must be used before any adjustments in level are made.

29. One of the goals of system optimization is naturalness, meaning that participants should sound like themselves. In *extremely rare* instances, a slight adjustment of the equalization may be needed to increase the articulation or intelligibility of a participant. For instance, if the talker's voice is extremely resonant, rolling off the low end will improve the clarity. Any such adjustments should be made *cautiously* and *minimally,* and the equalization should be returned to normal immediately afterward. It will not be necessary to make changes for each participant. Any adjustments to equalization should be made only after receiving approval from the Sound Department overseer or someone assigned by him.

30. *Reproducing Music and Drama Recordings:* Portions of the program will consist of reproducing music and drama recordings, and the mixer operator must be careful to run the system at the appropriate level for each. At conventions, musical preludes are scheduled both for the enjoyment of the audience and as an indication that all should be in their seats and ready to enjoy the program. The volume of such musical preludes should be kept at a reasonable, comfortable level, not one meant for drowning out conversation. At other times, music is intended to lead in the singing of Kingdom songs. The mixer operator must carefully listen to the audience to make certain that they are following the tempo of the song. If the audience is singing ahead of or behind the recorded program, the music level is likely too low to lead them properly. At the same time, the musical accompaniment must never overpower the singing. Consideration must also be given to the tenor of the song. A little increased volume may be appropriate for full-throated singing of the more lively and vigorous songs. However, many prayerful expressions would naturally be rendered with a somewhat softer voice and perhaps with slightly less volume. The Sound Department overseer would do well to share with all who operate the mixer the letter sent to all bodies of elders dated March 27, 2008, regarding music at theocratic events.

31. In addition to the mixer operator, an additional brother is needed to care solely for the operation of the primary and backup MP3 players. He will make sure that the proper tracks are cued as they are needed. To reduce the possibility of errors, the buttons on the front panel of the MP3 players should be used when starting or stopping playback. If there is any failure, the secondary MP3 player can take up where the break occurs so that there will be no interruption in the program. Both playback units should be located near the mixer, permitting close coordination between the two operators. The operator should carefully follow along with the songs as they are played, so that he will know when they have truly finished. As a precaution, rather than stopping the MP3 player at the conclusion of a song, it may be better for the mixer operator to turn down the corresponding mixer input. In this way, if an error is made, the sound from the playback device can be turned back up immediately.

32. At times, it may be necessary to make slight adjustments to the equalization for the dramas or musical portions of the program. For instance, horns have a tendency to exaggerate high frequencies, which can make stringed instruments and cymbals sound strident and harsh. An extremely slight roll-off of the high frequencies should be all that is needed to make the sound considerably more pleasant. Keep in mind that much work has gone into the preparation of the dramas and musical programs and that our desire is to reproduce the sound exactly as it was recorded. Therefore, any adjustments to equalization should only be made to compensate for equipment problems or building acoustics and not because of personal preference. Such adjustments to equalization should only be implemented after receiving approval from the Sound Department overseer or someone assigned by him.

Sound Checks During the Program

33. An additional check of the sound system should be performed during the first talk of each session. The brothers handling these checks will carefully observe whether there is enough level in each section and if intelligibility is sufficient. While sound pressure level meters are frequently used to check overall level, time should also be taken to sit in different sections and really listen to how the program sounds. These checks should be done as inconspicuously as possible. Never would we want to draw attention away from the program by making a spectacle of taking sound-level readings. Once the system has been checked for the session and deemed acceptable, there would be little reason for brothers to spend the remainder of the session making constant checks and minor adjustments. Generally, three checks per day (once before the doors open and one at the beginning of each session) will be sufficient to ensure quality sound.

CHAPTER NINE

System Troubleshooting

1. Certain basic principles are essential for creating a relatively trouble-free sound system. The most fundamental of these principles is prevention, which has been discussed in portions of previous chapters. The prevention of many problems is primarily related to good system design and careful installation. All installations should be meticulously wired, properly grounded, and made up of reliable, well-maintained equipment. Those responsible for sound transmission should be familiar with the system and all the connections. It is good to keep accurate documentation that shows the basic system layout. Items such as amplifier channels and cabling should be clearly labeled. Thorough system checks *each morning* before attendees arrive will help to ensure that the system will continue to function smoothly. Precautions such as these will not only reduce the likelihood of problems but also make locating and correcting such problems much easier.

2. Despite our best efforts, we can be sure that problems will arise from time to time. While this brief chapter cannot substitute for experience or the ability to think clearly under pressure, it is our hope that the basic principles covered can provide some assistance when problems do occur.

Check the Simplest Things First

3. When a problem occurs, the tendency is to overcomplicate matters. In actuality, considering just a few questions usually allows the cause to be quickly determined.

- **What?**—Is the problem a lack of sound or the addition of an unwanted sound?
- **Where?**—Does it affect all inputs/outputs or just one?
- **When?**—When did the problem begin? Is it intermittent or constant?

4. Here is where a calm heart will prove most helpful. When someone is panicked, even the simplest of problems can be difficult to diagnose. The vast majority of problems occur as a result of something extremely simple. When we are armed with that knowledge and the answers to the questions above, most problems can be quickly solved, as we shall see by considering the following two common situations.

No Sound

5. Nearly all pieces of modern audio equipment include not only power lights but also level meters of some sort. A quick glance at these indicators and meters will soon tell us where the signal stops. Is there a signal at the mixer? Is there a signal at the processing equipment? Is there a signal at the amplifier? Identifying the first device in the chain that is not showing a signal provides us with an excellent idea of where the problem may be located.

6. Once we have narrowed down the problem, we can identify the most likely culprit for the condition. For example, if all the power in a rack suddenly went out, would it be more likely that all the equipment had failed at the same time or that the circuit breaker had tripped? Obviously, things like tripped circuit breakers, blown fuses, and unplugged power cords are relatively common. Experience will teach us that certain parts of the system, particularly cables and connectors, are much more common points of failure than reliable, well-maintained pieces of equipment.

Unwanted Sound

7. In most instances, any unwanted sound is identified and removed during the course of installation and optimization. It is very rare for extraneous sound to present itself in the middle of a program, but when it does happen, it can be extremely difficult to track down. At times, someone with sufficient experience will immediately recognize the sound and correctly identify the problem. However, when this is not the case, the only recourse is to trace the signal. Since this process will likely affect all or part of the system, we must determine whether it will be more detrimental to let the sound continue or to begin tracing it out.

8. If the problem has just begun, we should first determine *what changed*. Was something moved? Is a different input being used? If a change has been made, immediately reverse it (if possible) and see if the problem disappears. If reversing the change corrects the problem, begin thinking of ways to work around the issue until it can be permanently resolved.

9. When tracing out unwanted sounds or signals, always work from the output to the input. Again, keep in mind the principle of checking the simplest things first. Even in the case of unwanted signals, cables and connectors are the prime suspects. Always endeavor to determine the *type* of sound that is intruding. Once the type of sound is identified, Figure 9.1 may be helpful in determining its cause.

TYPE OF SOUND	CAUSES
HUM	• Ground loops caused by poor ground connections • Inductive coupling between either components or cables (See Chapter 6) • Equipment malfunction
HISS	• Poor gain structure (See Chapter 7, paragraphs 3-13) • Equipment malfunction
DISTORTION	• Poor gain structure (See Chapter 7, paragraphs 3-13) • Microphone or speaker failure

Figure 9.1 Chart for Diagnosing Unwanted Sounds

CHAPTER TEN

Dismantling the System

1. Once a convention is complete, the sound system often has to be dismantled, packaged, and shipped to a storage location or another convention. Sound equipment is delicate and easily damaged. Dismantling the system in a careful, orderly way ensures that future conventions can benefit from its use. Those handling this important assignment should be concerned with the safety of personnel and equipment.

Specific Safety Concerns

2. For an overview of general principles of safety, it would be wise to review Chapter 5 before beginning the dismantling process. A review of this material should be conducted with the keymen used to oversee the various aspects of the work. Please also note the following two examples of situations that should receive special attention:

- Perhaps some or all of the equipment is scheduled to be transferred immediately to another convention. While due concern for time is appreciated, haste and carelessness often lead to personal injury or damaged equipment. Therefore, plan carefully and allow sufficient time to accomplish each task safely.
- At times, motorized equipment such as a forklift or boom lift is used. When using such equipment, safety measures such as seat belts, fall-protection harnesses, and a spotter or lookout must be employed.

Maintaining an Accurate Inventory

3. A conscientious, reliable brother should be assigned to maintain adequate records and an accurate inventory of each piece of the sound system. It is preferred that the brother who verified the receipt of the equipment be responsible for it during the dismantling process as well. As the convention progresses, he can update and verify his records if any equipment is added or replaced. Many branches supply a detailed list of equipment being used. If this is the case, the brother would work in harmony with the Convention Committee to communicate any changes to the branch immediately. At the conclusion of the convention, he should check each piece of equipment against the up-to-date checklist. He should account for all pieces of equipment, including hardware and manuals. If some equipment is stored locally, a detailed list should be made and given to the Convention Committee for future reference.

4. The brother assigned to inventory the equipment will also need to work closely with the Convention Committee when preparing the equipment for shipment. The Convention Committee may receive specific directions from the branch as to how the equipment should be shipped. Those shipping the equipment will need to know which piece, crate, or pallet is bound for which location. To do this, the brother caring for inventory should verify that each shipment is clearly marked with its destination as the equipment is being collected, inventoried, and properly packaged.

Coordinating the Work

5. Many brothers may be required to care for dismantling a sound system at a large convention. The sound overseer should clearly communicate to each key person or group what should be done and when. Specific direction should include when to begin dismantling, where to bring the equipment, who will receive it, how it is to be packed, and any other information that will help ensure that the equipment will be safely dismantled and dispatched to its proper destination. However, we should never jeopardize the safety of our brothers and/or equipment for the sake of expediency.

6. Whenever possible, the same crew that installed certain equipment should take responsibility for removing it. They will already be familiar with the equipment locations and mounting methods. In some instances, when the same site is used annually, the same crews install and dismantle the system each year. This makes the work easier for everyone, lessening the need for extensive written or verbal instructions.

7. Those caring for the actual dismantling of the equipment would endeavor to leave it in a presentable condition for the next convention. All temporary markings, labels, and tape should be removed. Any adhesive residue should likewise be removed. Screws, nuts, and bolts on equipment should be sufficiently tightened so they do not fall off during shipment. Use care not to overtighten these items and damage the equipment. Where practical, long lengths of wire and cable that have not been damaged can be saved for use at future conventions. If a piece of equipment was shipped in a container that was specially designed or labeled for that piece, care should be taken to return the piece to its assigned container.

Preparing to Ship the Equipment

8. While some equipment may be shipped as individual units, quite often several pieces are shipped together in containers and on pallets. It is extremely important that attention be given to the careful packing of all sound equipment for its protection during shipment. Items should never be transported loose inside any type of container but should be carefully packed to prevent their sliding and rolling about. During transport, sound equipment should at least be packed in cardboard cartons and should never be transported loose in a truck.

9. If pallets are required for shipping, items should be securely stacked on them, since unbalanced pallets have the tendency to tip. Loosely and precariously stacked items can fall from pallets, injuring personnel and damaging equipment. When stacking a pallet, begin by placing the unloaded pallet on level ground. For the greatest stability, it is best to place larger, heavier pieces at the bottom of the pallet. The lightest and most delicate pieces should be placed on the very top of the palletized load. Before the pallet is wrapped, each piece that is added should sit stably without the assistance of restraints such as ropes and metal or plastic bands. This will prevent any equipment from falling when the recipient unloads the pallet.

10. A pallet should never be stacked too high. The maximum height should be such that the equipment can be loaded and unloaded without the aid of a ladder. Those loading the pallet should check with the Trucking Department to determine the maximum cargo height permitted for the vehicles being used.

11. After any pallets have been securely stacked, the load should be secured with restraints such as ropes, metal or plastic bands, or plastic wrap. The overall weight and dimensions of each pallet should be recorded. This information should be given to those caring for transporting the equipment. Pallets can then be clearly labeled with their destination and any additional information, such as safety placards, needed to facilitate safe transport.

12. Once the sound overseer has completed the inventory of the equipment and has verified that the equipment is properly prepared for shipment, he can inform the convention overseer that the equipment is ready for shipment to the next destination(s).

APPENDIX 1

Electrical Fundamentals and Use of Decibels

1. In this appendix we will address some of the electrical fundamentals involved with designing and managing an efficient, reliable convention sound system. Each component in the system is designed to operate within parameters that are dependent on specific values and magnitudes. Although an extensive background in electrical engineering is not required, a basic knowledge of how these electrical elements of the sound system interact with each other will prove helpful.

2. Decibels (dB) are units of sound measurement frequently used in the sound-reinforcement industry and serve several worthwhile purposes. However, because of their logarithmic nature, for many they tend to fall into the category of mystery math. With a little explanation, decibels will be found to be more practical and useful than at first imagined.

3. In order to make practical application of these interrelated elements, a certain amount of math is required. To simplify working with the numbers involved, an inexpensive scientific calculator will prove most useful. In addition to the usual arithmetic functions, this calculator should have the ability to square a number and find its square root. A "log" key will also be needed, as well as a method for finding the antilog of a logarithm.

Electrical Measurements Using Ohm's Law

4. Basically, there are just two fundamental elements that govern how an electrical circuit performs: (1) *voltage* and (2) *resistance*. The relationship of these two basic elements establishes two additional quantities, or units, with which we regularly work: (3) *current* and (4) *power*.

5. The units of measurement for these four elements are defined as follows:

- *Volt:* The basic unit of *electromotive force,* or electrical pressure. One volt is the force necessary to cause a current of one ampere to flow through a resistance of one ohm.
- *Ohm:* The basic unit of *resistance,* or opposition, to electron flow. One ohm is the amount of resistance that will allow a current flow of one ampere when one volt is applied across it.
- *Ampere:* The practical unit for measuring the amount of *electrical current* flowing in a circuit. One ampere is the amount of electron flow that results when one volt is applied across a resistance of one ohm.

- *Watt:* The unit of *electrical power* that measures the amount of work being performed by an electrical circuit. This work, measured in watts, could include illumination from an electric lightbulb, mechanical motion produced by an electric motor, or sound radiating from a loudspeaker. One watt of power is dissipated when one volt causes one ampere to flow through a resistance of one ohm.

6. All electrical design and function involve these four basic units, which are related mathematically. Their relationship is defined in the set of basic equations derived from Ohm's law, published in 1827. When any two of these four values are known, the other two are easily calculated. First, to define the symbols:

Symbol	Unit
E	Voltage in volts
R or Z	Resistance (impedance) in ohms (Ω)
I	Current in amperes
P	Power in watts

7. These relationships are summarized by the graph in Figure A1.1. The equations on the outside are used to calculate the unknown value at the center.

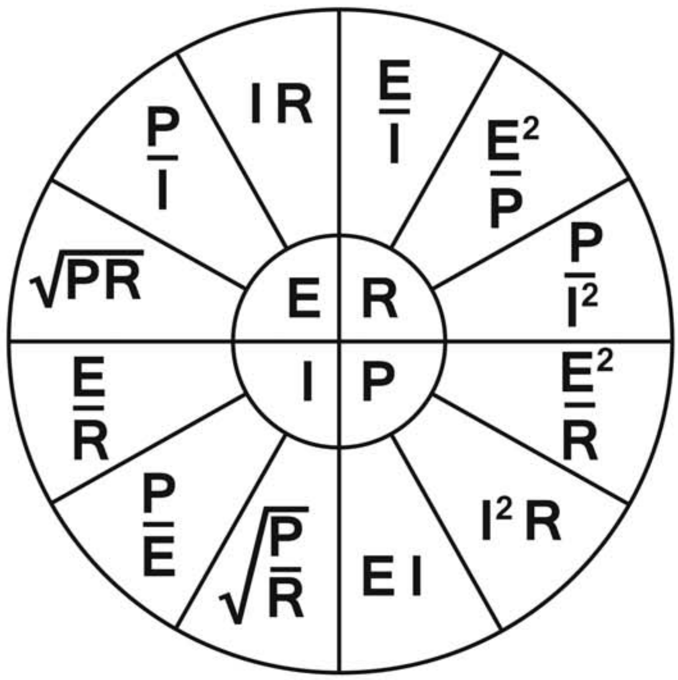


Figure A1.1 Ohm's Law Equations

8. Using the simple circuit in Figure A1.2, assume that the battery supplies 12 volts and the resistance is 6 ohms.

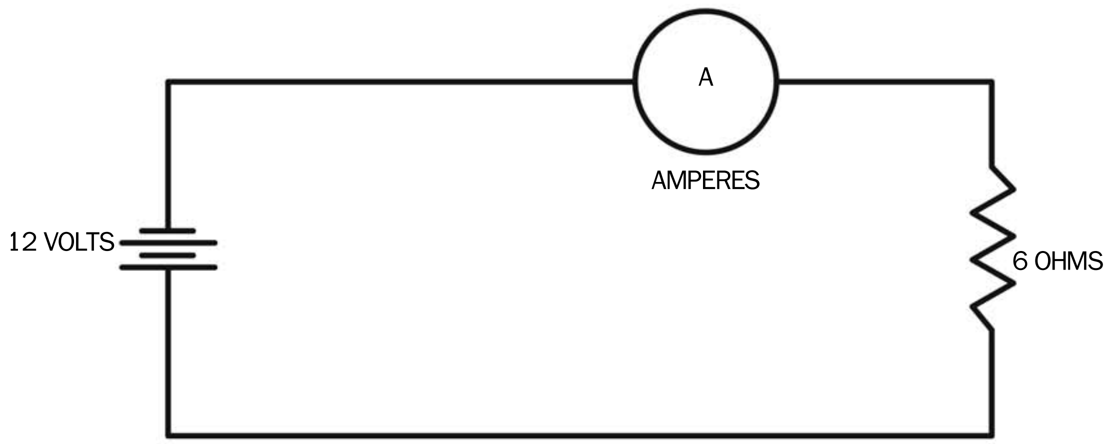


Figure A1.2 Basic Electric Circuit

How much current is flowing through the circuit?

I = E/R = 12/6 = 2 amperes

How much power is dissipated in the resistor?

P = E • I = 12 • 2 = 24 watts or P = E^2/R = 144/6 = 24 watts

9. Take note of the changes that occur when one value in a circuit is changed. When we increase voltage to 24 volts, note the changes to current and power. Let's look first at the current:

I = E/R = 24/6 = 4 amperes

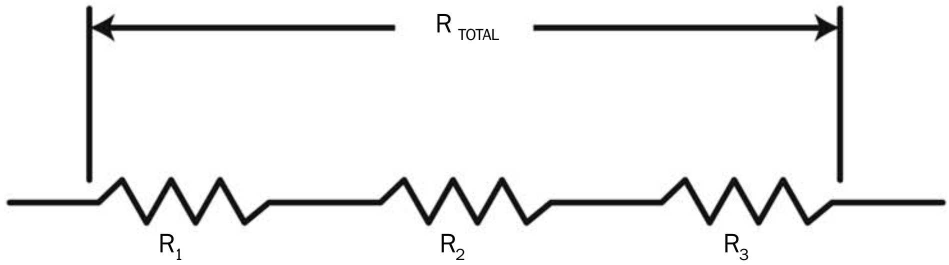
By doubling the voltage, we have also doubled the current in the circuit. But note what happens to the power. Since we have increased both voltage and current by a factor of two, the power is increased by a factor of 2^2, or 4:

P = E^2/R = 576/6 = 96 watts or P = E • I = 24 • 4 = 96 watts

10. In summary, the ratio of power change is equal to the *square* of the ratio of voltage change as long as *R* remains constant. This is implied in the equation $P = E^2/R$ from Ohm's law.

Series Circuits

11. A *series circuit* is formed when any number of resistors are connected so that there is only one path where the current can flow. To find the total resistance (R_T) for resistors in *series,* we simply add up all the resistance values, using the simple equation shown in Figure A1.3:



$$R_T = R_1 + R_2 + R_3$$

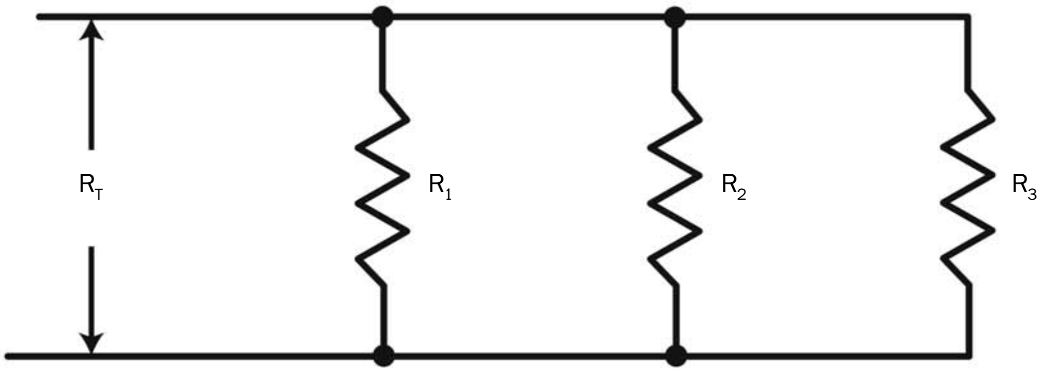
Figure A1.3 Formula for Calculating Resistance in a Series Circuit

If all resistance values in the series are equal, we can do the following:

$$R_T = R \cdot N \qquad (N = \text{number of equal resistance values})$$

Parallel Circuits

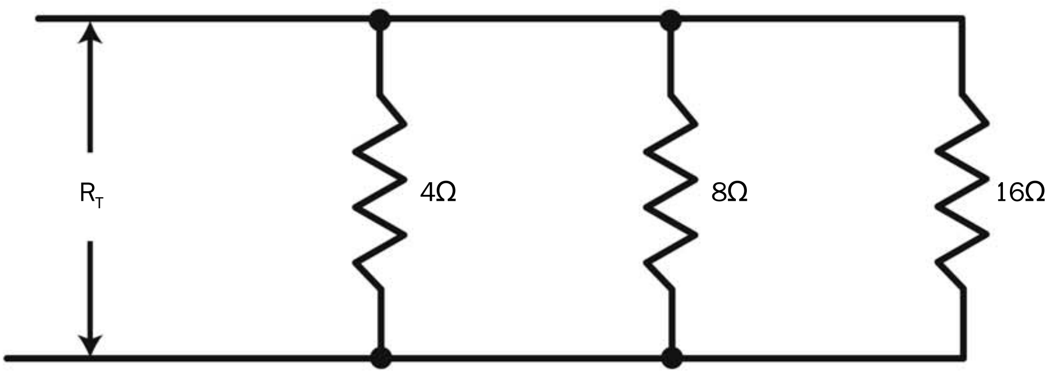
12. A *parallel circuit* is formed when two or more resistors are placed in a circuit so that current can flow through more than one path. The formula for finding the total resistance for resistors in *parallel* is shown in Figure A1.4a:



$$R_T = \frac{1}{\frac{1}{R_1} + \frac{1}{R_2} + \frac{1}{R_3}}$$

Figure A1.4a Formula for Calculating Resistance in a Parallel Circuit

13. This formula is a little more difficult to work manually, but a calculator with a "1/x" key makes it quite simple. Using the diagram above, we can fill in some values:



$$R_T = \frac{1}{\frac{1}{4} + \frac{1}{8} + \frac{1}{16}} = \frac{1}{.25 + .125 + .0625} = \frac{1}{.4375} = \underline{2.2857 \text{ ohms}}$$

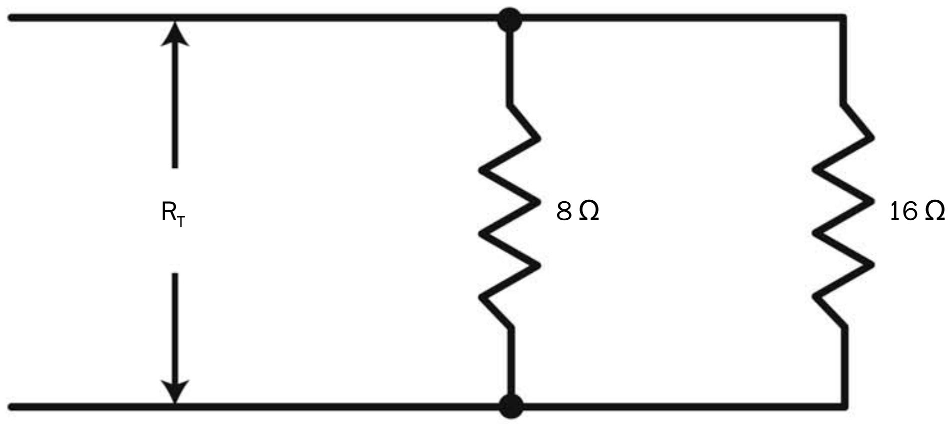
Figure A1.4b Example of Calculating Resistance in a Parallel Circuit

Thus, the total resistance in this example is 2.2857 ohms, which rounds up to 2.3 ohms. Remember that in a parallel circuit, R_T will always be less than the lowest single resistance value in the circuit.

14. If all resistance values in a parallel circuit are equal, things are a little simpler:

$$R_T = R/N \qquad (N = \text{number of equal resistance values})$$

15. For two parallel resistors of different values, the following simplified equation shown in Figure A1.5 applies:



$$R_T = \frac{R_1 \cdot R_2}{R_1 + R_2} = \frac{8 \cdot 16}{8 + 16} = \frac{128}{24} = \underline{5.3 \text{ ohms}}$$

Figure A1.5 Formula for Calculating Resistance in a Parallel Circuit With Two Different Values

Resistance Versus Impedance

16. The calculations just considered are based on ohms as units of electrical *resistance*. In reality, these calculations apply only to direct current (DC) circuits, such as automotive electrical systems or battery-operated circuits, and to alternating current (AC) circuits with strictly resistive loads.

17. Most of the calculations in audio system design, however, involve alternating current. The equations we have just learned still apply, but with one exception: We now replace the term *resistance* (*R*) with a new term—*impedance* (identified by the symbol *Z*), a value widely used in audio. *Z* now replaces *R* in the Ohm's law equations.

18. *Impedance* is defined as the total opposition a circuit offers to the flow of alternating current. For example, a loudspeaker may have an impedance of 8 ohms, a microphone mixer may have a 600-ohm output impedance, and a power amplifier may have 4-, 8-, and 16-ohm output impedances. Impedance includes the resistance of the circuit but may also include one or more additional values, known as *capacitive reactance* and/or *inductive reactance*. Capacitors and inductors are reactive circuit elements, storing AC energy rather than dissipating it as a resistor does. Although both capacitive and inductive reactance are measured in ohms the same as resistance, both vary with frequency, making their specific values more elusive. A complex load, such as a loudspeaker, represents a combination of a resistive element and one or more reactive elements. This combination is called *impedance*. It is good to note that the rules for determining series and parallel combinations of impedances are the same as for resistors.

19. There are, however, other important differences to be understood when working with DC resistance versus AC impedance. Resistance can be measured by a simple ohmmeter, whereas impedance often shows little relationship to ohmmeter readings. For example, measuring an 8-ohm loudspeaker with an ohmmeter may give a DC resistance reading of 5 or 6 ohms, depending on the power rating of the loudspeaker, whereas the actual AC impedance may be 8 ohms. A certain amount of inductive reactance likely explains the variation. But reactance cannot be measured with an ohmmeter. Since we are mainly interested in the resulting total impedance of the loudspeaker, referring to the manufacturer's specification sheet will generally satisfy our need for specific data. Measuring impedance requires an *impedance bridge* or *impedance meter*. This instrument impresses an AC voltage at a specified frequency across the circuit to be measured and indicates the impedance based on the amount of current that flows. An ohmmeter measurement, on the other hand, is based on a DC voltage and only tells us whether there is continuity. Its reading is not the measure of impedance and could actually be misleading. For example, should there be a series capacitor in an AC circuit, the ohmmeter would indicate an open circuit. Yet, the impedance could be correct for the circuit.

20. Accurate impedance values are of critical importance in matching loudspeakers to their amplifiers. Transferring the amplifier's power efficiently to the loudspeaker system requires a close impedance match. A serious mismatch can result in overloading the amplifier, raising the system's distortion, and/or severely wasting audio power. After each loudspeaker line is installed according to calculations, measuring the impedance of each line with an impedance bridge or meter may reveal connection errors, reversed transformers, and—sometimes—unauthorized taps.

21. As a sample problem, using the rules we learned in connection with series and parallel resistances, let us calculate the resulting impedance of the series-parallel loudspeaker system shown in Figure A1.6:

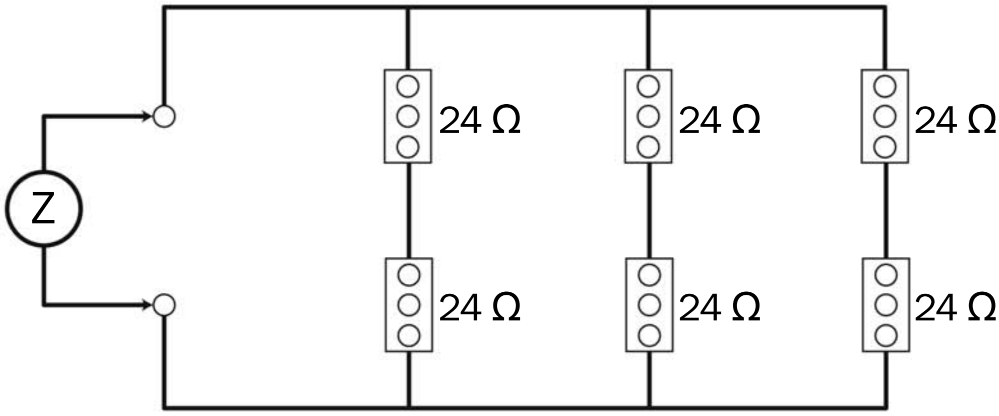


Figure A1.6 Six Sound Columns in a Series-Parallel Circuit

Each sound column has three 8-ohm speakers in series, calculated at 3 • 8 ohms = 24 ohms. Two 24-ohm columns in series add up to 48 ohms. Three 48-ohm pairs connected in parallel result in a total impedance of 16 ohms. This circuit would be a good match for the 16-ohm output of many mixer-amplifiers, where each sound column will receive equal audio power.

22. Generally, there are several ways to interconnect loudspeakers in order to achieve an efficient impedance match, including the use of suitable line transformers. Impedance matching is further discussed in Chapter 6, paragraphs 27 and 35.

AC Voltage Measurements

23. AC voltage can be expressed using three different values —*peak volts, peak-to-peak volts,* and *rms volts*. The relationship of these values is illustrated in Figure A1.7. The *peak* value is the maximum voltage reached at the highest amplitude of the voltage swing; the *peak-to-peak* (*p-p*) figure is measured from maximum positive to maximum negative excursion. But neither of these varying values represents the *power-effective* value of an AC voltage. As shown in the illustration, an AC voltage constantly changes value at a rate determined by the frequency of the signal. With each cycle, the voltage swings to a peak positive and then to a peak negative value, passing zero with each reversing excursion. Measuring this voltage with a voltmeter gives us what is known as its *root-mean-square* (*rms*) value, which is 0.707 times its peak value. The rms value is also its power-effective value, the value we most commonly use in our work. This is the value displayed by the AC voltmeter in our multimeter.

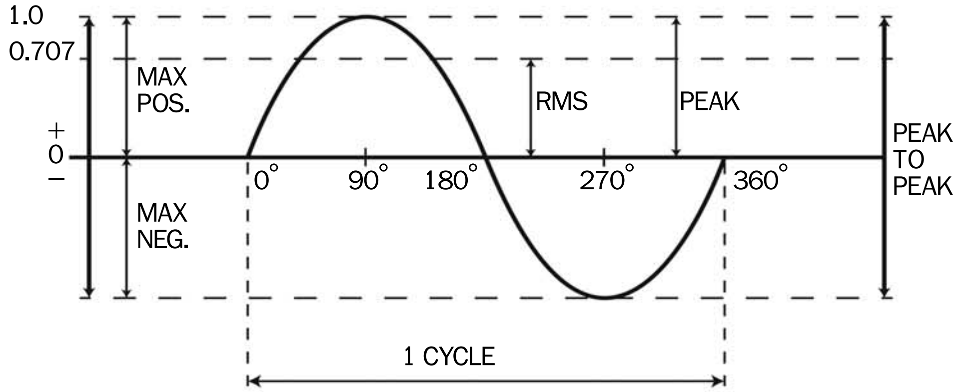


Figure A1.7 One Cycle of a Pure Sine Wave

24. Thus, when we say the local power line voltage is 120 rms volts, this AC voltage is delivering the same power to a lightbulb that 120 volts DC would. However, the figure of 120 rms volts represents a peak voltage of 170 volts and a peak-to-peak voltage of 340 volts.

25. The three voltage values are related mathematically, as follows:

AC peak volts = rms volts • 1.414
AC p-p volts = rms volts • 2.828
AC rms volts = peak volts • 0.707

For purposes of circuit design, a peak voltage may be needed when only the effective, or rms, value is known. To calculate the peak value, simply multiply the rms value by 1.4. Generally, the multipliers 1.4, 2.8, and 0.7, respectively, are sufficiently accurate for our calculations when determining peak volts, peak-to-peak volts, and rms volts.

26. The standard VU meter found on many mixers is used to indicate audio signal levels, but it may be considered an AC voltmeter, since it displays rms values. As a general rule, this meter is calibrated to read “0 VU” when the output signal corresponds to 1.2 rms volts. (“Peak reading” meters found in some modern audio equipment are not considered VU meters.)

Exponents and Logarithms

27. Exponents are mathematical tools used to show when numbers are being repeatedly multiplied by themselves. The process is symbolized by two numbers, a base and an exponent. The base is the number that is being repeatedly multiplied. The exponent is the number of times the base is multiplied. The exponent is usually shown as a superscript immediately to the right of the base. Let us consider a familiar example by using the number “10” as the base.

$$10^1 = \underline{10}$$
$$10^2 = 10 \cdot 10 = \underline{100}$$
$$10^3 = 10 \cdot 10 \cdot 10 = \underline{1{,}000}$$

28. The exponent is not always positive. When the exponent is negative, it is understood that the number being repeatedly multiplied is the reciprocal of the base (i.e., the number "1" divided by the base). For example:

$$10^{-1} = (1/10) = \underline{0.1}$$
$$10^{-2} = (1/10) \bullet (1/10) = 1/100 = \underline{0.01}$$
$$10^{-3} = (1/10) \bullet (1/10) \bullet (1/10) = 1/1{,}000 = \underline{0.001}$$

29. While the base can be any number, a base of 10 is often used in electronics and audio fields to express very large and very small numbers. Doing so can simplify measurements and complex calculations. Special names and prefixes have been assigned to the most commonly used numbers. These prefixes can be placed before a unit of measure to signify that the unit should be multiplied by the implied exponential value. Consider a few examples: A microphone may produce 0.003 volt. This can be expressed as "3 millivolts" and written "3 mV." A frequency of 6,600 hertz can be expressed as "6.6 kilohertz" and written "6.6 kHz." The table below lists some of the most commonly used exponential values of 10, their prefixes, symbols, and meaning.

Exponent	Value	Prefix	Prefix symbol	Example
10^{-6}	0.000001	micro	µ	1 µV = 1 millionth (0.000001) of a volt
10^{-3}	0.001	milli	m	1 mA = 1 thousandth (0.001) of an ampere
10^{-1}	0.1	deci	d	1 dB = 1 tenth (0.1) of a bel
10^{3}	1,000	kilo	k	1 kHz = 1 thousand (1,000) hertz
10^{6}	1,000,000	mega	M	1 MΩ = 1 million (1,000,000) ohms

30. Exponents can be used to express numbers that are not exact multiples of any specific base. For instance, if $10^1 = 10$ and $10^2 = 100$, then to express a number such as 50, we would have to raise the base 10 to a "power," or exponent, that falls somewhere between 1 and 2. That unknown exponent is called a *logarithm*. By definition, a *logarithm* is the power to which a base, such as 10, must be raised to produce a given number. In our example, we want to find the exponent to which 10 has to be raised to produce the number 50. This is written mathematically as follows: $logarithm = log_{10}(50)$. Notice that the base 10 is written as a subscript immediately to the right of the math term "log." The number that we are trying to produce is written in parentheses to the right of the term "log" and the subscript.

31. The best way to find a logarithm is to use the "log" key commonly found on most inexpensive scientific calculators. Our calculator will quickly tell us that *log 50 = 1.69897,* which rounds off to 1.7. Thus,

$$1.7 = \log_{10}(50)$$
$$10^{1.7} = 50$$

32. Consider a few more examples, and note the consistent pattern followed:

$0.7 = \log_{10}(5)$	in other words	$10^{0.7} = 5$
$1.7 = \log_{10}(50)$	in other words	$10^{1.7} = 50$
$2.7 = \log_{10}(500)$	in other words	$10^{2.7} = 500$
$3.7 = \log_{10}(5{,}000)$	in other words	$10^{3.7} = 5{,}000$

33. Now that we have seen how an exponent can manipulate the value of the base 10, we can proceed to work with *just* the exponents when the base number is assumed, or understood. Any number can be used as a base, but 10 will be assumed for our purposes to match the function of the "log" key on a scientific calculator.

34. In our example above, 1.7 is the logarithm, or log, of 50. Conversely, 50 is the antilog of 1.7. Just as a "log" key is commonly found on most scientific calculators, an antilog key, or "10^X" key, enables us to find the antilog of an exponent. Your calculator's instruction booklet may give some additional tips for simple methods of solving what may appear to be complex mathematical problems.

Learning to Work With Decibels

35. The decibel (1/10 bel) is an important application of the logarithm. Often abbreviated *dB,* the decibel was introduced by the telephone industry in the 1920's to express power gains and losses. Since then, its use has spread to include voltage ratios, noise and distortion levels, and, in the acoustic field, sound pressure levels and so forth. Basically, the decibel is a means of expressing a power ratio or a comparison of two power values. As will be seen, even when applied to voltage ratios, the decibel still defines the effect of voltage changes on power.

36. *Power ratios*—Let us look for a moment at how decibels relate to audio power. To express power changes or ratios in dB, we use the following formula:

$$\text{dB} = 10 \log P_1/P_2$$

Here we are comparing two power levels, where P_1 is the higher value in watts and P_2 is the lower value. Our change is thus based on the *ratio* of these two levels. We must remember that the dB power formula *always* requires a multiplier of 10 ("10 log") so that we will arrive at an answer in decibels, rather than in bels.

37. Sample problem: One watt is how many dB above one milliwatt (1 mW = .001 watt)?

$$\mathbf{dB = 10 \log P_1/P_2 = 10 \log 1/.001 = 10 \log 1{,}000 = 10 \cdot 3 = \underline{+30\ dB}}$$

Thus, one watt is +30 dB above one milliwatt, or we could say that one milliwatt is −30 dB below one watt. Note that the larger number of the ratio is generally divided by the smaller number. This simplifies the calculation, giving a positive logarithm in all cases. By simply adding a plus (+) or a minus (−) sign, we show increase or decrease.

38. The following nomogram (Figure A1.8) can be used to work problems based on the formula for power ratios. Observe that the upper scale is labeled "dB Above and Below a One-Watt Reference Level." This nomogram will prove very helpful in working out the power requirements for different loudspeakers.

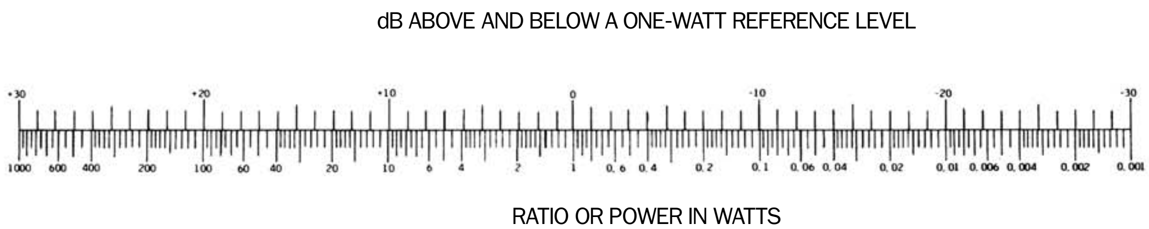


Figure A1.8 Nomogram for Determining Power Ratios Directly in dB

39. Here are a few practice problems:

1. What is the power level of 4 watts, relative to 1 watt? Opposite 1 watt on the nomogram, read 0 dB; opposite 4 watts, read 6 dB. The difference between the two values is 6 dB.
2. What is the power change in dB from 10 watts up to 40 watts? Opposite 40 watts on the lower scale, read 16 dB; opposite 10 watts, read 10 dB; again, the difference defines a power change of +6 dB. If we decreased power from 40 watts to 10 watts, the change would be −6 dB.
3. A power change from 0.1 watt to 3 watts results in what level change? (+15 dB)
4. Try another change from 1 watt to 10 watts (+10 dB) or 20 watts to 200 watts (+10 dB). (By carefully studying the nomogram, you can observe that any 10:1 power ratio will always correspond to

a 10 dB level difference. A 2:1 power ratio results in a 3 dB level difference. A 6 dB level change indicates a 4:1 power ratio change.)

5. The following problem is a little more difficult: An amplifier has an input impedance of 600 ohms with an input signal of 0.775 volt. Its output is 40 volts, terminated by an 8-ohm load. What is the power gain in dB? Using the formula learned earlier, $P = E^2/Z$, we calculate both input and output powers individually and then compare them on the nomogram:

$$P_{in} = E^2/Z = 0.775^2/600 = 0.600625/600 = 0.001 \text{ watt (1 mW)}$$
$$P_{out} = 40^2/8 = 1{,}600/8 = 200 \text{ watts}$$

From the nomogram, we read: −30 dB at 0.001 and +23 dB at 200. Thus, the power gain is 23 − (−30) = 53 dB.

40. In this power nomogram, 0 dB is opposite one watt, providing power ratios above and below one watt. Since the input power in this example is −30 dB below one watt and the output power is +23 dB above one watt, it is necessary to add the two decibel figures to obtain the total power gain for this particular amplifier. Remember, the power nomogram can be used only for comparing *power quantities*. If input/output data is given in volts only, we cannot calculate power gain, since a voltage alone does not represent a specific power. We must first determine the input and output impedances and then calculate the powers accordingly. Note that input and output impedances are rarely the same.

41. *Voltage ratios*—We can also use decibels to express voltage comparisons across the *same* impedance value by using this equation:

$$\text{dB} = 20 \log E_1/E_2 \text{ (with Z unchanged)}$$

E_1 represents the higher value in volts and E_2 the lower value. Here again we must find the log of a ratio, but our multiplier now becomes 20. Here are a few examples:

1. We raise the output of a line amplifier from 1 volt to 10 volts. What is the increase in dB?

$$\text{dB} = 20 \log 10/1 = 20 \log 10 = 20 \bullet 1 = \underline{+20 \text{ dB}}$$

2. We now increase the output from 10 volts to 20 volts. How many dB would this be?

$$\text{dB} = 20 \log 20/10 = 20 \log 2 = 20 \bullet 0.3 = \underline{+6 \text{ dB}}$$

3. What is the overall gain from 1 volt to 20 volts?

$$\text{dB} = 20 \log 20/1 = 20 \log 20 = 20 \cdot 1.3 = \underline{+26 \text{ dB}}$$

4. We also could have reached this total by simply adding the results of the first two steps: 20 + 6 = +26 dB.

42. As a memory aid, it may be helpful to explain why we use a *20 log* multiplier to determine voltage ratios. Going back to Ohm's law and the relationship between voltage and power, we recall that $P = E^2/R$, which tells us that power will vary with the square of the voltage for any given resistance or impedance. It is of interest to note that if we want to square a number using logarithms, we can simply multiply its log by two and find the antilog of the result. Now, combining these two ideas: Since our basic dB formula for *power ratios* is $dB = 10 \log P_1/P_2$, we can adapt the same formula for *voltage ratios* by simply adding the multiplier *2* to square the ratio, so that it would read: $dB = 10 \cdot 2 \cdot \log E_1/E_2$, which is the same as:

$$\text{dB} = 20 \log E_1/E_2$$

Thus, calculating *voltage* ratios *always* requires a *20 log* multiplier.

43. To simplify converting voltage ratios, we can use the following decibel nomogram:

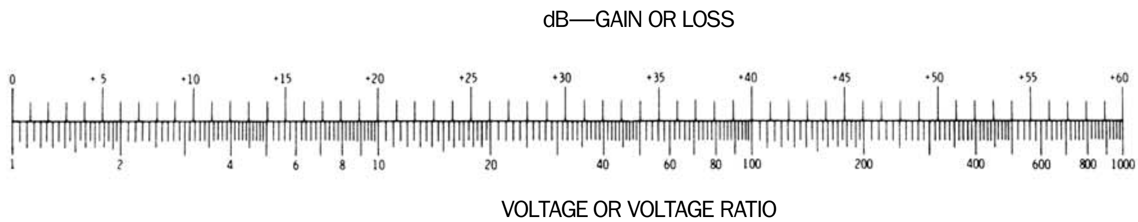


Figure A1.9 Nomogram for Determining Voltage Ratios Directly in dB

44. Find the answers to the above problems on the voltage nomogram. Here we must remember that to find a ratio using this nomogram, voltages must be measured across the *same* impedance value, even when comparing voltages in different circuits.

45. Here are a few guidelines for managing decibels:

1. Electrical and audio *power calculations* expressed in dB require a *10 log* multiplier.

2. *Voltage ratios* and certain acoustic relationships require a *20 log* multiplier.

3. Any 10:1 change in power will result in a 10 dB change in level.
4. Doubling *power* represents a *3 dB* increase in level ($10 \log 2 = 3$ dB).
5. Doubling *voltage* represents a *6 dB* increase in level ($20 \log 2 = 6$ dB).

46. Or stated another way, when voltage is doubled (with R or Z remaining the same), then power is increased four times, an increase of 6 dB ($10 \log 4 = 6$ dB).

Decibel Reference Levels

47. By this time, a very important fact relative to decibels should be apparent: Unlike the volt, ohm, or watt, the decibel in itself has no absolute value. Consider this example: Someone may say, 'The maximum output of my mixer is +18 dB.' What voltage would we expect to measure at +18 dB? Actually, that statement is ambiguous, since a reference for "0 dB" is not specified. However, when a standard reference value is specified (or understood) for "0 dB," then any number of dB above or below that stated or implied "zero" reference may be used to describe a specific quantity. Over the years, several reference levels for audio signals have been established, and it may be helpful to become familiar with the most common of these.

48. *dBm*—The term *dBm* expresses an electrical power level and is always referenced to one milliwatt (.001 watt), as indicated by the suffix "m." Thus, 0 dBm = 1 milliwatt. This notation was originally devised for the typical 600-ohm telephone line. In many parts of the world, this has been the standard for the telephone industry and radio broadcasters for decades. It so happens that a voltage of 0.775 rms volt applied to a 600-ohm line dissipates a power of one milliwatt. Thus, 0.775 volt represents one milliwatt (0 dBm) but *only* when the circuit impedance is 600 ohms. For any other impedance, 0.775 volt could not represent one milliwatt and thus is no longer truly related to "dBm."

49. With modern audio equipment, however, the concept of requiring a specific power level interface actually serves little purpose and often tends to complicate matters. Power levels are not a consideration, as we are not transferring power—only a signal voltage. The exception, of course, is where power amplifiers are driving loudspeaker systems—where *watts,* rather than dB quantity, is the term commonly used. The line-level input impedance of most modern professional audio equipment is usually much higher than 600 ohms, falling somewhere between 10k ohms and 50k ohms. (This is the concept followed in interfacing nearly all consumer-grade components, the majority of which have unbalanced inputs.) Another advantage of the high-impedance input concept is that it often permits more than one input to be safely parallel-connected to the same output.

50. *dBu*—To eliminate confusion when attempting to interface audio power levels, the professional audio field has adopted another term, *dBu*. Please note, though, that dBu is not a power level. It is a voltage level in which 0 dBu is always referenced to 0.775 volt without regard for circuit or load impedance. Thus, it can be said that the voltage represented by 0 dBu is equivalent to 0 dBm—but *only* when the dBm figure is derived with a 600-ohm load.

51. However, there can be a significant difference between dBu and dBm specifications. For example, one mixer may specify a +20 dBm output. This mixer should be capable of providing a level of 7.75 rms volts into a 600-ohm load, which is 20 dB above 0 dBm. However, when a mixer's output is rated in dBu, it is advisable to look more closely at the manufacturer's specification sheet. The specification for another mixer may read: "This mixer's maximum output level is +20 dBu into 10k ohms or higher impedance." Both mixers are capable of producing the same output voltage, but not the same power. While both can feed a 10k-ohm input, only the first mixer is capable of driving a 600-ohm input. The second mixer is obviously designed for a higher impedance. Connecting it to a 600-ohm load would not be recommended, as this would probably overload and possibly damage the unit.

52. Incidentally, the "u" in "dBu" stands for *unloaded,* a term engineers use to describe an output that works into no load or an insignificant load, such as the typical high-impedance inputs of modern audio equipment. A nonloading input is often referred to as a *bridging input* when its impedance is at least ten times that of the output source driving it.

53. *dBv*—For a time, the reference level *dBv* (lowercase *v*) was adopted by some manufacturers in an effort to offset some of the inconsistencies with dBm. Like 0 dBu, 0 dBv is referenced to 0.775 volt and is not dependent on load impedance. However, dBv never gained widespread use in the industry, no doubt because of possible confusion with the audio reference level dBV.

54. *dBV*—Some manufacturers, especially of consumer-type equipment, may use the notation *dBV* (capital *V*), which is another accepted way of expressing a specific voltage level. Generally, 0 dBV is referenced to one volt, without regard to circuit impedance. Since 0 dBV (1 volt) and 0 dBu (0.775 volt) are referenced to different voltages (not powers), the decibel difference between them is 2.2 dB. Thus, to convert dBV to dBu, we simply add 2.2 dB to the dBV value. To convert dBu to dBV, we subtract 2.2 dB from the dBu value.

55. *VU*—Many microphone mixers are equipped with a meter for indicating the instantaneous audio output level. Known as a VU meter (*VU* standing for volume units), this meter is usually calibrated to read "0" when the output level is +4 dBu. This level, often referred to as "0 VU" or "+4," represents 1.225 rms volts, a level 4 dB higher than 0 dBu (0.775 volt). As an industry standard, this is often the preferred level for interconnecting the components of a professional sound system. It defines a specific voltage level from the mixer and is the standard used by many sound engineers.

56. The chart below provides a summary of the decibel reference levels.

Reference Level	
dBm	0 dBm = 1 milliwatt; 0.775 volt in a 600-ohm circuit
dBu	0 dBu = 0.775 volt, not dependent on load impedance (not a power level)
dBv	0 dBv = 0.775 volt, not dependent on load impedance (obsolete)
dBV	0 dBV = 1.0 volt, not dependent on load impedance
VU	0 VU = 1.2 volts (equivalent to +4 dBu), not dependent on load impedance

Sound Pressure Level (SPL)

57. In the design and operation of a sound system, one of our most important goals is to provide a sufficient sound level for everyone in the audience to hear well. Rather than attempting to measure the acoustic power produced by a loudspeaker, it is much simpler to measure its sound pressure level using a sound-level meter. This relatively inexpensive device will immediately provide this data in decibels for any given location at the convention site.

58. It should be noted that a change in sound pressure level is analogous to a change in voltage (electrical pressure) in an audio circuit. Thus, to express a dB-SPL change in level requires a *20 log multiplier,* the same as for voltage changes. The formula for SPL change then becomes:

$$\text{dB-SPL} = 20 \log D_f / D_n$$

D_f represents the farther distance and D_n the nearer distance. The equation is the same, whether *both* dimensions are in inches, feet, or meters.

59. To illustrate, if we measured 90 dB-SPL 10 feet from a source and then moved to a point 12 times that distance, or 120 feet, what should our sound pressure level be in dB?

$$\text{dB-SPL}_f = \text{dB-SPL}_n - 20 \log D_f / D_n$$

First, we calculate the loss between the two distances:

$$\text{dB} = 20 \log 120/10 = 20 \log 12 = 20 \cdot 1.08 = \underline{21.6 \text{ dB}} \text{ loss}$$
$$\text{Thus: dB-SPL}_f = 90 - 21.6 = \underline{68.4 \text{ dB-SPL}}$$

60. Here we can use a nomogram similar to that illustrated in Figure A1.9 for voltage ratios. The lower scale in Figure A1.10 allows us to ascertain dB gains or losses based on distance ratios or actual measured distances (in inches, feet, meters, and so forth).

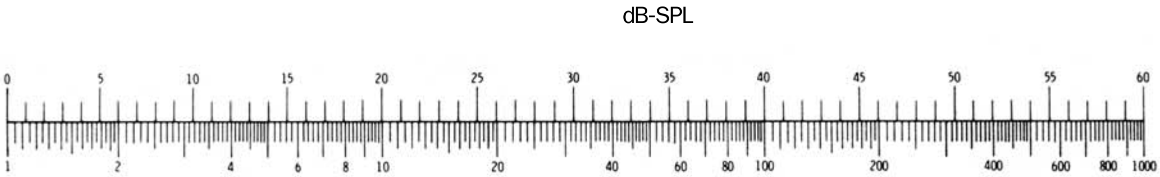


Figure A1.10 Nomogram for Determining Relative SPL Directly in dB

1. Find the dB loss for the problem cited above. (Since our distance ratio is 12, the loss is <u>21.6 dB</u>.)
2. What is the SPL loss between 4 feet (1.22 m) and 80 feet (24.38 m)? (The ratio is 20, so the loss is <u>26 dB</u>.) Or read 12 dB opposite 4 feet and 38 dB opposite 80 feet. The loss will be the difference. (38 − 12 = <u>26 dB</u>)
3. What increase in SPL level at the microphone would we expect when the working distance is reduced from 10 inches (25.40 cm) to 4 inches (10.16 cm)? (Since the ratio is 2.5, level will increase by <u>+8 dB</u>.)
4. Some older specification sheets reference loudspeaker sensitivity at four feet instead of one meter. What is the SPL change from one meter (3.28 ft) to four feet (1.22 m)? (<u>−1.7 dB</u>)

Watts and SPL

61. We will now consider practical examples of how we can use this information. If we know the dB-SPL that our loudspeaker can produce at a specific distance with a specified power applied, we can calculate the SPL it would produce at any given power. Or, working backward, if we require a certain sound level at a distant row of seats, we can calculate the power our loudspeaker would require to produce that level. To do this, we will need one important specification for our loudspeaker—its *sensitivity,* usually given in decibels.

62. For example, the specification sheet for a particular loudspeaker notes the following: "96 dB-SPL (1 watt @ 1 meter)." This means that with one measured watt of pink noise applied to this speaker, it will produce a level of 96 dB-SPL, measured on the speaker's axis at a distance of one meter. Some manufacturers may base their specification on a measured distance of four feet rather than one meter, or a different power level may be used. As noted by our last example above, the dB loss from one meter to four feet is −1.7 dB, which can easily be factored in when required. By utilizing the equations we have learned, we can convert the given specifications to a consistent standard.

63. To get acquainted with the math involved, let us work out a practical problem: Assume that we have a loudspeaker with a rated sensitivity of 96 dB-SPL (1 watt @ 1 meter). We are outdoors and determine that we require an SPL of 80 dB at the last row of seats, which is 18 meters (59 ft) distant from the loudspeaker. How much power is required to drive our loudspeaker at this level?

64. First, we must calculate the acoustic loss due to the inverse square law over the 18-meter distance. When we add this loss to the required 80 dB, we will know what SPL our loudspeaker must produce at one meter. From the SPL nomogram (Figure A1.10), we read 25.1 dB opposite 18 meters. Thus, the loss at 18 meters will be about 25 dB, or to calculate our loss mathematically:

$$\mathbf{dB = 20 \log D_f/D_n = 20 \log 18/1 = 20 \log 18 = 20 \bullet 1.255 = \underline{25.1\ dB}}$$

Therefore, we need 80 + 25 = <u>105 dB-SPL</u> at one meter from the loudspeaker.

65. Since we know that our loudspeaker produces 96 dB-SPL at one meter with a one-watt input, to attain the 105 dB-SPL will require a few more watts. How many more? First, we must determine how many dB will be required above the one-watt level.

$$\textbf{Additional level required} = \mathbf{105.1\ dB - 96\ dB = \underline{9.1\ dB}}$$

66. How much power is required to raise the SPL of this loudspeaker an additional 9.1 dB? If we refer to the dB power nomogram (Figure A1.8), the answer is easy. From the center of the upper scale ("0"), we move left to +9.1 dB. The figure opposite, on the lower scale, is 8. Thus, our loudspeaker will require 8 • 1 watt, or <u>8 watts,</u> to attain the additional SPL. That power increase should give us the 80 dB needed at 18 meters. By basing our sensitivity specification on one watt, we can drop the ratio and read the new power ratio directly from the lower scale. This nomogram should prove helpful.

67. For math enthusiasts, the following formula can be used to find power (watts) when the change in dB is known:

$$\mathbf{P_1 = P_2 \bullet \text{antilog}\ 10^{dB/10}}$$

Now we insert the known figures (P_2 = 1 watt; dB = 9.1):

$$\mathbf{P_1 = 1 \bullet \text{antilog}\ 10^{9.1/10} = 1 \bullet \text{antilog}\ .91 = 1 \bullet 8.1 = \underline{8.1\ watts}}$$

68. Note that it was necessary to change our dB figure back to a logarithm by dividing it by 10 (9.1/10 = .91). Now, what is the antilog of .91? Using our scientific calculator, we find it is 8.1, which represents the ratio of increased power needed. Simply multiplying the sensitivity of one watt by this ratio gives us the actual power required.

69. If we are driving ten loudspeakers at this level, then we will need a total of 80 watts (10 • 8 watts) from our amplifier, plus some headroom for peaks. How much headroom? To achieve a comfortable headroom of 6 dB below clipping, how much power do we need? Earlier in this discussion we learned that a 6 dB increase in level will require a fourfold increase in power. (See the chart below.) To achieve this safely, our amplifier should be capable of producing at least 320 watts (4 • 80 watts) of audio power and our loudspeakers must be capable of handling 32 watts each. (More information on headroom can be found in Chapter 4, paragraph 55.) Perhaps from this brief consideration, we can see that we sometimes need to do a little homework to achieve a stable and reliable sound system.

Common Decibel Ratios

70. Following are some common dB notations:

Power Ratio	dB Change	Voltage or Distance Ratio (SPL)
2:1	3 dB	1.4:1
4:1	6 dB	2:1
10:1	10 dB	3.16:1
100:1	20 dB	10:1
1,000:1	30 dB	31.6:1
10,000:1	40 dB	100:1
100,000:1	50 dB	316:1
1,000,000:1	60 dB	1,000:1

71. We hope this discussion of electrical fundamentals and the use of decibels will assist you in planning and engineering successful sound systems at future assemblies and conventions. The nomograms included herein will provide quick answers to many of the decibel problems you are likely to encounter. For convenience, these nomograms along with various decibel formulas are found in Appendix 7.

APPENDIX 2

Physical Principles of Sound

1. To most persons, sound is simply what we hear. In the field of sound reinforcement, however, we perceive sound as much more than an auditory sensation. It is also a controllable and predictable form of energy—a physical phenomenon that adheres to well-established principles. These principles are based on Jehovah's laws of creation, laws that we hope to live with for a very long time. Since sound is invisible to the eye, understanding a few of the basic physical laws that govern the behavior of sound will assist us with designing and planning effective sound systems and arriving at successful solutions to specific acoustic problems that we encounter from time to time.

What Is Sound?

2. Simply expressed, sound is a series of very rapid pressure variations transmitted through the medium of the air surrounding us. In ordinary conversation, these vibrations are incredibly small, typically varying the atmospheric pressure by approximately one millionth of a pound per square inch. When these minute pressure vibrations reach our eardrums, we perceive them as discrete, or separate, intelligible sounds.

Figure A2.1 Expanding Ripples in Water Are Similar to Sound Waves in Air

3. In certain respects, radiating sound waves can be compared to the expanding waves that result when a droplet of water falls into a quiet pool of water (Figure A2.1). The alternating crests and troughs, illustrated in Figure A2.2, correspond to the variations in pressure generated by a vibrating body in contact with air. That body could be a person's vocal cords, the sounding board of a piano, or a loudspeaker cone. The height of these waves and the distance they will travel are determined by the strength of the disturbance, illustrating acoustic *amplitude,* or *intensity.* For example, a small droplet creates small ripples that travel only a short distance before their energy is spent. Dropping a larger amount of water creates a greater disturbance, generating waves of greater height that travel much farther across the water's surface. These may reflect off a boundary surface at the edge of the pool and continue in an opposite direction, simulating the effects of reflected sounds that we perceive as echoes.

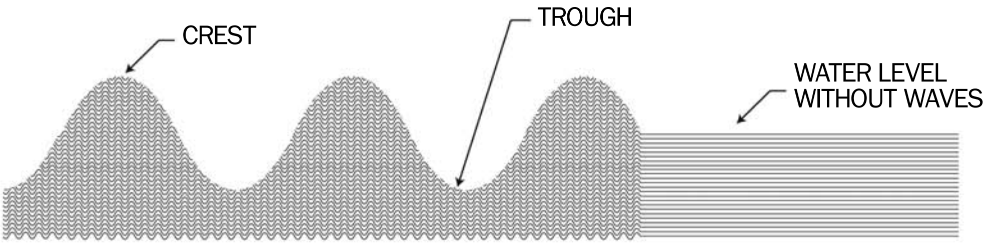


Figure A2.2 Wave Excursions Created by an Object Falling Into the Water

4. However, there are two important dissimilarities between the water analogy and real sound waves in the elastic medium of air. (1) Air particles excited by a vibrating sound source do not literally move up and down like waves on the surface of water. As the vibrating surface moves forward in a positive direction, the particles of disturbed air actually bump one another as they pass their positive energy from particle to particle, creating a positive pressure wave. As the vibrating surface moves backward in a negative direction, the pressure wave is replaced by a rarefaction, or decrease in density and pressure, which is then followed by another positive pressure wave, and so on. The result is a series of wave fronts alternately compressed and rarefied in the air. (2) The water surface illustrates the expanding disturbance only on the horizontal plane, whereas in reality sound is omnidirectional, radiating in *all* directions as a series of expanding spheres, although rarely with equal intensity in all directions.

Physical Characteristics

5. Sound has several measurable characteristics, which include *frequency, level, acoustic loss due to distance, velocity, wavelength,* and *phase*. Features of the acoustic environment both outdoors and indoors will also significantly affect the behavior of sound. These are discussed in greater detail in Appendix 3.

6. *Frequency:* Everyone is familiar with the low and high tones of voice and music, sometimes referred to as bass and treble sounds. To define these tones, we relate them to *time,* bringing us to an extremely important dimension in the field of sound—*frequency*—the *number* of pressure waves, or cycles, to reach our ears in a one-second interval. For that reason, frequency was once expressed as *cycles per second* (*cps*). The unit *hertz* (*Hz*) is now used to define frequency in cycles per second. As illustrated in Figure A2.3, one *cycle* is graphically defined by the interval from one point on the waveform, through all its excursions, to the same point on the next cycle of the waveform.

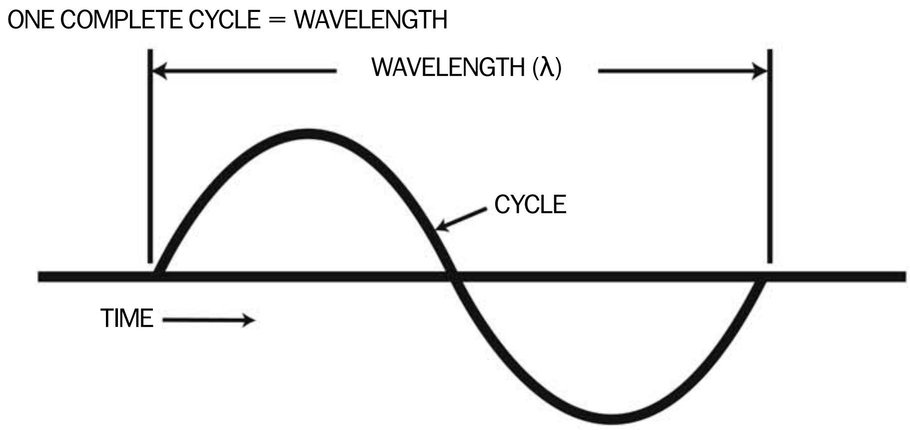


Figure A2.3 One Complete Wave Cycle

7. A low, or bass, tone has a low frequency. The lowest frequency that our ears interpret as a discrete sound is considered to be about 20 Hz. A sound described as "high pitched" will have a high frequency. The more pressure waves that occur each second, the higher the frequency (or pitch). The highest frequency perceived by humans is around 20,000 hertz (or 20 kilohertz [kHz]), a range generally limited to people under 25 years of age. (The prefix "kilo," meaning "1,000," is often used to express larger numbers of hertz. Thus, "20,000 Hz" can simply be written as "20 kHz.")

8. It will be helpful for the soundman to become familiar with the frequency range of common sounds that we regularly encounter. Figure A2.4 illustrates the spectrum of audible frequencies that relate to the musical scale, which is familiar to many. The frequencies in hertz shown below the piano keyboard refer to the *fundamental,* or predominant, frequency of each tone in the scale. You will observe that the range of the human voice falls about midscale. The fundamental tones of the male and female voice, as indicated by the solid lines, include only about four octaves of the acoustic spectrum, beginning around 90 Hz for a bass voice and extending to around 1,400 Hz for a soprano

9. It may surprise some to learn that the fundamental frequency of the highest note on the piano keyboard is only a little over 4 kHz, specifically, 4,186.01 Hz. Observe that the range of many orchestral instruments is shown to extend as high as 16 kHz, as indicated by the shaded lines. These high-frequency musical sounds are

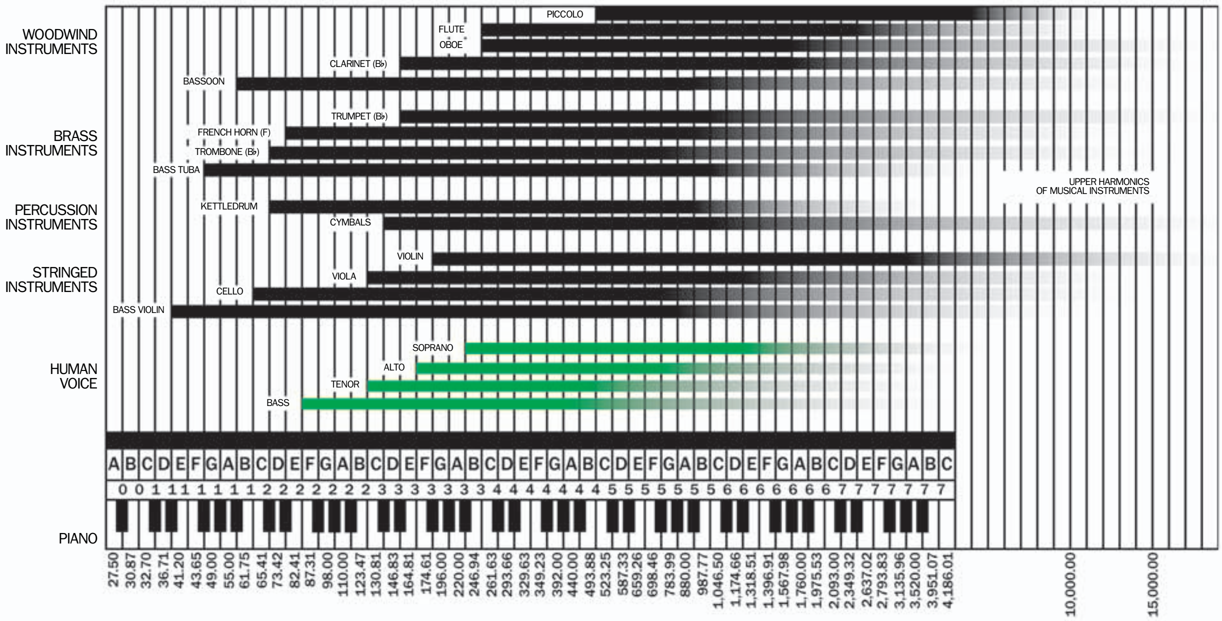


Figure A2.4 Frequency Ranges of Musical Instruments and the Human Voice

known as *partials,* or *overtones.* They include harmonics, which are multiples of a predominant fundamental frequency, as well as other frequencies not necessarily even harmonics of the fundamental. Overtones vary greatly in intensity and character, depending on the resonances peculiar to the particular instrument. Thus, middle C (261.63 Hz) on a piano sounds uniquely different from middle C played by a violin, French horn, or bassoon. Even though each instrument is capable of producing the same tone, or fundamental frequency, a unique combination of resonances and overtones gives each instrument its own distinctive timbre and character. Similarly, variations in character are heard in the human voice. Each of us has a voice that is unique, each with its own identifiable characteristics. A quality sound system capable of accurately reproducing this broad range of distinctive tones and overtones is often referred to as a *high-fidelity* system. Two other important topics related to frequency will now be discussed, namely, the octave and the articulation range.

10. (1) *The Octave*

- Important to our comprehension of relative frequencies is understanding the *octave.* To the musician, an octave consists of two harmonic tones eight notes apart on the musical scale, as illustrated in Figure A2.4. In the acoustic and audio world, we relate the octave to frequency, so this same octave is conceived of as the interval between any two frequencies having a ratio of 2:1. For example, find A4 (440 Hz) near the center of the keyboard in Figure A2.4. This is “concert A,” the frequency, or “pitch,” as it is known to musicians, universally used to tune the orchestra. If we move up the scale one octave, we come to

A5, with a frequency of 880 Hz, double that of A4. Doubling the frequency again to 1,760 Hz raises us another octave to A6, and so on. If we divide the frequency of A4 (440 Hz) by two, we have 220 Hz, which is the fundamental frequency of A3, one octave lower on the scale, and so on. The lowest tone on the piano keyboard is A0, with a fundamental frequency of 27.5 Hz.

- This helps us to see the mathematical progression followed by the octave scale, where each succeeding octave step upward covers twice the number of frequencies as the step before it. Our ears accept any doubling or halving of a frequency as an octave. Thus, the step from 80 Hz to 160 Hz is one full octave, a step of 80 frequencies. Likewise, the step from 8 kHz to 16 kHz is still only one octave, although it includes 100 times the number of frequencies—a step of 8,000 frequencies!
- Note that this mathematical progression follows a *logarithmic scale* rather than a *linear scale,* where the divisions are of equal value, as seen on a common ruler. The logarithmic scale occurs frequently in the sound field. This is not unusual in nature, for the ear's sensitivity to sound and the eye's sensitivity to light follow a logarithmic curve much more closely than a linear one. Understanding the octave and its relationship to the audible *frequency spectrum* is of considerable value in managing audio and acoustic values.

11. (2) *Articulation Range*

- Since the nature of our spiritual programs features the spoken word, speech clarity is a key requirement for a successful sound system. Most critical to the intelligible reproduction of speech is the *articulation range,* the octave from 2 kHz to 4 kHz, which contains most of the consonantal and sibilant sounds essential for clear articulation. This is discussed in greater detail in Appendix 4.

12. *Sound Pressure Level* (*dB-SPL*): In the science of sound, several units of measurement have been devised to measure sound intensity and level. The *decibel,* often abbreviated *dB,* is most commonly used in the sound industry and has proved quite adequate for our purposes. Since decibels are also used to define other numerical ratios, such as audio power and voltage, it is customary to add the suffix "SPL" (Sound Pressure Level) to the dB measurement when it defines sound level, thus, *dB-SPL.* A professional sound-level meter display reads in dB-SPL. (dB-SPL levels are referenced to the *threshold of hearing,* which is considered to be 0 dB-SPL. For a more detailed discussion of the decibel, please see Appendix 1.)

13. Figure A2.5 illustrates the approximate dB-SPL of various common sounds. For example, near the center of the scale, we observe that the sound pressure level for normal conversation (measured at three feet [0.91 m] from the source) is about 65 dB-SPL.

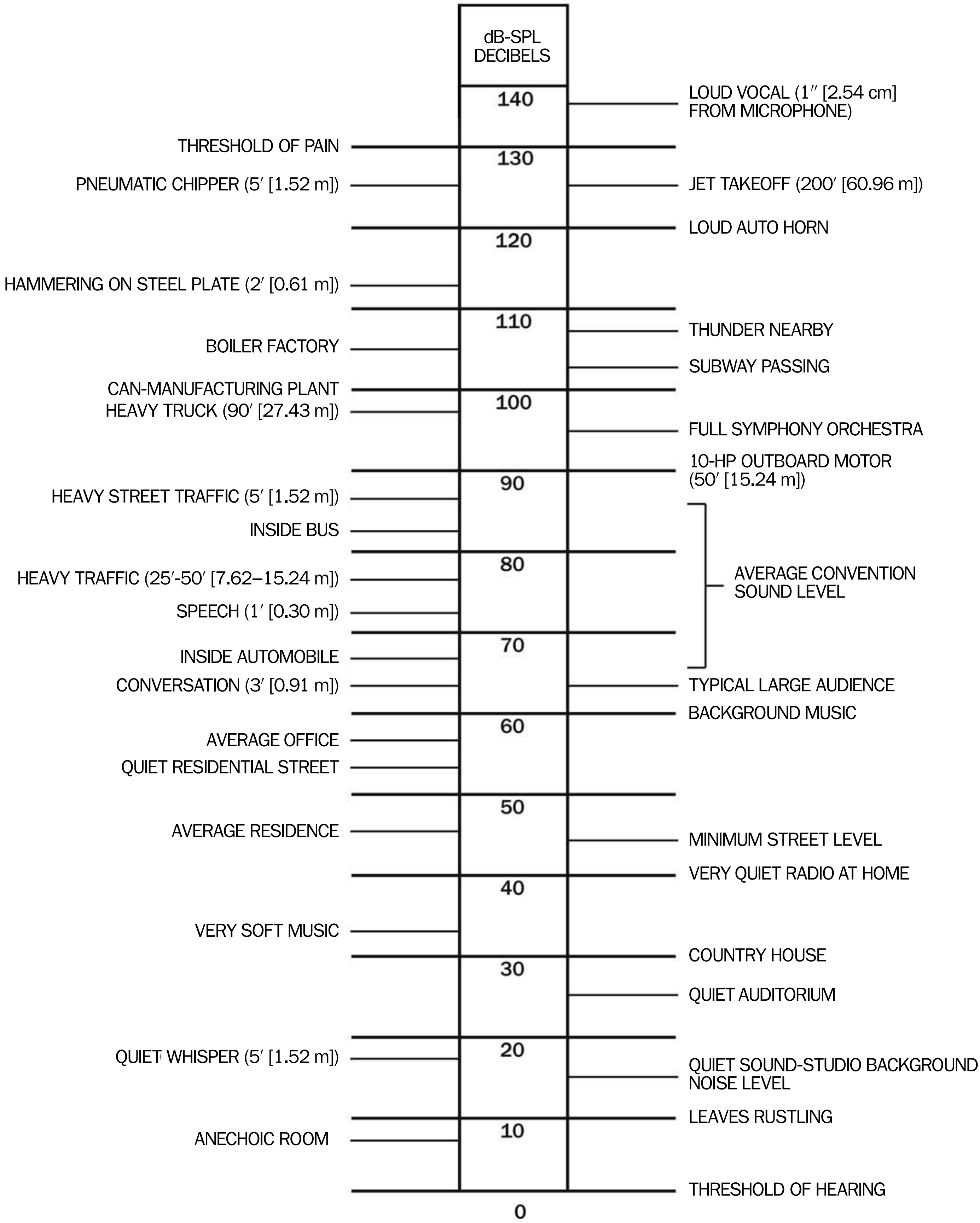


Figure A2.5 Chart of Relative SPL Levels of Common Sounds

The sound level at a convention (measured at the seats) might average between 70 and 75 dB-SPL or a little higher, depending on the ambient, or surrounding, noise. Indeed, the character of the acoustic environment, the distances involved, and the ambient noise will all have a significant effect on the SPL demands on our sound system.

14. *Sound Level Versus Loudness:* The terms “sound level” and “loudness” are not synonymous. *Sound level* is a physical dimension that can be measured with a sound level meter. *Loudness,* on the other hand, is a subjective measure of sound sensation to our ears, which can be influenced by a variety of factors. The conclusion that a sound is “too loud,” “too soft,” or “loud enough” depends on such variables as ambient noise level, personal hearing acuity, and personal preference. The word *volume* is often used as a synonym for loudness.

15. *Acoustic Loss Due to Distance:* From experience we know that as we walk away from a person who is speaking, the sound of his voice will decrease in volume until it eventually becomes lost in the surrounding noise or disappears below our threshold of hearing. Conversely, as we walk toward that person and distance *decreases,* the intensity of his voice *increases.* What we may not realize is that this change in sound intensity is fairly predictable, for it follows a natural law known as *inverse square law.* Briefly, the law states that in *free space,* where there are no obstructions or reflections, sound level will *vary inversely with the square of the distance change.*

16. It might be helpful to recall the illustration at the outset of this appendix where it was stated that, in certain respects, radiating sound waves can be compared to the expanding waves that result when a droplet of water falls into a quiet pool. Later, it was explained that one of the differences is that the water surface illustrates the expanding disturbance *only* on the horizontal plane, whereas in reality sound is omnidirectional, radiating in *all* directions. With that in mind, consider Figure A2.6. As noted earlier, sound waves propagate spherically, as illustrated by the expanding circles. Using the typical solid angle shown, note that all the sound that passes through the small box at radius *d* also passes through the boxes at *2d, 3d,* and *4d.* However, we note that as the radius increases, the area of the boxes increases in size according to the *square* of the distance change, so that when we double the radius at *2d* the area of the box *increases* by a factor of 4 (2^2) and sound intensity *decreases inversely* to only one fourth. In the box at *3d,* sound intensity drops further to one ninth. At *4d,* intensity drops to one sixteenth, and so on.

17. Here is a common and very useful rule: *Sound level varies 6 dB with each doubling or halving of the distance.* To illustrate: If we measure the sound pressure level at 20 feet (6.10 m) from a

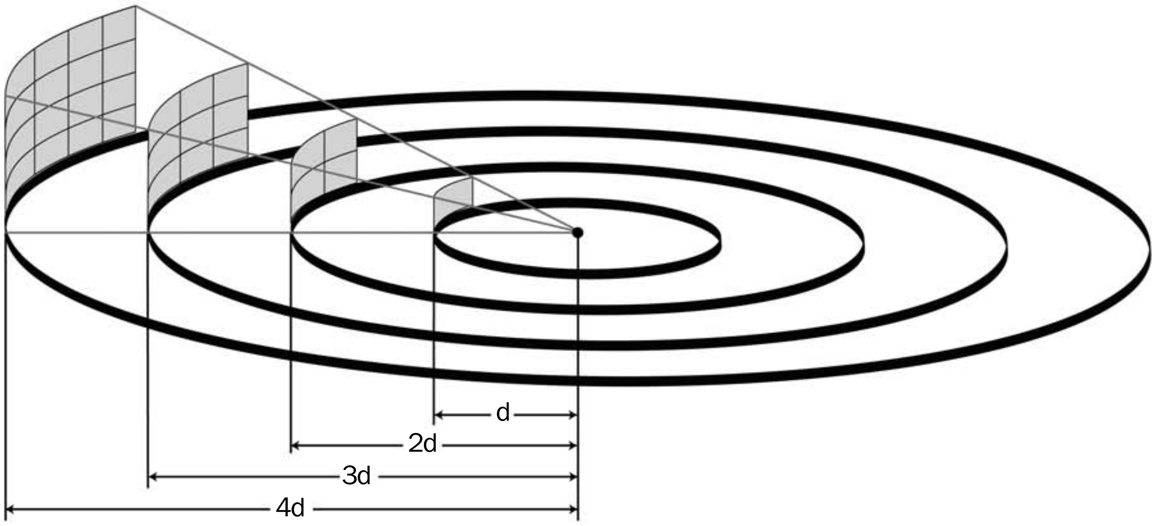


Figure A2.6 Sound Intensity Varies With the Square of the Distance Change

sound source and then move to one half that distance, or 10 feet (3.05 m), decreasing by a factor of two, we find that our sound level increases by *approximately* 6 dB, or four times (2^2) the level at the first location. Moving the other direction, if we are outdoors and measure the sound 20 feet from a loudspeaker and then move to a point 80 feet (24.38 m) from the loudspeaker (four times the first distance), we would expect the sound level to drop to one sixteenth of the source level (4^2), a loss of *approximately* 12 dB (20 feet [6.10 m] to 40 feet [12.19 m] = 6 dB loss; 40 feet [12.19 m] to 80 feet [24.38 m] = 6 dB loss; total loss = 12 dB). We use the term "approximately" because inverse square law technically applies only in *free space,* an area free of obstructions and reflections. Rarely do we encounter such a space, for even reflections from some type of existing floor can alter the precise application of inverse square law. Still, this law provides a useful way to estimate sound level under many circumstances. Other methods for predicting acoustic gains and losses, such as the use of nomograms, are illustrated in Appendix 1.

18. *Sound Velocity and Delay:* It is a well-known fact that sound does not travel through the air instantaneously but, rather, at a fairly constant speed, or *velocity.* This velocity is defined as approximately 1,130 feet, or 344 meters, per second. Thus, a clap of thunder that is heard one second after a lightning flash is seen indicates that the lightning, happily, is about 1,130 feet away from us. Although the velocity, or speed, of sound varies slightly with changes in temperature, humidity, and atmospheric pressure, the figures given here for a temperature of 70°F. (21°C.) are adequate for our purposes. Here are simple equations for determining sound delay for any given distance:

Delay (seconds) = Distance (feet) ÷ 1,130 or Distance (meters) ÷ 344

When we prefer to represent delay time in *milliseconds,* the following equations are useful:

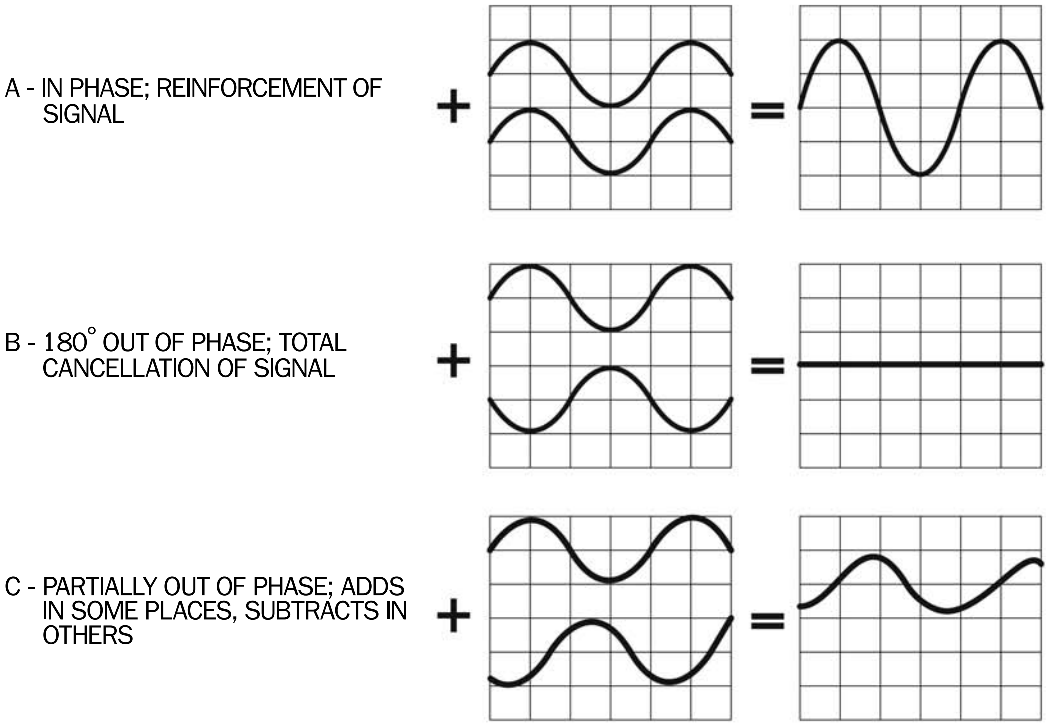
Delay (milliseconds) = Distance (feet) • .885 or Distance (meters) • 2.9

19. *Wavelength:* A *wavelength* is the *physical distance* between two successive sound waves, as illustrated in Figure A2.3. Visualize a tone having a frequency of 100 Hz, where a block of 100 complete wave cycles will pass a fixed point during a one-second interval. If the tone were 1 kHz, ten times as many cycles would pass that same point in one second. Since the velocity of sound in air may be assumed to be the same over the entire spectrum, we can conclude that wavelength varies inversely with frequency. Thus, a wavelength for a 100 Hz tone is 11.30 feet (3.44 m), whereas for a 1 kHz tone, a wavelength is only 1.13 feet (34.44 cm). A wavelength for a 10 kHz tone is a mere 1.36 inches (3.45 cm). Wavelengths for the aural spectrum of 20 Hz to 20 kHz range from 56.50 feet (17.22 m) for 20 Hz to 0.68 inch (1.73 cm) for 20 kHz. Here are the simple equations for determining the wavelength for any given audible frequency (f = frequency in hertz):

Wavelength (feet) = 1,130 ÷ f or Wavelength (meters) = 344 ÷ f

20. The concept of relative wavelength is important to our understanding of the refraction of sound around obstacles of various sizes. Wavelength also dictates the lowest frequency that can be satisfactorily managed by a horn of a specific length. For example, a small horn may provide satisfactory control over the relatively short wavelengths of higher frequencies but virtually no control over the longer wavelengths of bass frequencies.

21. *Phase:* This is another important characteristic of our understanding of sound, since it affects how sounds sum together. Again, using a simple sine wave, we can visualize graphically the relative effect that phase relationship has on the combining of sound waves. If we take two coherent waveforms of the same frequency, shape, and amplitude and start them simultaneously, the resulting waveform will be of the same frequency and shape but twice the amplitude of the originals, as shown in Figure A2.7a. These two waveforms are said to be at 0° phase with each other, and we sometimes refer to them as *in phase* or *at minimum phase.* Thus, *in-phase* coherent signals will always *reinforce* each other. If we take two coherent waveforms identical to those just described, except that they are 180° *out of phase,* each will cancel the other out, resulting in no sound at all, as seen in Figure A2.7b. Here we can visualize that at any instant, the positive pressure of one wave is exactly balanced by the corresponding negative pressure of the other. When there is no variation in air pressure, there is no sound. Thus, we can conclude that *in-phase* signals reinforce; *out-of-phase* signals cancel out. Where two identical signals are partially out of phase, as in Figure A2.7c, a partial cancellation or reinforcement takes place and the resultant signal is somewhere between zero and twice the original signal level, depending on the phase difference.



Figures A2.7a, b, c Combining Sine Waves of Equal Frequency and Amplitude

22. The effects of phase relationships are by no means limited to the acoustic environment. These considerations are perhaps even more important in the audio or electrical portion of the sound system. Correct polarity should always be observed, especially in the connection of electro/acoustic transducers, such as microphones and loudspeakers. The acoustic environment will be discussed further in Appendix 3.

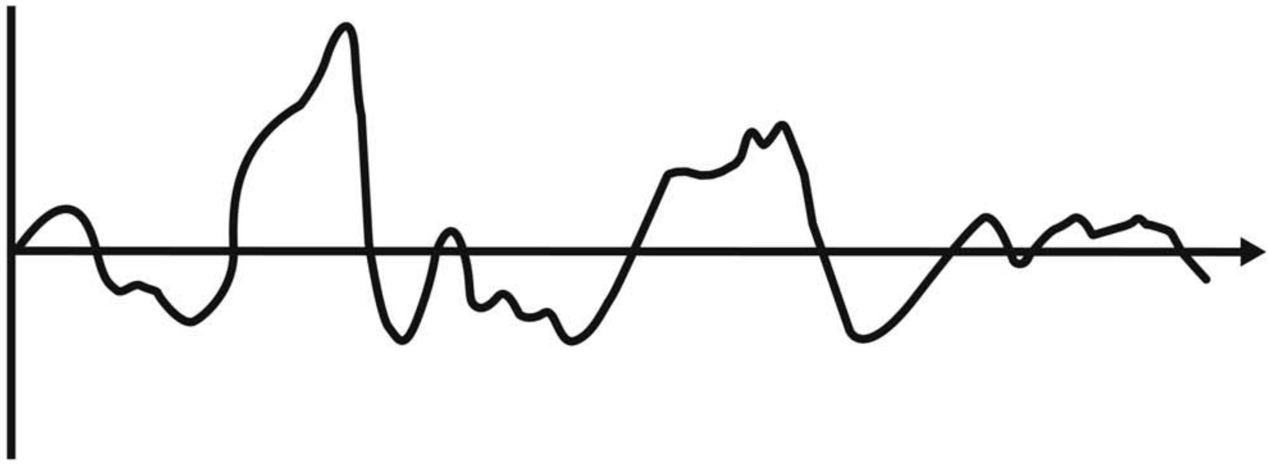


Figure A2.8 Complex Waveform Generated by a Single Spoken Word

23. Up to this point, we have been looking at sound waves in their simplest form, the sine wave, a waveform usually confined to the lab or test bench. In reality, nearly all the sounds we hear are *complex* in nature, consisting of a fusion of many frequencies, amplitudes, and phase relationships. Consider the type of wave created by a single spoken word in Figure A2.8. Since such waveforms are seldom symmetrical and often do not repeat, it is difficult, if not impossible, to divide them neatly into cycles or to categorize them as to frequency.

24. It is hoped that a better understanding of the fundamentals of sound will assist you to cope with the acoustic environment unique to each specific convention site, as well as assist in predicting and managing the behavior of the acoustic energy produced by the sound system.

Coping With the Acoustic Environment

1. A correlation always exists between the physical characteristics of sound and the acoustic nature of the environment as defined by its atmosphere, its dimensions, and the nature of its surfaces. Other significant factors include the characteristics of the loudspeakers selected and the manner in which they are installed and operated—all of which combine to create a unique sound system.

The Outdoor Environment

2. While sound outdoors in open areas is governed mainly by inverse square law, as described in Appendix 2, it is also good to be aware of other forces that can adversely affect a sound system's performance.

3. *Losses Due to Low Relative Humidity:* Over large distances, the atmosphere itself becomes increasingly resistive to sound, especially as frequency increases. Low *relative humidity* contributes to an additional loss of high frequencies because dry air is more resistive to sound than moist air. The lower the humidity, the greater the loss, as illustrated in Figure A3.1.

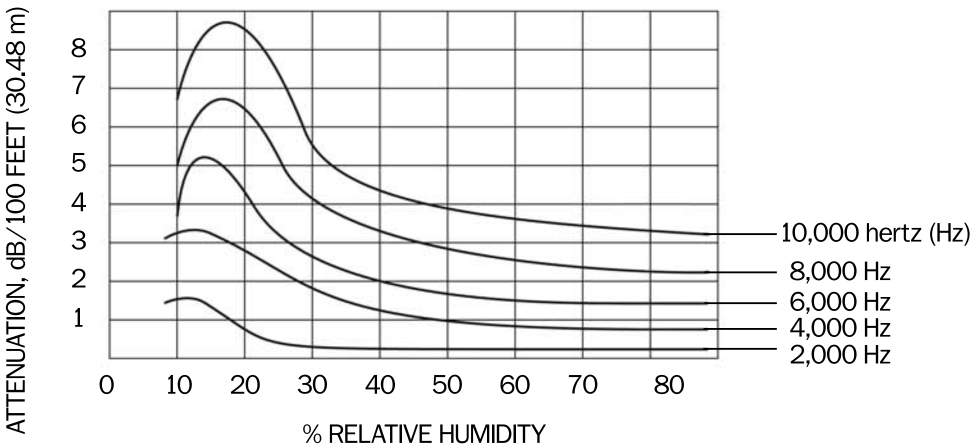


Figure A3.1 Attenuation Due to Variations in Relative Humidity

4. When relative humidity drops to 20 percent or lower, substantial attenuation of high frequencies (4 kilohertz [kHz] and above) can be anticipated, increasing with distance and rising frequency. Such losses can often be offset by equalization, but care must be exercised to avoid destroying the tonal balance for listeners seated near the loudspeakers.

5. *Refraction of Sound:* Directional changes in sound, known as *refraction,* are introduced when sound passes between zones of different temperatures. The velocity of sound increases slightly with rising temperature (1,130 ft/sec at 70°F [344.42 m/sec at

21.11°C]; 1,141 ft/sec at 90°F [347.78 m/sec at 32.22°C]). As illustrated in Figure A3.2, an upward "bending," or refraction, of the sound path occurs when warmer air is below cooler air. If the temperature zones are inverted so that cooler air is below a layer of warmer air, the sound path will be refracted downward. This effect can be observed outdoors, especially when loudspeakers are located near the ground. Mounting loudspeakers four feet (1.22 m) or more above the ground will help to minimize this problem.

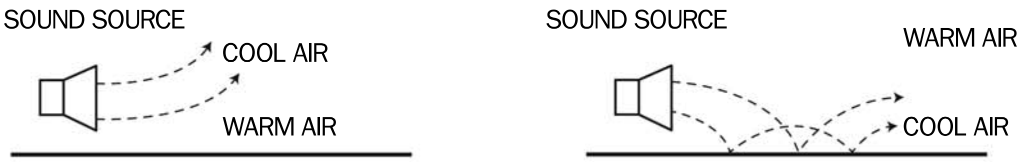


Figure A3.2 Sound Refraction Caused by Temperature Differences Between Ground and Air

6. Acoustic refraction may also be encountered because of wind velocity, especially where sound is projected over large distances, as illustrated in Figure A3.3. While moderate breezes have little effect, strong breezes and wind gusts can produce a distracting fading effect, degrading overall level and clarity. Increasing wind velocity may also increase ambient noise, and the sound system may be called on to compensate for this. As illustrated, wind velocity is usually slower nearer the ground.

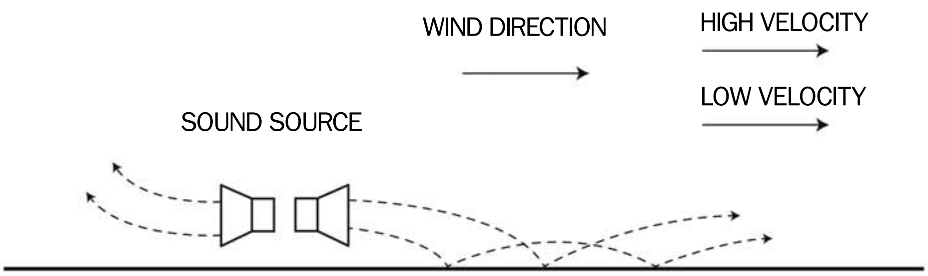


Figure A3.3 Effect of Wind Velocity on Sound Propagation

Behavior of Sound Indoors

7. The majority of our conventions utilize indoor venues, where we encounter additional acoustic factors that govern sound behavior. These include the dimensions of the room, its volume, its architecture, its shape, and the materials used in its construction. All the factors mentioned earlier are still at work, but these additional complex factors will surely influence the performance of our sound system. One of the chief obstacles to good clarity and intelligibility is *masking noise,* which can include any of the following: (1) excessive reverberation; (2) delayed high-level reflections (echoes); (3) late sound from distant loudspeakers; (4) incorrect equalization, such as excessive bass or weak high frequencies; and (5) distortion. When one or more of these negatives becomes prominent, intelligibility will suffer. The latter four maladies can usually be avoided or corrected by good system design, signal delay, and/or proper equalization. *Excessive reverberation,* on the other hand, can be more challenging, depending on the acoustic character of the environment.

Acoustic Reflection and Absorption

8. When the surfaces and furnishings of a large meeting room are composed of soft, sound-absorbent materials—acoustic ceiling and panels, carpeted floor, padded seats, and so forth—much of the sound is absorbed, rather than reflected, and reverberation is not likely to be a serious problem. While this could describe many Kingdom Halls and well-designed Assembly Halls, only a few of our rented convention facilities fit that description. In many of our convention venues, boundary surfaces (walls, floors, and ceilings) are constructed of hard and acoustically reflective materials, such as concrete or plaster. Seats are often composed of wood, molded fiberglass, or even concrete—all surfaces that are highly reflective to sound. Most of the acoustic energy impinging on them is reflected again and again until it is eventually absorbed by the room's objects or by the air itself. These profuse reflections combine with the natural delay in the movement of sound to create a condition known as *reverberation,* a form of masking noise that often impairs speech intelligibility severely for large portions of the audience.

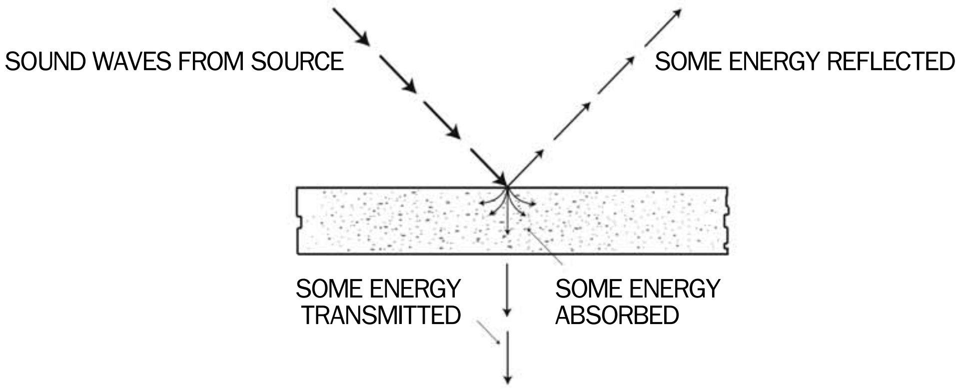


Figure A3.4 Distribution of Acoustic Energy as It Strikes a Boundary Surface

9. As illustrated in Figure A3.4, when sound energy strikes a surface, some energy is reflected, some is absorbed, and some may be transmitted through the material. These reactions vary greatly with the nature of the material itself, as well as with frequency and angle of incidence. In coping with the behavior of sound in an enclosed room, it is helpful to be aware of how certain materials react to sound energy impinging on them.

10. The chart in Figure A3.5 illustrates some of the more common architectural materials and surfaces we are likely to encounter. The average *absorption coefficients* are given for three bands of frequencies, centered at 125 Hz, 1 kHz, and 4 kHz. We readily observe that the amount of acoustic absorption for many of the materials traditionally used for acoustic treatment varies greatly over the frequency spectrum. The figure given as the coefficient can be assumed to be an average *percentage* of acoustic energy *absorbed* (or *transmitted*) by the material for that frequency band. The remainder is *reflected* back into the room. For example, carpet is often considered to have good sound absorption. Note that heavy carpet on concrete absorbs most of the high-frequency energy at 4 kHz and above, reflecting only about 35 percent of the sound. At 1 kHz the same carpet absorbs about 37 percent and reflects 63 percent, but at 125 Hz, only about 2 percent of the energy is absorbed, reflecting 98 percent. Installing felt or foam rubber backing under the carpet improves its absorption of low frequencies considerably. Ordinary window glass, on the other hand, reflects nearly all the high-frequency energy at 4 kHz and above but "absorbs" (transmits) considerable low-frequency energy. Heavy plate glass reflects nearly all frequencies.

11. Note, too, that the method of installing certain materials, such as acoustic tile, has a significant effect on their acoustic performance. Most porous materials display a dramatic increase in absorption, especially at lower frequencies, when an air space is maintained between the acoustic material and the reflective boundary surface behind it—an important fact to remember when planning the acoustic design of a Kingdom Hall or an Assembly Hall. Most manufacturers of acoustic materials are very willing to supply acoustic data for their products.

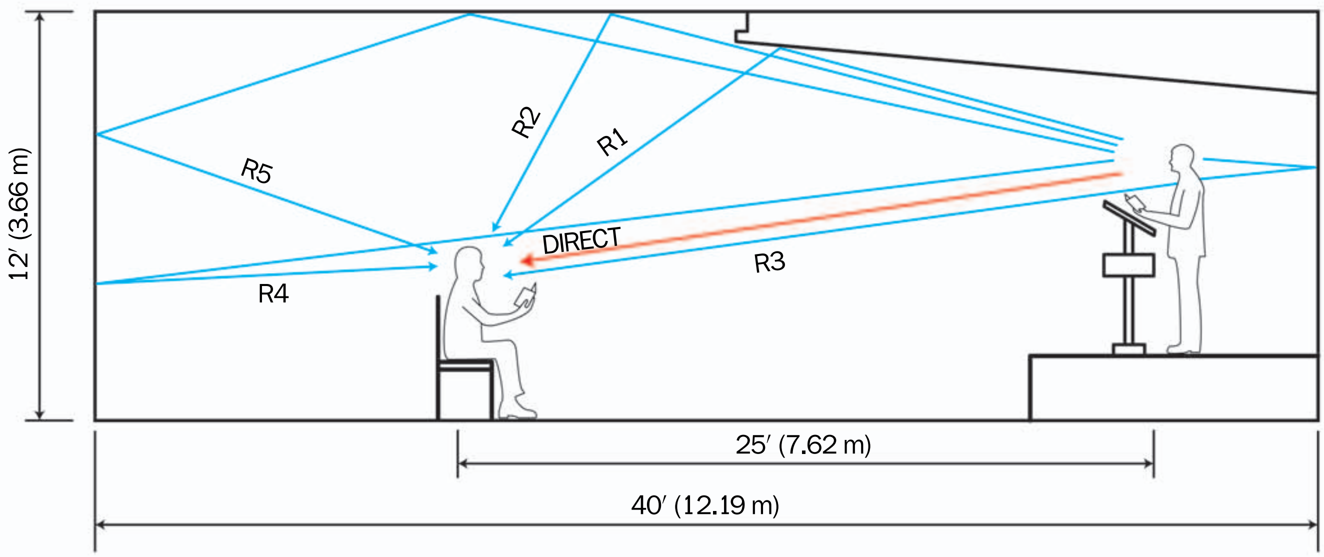
12. As shown, all construction materials reflect some of the acoustic energy that reaches them. In practical terms, there are really no perfect absorbers or reflectors of sound. However, having some idea of the relative acoustic characteristics of various materials will aid us in predicting the acoustic performance of a given environment.

MATERIAL	ABSORPTION COEFFICIENTS		
	125 Hz	1 kHz	4 kHz
Acoustic tile, 5/8″ (1.59 cm), cemented to plaster or concrete	.15	.70	.65
Acoustic tile, 5/8″ (1.59 cm), fastened to 1″ (2.54 cm) furring strips	.25	.70	.65
Acoustic tile, 5/8″ (1.59 cm), suspended ceiling, 16″ (40.64 cm) space above	.50	.75	.65
Audience area, fully occupied	.50	.95	.85
Audience area, unoccupied (upholstered theater seats)	.45	.90	.70
Brick, unglazed	.03	.04	.07
Carpet, heavy, on concrete	.02	.37	.65
Same, on 40-ounce (1.13 kg) hair felt or foam rubber	.08	.69	.73
Concrete block, coarse, unpainted	.36	.29	.25
Concrete block, painted	.10	.07	.08
Concrete, poured	.01	.02	.03
Fabrics, draperies			
Light velour, 10 ounces (.28 kg) per square yard (.84 sq m), hung straight, in contact with wall	.03	.17	.35
Heavy velour, 18 ounces (.51 kg) per square yard (.84 sq m), draped to half area	.14	.72	.65
Cotton, draped to half area	.07	.80	.50
Floors			
Concrete or terrazzo	.01	.02	.02
Linoleum, asphalt, rubber or cork tile on concrete	.02	.03	.02
Wood	.15	.07	.07
Glass			
Large panes of heavy plate glass	.18	.03	.02
Ordinary window glass	.35	.12	.04
Gypsum board, 1/2″ (12.70 mm), nailed to 2 x 4's (38 x 89 mm) 16″ (40.64 cm) on center	.29	.04	.09
Marble or glazed tile	.01	.01	.02
Plaster, rough finish on lath	.02	.05	.03
Same, with smooth finish	.02	.04	.03
Plaster, smooth finish on tile or brick	.01	.03	.05
Plywood paneling, 1/4″ (.64 cm), 2″ (5.08 cm) air space	.30	.10	.07
Plywood paneling, 3/8″ (.95 cm) thick	.28	.09	.11
Water surface, as in a swimming pool	.01	.01	.02

Figure A3.5 Absorption Coefficients for Various Architectural Materials

“Early” and “Late” Reflections

13. Figure A3.6 illustrates a room with dimensions that may closely approximate many Kingdom Halls. Depicted are the paths of reflected sound from the various boundary surfaces of the room, such as the ceiling and walls. The longest reflection path, labeled R5, reaches the listener 20 milliseconds (ms) after the direct signal arrives from the talker. Even though these reflected signals arrive somewhat later than the direct sound, the delay is well below 35 milliseconds. Thus, the listener perceives these reflections as one sound, since they actually reinforce the direct signal. Room reflections under 35 milliseconds, sometimes referred to as *early reflections,* serve to enhance sound quality as well as reinforce the overall level. This explains why, even without a sound system, it is much easier to talk to a group of people in a room rather than outdoors. Not only do the walls and ceiling keep extraneous noise out of the room but they also capture much of the acoustic energy from the talker that would normally be lost in space, reflecting it back into the room and reinforcing the direct sound of his voice. This advantage, known as *room gain,* can significantly reduce the audio power needed to achieve a desired sound level.

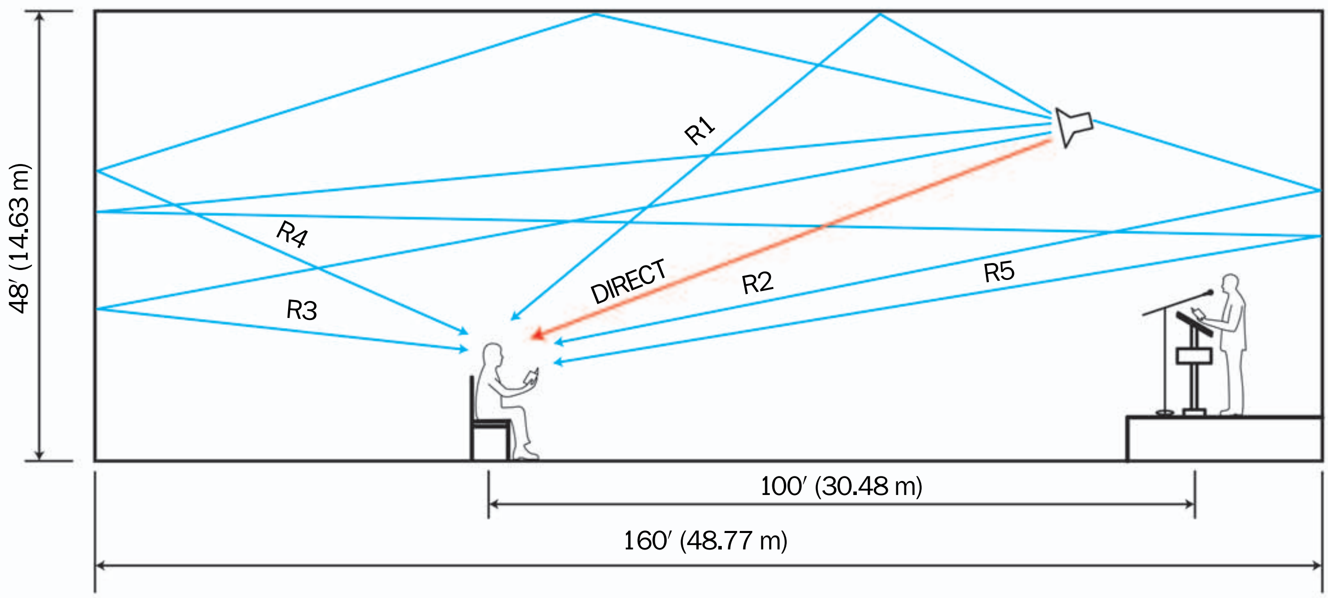


SOUND PATH	DISTANCE	DELAY
DIRECT	25 feet (7.62 m)	22 ms
R1	27 feet (8.23 m)	24 ms
R2	28 feet (8.53 m)	25 ms
R3	35 feet (10.67 m)	31 ms
R4	45 feet (13.72 m)	40 ms
R5	47 feet (14.33 m)	42 ms

All reflections shown are early reflections, arriving at the listener within 35 milliseconds after the direct sound

Figure A3.6 Early Reflections in Relation to Direct Sound in a Small Room

14. If we multiply all the dimensions of our hypothetical room by four, we will have a room the size of a large hall or perhaps an Assembly Hall auditorium, as illustrated in Figure A3.7. Because of the room's size, we have installed a loudspeaker to assist the talker in reaching his audience, which is now much farther away. Since our listener is about 80 feet (24 m) from the loudspeaker, we can calculate that it will require about 71 milliseconds for the direct signal from the loudspeaker to reach his ears. Reflection R1 arrives at the listener 22 milliseconds after the direct signal. This is an early reflection arriving within the 35-millisecond limit and therefore creates no problem with clarity. But note the problems that develop because of other sound reflections within the room. Reflection R2 arrives 53 milliseconds later; reflection R3, 90 milliseconds later; reflection R4, 106 milliseconds later; and reflection R5, although weaker, arrives 283 milliseconds later. Reflections R2-R5 are *late reflections,* arriving at our listener's ears well past the 35-millisecond limit for good articulation. This sets up a condition that can severely degrade clarity and intelligibility.



SOUND PATH	DISTANCE	DELAY
DIRECT	80 feet (24.38 m)	71 ms
R1	105 feet (32.00 m)	93 ms
R2	140 feet (42.67 m)	124 ms
R3	182 feet (55.47 m)	161 ms
R4	200 feet (60.96 m)	177 ms
R5	400 feet (121.92 m)	354 ms

R2, R3, R4, and R5 are late reflections, contributing to the diffuse reverberant field reducing intelligibility

Figure A3.7 Late Reflections in Relation to Direct Sound in a Large Room

15. While our drawing illustrates only four late reflections on a single plane, in reality our listener is bombarded by a fusion of late reflections from all directions—ceiling, floor, walls, and so on—all arriving at his ears at different times determined by the length of the path the sound has traveled. The buildup of masking noise in a reverberant room is relatively slow because of the transit time required by sound reflecting from surface to surface. These late reflections may continue to multiply in number, producing that unintelligible combination of sounds we call *reverberation*. Unless our loudspeaker is sufficiently directional, its direct signal may well be masked by a profusion of diffused reflections, making speech comprehension very difficult, if not impossible, for our listener. In this room we have encountered a potential obstacle to articulate sound that was not present in the smaller room.

Reverberation Time (RT_{60})

16. *Reverberation time,* or RT_{60}, is defined as the time required for a sound in a room to decay 60 decibels (dB), or to one millionth of its original intensity. In very rough human terms, it is the approximate time for a very loud sound to decay to inaudibility. As illustrated in Figure A3.8, a two-second reverb time requires two seconds for a sound impulse of 100 dB-SPL to decay by 60 dB, or to 40 dB-SPL. Rarely, however, do practical circumstances allow for measuring a full 60 dB of decay. As the reverberant field disappears into the ambient noise present in the room, instruments designed to measure RT_{60} usually measure only the first 20 or 30 dB of decay, then extrapolate the actual RT_{60} from this measurement, as illustrated in Figure A3.9.

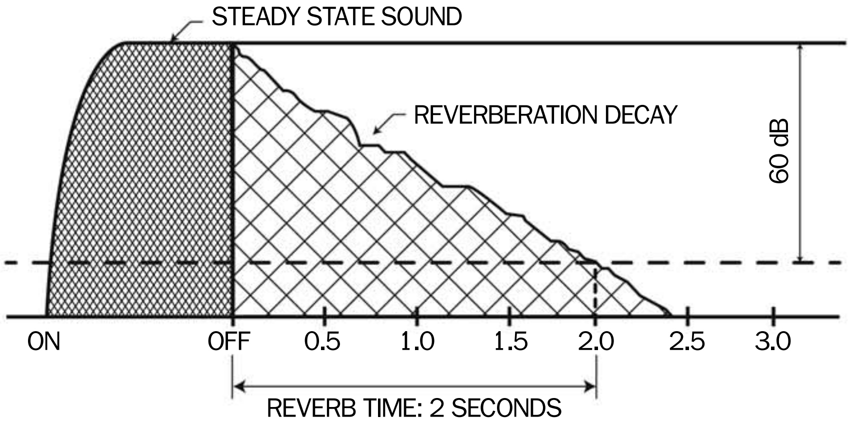


Figure A3.8 Sound Decay of a Typical Room With an RT_{60} of Two Seconds

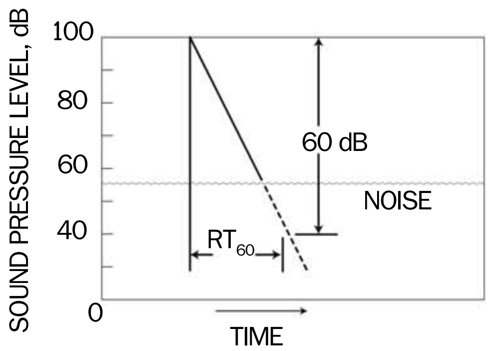


Figure A3.9 The Slope of the Limited Decay Extrapolated to Determine Actual RT_{60}

17. As we have seen, the rate of decay is governed mainly by three factors: (1) The loudness of the energy injected into the room, (2) the amount of acoustic absorption in the room, and (3) the distance that sound must travel between reflecting surfaces. The farther apart these surfaces, the longer it takes for reflections to travel from one surface to another and the longer reverberation persists in the room before it is absorbed or decays to inaudibility. Often, a simple handclap can provide an indication of the potential problems that can be anticipated from reverberation. If the sound of our handclap is still audible after two seconds, we can be certain that the reverberation is significant, for the actual RT_{60} is considerably greater than the reverb time perceived by our ears. On the other hand, when the sound impulse generated by our handclap disappears within a second or less, we can be grateful that reverberation will probably not be a serious handicap to a speech program.

18. The following chart may be helpful in anticipating the approximate effect that RT_{60} will have on system performance.

REVERB TIME (SECONDS)	TYPE OF ROOM	MUSIC	SPEECH
0.5	DEAD	DRY	EXCELLENT
1.0	MODERATELY DEAD	DULL	GOOD
2.0	MODERATE	GOOD	FAIR
4.0	LIVE	FAIR	POOR
8.0	VERY LIVE	POOR	PROBLEMS!

19. A reverberation time of one second or less can be considered a very satisfactory acoustic environment for speech reinforcement, since *direct sound* and *early reflections* make up the greater portion of the sound field. As illustrated in Figure A3.10a, the reverberation in this room disappears fairly quickly, creating little problem with intelligibility. On the other hand, many convention venues, both large and small, are plagued with reverberation problems that are more severe. Reverberation times of several seconds are not uncommon, sometimes in excess of eight seconds. When a large stadium is located partially outdoors, the presence of a

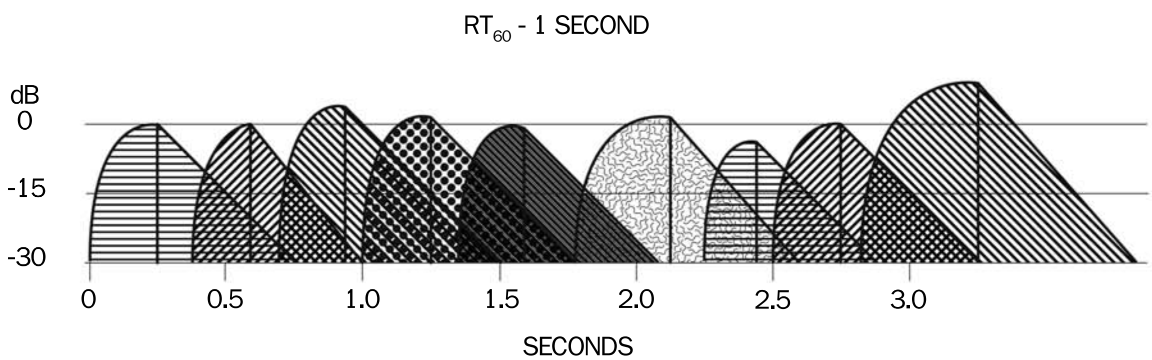


Figure A3.10a Clean, Articulate Speech Is Possible With a Short Reverb Time

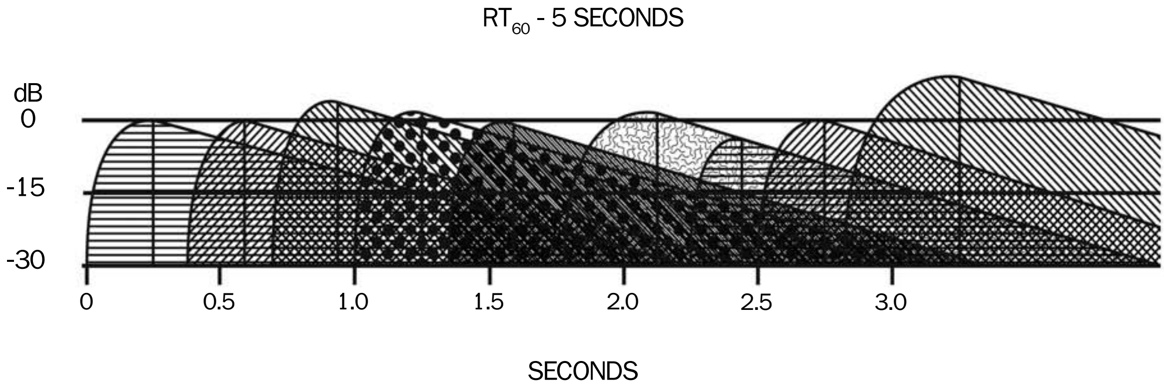


Figure A3.10b A Long Reverb Time Degrades Clarity and Intelligibility

concrete floor, reflective walls, and possibly a steel roof creates an immense room, even though it is not totally enclosed. That "room" will likely possess many of the undesirable characteristics that contribute to severe reflections and reverberation.

20. Why does this occur? Figure A3.10b illustrates the acoustic condition created by a five-second reverb time. The first syllable uttered by the talker begins to "charge" the room with sound, similar to the charging of a capacitor. The acoustic charge then slowly decays, or "discharges," at a rate dictated by the room's RT_{60}. Since the average talker utters about three syllables per second, sound from the second, third, fourth, and succeeding syllables is added to the energy generated by the first syllable still decaying in the room. After a few seconds, the reflected energy generated by many syllables induces a combination of reverberating sounds that persist throughout the room as a continuous diffusion of noise. This diffusion is referred to as the *reverberant field* and has the same masking effect on intelligible speech as other forms of noise, masking the consonantal and sibilant sounds so essential to good articulation. Paradoxically, it is the sound system itself that induces this reverberant field! While the examples diagrammed above may not be technically accurate in all circumstances, they do serve to contrast the effects of short and long reverberation times on typical sound behavior.

The Direct Field Versus the Reverberant Field

21. The mention of the reverberant field introduces another very important concept to our analysis of the behavior of sound indoors. Figure A3.11 depicts the two sound fields generated by the sound system: (1) the *direct field* governed by inverse square law and (2) the *reverberant field* determined by the acoustic character of the room. If we are near a loudspeaker, the existence of any amount of reverberation is of little concern. However, as we move away from the sound source, the level of direct sound decreases according to inverse square law, just as it does outdoors. If the room is sufficiently large and reverberant, there will be a gradual loss of articulation as we move farther into the reverberant field, which remains at a fairly constant level. As direct sound continues to decrease, reverberation may degrade articulation to the point where speech clarity is seriously impaired. Listeners seated in the direct field created by the loudspeaker will experience no problem with articulation loss. However, those seated in the reverberant field will experience reduced clarity.

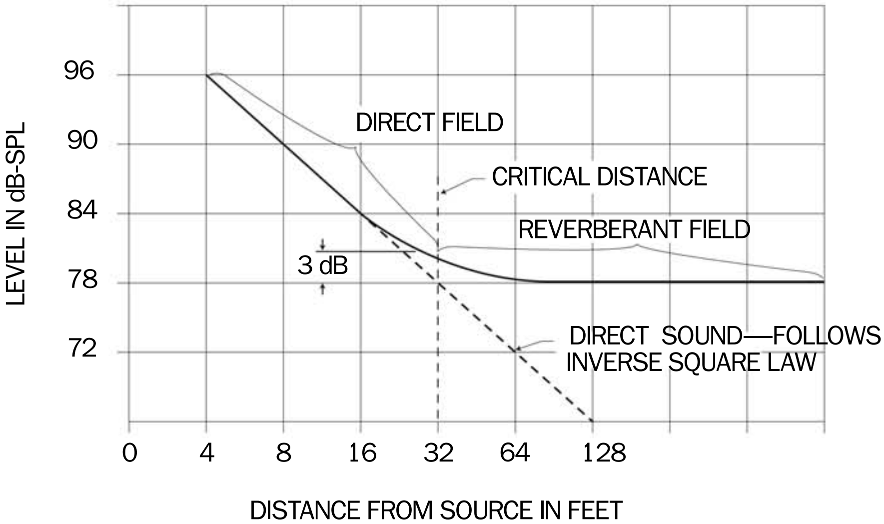


Figure A3.11 Attenuation of Direct Sound and the Reverberant Field

22. This interaction between the direct and reverberant fields is characteristic of all large uncomplicated rooms. It can be easily demonstrated by amplifying some pink noise of one octave bandwidth at a steady loudness. A sound-level meter is used to measure the sound pressure level (SPL) induced in the room. If we move slowly from a distant point in the reverberant field toward the source of sound, we will note that the SPL remains fairly constant. Eventually, though, the SPL will begin to climb until it reaches a level of 3 dB above the reverberant field reading. The direct and reverberant fields are equal at this point, with the summation of the two equal fields resulting in a 3 dB increase in level. As shown in Figure A3.11, this point indicates what is known as the *critical distance* (D_C), a useful concept in indoor sound reinforcement.

23. Critical distance is directly related to the acoustic characteristics of the room. By illustrating two reverberant fields, one for a relatively live reverberant room and the other for a relatively dead room, Figure A3.12 shows how the level of the reverberant field generated by the sound system can affect D_C. As noted above, at D_C the two fields are equal and thus the ratio is unity. At one-half D_C, the direct field would be 6 dB *higher* than the reverberant field. At double the D_C, the ratio is inverted and the direct field is

6 dB *lower* than the reverberant field. At three times the D_C, the direct field would be down about 9.5 dB, and so on. Experiments have shown that three times the D_C is about the maximum that can be tolerated before articulation loss becomes unacceptable.

24. It must be noted that D_C does not represent a fixed position but, rather, varies around an *average* position that is constantly changing according to the loudness of the amplified program. If the talker pauses long enough for the reverberant field to die away, his first few words thereafter will be entering minimal reverberation and so will be heard clearly at a greater distance. As he continues speaking, the reverberant field will again build up to an average level and this level will tend to follow the dynamic excursions of the talker. If he should suddenly lower his voice, the continuing reverberation will obscure his words for a few seconds until it drops to the new level. In practice, excessive reverberation seems to degrade the articulation of a very dynamic talker more than it does that of one who speaks at a more uniform level.

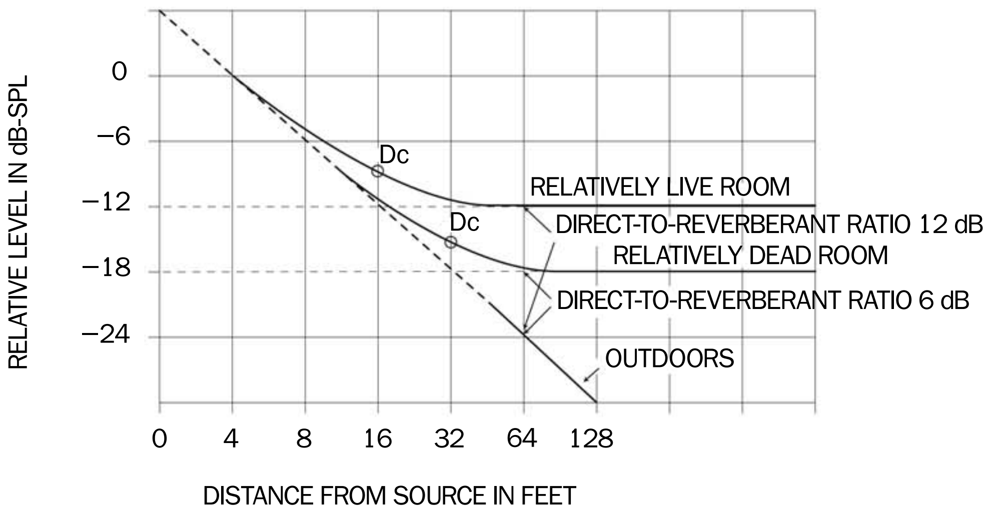


Figure A3.12 Effect of Reverberant Field Level on Critical Distance and Direct-to-Reverberant Ratio

25. How much *articulation loss* is acceptable? Listening tests conducted by researchers have suggested that a 10 percent articulation loss is considered the *maximum* that can be tolerated for speech over extended periods. As the amount of "lost data" increases, the brain must work harder to assemble meaningful thoughts from partial information, resulting in what is sometimes referred to as *listener fatigue*. The harder the brain must work to achieve understanding, the sooner fatigue sets in and attention wanders. When intelligibility is only borderline for many listeners with normal hearing, those with less-than-average hearing ability will have a greater problem hearing the program.

26. Will raising the level of the sound system solve the problem? Very seldom. Raising the overall sound level usually worsens the situation by increasing the level of the reverberant field, thus affecting more listeners. While the actual RT_{60} does not change, reverberation appears to linger in the room longer before it decays below audibility or below the ambient noise level of the room. Over the years, many studies have attempted to define a specific formula for establishing the point where intelligibility becomes a serious problem, but these mathematical approaches have not been totally satisfactory. Indeed, intelligibility, or speech clarity, is determined not only by the room's RT_{60} but also by the level of the reverberant field, the ambient noise level, the enunciation and voice quality of the talker, and the hearing acuity of the listener. All of these factors must be weighed when making a subjective judgment of sound intelligibility.

Some Helpful Suggestions

27. Thus, we have seen that the reverb time of a rented facility is generally governed by a variety of factors, most of which are beyond our control. Normally, we are not going to install large quantities of acoustic treatments in a rented facility in order to reduce its RT_{60}. Therefore, the course of wisdom would suggest that we take every reasonable precaution during our design and installation to reduce sound reflections and keep the reverberant field level as low as possible. This will increase the critical distance and improve articulation for many listeners seated in borderline areas. Often, taking several small steps can lead to a significant improvement in articulation. Here are a few suggestions:

(1) ***Select loudspeakers that are best suited for the application.*** Loudspeakers should have controlled directivity, so that sound reaches the audience with minimal reflections from walls, ceilings, floors, and other surfaces. Highly directional loudspeakers may help to reach this goal by reducing echoes and reverberation. Small sound columns with wide dispersion patterns probably will not project sufficient direct sound to the most distant rows of seating. It is always helpful to be familiar with the characteristics of the loudspeakers used.—See the discussion on loudspeaker specifications in Chapter 4, paragraphs 21-27, and Appendix 5.

(2) ***Aim to minimize the first reflection.*** During a speech program, every effort should be made to minimize the *first* reflection. Since a loudspeaker's strongest signal is generally along its central axis, loudspeakers should be aimed at the area of greatest absorption—the audience—rather than at nearby reflective surfaces, such as concrete walls, floors, or stairs, or into empty seating sections. This basic principle may seem intuitive, but it cannot be overemphasized.

(3) ***Reduce the distance from the loudspeakers to the listeners.*** While moving loudspeakers closer to the audience may require more loudspeakers, it also places more listeners within the direct field of a loudspeaker, providing them with better articulation. It may also permit the overall sound level to be lowered, which will not only reduce the level of the reverberant field but at the same time reduce the potential for acoustic feedback.

(4) ***Position the microphone for the best pickup of direct sound.*** Depending on its placement, the talker's microphone may be picking up considerable reverberation, only to amplify it through the sound system, thus contributing to the level of the reverberant field. Use a cardioid microphone with a pick-up pattern that rejects most of the unwanted off-axis sound. Moving the microphone closer to the talker helps to improve the ratio of direct to reverberant sound being picked up.

(5) ***Zone the seating and loudspeakers logically.*** Seating and loudspeakers should be zoned so that a fairly densely seated audience can be easily and adequately covered. Experienced soundmen know that when high levels of sound are aimed into seating sections that are nearly empty, a multitude of reflections adds significantly to the overall reverberant field, contributing to degraded intelligibility for many listeners. On the other hand, it is not unusual for a full house to reduce RT_{60} by two or more seconds.

(6) ***Attenuate bass frequencies and add a "presence peak" in the articulation range.*** As explained in Appendix 4, bass frequencies contribute little to speech intelligibility but, rather, tend to reinforce the reverberant field and its level of interference. In addition, bass frequencies usually are nondirectional and not well controlled by the loudspeaker system, nor are they readily absorbed by objects in the room. In Chapter 7, proper equalization for speech was discussed. The methods described there will typically be suitable for most convention venues. However, when confronted with a *particularly troublesome* reverberant field, even more aggressive equalization may be necessary. In such cases, the gradual attenuation of low frequencies may need to start as high as 500 to 600 Hz. If necessary and if our headroom before feedback allows for it, intelligibility can be further improved by introducing a presence peak of 3 to 6 dB in the region of the articulation range between 2 kHz and 4 kHz. Be aware that certain cardioid microphones are designed with a built-in presence peak that provides a certain amount of high-frequency boost, which can prove helpful in difficult circumstances. Take care, however, to ensure that sound quality does not become harsh and strident.

(7) ***Lower the overall sound level.*** It should be apparent that if we inject *less* energy into the room, the reverberant field will decay below the threshold of interference sooner. Operating the sound system louder than necessary raises the level of the reverberant field, increasing the interference caused by reverberation. Experienced soundmen know that excessive volume degrades intelligibility in a highly reverberant room. Articulation can be optimized by maintaining the level no higher than is necessary to cover the audience adequately. Less volume leads to less reverb, which leads to improved sound. Who would have thought that intelligibility could be improved by turning down the volume!

(8) ***Consider installing a low-level distributed system.*** If the level of the reverberant field is still considerably higher than is ideal for speech applications, attention must be given to a system design that will position listeners closer to the loudspeakers. In such cases, a low-level distributed system may be recommended. This type of system utilizes a larger number of loudspeakers operating at a lower level, possibly augmented by suitable signal delay.—See the discussion on signal delay in Chapter 4, paragraphs 36 to 39.

28. It is hoped that this discussion has served to clarify the behavior of sound in relation to loudspeaker systems and the acoustic character of the sites where they are used. This knowledge will help conscientious brothers to cope with specific acoustic problems and to design, install, and operate intelligible sound systems in many difficult convention venues.

Appendix 4

The Gifts of Speech and Hearing

1. Among the many gifts that Jehovah has lovingly given to humans is the gift of spoken communication. To utilize this gift effectively, he has graciously endowed us with two amazing mechanisms —speech and hearing—that are highly sophisticated and complex systems that truly reflect our Creator's awesome wisdom. Even after centuries of research, just how the human brain converts intelligent thought into speech is still not fully understood. As lexicographer Ludwig Koehler wrote: "What actually happens in speech, how the spark of perception kindles the spirit . . . to become the spoken word, eludes our grasp. Human speech is a secret; it is a divine gift, a miracle." (*it*-2, p. 201) Yet, we easily comprehend the sounds of intelligible speech, as well as the parameters that speech occupies within the audible spectrum—valuable information that can surely enhance our ability to amplify and distribute the spoken word effectively to others. Equally important is an understanding of how our ears gather acoustic information from the atmosphere around us and convey the sensation of sound and thought to our brains. A brief look at these faculties will certainly assist us in the management and operation of successful sound systems.

The Gift of Speech

2. Just what is speech? Simply expressed, speech is a series of *coded sounds* drawn from the vocabulary of words associated with one of earth's approximately three thousand tongues. Words are the tools of language—simple or complex sound patterns that convey intelligent thought and meaning. From infancy, our minds have been trained to identify the coded sounds or words unique to a specific language. Indeed, we have learned not only to communicate with language but also to think and reason in terms of language. Can you conceive a thought that does not have a word or words to identify or describe it? In most modern languages, written words consist of alphabetical characters arranged to represent the phonetic sound of the words so that when we read, we are literally reading sounds. This should not be surprising, since the spoken word preceded written text.

3. Further testifying to the wisdom and efficiency of our Creator, Figure A4.1 depicts a very efficient system capable of handling multiple functions, including breathing, eating, and speaking. As a vocal mechanism, this amazing system is capable of articulating an unlimited array of speech sounds unique to each spoken language. Our vocal cords, located in the larynx, or voice box, produce the fundamental tones, ranging from around 90 to about 1,400 hertz (Hz), while our mouth, tongue, teeth, and lips modify and shape these tones into discrete syllables or words. Read this complete sentence aloud and note the precise, split-second timing of the gymnastics performed by the various components of the speech mechanism as they accurately produce the sound of each syllable. The cavities of our mouth, throat, chest, and sinuses serve to amplify and reinforce our voice, producing resonances that enrich its quality and help to make it uniquely characteristic of us. —For further information, see *Ministry School,* Study 29, pages 181-185.

Vowels and Consonants

4. Most of the energy in speech is utilized to produce *vowel* sounds. The vowels *a, e, i, o, u,* and *y,* used singly or in combination, provide a broad variety of sounds in themselves. However, intelligible speech also requires *consonants,* which shape and modify the vowel sound patterns.

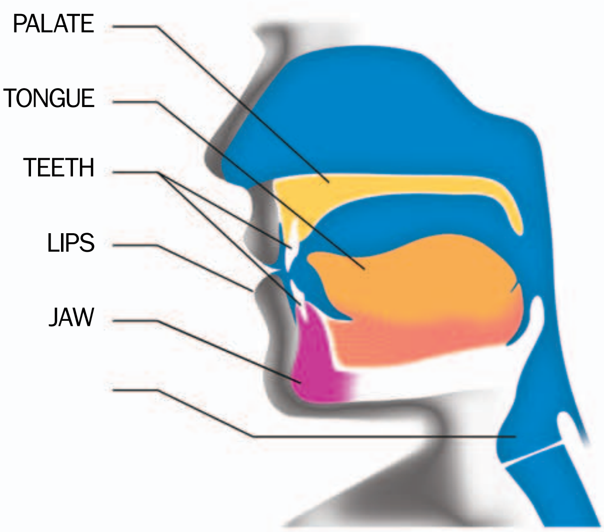


Figure A4.1 Anatomy of the Speech Mechanism

Consonantal sounds are sometimes referred to as *transients.* For example, if we pronounce a series of similar words such as *back, bad, bam, ban, bat, bath,* we observe not only that the initial transient and vowel sound—*ba*—is the same in each but also that it predominates in duration and loudness, whereas the consonantal sound needed to distinguish each word (*ck, d, m, n,* and so on) is of extremely short duration and is spoken at a lower level than the vowel. Words like *buy, die, guy, my, pie,* and *tie* have the distinguishing transients at the beginning of the word. But regardless of where they fall, the identifying consonants not only are shorter but are pronounced at a lower level. These facts emphasize the need for good listening conditions and adequate sound level so as to achieve clarity and precise acoustic articulation.

Sibilants and Plosives

5. Sibilants are consonants that contain a hiss or whistle when pronounced, such as the sounds produced by the letters *s, sh, f, v, z, j,* soft *g* (as in *courage*), or *ch* (as in *church*). By passing the breath through restrictions in the speech mechanism, a certain amount of controlled noise is generated, and thus these sounds are sometimes referred to as *fricatives*. Other consonants, such as *p, b, t, d,* and *k,* are known as *plosives* and are formed by sharp or abrupt bursts of breath passed through the speech mechanism. Plosives often produce distracting breath "pops" when certain microphones are positioned too close to the one speaking.

Articulation Range

6. Articulation and clarity of amplified speech therefore depend primarily on the ability of the sound system to reproduce *consonantal* and *sibilant* sounds faithfully. Take particular note of the fact that nearly all consonantal sounds are concentrated in the upper portion of the speech frequency spectrum, especially the octave from 2 to 4 kilohertz (kHz). This octave can be referred to as the *articulation range* and is essential to the perception of intelligible speech. When a person speaks to us in a whisper without using his vocal cords, we are hearing only these higher frequencies. Yet we are amazed at the high degree of articulation carried by this relatively narrow band of frequencies. While most of the *acoustic power* of speech falls well below 2 kHz, surprisingly little of that energy contributes to articulation. Yet, these lower frequencies are essential, for they not only supply power to the voice but also contribute to its unique character. Without them, speech would sound thin and weak.

Bandwidth Requirements for Speech

7. Thus, we can conclude that the *minimum* band of frequencies required to amplify speech with good *articulation,* adequate *power,* and *naturalness* is not as broad as some might expect. A bandwidth spanning from around 90 Hz to 6 kHz is quite adequate for intelligible speech. While this is not a difficult requirement for modern audio components, our goal is to deliver this frequency range to every seat with minimum interference. When we consider that music is also part of our worship, we recognize that the bandwidth should be increased in order to provide good music fidelity.—See Figure A2.4 in Appendix 2.

The Gift of Hearing

8. To balance our consideration of the characteristics of speech, we must also discuss some vital facts related to the marvelous gift of hearing. As with the power of speech, the faculty of hearing reflects incredible design and ingenuity that magnify the Creator's wisdom. Commenting on the marvelous human hearing system, F. Alton Everest, acoustician and author, stated: "In considering the human hearing system in any depth, it is difficult to escape the conclusion that its intricate functions and structures indicate some beneficent hand in its design. Scientific investigations of how our hearing system really works are continually revealing new marvels, even more awe inspiring than earlier fragmentary knowledge." —*Handbook for Sound Engineers,* 1991 Ed., p. 27.

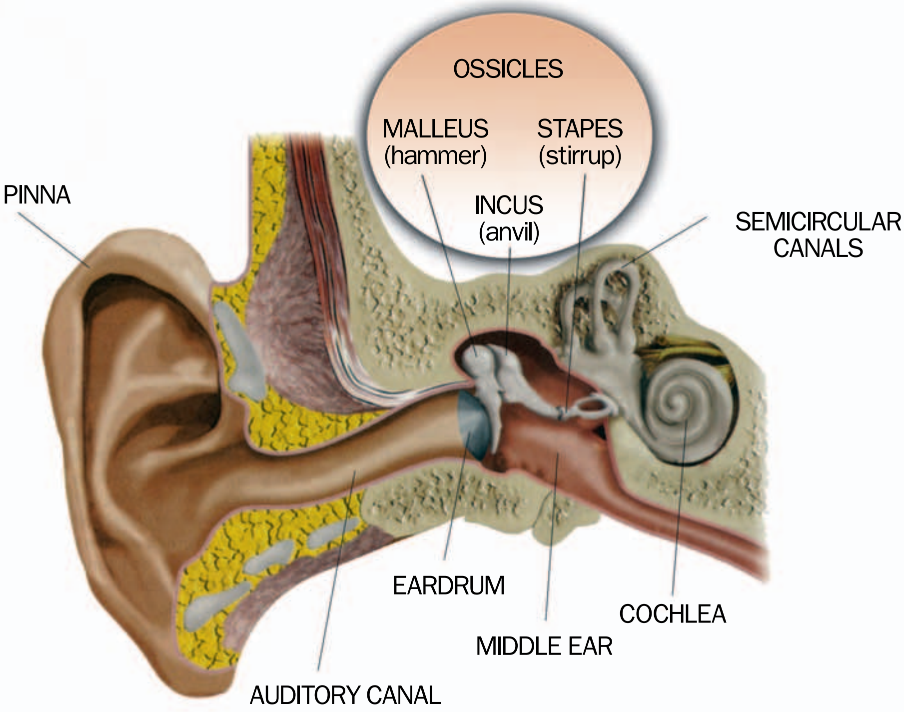


Figure A4.2 The Anatomy of the Ear

9. As illustrated in Figure A4.2, the human ear is composed of four major parts. The outer ear consists of the *pinna,* the convoluted visible part of the ear, and the *auditory canal,* one end of which is open to the outside world while the other end is terminated by the *tympanic membrane,* or *eardrum.* Sound waves travel through the auditory canal to the eardrum, causing it to vibrate in unison with the pressure variations. In the air-filled *middle ear,* the *ossicles,* an assembly of three tiny bones named the *malleus, incus,* and *stapes* (also known as the *hammer, anvil,* and *stirrup*) provides a mechanical linkage to transfer the minute sound vibrations from the eardrum to the *inner ear,* where the *cochlea,* a marvelous sensory transducer embedded in the bone, translates the mechanical vibrations into nerve impulses and sends them to the auditory cortex of the brain, where we perceive them as sounds. This brief description is an obvious oversimplification of an ingenious system of astounding complexity and sophistication. Even after a century of painstaking research, much of the complex psychoacoustic function of our ears in concert with the brain still remains shrouded in mystery.—See the September 22, 1997, *Awake!,* pages 21-23.

10. The pinna, or external part of the ear, with its irregular convolutions was long regarded by many either as a vestigial organ or as a simple sound-gathering device. It is a sound-gathering device, but recent research shows the pinna to be a highly sophisticated acoustic structure that contributes greatly to our ability to localize the sources of sounds around us. These two fleshy acoustic barriers mounted on our heads work in concert to enable us not only to distinguish sounds arriving from the right, left, front, or rear but also to determine from what angle above, below, in front of, or behind us the sound originated. Measurements show that the pinna increases the sound pressure level (SPL) at the entrance of the auditory canal by several decibels (dB) in the all-important articulation range between 2 and 4 kHz, the range we most depend upon for speech communication. This complex design testifies to the love and wisdom of its Designer!

11. As further evidence of intelligent design, the auditory canal has a length of about 1.18 inches (3 cm), which happens to be one fourth of the wavelength of 2,870 Hz. This causes a "pipe effect," which provides pressure amplification, so that in the region of 2-4 kHz, the acoustic pressure at the surface of the eardrum is about 10 dB higher than at the opening of the canal. This contributes to a long-known but little-understood fact, namely, that the sensitivity of the human ear is greatest in this important region of mid-frequencies—hardly an evolutionary accident.—See Figure A4.3.

12. Similarly, studies of middle- and inner-ear construction reveal astounding complexity and engineering excellence, which are beyond the scope of this manual. Modern engineers acknowledge that the ear is designed perfectly for its purpose. It could not have been done better—further evidence of the awesome wisdom and design in creation.

Hearing Sensitivity

13. Consider, too, the incredible sensitivity of our ears, as revealed by acoustic research. In a very quiet environment, such as an anechoic chamber, a person with healthy ears can hear the sound of his blood coursing through his blood vessels. Under laboratory conditions the human ear can detect the softest sound known to humans, the rain of air particles impinging on the eardrums, a motion incredibly small—about one hundredth of a millionth of a centimeter. Thus, human hearing is sensitive enough to detect the very threshold of hearing. Any sound softer than this would be drowned out by the molecular noise of the air itself.

Equal-Loudness Contours

14. Research has established that instead of being uniform, the sensitivity of our ears to audible sound varies according to the frequency and loudness of the sound. This is seen in the set of Robinson-Dadson *equal-loudness* contours illustrated in Figure A4.3. Here, each contour represents the sound pressure level required to keep the *apparent* loudness constant, or equal, over the audible spectrum. The contours are plotted in 10 dB steps around a 1 kHz center frequency and illustrate the varying contours of our hearing sensitivity, beginning at 0 dB-SPL—considered the threshold of hearing—and continuing up to 120 dB-SPL, which is VERY LOUD indeed!

15. Looking at the bottom contour, which illustrates the weakest sound we can detect, we observe that our ears are far more sensitive in the 1 to 6 kHz region than at either extreme. A bass tone, such as 50 Hz, must be nearly 50 dB louder than a 3 kHz tone in order to be barely perceptible. This means that the ear's sensitivity to bass frequencies is much lower than it is to mid-frequencies —certainly a blessing in this age of noise pollution. We note, however, that as sound levels increase above the conversational level of 70 dB, the response of our ears tends to flatten out.

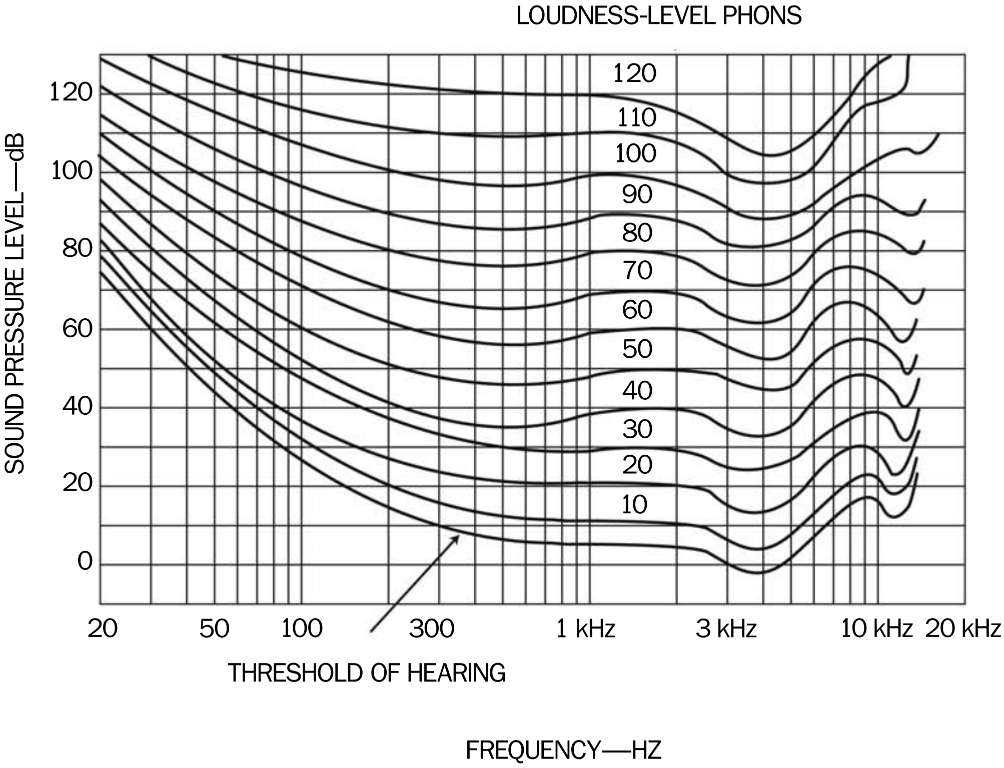


Figure A4.3 Equal-Loudness Contours

16. Again, is it any surprise that the ear's greatest sensitivity at all levels is in that all-important articulation range between 2 and 4 kHz, the band most critical to the perception of intelligible speech? For this we can be thankful!

17. From the set of equal-loudness contours (Figure A4.3), we can visualize the lower and upper boundaries of aural perception, from the threshold of hearing at the lowest levels to the threshold of pain at the highest levels. As shown by the shaded areas in Figure A4.4, music and speech do not utilize the entire auditory area, which helps to illustrate the relatively limited parameters needed for acceptable speech reinforcement.

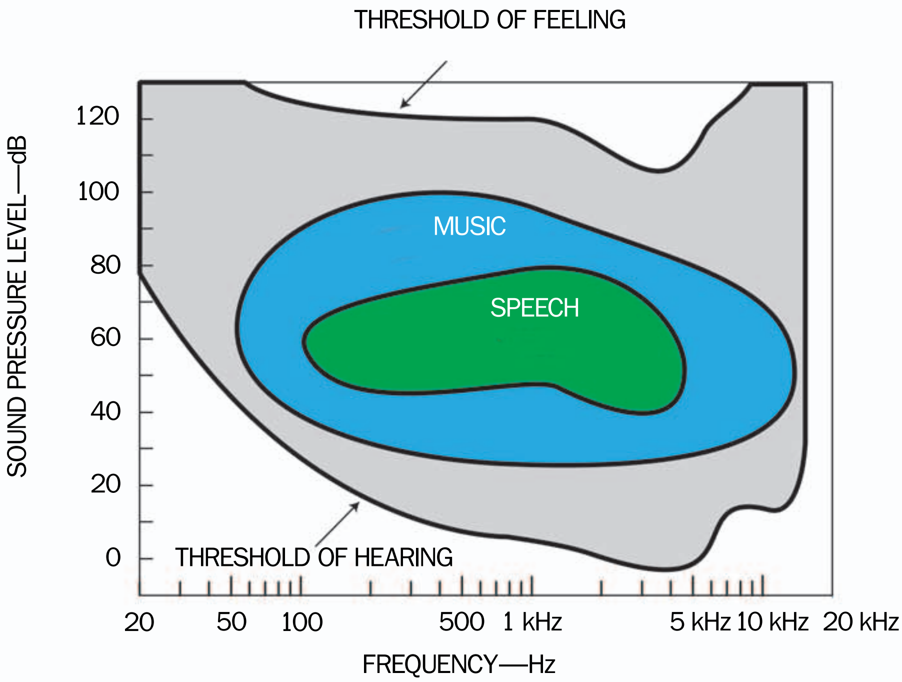


Figure A4.4 The Auditory Area for Music and Speech

Hearing Loss Associated With Age

18. As noted in Appendix 2, paragraph 7, the oft-published figures of 20 Hz to 20 kHz delineating the range of human hearing are generally valid only for young people with undamaged ears. As imperfect humans, however, our hearing acuity, especially at high frequencies, diminishes significantly as we grow older. This loss in hearing sensitivity associated with age—known as *presbycusis*—is illustrated in Figure A4.5. We note that the average loss of hearing sensitivity above 1 kHz is somewhat more acute for men than for women. Thus, statistics confirm that the normal hearing curve for a 65-year-old person will likely be somewhat different from that of a 25-year-old.

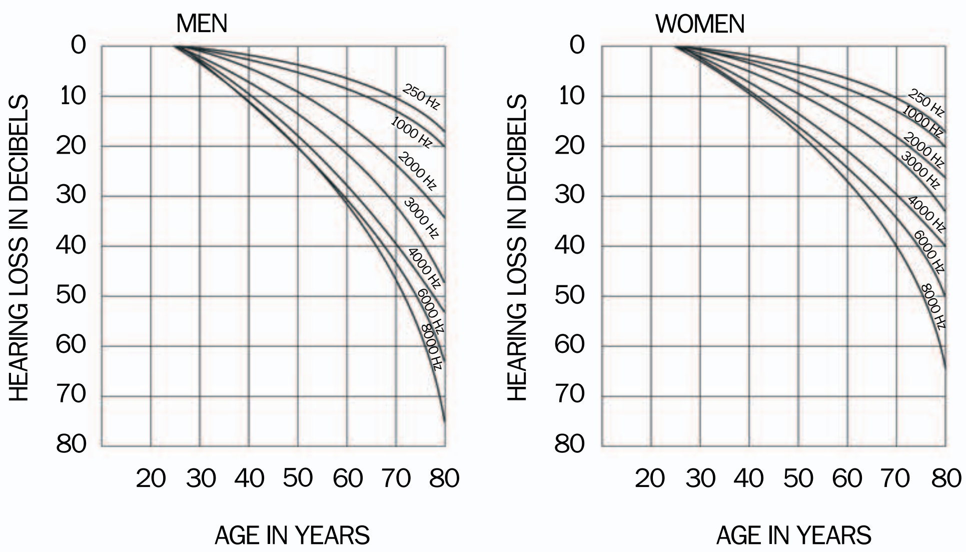


Figure A4.5 Hearing Loss Associated With Age

19. These factors should assist us in considering the acoustic needs of our audience and may also help us to understand certain specific problems that we encounter from time to time. We do well to be aware of the limits imposed by our own human imperfections, as well as those of our brothers, when adjusting the level and tonal character of our sound system. Because of reduced sensitivity, especially at higher frequencies, older persons will likely find hearing more difficult, especially where significant reverberation or noise is present. When older and infirm brothers are located in a special seating area, would it not be considerate to give special attention to the volume and clarity of sound in that area?

The Psychoacoustic Phenomenon

20. With this background, we observe that the auditory process utilizes far more than a pair of sensitive ears. What we have covered thus far does not explain how the *mind* responds to what the ear captures—and how it fills in what the ear, for one reason or another, fails to capture. The study of the ear-brain relationship falls into the science of *psychoacoustics,* and much has yet to be learned about this marvelous phenomenon. We can be thankful that each of us possesses a God-given ability to direct our hearing faculties *selectively* toward sounds we choose to hear while rejecting a bedlam of interfering sounds that may actually be louder. Up to a point, *we hear only what we want to hear*. Thus, we can carry on a comfortable conversation with another person across the room, such as at our Kingdom Hall, while dozens of people are talking around us. Or a mother will hear her infant child crying in an adjoining room while the rest of us may be totally oblivious. No microphone has ever been devised that can even approximate the ear-brain capability. Does this explain why we may thoroughly enjoy hearing a fine talk from our seat in the audience and yet find that the recording we made at the same location is scarcely intelligible? Our microphone simply has no brain! This may also explain why the audience at a convention where the sound is less than optimum can afford to be so forgiving.

Achieving Successful Communication

21. Many factors affect the quality and intelligibility of amplified speech, not the least of which are the articulation and voice quality of the one speaking into the microphone. In a difficult acoustic environment, the person who speaks very rapidly or enunciates poorly may contribute significantly to a loss of clarity. In a highly reverberant area, we can appreciate why rapid-fire speech will be more difficult to understand than speech delivered at a slow, deliberate pace. It is only fair to the soundman to acknowledge that even the finest sound system can do little to compensate for poor articulation or blurred enunciation on the part of the talker. On the other hand, a slower pace accompanied by precise articulation may help even a borderline sound system to perform acceptably well.

22. In summary, the overall success of our convention hinges largely on the effectiveness of the sound system in communicating adequate, articulate, and comfortable sound to everyone in the audience. Thus we are enabled to make the best use of the gifts our loving Creator has given us to strengthen our faith in his promises and to share the good news of his Kingdom with others.

Appendix 5

Specifying Audio Equipment for Conventions

1. This section provides helpful guidelines for choosing proper equipment for convention sound systems. Interpreting manufacturers' specifications can be difficult, sometimes resulting in our obtaining a piece of equipment that does not meet our expectations or that has more features than we need. The following simple, brief explanations of key specifications should do much to assist in the decision-making process. Typically, consumer equipment is not recommended for convention use due to its construction, connections, and output level. To maximize reliability, professional sound equipment with balanced inputs and outputs is preferred. MP3 players would be an exception to this, however, since professional models are often available only at an exorbitant cost while offering relatively few benefits. Sound equipment should be obtained from well-known manufacturers who guarantee their equipment. It is wise to give careful thought to specifying equipment so that dedicated funds are not wasted.

2. In contrast to the simple descriptions we will give for most audio components, a more extended discussion of transformers has been included in this section, as they are somewhat of a mystery to many and were not thoroughly covered in other parts of this handbook. In most cases, transformers are already part of another piece of equipment, while in other instances a transformer may need to be obtained for a specific purpose, such as isolation. Most modern transformers are packaged in such a way as to make installation extremely simple, and little thought need be given as to how they are actually wired. Nonetheless, this information has been included for those desiring a deeper understanding or for circumstances that may require it.

Microphones

3. Microphones play a vital role in sound reinforcement, and selecting the correct one is necessary for natural, intelligible sound. The two most common types of microphone are dynamic and condenser.

4. A dynamic microphone transducer is simply constructed and robust and is capable of handling extremely high sound levels while being relatively unaffected by extremes of temperature or humidity. Dynamic microphones of high quality have good frequency response and are excellent for reproducing speech, making them the preferred choice for convention use.

5. Condenser microphones are highly sensitive, have a flat frequency response, and usually sound more natural at higher frequencies. Because of their design, they can be manufactured smaller than most dynamic microphones. Condenser microphones are not generally used for convention sound, since they are not as rugged as dynamic microphones, require phantom power, are prone to radio frequency interference (RFI), and are typically more expensive.

6. When selecting a microphone, look for a low-impedance dynamic microphone with a cardioid pickup pattern. The frequency response will not be perfectly flat; many vocal microphones have an intentional peak in the upper frequencies. This is sometimes called a presence peak and is designed to increase intelligibility and diminish the proximity effect (bass boost) that occurs when cardioid microphones are positioned close to the talker. The cardioid (unidirectional) pickup pattern is preferred, since it helps reduce feedback by rejecting unwanted sounds coming from directly behind it.

Microphone Cables and Cable Snakes

7. Even with good design, proper installation, and high-quality audio equipment, an entire sound system could be degraded or completely silenced by bad cables or connectors. Since cables play such a vital role in signal transmission, it is not prudent to try to save money by purchasing cheap cables. This is especially true when we consider how inexpensive cable is compared with the overall cost of the electronic components used in a sound system.

8. As a minimum requirement, microphone cables should be constructed of shielded two-conductor 22-gauge stranded wire. The outer jacket should be constructed of durable, pliable rubber for greater flexibility and protection. The connectors terminating the cable ends should be well made.

9. A cable snake has multiple microphone cables within a rubber or plastic jacket, with one end terminating in a stage box with chassis-mounted XLR connectors and the other end providing individual XLR connectors for connection to the audio mixer. This allows for quick, clean installation, and the outer jacket provides protection for multiple cables. The cable snake should be constructed with high-quality materials similar to those described above. A 12-input cable snake works well for our use, as this will give a couple of spares in case a cable or connector breaks.

MP3 Players

10. A simple MP3 player works well for convention sound. The interface should allow the operator to use the buttons on the faceplate to queue, play, and stop songs manually. A digital display that shows the track number and elapsed time of the song being played is also useful.

Mixers

11. Audio mixers come in a wide range of prices and complexity. For convention use, a large console mixer would be excessive, since such mixers are designed to provide many extended functions that are unnecessary for our programs. A simple mixer of high quality is all that is needed.

12. At some conventions, it may be possible to use a gated (sometimes called automatic) mixer. Such mixers have sophisticated electronics that determine which microphones are being used and switch automatically between them. In no way do these mixers replace human operators, but they can offer a measure of assistance in situations involving large numbers of open microphones. This feature is therefore especially useful for demonstrations and interviews that have multiple participants. It should be noted, however, that some budget gated mixers can be more problematic than helpful. Older and budget models are sometimes slow to react, often gating late and missing the first syllable spoken.

Digital Signal Processors (DSPs)

13. Since all signal processing takes place within the digital signal processor unit, any DSP should be thoroughly investigated before being selected. You will want to make sure that it provides for all your processing needs, such as gain control, polarity switching, limiters, compressors, parametric equalizers, cut/shelf filters, feedback reducers, delays, crossovers, and splitters. Since some DSPs are designed for stereo or left/center/right systems and our sound design is for a mono system, it is very important to verify that any input can be sent to any output. Verify that the DSP can provide the maximum delay you will require at the convention venue on each individual output channel.

Loudspeakers

14. Loudspeaker selection is dependent on the overall system design. Regardless of the loudspeaker used, it should be highly sensitive, allowing for the greatest sound pressure level while drawing minimal power. For two-way speakers, the crossover frequency should ideally be below the articulation range. The speakers should be of rugged construction. The specific application determines the type of loudspeaker that should be chosen.—For more information, please see Chapter 4, paragraphs 21 to 27.

Audio Transformers

15. For the components of a sound system to interface safely and reliably, each device must be operated within proper electrical parameters. Modern professional audio components are generally designed to interface correctly with other devices in the system, often by means of quality audio transformers. Transformers often serve to interface microphones with their preamplifier (mixer), interface mixers with other low-level devices, and transfer audio power from power amplifiers to the loudspeakers they drive. Incorrect loading and/or incorrect signal levels can introduce a variety of audio problems, including distortion, noise, hum, oscillation, and excessive signal loss, and may even damage certain types of equipment. Transformers help to match the components of a sound system properly. Since we will likely encounter the need for audio transformation somewhere in a large sound system, a brief discussion of audio transformers and their unique characteristics will be helpful.

16. An audio transformer consists of two or more windings of wire on a core of laminated steel or powdered iron. These windings are known as the *primary,* to which an audio signal is fed, and the *secondary,* from which the signal is taken—either of which may consist of one or more windings on the same core. Taps are often provided on each winding to allow the transformer to match a greater variety of parameters in an audio circuit.

Low-Level Transformers

17. Low-level transformers are frequently used in microphone circuits and other low-level applications. Not only do they ensure the stability of balanced circuits but they also provide DC and ground isolation when needed. Looking at the three diagrams in Figures A5.1, A5.2, and A5.3, we note that a different impedance is shown for each primary and secondary winding. Transformers are also used to change voltages. These changes are referred to as *ratios,* and it is important to understand the relationship between *voltage ratios* and *impedance ratios*. This relationship may be simply stated as follows: *The impedance ratio of a transformer equals the voltage ratio squared.*

Stated algebraically: $Z_1/Z_2 = (E_1/E_2)^2$ **or** $E_1/E_2 = \sqrt{Z_1/Z_2}$

18. In Figure A5.1, we observe that the impedance ratio is 15,000:600, or 25:1 (15,000/600 = 25). What is the voltage ratio? Using the second equation above, we determine that the voltage ratio would be the square root of 25:1, the impedance ratio. The re-

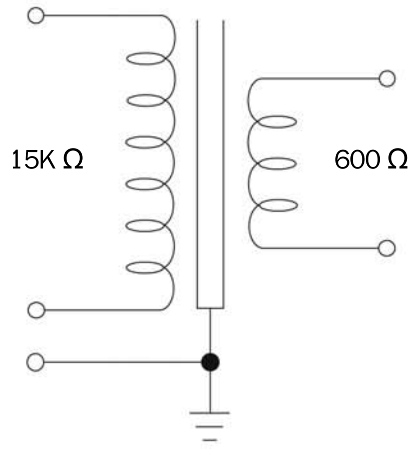


Figure A5.1 Typical Low-Level Matching Transformer sult is 5:1. Thus, with 5 root-mean-square (rms) volts across the 15,000-ohm winding, we should read 1 volt across the 600-ohm winding. Conversely, if the transformer is reversed (as in Figure A5.2) and 1 rms volt is applied to the 600-ohm winding, we would expect to read 5 volts across the 15,000-ohm winding. This voltage ratio of 5:1 is also equal to the ratio of turns of wire in the two respective windings, meaning that the 15,000-ohm winding has five times as many turns of wire as the 600-ohm winding. Thus, the *voltage ratio* and *turns ratio* in transformers are the same, and either can be used in the original rule.

19. Note that the two windings of the transformer in the illustration are not directly connected together. This illustrates another distinct advantage provided by the use of a transformer, namely, an effective DC *isolation* between the two circuits that it joins. In audio we are often obliged to use both balanced and unbalanced circuits in transmitting an electrical signal from one component to another or from one system to another. Balanced circuits cannot be changed to unbalanced circuits, or vice versa, according to whim. Rather, these electrical relationships must generally be maintained to obtain satisfactory results. As we note in Figure A5.2, one leg of the unbalanced circuit is connected to the system ground. If a balanced line becomes connected to an unbalanced circuit at some point, both circuits will become unbalanced, defeating the advantages provided by a balanced circuit and probably introducing serious problems with hum and/or noise. As a rule, to minimize the pickup of hum, RFI, and noise along the way, an audio program being sent to a distant point is always sent via a *balanced* line. It can also be assumed that telco (telecommunications company) lines are always *balanced* lines. Providing the isolation and balanced status needed usually requires the use of an audio transformer.

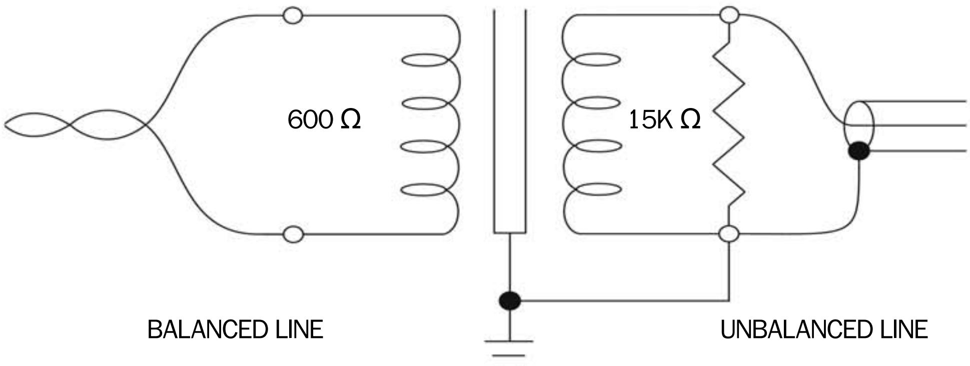


Figure A5.2 Typical Matching Transformer Used to Isolate Circuit

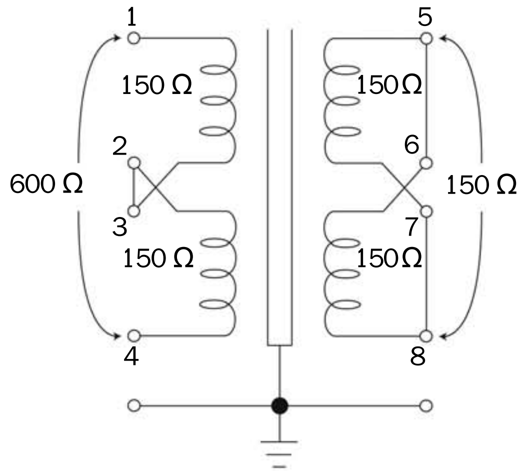


Figure A5.3 Microphone- or Line-Level Matching Transformer

20. The transformer shown in Figure A5.3 is typical of a quality line-to-line or mic-to-line transformer that can be used in a variety of ways. As shown in the figure, it can match 600 ohms to 150 ohms. However, we note that both the primary and secondary each have two windings. On the left side, the two 150-ohm windings are connected in series, thus giving the series winding twice the number of turns as the two windings on the right side, which are connected in parallel and act as one winding. Thus, the turns ratio (and voltage ratio) of this transformer as shown is 2:1, while its impedance ratio is $2^2:1^2$, or 4:1.

21. The transformer in Figure A5.3 could easily be adjusted to match 600 ohms to 600 ohms (1:1) simply by connecting the windings on the right in series, just as was done with those on the left. In a 1:1 (line-to-line) configuration, it would provide the advantage of DC or ground isolation between two audio systems without any change in voltage. The ability to balance or unbalance audio circuits or to break potential ground loops by the use of appropriate transformers is a very important engineering concept. In many circumstances, a transformer may be the only practical way to connect a balanced line to equipment with unbalanced inputs or outputs.

Transformers for Constant-Voltage Systems

22. Since the purpose of any sound system is to increase, or amplify, the power of an audio signal, the efficient transfer of power from the amplifier to the loudspeakers is of primary concern. A basic electrical law states that *maximum power is transferred when the impedance of the load*—the loudspeaker system—*equals the impedance of the power source*—the power amplifier. Achieving this is accomplished by (1) selecting series-parallel connections or (2) utilizing a constant-voltage system using transformers. The advantages and disadvantages of each system have been discussed in detail in Chapters 4 and 6.

23. The constant-voltage concept is simply another means of *impedance matching, but in this case* it is designed around power increments in watts, rather than ohms. For the layman, this method is much simpler than the complex calculations required to figure impedances in ohms. Since all power amplifiers give their power output rating in watts, all that is required to achieve a proper impedance match in a constant-voltage system is to make certain that the sum of all wattage taps never exceeds the power rating of the amplifier. If the total load in watts becomes greater than the amplifier's power rating, the amplifier will become overloaded and some adjustment must be made. This may be accomplished by reducing the number of loudspeakers in the circuit or by connecting each loudspeaker to a lower-wattage tap on its transformer. What if the total load on the amplifier is *less* than its rated power? It simply means that the load will not be able to draw the full power available. In reality, this is rarely a problem, since we seldom operate an amplifier at full power. If that were done, clipping and distortion would likely occur.

24. The following are a few caveats to consider when specifying or applying audio transformers:

- Transformers do not give us "something for nothing." They are passive devices that cannot increase available power. Since no transformer is 100 percent efficient, there will always be a slight power loss, known as *insertion loss,* when a transformer is inserted into any circuit. Many manufacturers specify the insertion loss for their transformers in decibels (dB) or as a percentage. A loss of 0.5 dB represents about a 10 percent loss in power, or an efficiency of 90 percent. Thus, if an amplifier is delivering 100 watts into a line using transformers with 90 percent efficiency, no more than 90 watts can reach the loudspeakers. This inefficiency is trifling when compared to the very low efficiency of loudspeakers, but it may affect the number of loudspeakers that we can connect to a line.
- The efficiency, frequency response, and cost of a transformer increase with core size and weight. Frequency response and power-handling capacity are functions of the size and quality of the laminated core as much as of the windings themselves. A quality 70-volt transformer capable of handling substantial power and having a broad frequency response must necessarily have a hefty core. It will not have a bargain price tag. On the other hand, economy transformers with small cores may perform well in limited-range paging systems, but using them on broadband program material at high power may severely overload the system, resulting in distortion and possible damage to the amplifier. This can occur despite the soundman having made certain that the total wattage taps do not exceed the amplifier's rated power.

- Where substantial power is to be transformed and line isolation is not a requirement, improved efficiency can be obtained by utilizing autotransformers (with a single winding) rather than standard dual-winding transformers.
- You may have noted that transformer impedances do not *sum* and *divide* like resistors. For example, series windings do not "add" like resistors in series; instead, they are calculated according to the square of the number of turns. Hence, as noted in Figure A5.3, the two 150-ohm windings in series form an impedance, not of 300 ohms, but of 600 ohms. Similarly, windings in parallel do not divide as parallel resistors would. Parallel *identical* windings on the same core function as one, without altering impedance, but can handle twice the current that a single winding can. Dissimilar windings on the same core must NEVER be paralleled, as this would seriously alter the characteristics of the transformer.
- Care must be exercised to make certain that windings on the same core are connected in phase. Out-of-phase windings will result in faulty performance by the transformer. Be sure to follow the manufacturer's specifications.

Power Amplifiers

25. The purpose of a power amplifier is to raise the power level of the audio signal to a level that is sufficient to drive the loudspeaker system. There are many different types of amplifiers, but those that are most suitable for large conventions are generally dual-channel high-power audio amplifiers.

26. The following paragraphs will explain some of the critical specifications associated with amplifiers so that you can interpret what the various measurements mean. This will help you to make sure that you do not place a greater load on the amplifier than it is capable of handling and also to make fair comparisons between models when you are selecting an amplifier.

27. *Power Output:* Does the amplifier have enough power? Amplifiers that provide clean high-output power are readily available and affordable. How much power the amplifier is capable of producing is defined by its power rating. There are two ways that manufacturers list power ratings: peak and rms. It is best to look for the rms power rating, since it represents the *continuous* power output. On the other hand, the peak power rating is based on a short *burst of power* that can make the amplifier rating look deceptively high. Also, it is important to verify that the power rating is given per output channel.

28. *Bandwidth:* This specification lets us know the range of frequencies over which the amplifier is able to maintain its rated power. Amplifiers used for conventions should be able to reproduce frequencies between 20 hertz (Hz) and 20 kilohertz (kHz) at their rated power. Some manufacturers will list their power rating as "x watts @ 1 kHz," which means that they only tested one specific frequency to get the power rating. This makes the power rating appear higher than it would have if it had been tested over the full spectrum of sound.

29. *Total Harmonic Distortion* (*THD*): At some point, all amplifiers introduce distortion. When considering the amplifier's THD, you are looking for a low number. A distortion figure of 1 percent or lower is acceptable for convention use.

30. *Amplifier Outputs:* Many amplifiers are designed with output impedances of 4, 8, or 16 ohms, and if properly matched, allow their rated power to be fully transferred to the load. Typically, these would only be used with a *series-parallel* system. For conventions, high-power amplifiers that have constant-voltage outputs are preferred. Such high-quality amplifiers simplify our design and installation and ensure that we will have the power we need.

31. Reliability is a very important factor in the selection of power amplifiers. A professional or commercial design should be rugged and reliable, both mechanically and electrically. Usually, these qualities go together. In addition, it should have a good protective circuit on its output. This circuit should protect the amplifier against a full direct short, open circuit conditions, thermal overheating, ultrasonic feedback, and RFI.

APPENDIX 6

Audio Equipment Testing and Maintenance

1. Depending on how the sound equipment has been obtained, it *may* need to be tested. If the sound equipment is on loan from the branch, it will have been tested before being sent out for convention use. Nonetheless, it is a good practice to check each piece of equipment before each convention to verify that nothing was damaged during previous use, shipping, or handling. It is important to have reliable test equipment. This equipment will be very helpful not only for testing but also for troubleshooting if problems arise.

2. The following methods of testing are not always possible and may not be necessary in every case; nevertheless, it is always beneficial to know and understand basic testing procedures. Five categories of equipment will be discussed: (1) input devices, (2) front-end equipment, (3) amplifiers, (4) loudspeakers, and (5) mounting equipment.

3. First, we will discuss the tools and meters needed to test sound equipment. Note that some of the meters discussed will also prove helpful when setting up and calibrating a convention sound system.

Common Test Equipment

4. *Multimeter, or Volt-Ohm Meter* (*VOM*): The multimeter is a versatile test instrument designed to measure AC and DC voltage as well as resistance. The ability to check continuity and measure audio voltages accurately throughout the system is critical for setup and troubleshooting. A good multimeter should therefore be considered essential for proper installation of a sound system.

Figure A6.1 Fluke 87 Multimeter
(Reproduced with permission by Fluke Corporation)

5. *Impedance Meter* (*Z Meter*): To verify wattage accurately on installed loudspeaker systems, an impedance meter can be used. Why is this used instead of an ohmmeter? An ohmmeter measures the opposition to DC voltage; however, a speaker line carries AC voltage. AC circuits have an additional form of resistance called reactance. The amount of reactance depends on the frequency being injected into the circuit. An impedance meter generates an AC signal that is fed through the circuit in order to measure it correctly.

Figure A6.2a Gold Line ZM1 Impedance Meter
(Reproduced with permission by Gold Line)

Figure A6.2b TOA ZM-104A Impedance Meter
(Reproduced with permission by TOA Electronics, Inc.)

6. *Sound Pressure Level (SPL) Meter:* An SPL meter is designed for the sole purpose of measuring sound pressure level accurately. Most SPL meters can be set to react either slowly or quickly to changes in level and can also be switched between weighted and unweighted scales. Weighted scales are necessary, since they approximate actual human hearing more closely. The *A-frequency-weighting* scale is best used only when measuring low-level signals between 40 and 55 decibels (dB). For general SPL measurements in seating sections at a convention, the *C-frequency-weighting* scale should be used. In taking SPL measurements, keep in mind that the microphone should be held approximately at the ear level of the audience and pointed toward the sound source.

Figure A6.3 Galaxy Audio CM-160 SPL Meter
(Reproduced with permission by Galaxy Audio Inc.)

7. *Real-Time Audio Analyzer:* A real-time audio analyzer can be used to visualize frequency response. This device would primarily be used during the initial optimization of the system, allowing troublesome frequency peaks to be identified quickly. Since highly portable real-time analyzers have become available at a reasonable cost, these are now more readily available for our use at conventions. Most of these devices also function as SPL meters and sometimes even as signal generators.

Figure A6.4 Phonic PAA3 Real-Time Audio Analyzer
(Reproduced with permission by Phonic Corporation)

8. *Audio Signal Generator:* Audio test signals are a necessity for optimizing a system and testing individual pieces of equipment. The signal generator must have an adjustable output level and should be able to provide at least pink noise and sine waves at specific frequencies. When a signal generator is not available, an audio test-signal CD or MP3 can be used.

Figure A6.5 NTi Audio Minirator MR2 Audio Signal Generator
(Reproduced with permission by NTi Audio)

9. *Oscilloscope:* An oscilloscope allows us to observe the waveform of an audio signal visually. This can help to determine the point at which a signal begins to clip, or it can help to isolate a source of hum or other noise. Oscilloscopes are therefore quite useful in the testing and repair of equipment. However, while useful, they are not essential to the installation of a sound system.

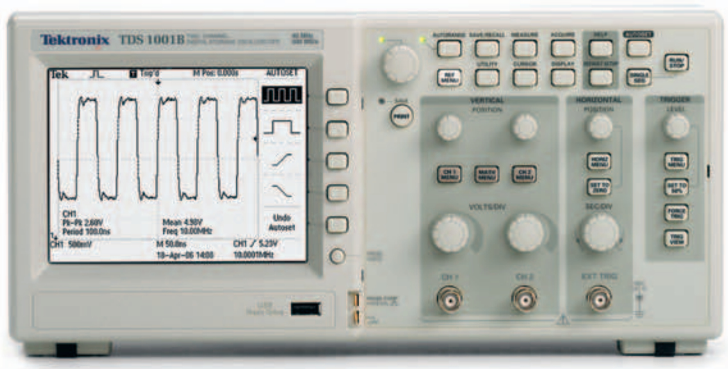


Figure A6.6 Tektronix TDS 1001B Oscilloscope

Testing Cables

10. Verifying that all cables and connectors are in good condition is critical when testing and setting up a sound system. Often, what appears to be a fault in the equipment is actually nothing more than a bad cable or connector. In view of this, the cables and connectors should be tested separately from the equipment, using either a cable-testing device or a multimeter.

Input Devices

11. The two most common input devices used for conventions are MP3 players and microphones.

12. When testing an MP3 player, each file should be played in its entirety to ensure that every track will play without error.

13. Microphones can be tested in a variety of ways. The simplest test is a comparison test. To perform this test, a microphone is plugged into the sound system. As someone speaks into the microphone, listen to the sound produced. Repeat this process with several microphones or with one microphone that is known to be good. For example, if a convention will use six to eight microphones, these can be compared with one another. The microphones should have similar, consistent sound quality. If one microphone sounds different, it likely has a problem. Special notice should be taken of the lower, or bass, frequencies. A defective microphone will often still work but will have a high-pitched frequency response, capturing little or no bass sound. If all the microphones sound similar to each other, this is usually an indication that all of them are good.

14. Other things to check for when testing a microphone are loose connectors or a loose cartridge. Shake the microphone gently and listen. Excessive knocking may indicate a loose cartridge inside. Next, gently try to wiggle the connector at the base of the microphone. If it is loose, it likely needs repair.

Front-End Devices

15. The most common types of front-end equipment include mixers, equalizers, digital delays, feedback reducers, distribution amplifiers, digital signal processors (DSPs), and FM transmitters. We will discuss common testing methods for each of these processing devices. Not all devices can be tested with these exact methods, but the following overview will provide principles that can be applied to a variety of situations and setups.

16. *Mixers:* Testing a mixer involves checking all channel inputs and their potentiometers. A line- or mic-level signal can be used, depending on the manufacturer's specifications. To begin, progressively input a one kilohertz (kHz) sine wave into each channel of the mixer, using an audio signal generator. Connect a multimeter and an oscilloscope to the output. As the signal is connected to each channel, turn up the potentiometer for the channel. Monitor the output on the oscilloscope. The sine wave should increase smoothly in amplitude. If the signal increases in a jerky pattern on the oscilloscope, this means that the potentiometer is noisy and needs to be cleaned with contact cleaner. Also, the maximum output voltage before clipping should be noted on the multimeter. This voltage may need to be converted to dB in order to compare it to the manufacturer's specifications. (Refer to Appendix 7, Figure A7.8.) If the mixer has a VU meter, the multimeter and oscilloscope can be used to calibrate the output to the correct voltage or decibel level.

17. *Equalizers:* The procedure for testing an equalizer is similar to that for testing a mixer. Input a signal of known frequency into the equalizer. Check the equalizer's output with a multimeter and an oscilloscope to make sure it falls within the manufacturer's specifications. All potentiometers should be tested for noise. The potentiometer for each frequency band should be adjusted to its maximum and minimum levels as a square wave is fed into the equalizer. Monitor the output with the oscilloscope. As each potentiometer is adjusted, the square wave will change shape. The changes should be smooth and consistent.

18. *Digital Delays:* Digital delays can be tested by comparing the timing of the input signal to that of the output. A sine wave can be used as an input signal. Both the input and output signals should be monitored on an oscilloscope as the delay is adjusted. While the delay is being adjusted, the sine wave associated with the input should remain steady. However, the sine wave associated with the output should move left or right on the oscilloscope. The direction of movement will depend on whether the delay timing is being increased or decreased. Lastly, the maximum output voltage before clipping should be noted on the multimeter. Compare this with the manufacturer's specifications.

19. *Feedback Reducers:* Feedback reducers are tested to make sure that any repetitive frequencies found in the input signal are being effectively eliminated from the output signal. An audio signal generator is a good input source for this test. Use an oscilloscope to monitor the output of the feedback reducer. When the oscilloscope is connected to the output, reset all filters. Using the signal generator, input a sine wave into the feedback reducer. The sine wave should be seen momentarily on the oscilloscope and should quickly minimize to a negligible voltage. Adjust the frequency on the signal generator to a variety of frequencies in the audio range. The feedback reducer should quickly adjust its filters so that the output voltage is nearly zero. All filters should be tested to make sure the unit locks onto and reduces successive feedback frequencies. Once the test is complete, all filters should be reset.

20. *Distribution Amplifiers:* Distribution amplifiers have line-level and/or mic-level inputs and outputs that can be adjusted per channel. As with equalizers, testing these units is similar to testing a mixer. Some distribution amplifiers may have one or two inputs that can be connected to any number of outputs. A sine wave signal should be fed into each input and verified at each output selected. Using an oscilloscope, monitor the output waveform for clean, smooth transitions as the potentiometer for each output is checked.

21. *Digital Signal Processors:* The digital signal processor is a piece of hardware that can simulate all the processing devices discussed so far and many more. A DSP can electronically alter an audio signal to appear as if it had been processed by one or more of these devices. Depending on the brand and specifications of the DSP, it can have multiple inputs and outputs. Normally, a computer connected to the DSP uses software from the manufacturer to load various processing effects. The computer program can also be used to route any of the DSP's inputs to any or all of its outputs. The routing is the first thing that should be checked. Connect the computer to the DSP. Use the computer program to route each input to each output separately. Try a number of input and output routing combinations. Again using the program, simulate several processing devices on each input and output. Usually, the DSP will not have real potentiometers but virtual ones that are adjustable on the computer by means of software. Still, these should be checked to make sure they respond smoothly to level adjustments. Also, the maximum output voltage should be checked and all the outputs should be consistent when set to the same levels.

An oscilloscope and a multimeter should be used to check all outputs. If the DSP allows settings to be saved and/or loaded via the computer, this should also be done. A DSP, like a computer, is a complex piece of hardware. Therefore, it should be allowed to run for a while during the testing process to ensure that no aspect of the internal hardware and software is corrupted.

22. *FM Transmitters:* FM transmitters are often used during a convention to make the program more accessible to those who are hard of hearing and those who are working in departments away from the main listening area. To test an FM transmitter, start by feeding a known good audio signal into the unit. It is best to use some form of prerecorded music or speech for this purpose. Next, set the unit to a frequency that has no interference in the area being tested. Then set a known good FM receiver to the same frequency and listen for the audio signal that was input into the transmitter. The sound should be clear and intelligible. The manufacturer's specifications should be checked to see how far the unit will transmit the signal. It is good to keep in mind that concrete walls and steel can reduce the transmission distance, so the unit should be tested in a relatively open area.

Power Amplifiers

23. The next pieces of equipment to check are the amplifiers. First, the amplifier should be checked for any physical damage or loose hardware. Next, check the amplifier for functionality. The following pieces of test equipment are recommended: a signal generator, a multimeter, an oscilloscope, and a dummy load (a power resistor, such as an eight-ohm resistor rated for 250 watts).

24. To test an amplifier, the maximum voltage needs to be measured across a known load. The total power output in watts can then be calculated and compared to the manufacturer's specifications. To measure the maximum undistorted output of a power amplifier accurately, set any tone controls to "flat." Use a one kHz sine wave at line level as the test signal and raise the level on the amp until the amplifier begins to clip, or distort, as illustrated in Figure A6.7. Special probes may be needed to ensure that the maximum output voltage can be seen on an oscilloscope. Then reduce the level to a point just below the clipping threshold, measure the AC voltage across the load resistor, and calculate the power output using Ohm's law.—Please see Appendix 7, Figure A7.1.

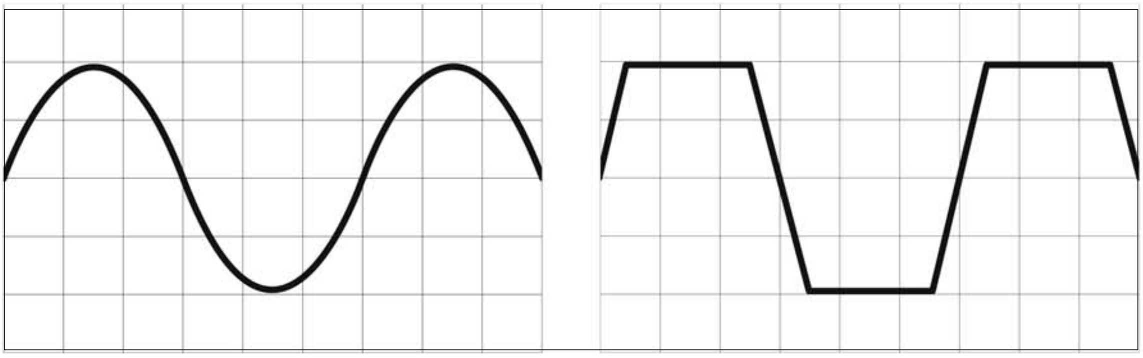


Figure A6.7 Sine Wave Examples

25. If the load resistor can handle the wattage, it may be beneficial to let the amplifier run at the maximum voltage for 30 to 60 seconds to make sure the amp runs normally even in warmer conditions. Please note that if the amplifier has a constant voltage output (as in the case of a 70-volt [70V] amplifier), it may not be feasible to test the amplifier for maximum output, since a line-matching transformer would be needed for the test. In this case, it may be sufficient merely to test the amplifier with a high-wattage resistor to make sure the output works under load.

Loudspeakers

26. There are several different types of loudspeakers that could be discussed. We will first focus on testing full-range loudspeakers. A speaker cabinet may contain one or more full-range loudspeakers. Others may be a two-way system consisting of high- and low-frequency transducers. These types of loudspeakers can be tested the same way, since they both produce full-range audio.

27. Most manufacturers will provide a sensitivity specification for the loudspeaker. Sometimes it is possible to test a loudspeaker in such a way that its actual SPL response can be compared to the manufacturer's specifications. For this sort of test, a known good audio amplifier along with an SPL meter should be used. If possible, test the loudspeaker in an open, nonreflective environment. This will prevent the SPL reading from being affected by reflections from surfaces in the room. Place the SPL meter's microphone in front of the face of the loudspeaker at the distance given in the manufacturer's specifications. A pink-noise generator should be fed into the amplifier. The amplifier's output should then be connected to the loudspeaker being tested. A multimeter should be used to measure the voltage across the speaker terminals. Using Ohm's law, determine the power being supplied to the loudspeaker. When the input power matches the manufacturer's specification, record the SPL output of the loudspeaker. Compare the actual reading to the manufacturer-specified SPL. If the level has dropped more than 3 dB, it may indicate that the loudspeaker is damaged.

28. If this test cannot be performed, a simpler test may be done. If necessary, a loudspeaker can be tested by simply listening to it. It is important to have a control standard, a known good loudspeaker that is the same model as the unit under test. The loudspeaker can be connected to an amplifier playing full-range audio, preferably music. The response should be compared to the standard. Pay particular attention to the low and the high frequencies output by the loudspeaker, especially if it is a two-way loudspeaker.

29. The loudspeaker should also be checked for physical damage. A quick shake of the loudspeaker can ensure that all the speakers in the cabinet are secure. Also, if the loudspeaker has screws on the connection terminals, these should be properly tightened to prevent them from vibrating loose and falling off.

30. As with the testing of a full-range loudspeaker, a horn driver can be tested for adequate level by using an SPL meter. Please note, however, that a driver should never be tested without being mounted to a horn. Additionally, low frequencies should *never* be used to test a horn driver, as they will damage it. The specifications of the driver should be checked to make sure that no frequency below the driver's range is used. Pink noise should be processed by a high-pass filter to test the driver in conjunction with a horn. The SPL meter's microphone should be set 3.28 feet (1 m) away from the face of the horn. Compare the test readings with those of a known good driver.

Audio Transformers

31. Checking line-matching transformers is much easier than testing a loudspeaker. The only step required is to check the windings of the transformer for continuity. Connecting the transformer to an amplifier with a constant voltage output (like a 70V system) and then connecting a low-impedance loudspeaker to the output of the transformer will test whether the windings are good. All taps on the transformer should be checked. As the higher-wattage taps are tested, an increase in volume should be noted.

Loudspeaker Lifts

32. Often, events that rely on sound reinforcement can achieve adequate coverage of the facility only when the loudspeakers used are sufficiently lifted and aimed. There are many devices that can be used to elevate loudspeakers. Two of the most common are equipment lifts and tripods. For the purpose of this discussion, equipment lifts and tripods will be referred to simply as lifts. The devices being considered in this section do not use motors, hydraulics, or pneumatics to hoist equipment.

33. Lifts can range from fairly simple devices composed of a couple of telescoping metal tubes to complex systems using cranks, cabling, and winches. A substantial cost would likely be associated with either renting or purchasing the more complex models. Additionally, each lift might potentially suspend equipment weighing several hundred pounds (or kilograms) above the audience. It is therefore both a matter of cost and, more important, a matter of safety to review some general principles about inspecting and maintaining these units. While there are too many styles, brands, and models to comment on every possible situation, it will prove beneficial for us to consider briefly (1) how to clean lifts, (2) what to look for when inspecting them, and (3) how to perform routine maintenance.

34. *Cleaning Lifts:* Most lifts use some sort of metal for their central support as well as for their extremities. While some kinds of metal are resistant to corrosion, a yearly cleaning can help to prevent even the most reactive metals from being weakened because of corrosion. Cleaning will also prevent a lift's parts from binding because of dirt and adhesive residue.

35. While a lightly moistened rag can be used to remove dust, the best cleaners for metal parts are typically chemical sprays made from petroleum distillates. Some common brand names are WD-40 and LPS. These compounds will remove small amounts of corrosion from metal, protect metal from further corrosion by displacing moisture, and dissolve adhesive residues.

36. It is important to use these chemicals only in a well-ventilated area, as some can pose considerable health hazards and are flammable. Also, even a few drops can make floors slippery. Therefore, care should be used to apply the chemical in an orderly manner, away from general traffic. It is best to spray a small amount onto a clean rag and to wipe every metal surface and part until no rust or residue remains. These chemicals will not, however, repair heavily corroded sections of metal.

37. *Inspecting Lifts:* Before ever using any lift, it is a good practice to become familiar with its many parts and how it operates and to inspect each element of the unit thoroughly. The inspection of a lift should begin with reading the manual and all warning/information labels affixed to the unit. If a manual did not come with the lift, the manufacturer or online sources may be able to provide the required instructions. Following these instructions not only can prevent misuse but will likely prove helpful in solving problems that may arise.

38. After reading any manuals and labels, visually inspect each lift. Does it show signs of major corrosion? Are there parts missing, such as bolts or rivets? Is the unit significantly dented? If the lift uses cabling, does the cabling look sound or are there frays and kinks?

39. If the lifts appear to be sound, they should be taken through a full operational check before being used. Each part of the lift should be tested independently. Does each part move without binding? Will the unit rise to its full height? Some units have wheels and casters; do they roll smoothly? Do *all* the safety mechanisms work properly?

40. If the branch owns the lifts and you have permission to repair them, correct any obvious deficiencies before even attempting to hoist equipment. If you do not have permission to repair the lifts, take detailed notes of all problems and, if possible, take pictures. Give all pertinent information to convention oversight, who will contact the branch for direction.

41. *Maintaining and Repairing Lifts:* Routine maintenance will be performed on every branch-owned lift each year. When a lift is broken or malfunctioning, efforts should be made to repair or replace the damaged parts and to verify that the lift is safe for use in the coming year. All lifts that cannot be repaired should be discarded or used solely for spare parts.

42. It is the course of wisdom to use brothers with mechanical experience when performing maintenance work on more complex systems. If such ones are not available and a repair is needed, contact the branch office for direction.

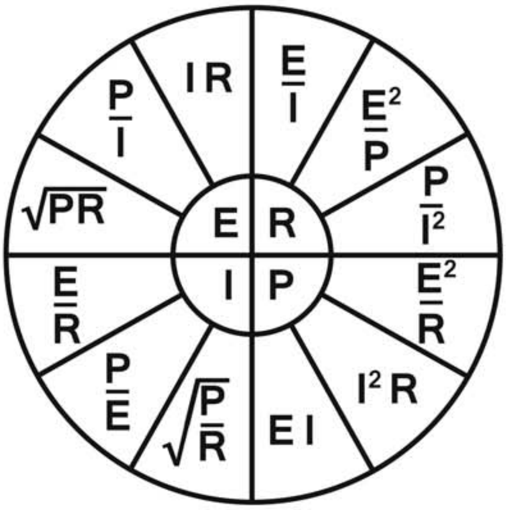
43. For the most part, this maintenance can be done with basic mechanical tools, such as a ratchet set, screwdrivers (flathead and Phillips), and standard silicone lubricants. All moving parts should be lubricated so that they move freely. All nuts and bolts should be checked. Nuts and bolts that show evidence of cross threading or metal fatigue or that have a weight rating less than that of the lift's maximum load should be discarded and replaced.

44. Lifts that use cabling to hoist loads require special attention. Winches, pulleys, and cranks require thorough cleaning every few years. The manufacturer's specifications should be consulted for recommended maintenance timetables and replacement materials. During the thorough cleaning, the grease inside the winch mechanism should be completely removed and replaced. This is vital to make sure that the raising, lowering, and braking mechanisms will not malfunction. Hardened and impacted grease can cause gears and braking mechanisms to bind.

45. Cabling should also be completely inspected every year. Frays, kinks, and breaks in metal cabling will compromise the lift's load-bearing capability. Each country or locality may have its own standards for what is acceptable. However, as a rule, it is better to replace any questionable part, as the safety of all who approach or operate these lifts is involved.

APPENDIX 7

Useful Data, Formulas, and Charts



Symbol	Unit
E	Voltage in volts
R or Z	Resistance (impedance) in ohms (Ω)
I	Current in amperes
P	Power in watts

Figure A7.1 Ohm's Law Chart and Symbol Definitions
(Please see Appendix 1:6-10.)

For two resistance values:

$$R_T = \frac{R_1 \cdot R_2}{R_1 + R_2}$$

For two or more resistance values:

$$R_T = \frac{1}{\frac{1}{R_1} + \frac{1}{R_2} + \frac{1}{R_3}}$$

Figure A7.2 Parallel Resistance Formulas
(Please see Appendix 1:12, 15.)

U.S. System: **Delay (seconds) = Distance (feet) ÷ 1,130**

Delay (milliseconds) = Distance (feet) • .885

Metric: **Delay (seconds) = Distance (meters) ÷ 344**

Delay (milliseconds) = Distance (meters) • 2.9

Figure A7.3 Acoustic Delay Formulas
(Please see Appendix 2:18.)

U.S. System: **Wavelength (feet) = 1,130 ÷ f**

Metric: **Wavelength (meters) = 344 ÷ f**

(f = frequency in hertz)

Figure A7.4 Wavelength (λ) Formulas
(Please see Appendix 2:19-20.)

U.S. System to Metric:			Metric to U.S. System:		
1 inch	=	2.54 centimeters	1 millimeter	=	0.0394 inch
	=	25.4 millimeters			
			1 centimeter	=	0.3937 inch
1 foot	=	0.3048 meter		=	0.0328 foot
	=	30.48 centimeters			
	=	304.8 millimeters	1 meter	=	39.37 inches
				=	3.281 feet
1 mile	=	1.6093 kilometers			
	=	1,609.3 meters	1 kilometer	=	0.621 mile

Figure A7.5 Common Conversions Between
U.S. System and Metric

Exponent	Value	Prefix	Prefix symbol	Example
10^{-6}	0.000001	micro	μ	1 μV = 1 millionth (0.000001) of a volt
10^{-3}	0.001	milli	m	1 mA = 1 thousandth (0.001) of an ampere
10^{-1}	0.1	deci	d	1 dB = 1 tenth (0.1) of a bel
10^{3}	1,000	kilo	k	1 kHz = 1 thousand (1,000) hertz
10^{6}	1,000,000	mega	M	1 MΩ = 1 million (1,000,000) ohms

Figure A7.6 Prefix and Symbol Chart for Exponents
(Please see Appendix 1:29.)

Reference Level	
dBm	0 dBm = 1 milliwatt; 0.775 volt in a 600-ohm circuit
dBu	0 dBu = 0.775 volt, not dependent on load impedance (not a power level)
dBv	0 dBv = 0.775 volt, not dependent on load impedance (obsolete)
dBV	0 dBV = 1.0 volt, not dependent on load impedance
VU	0 VU = 1.2 volts (equivalent to +4 dBu), not dependent on load impedance

Figure A7.7 Reference Chart for Decibel Levels
(Please see Appendix 1:47-56.)

Power: To calculate dB change due to power: $dB = 10 \log P_1/P_2$
To calculate power ratio when dB change is known: $P_1/P_2 = \text{antilog } dB/10$

Voltage: To calculate dB change due to voltage: $dB = 20 \log E_1/E_2$
To calculate voltage ratio when dB change is known: $E_1/E_2 = \text{antilog } dB/20$

Distance: To calculate relative sound pressure level (SPL) loss due to distance: $dB = 20 \log D_f/D_n$

To calculate distance ratio when SPL loss is known: $D_f/D_n = \text{antilog } dB/20$
(f = far; n = near)

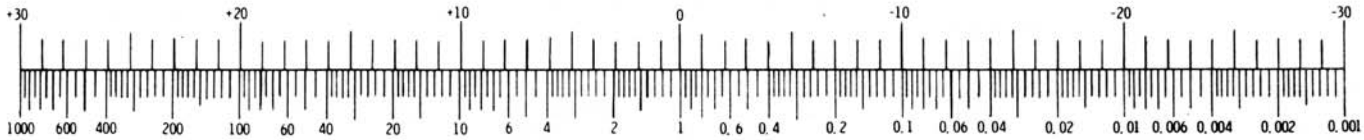
Figure A7.8 Decibel Calculations for Power, Voltage, and Distance
(Please see Appendix 1:36, 37, 41, 42, 58.)

Power Ratio	dB Change	Voltage or Distance Ratio (SPL)
2:1	3 dB	1.4:1
4:1	6 dB	2:1
10:1	10 dB	3.16:1
100:1	20 dB	10:1
1,000:1	30 dB	31.6:1
10,000:1	40 dB	100:1
100,000:1	50 dB	316:1
1,000,000:1	60 dB	1,000:1

Figure A7.9 Decibel Ratio Chart
(Please see Appendix 1:70.)

For determining POWER RATIOS directly in dB

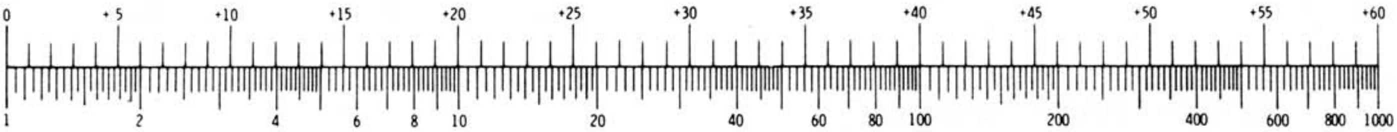
dB ABOVE AND BELOW A ONE-WATT REFERENCE LEVEL



RATIO OR POWER IN WATTS

For determining VOLTAGE RATIOS directly in dB

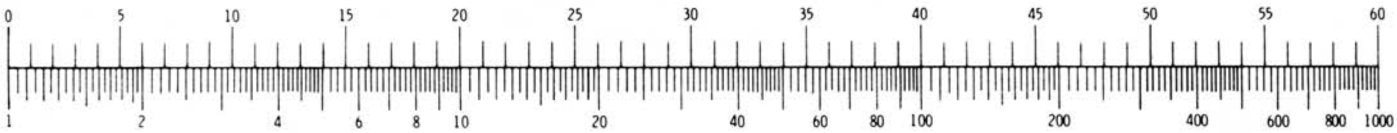
dB—GAIN OR LOSS



VOLTAGE OR VOLTAGE RATIO

For determining RELATIVE SPL directly in dB

dB-SPL



DISTANCE OR DISTANCE RATIO

Figure A7.10 Nomograms for Determining Power Ratios, Voltage Ratios, and Relative Sound Pressure Levels (SPL)
(Please see Appendix 1, Figures A1.8, A1.9, and A1.10.)

When two levels differ by:	Add this dB-SPL to the higher value:
0 to 1 dB-SPL	+ 3 dB-SPL
2 to 3 dB-SPL	+ 2 dB-SPL
4 to 8 dB-SPL	+ 1 dB-SPL
9 or more dB-SPL	0 dB-SPL

Figure A7.11 SPL Summing Chart
(Please see Chapter 4:53.)

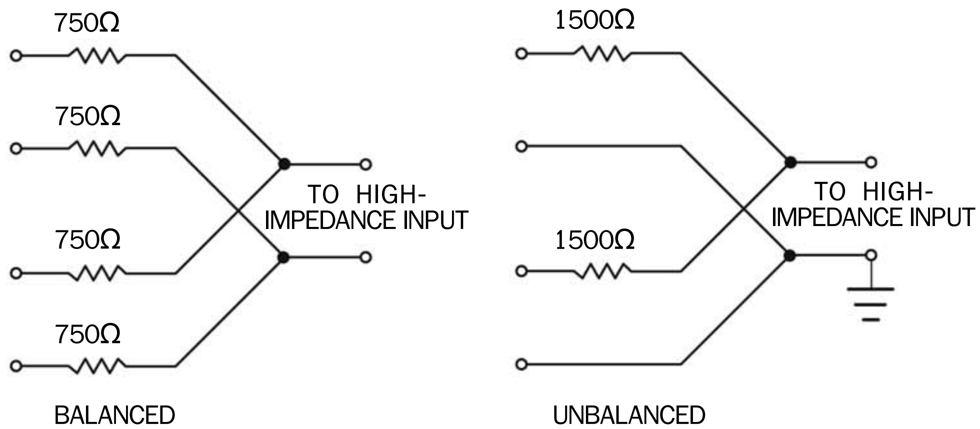


Figure A7.12 Summing Diagram for Balanced and Unbalanced Circuits

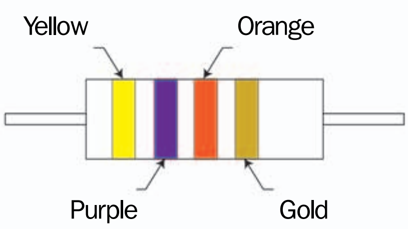
POWER LOSS DUE TO LOUDSPEAKER LINES

					FEEDER-PAIR LENGTH (FT) FROM AMPLIFIER TO LOAD FOR APPROXIMATELY 0.5 dB LOSS IN POWER (11%)								
WIRE SIZE AWG	NEAREST BRITISH SWG	DIAMETER (mm)	OHMS PER 1000′ PAIR (304.80 m)	MAX SAFE POWER AT 70V	4Ω	8Ω	16Ω	POWER AND LOAD IMPEDANCE AT 70V					
								30W 167Ω	40W 125Ω	60W 83Ω	100W 50Ω	250W 20Ω	400W 12.5Ω
10	12	2.59	2.0	1750W	120	240	475	4930	3690	2450	1475	590	370
12	14	2.05	3.2	1400W	75	150	295	3080	2300	1530	920	370	230
14	16	1.63	5.2	1000W	45	90	180	1895	1420	945	570	225	140
16	17-18	1.29	8.0	420W	30	60	120	1230	920	610	370	150	90
18	19	1.02	13.0	210W	15	35	70	760	565	380	230	—	—
20	21	0.81	20.6	70W	10	25	45	480	360	240	—	—	—
22	23	0.65	32.6	35W	7	15	30	300	—	—	—	—	—

Figure A7.13 Power Loss for Speaker Wire (Please see Chapter 4:55.)

Color	1st Figure	2nd Figure	Multiplier	Tolerance
Black	..	0	1	..
Brown................................	1	1	10	1%
Red....................................	2	2	100	2%
Orange	3	3	1,000	3%
Yellow	4	4	10,000	4%
Green	5	5	100,000	..
Blue	6	6	1,000,000	..
Purple................................	7	7	10,000,000	..
Gray..................................	8	8	100,000,000	..
White................................	9	9	1,000,000,000	..
Gold..................................	..	..	0.1	5%
Silver	..	..	0.01	10%
No Color...........................	..	..	....	20%

For example, a resistor shows four bands of color that, starting at one end, appear as yellow, purple, orange, and gold.



Thus, the resistance is 47,000 ohms and the tolerance is 5%.

Figure A7.14 Resistor Color Code Chart

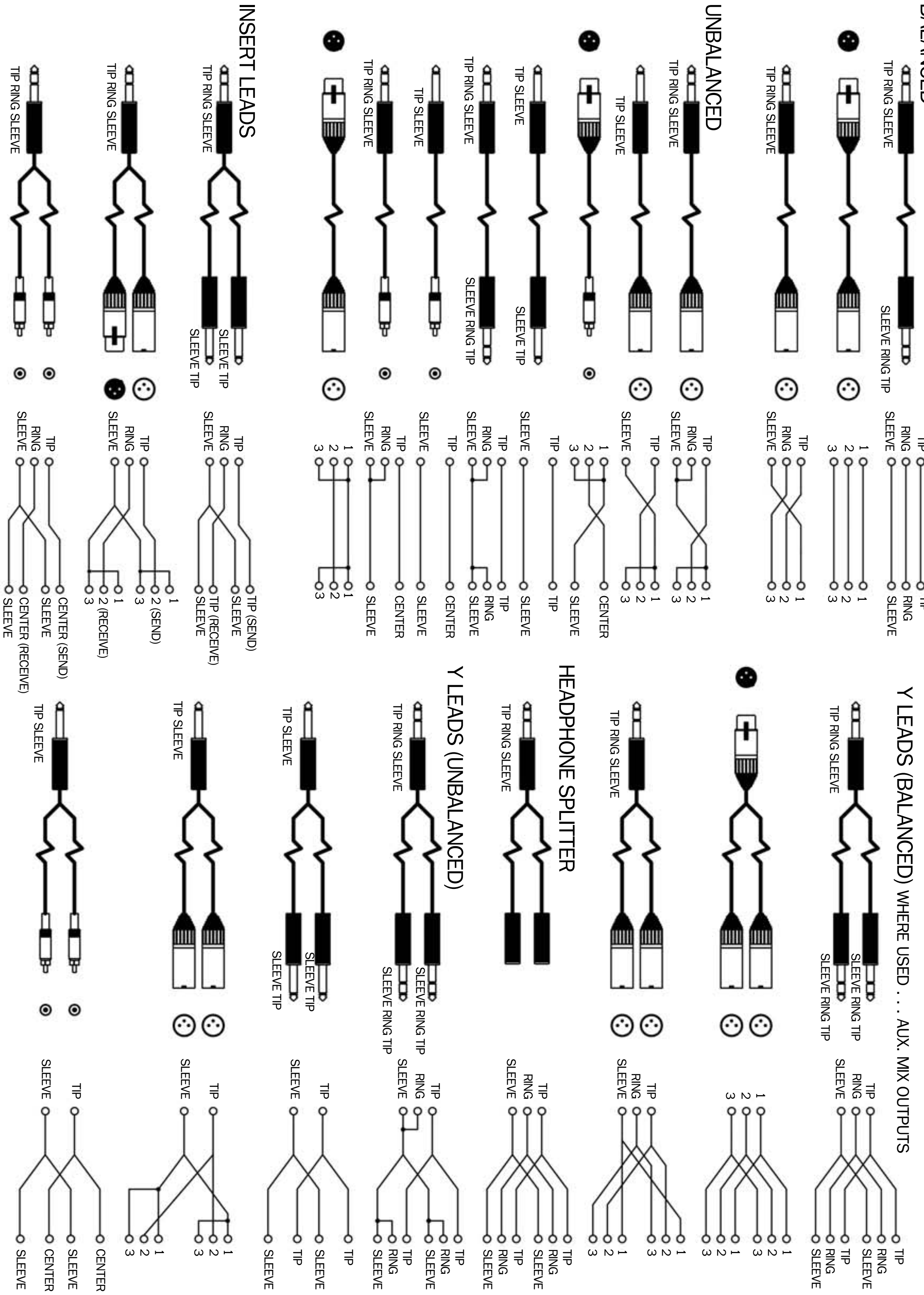
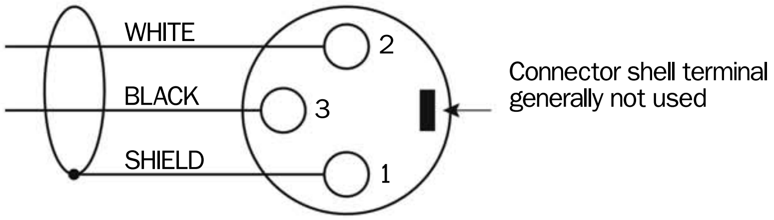


Figure A7.15 Chart to Construct Various Cable Jumpers (Please see Chapter 6:32.)

Wiring and color code for 3-pin audio connectors (XL, A3F, A3M, etc.)



Wiring and color code for 1/4″ phone plugs and jacks

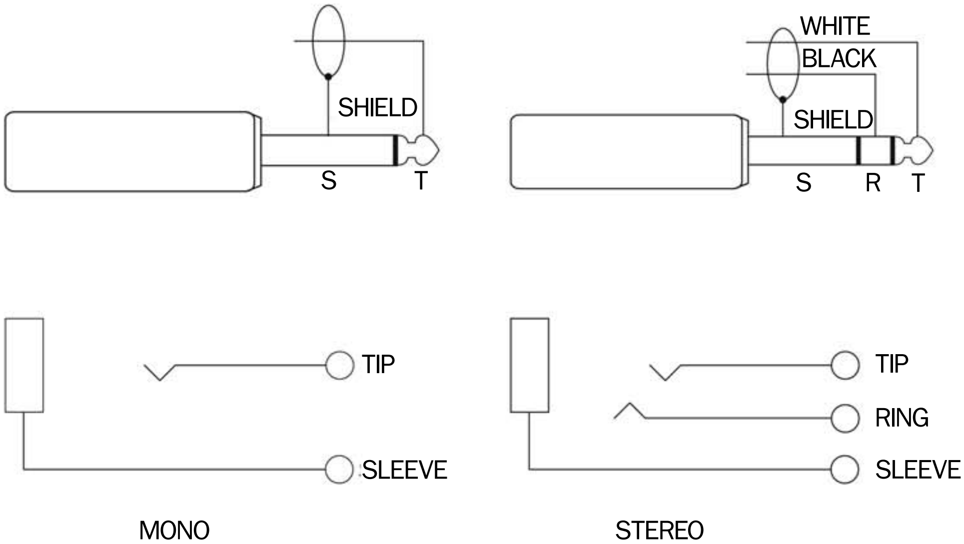
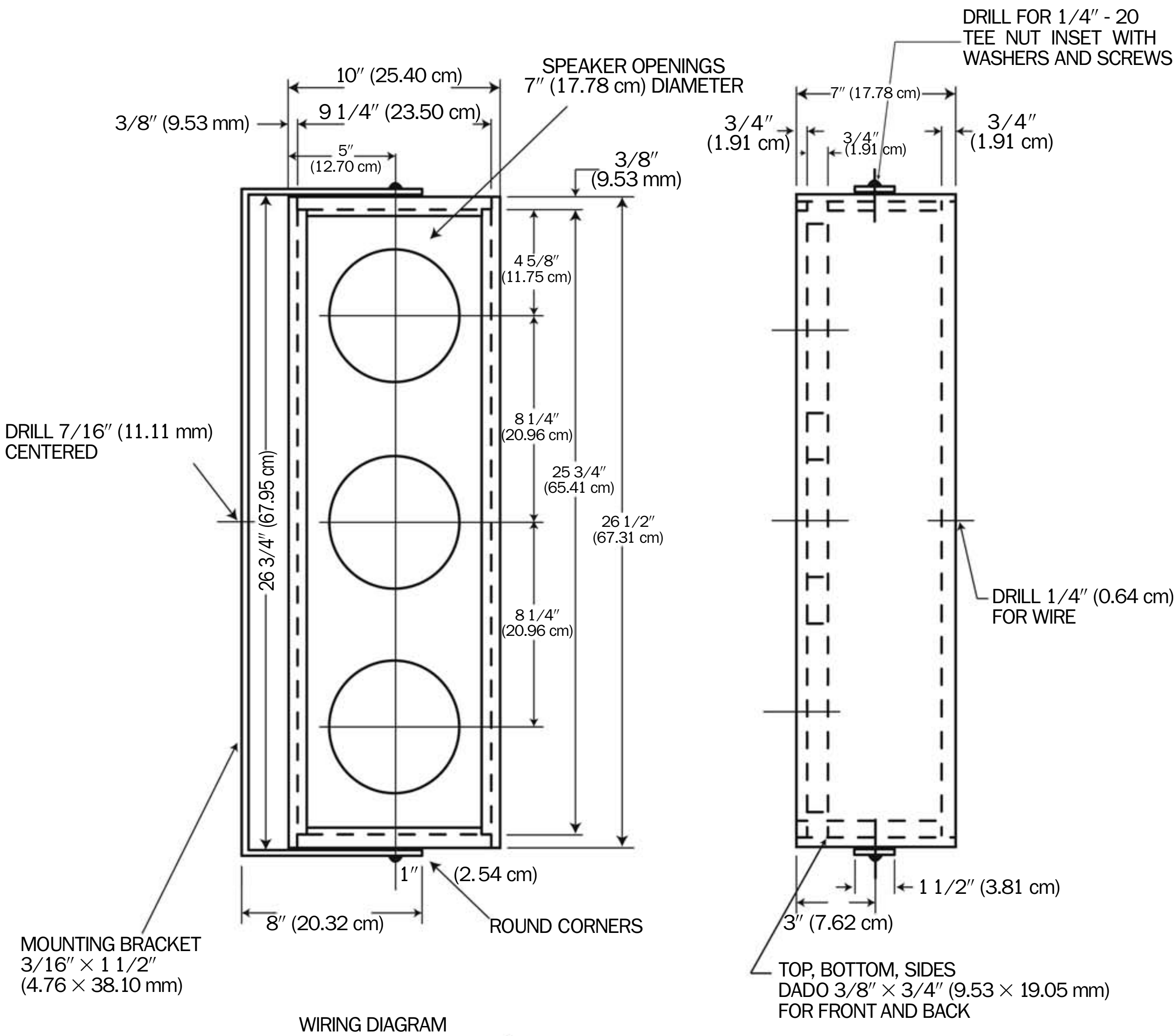
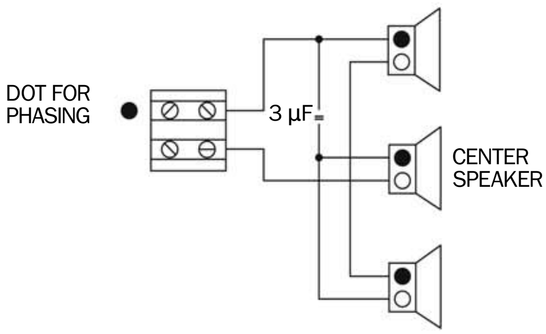


Figure A7.16 Cable Connections for XLR and 1/4″ Phone Plug
(Please see Chapter 6:32-34 and Figure A7.15.)



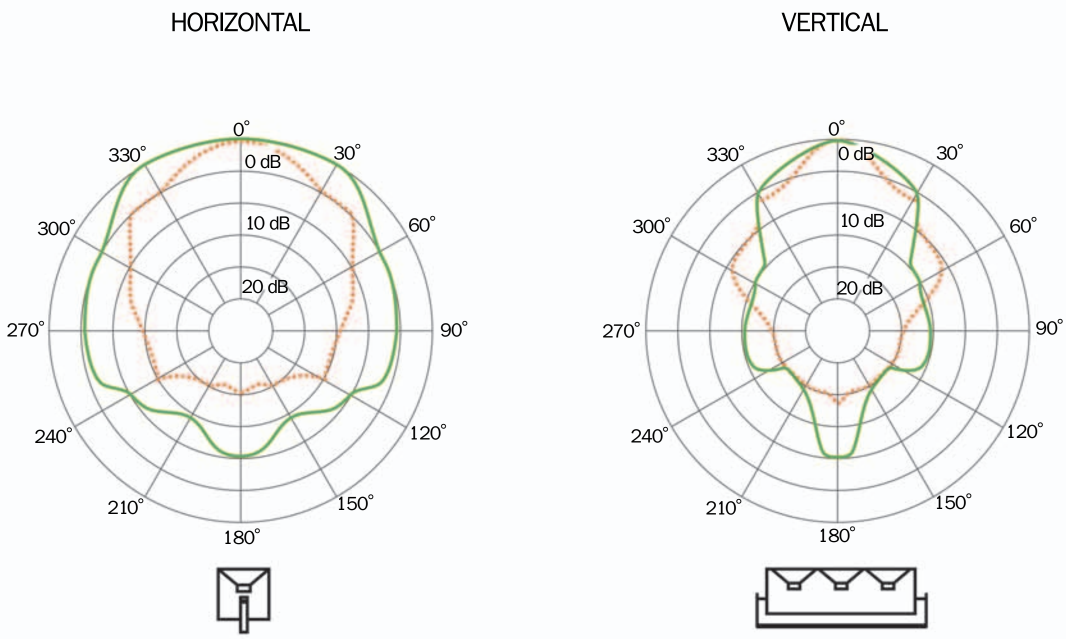
WIRING DIAGRAM



NOTES:
1. ALL MATERIAL 3/4″ (1.91cm) STOCK
2. LOCATE PANEL FOR FRONT-MOUNTED SPEAKERS
3. USE CAULK OR PAPER GASKET UNDER SPEAKER RIM FOR SEAL
4. GRILLE CLOTH STRETCHED AND STAPLED ACROSS FACE
5. PACK BOX LOOSELY WITH DENSE FIBERGLASS BLANKETS
6. BOX TO BE AIRTIGHT
7. LOCATE TERMINAL BLOCK ON BACK PANEL

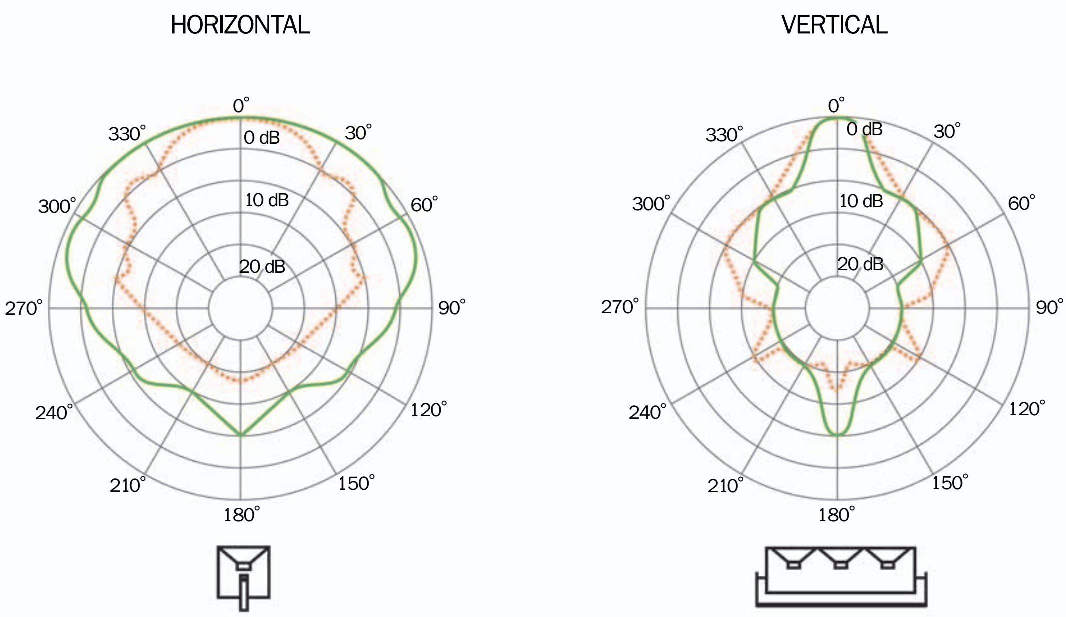
8″ (200 mm) THREE-SPEAKER COLUMN		
ITEM	NUMBER REQUIRED	PARTS LIST
1	ONE EACH	FRONT AND BACK 3/4″ × 9 1/4″ × 25 3/4″ (1.91 × 23.50 × 65.41 cm)
2	ONE EACH	TOP AND BOTTOM 3/4″ × 7″ × 9 1/4″ (1.91 × 17.78 × 23.50 cm)
3	2	SIDE 3/4″ × 7″ × 26 1/2″ (1.91 × 17.78 × 67.31 cm)
4	3	8″ (200 mm) PM SPEAKER—10-oz. (0.28 kg) MAGNET
5	1	3 µF CAPACITOR (NONPOLARIZED)
6	1	2 TERMINAL JONES BARRIER STRIP

Figure A7.17 Construction Details for Watchtower Sound Column
(Please see Chapter 4:21.)



Watchtower sound column - 8″ (200 mm) speakers

Sensitivity (1W @ 4 ft [1.22 m]): 94 dB-SPL(A)



Two 8″ (200 mm) Watchtower sound columns, stacked vertically

Sensitivity (1W @ 4 ft [1.22 m]): 94 dB-SPL(A)

400-800 Hz: 2,000-4,000 Hz:

Figure A7.18 Polar Plots for Watchtower Sound Columns
(Please see Chapter 4:21.)

Proper Microphone Techniques

When the brother speaking is standing at a lectern, the microphone should typically be level with his chin and angled about 30 degrees upward toward his mouth at a distance of four to six inches (10 to 15 cm).

Figure A7.19 Proper microphone position at the lectern (Please see Chapter 8:15-17.)

When a participant is using a handheld microphone, the microphone should be held with the microphone surface slightly below the lips at chin level and at a distance of four to six inches (10 to 15 cm) from the mouth. The talker should speak directly over the top of the microphone.

Figure A7.20 Proper handheld microphone position (Please see Chapter 8:18-20.)

Basic Guidelines

- Any cellular phone taken on the stage must be powered off.
- Do not pull the lectern away from the microphone stand.
- The microphone should generally be about four to six inches (10 to 15 cm) from your mouth.
- Project your voice. Use a little more volume and intensity than you would in conversation.
- When addressing other participants on the stage, always turn your head in such a way that you are still speaking into the microphone.
- If speaking softly for effect, move very close to the microphone. If speaking loudly, pull away.
- If you hear popping from breath noise, speak across the microphone rather than directly into it.
- If you hear feedback, move closer to the microphone.
- If you need to clear your throat or have the urge to cough or sneeze, be sure to turn your head away from the microphone.

Further Reading

Additional information on audio engineering and sound reinforcement can be found in such references as:

- JBL *Audio Engineering for Sound Reinforcement*
- *Microphone Techniques for Live Sound Reinforcement*
  —A Shure Educational Publication
  (Free PDF at http://www.shure.com/americas/support/)
- Yamaha *Sound Reinforcement Handbook*

APPENDIX 8

Glossary

ABSORPTION – (1) The dissipation of sound energy into other forms of energy, such as heat, as a result of the sound waves interacting with matter. (2) The ability of a material to absorb sound energy, thereby reducing the amount of energy reflected.

ABSORPTION COEFFICIENT – A measure of the sound-absorbing qualities of a given material at a particular frequency and affected by angle of incidence. When it is given as a number between 0 and 1, "0" indicates that the material is completely reflective and "1" indicates that it is completely absorptive. The measure is sometimes given as a percent of the absorption of open space (100%).

ABSORPTION UNIT (U) – A measure of acoustic absorption; equal to one square foot (1 sq m) of totally absorptive material. The total U for a specific surface is obtained by multiplying its total area (in square feet [sq m]) by the established absorption coefficient for the surface material at a specified frequency. Sometimes measured in sabins.—See also "Sabin."

ACOUSTIC GAIN – The increase in sound level (often expressed in decibels) provided by a sound-reinforcement system as compared with the level when the system is not in operation.

ACOUSTICS – The science of sound. Can also refer to the effect that a given environment has on sound.

ACTIVE DEVICE – A component that has the ability to amplify a signal by utilizing a power source other than the input signal.

ACTIVE EQUALIZER (or Filter) – See "Equalizer."

ALTERNATING CURRENT (AC) – A flow of electricity that reaches a maximum in one direction, reverses itself, passes through zero until it reaches a maximum in the opposite direction, reverses itself again, and repeats the cycle continuously.

AMBIENCE – The distinctive acoustic character of a given room or enclosure, resulting from reverberation and reflections from walls, ceiling, and other surfaces.

AMBIENT NOISE – Unwanted background noise surrounding a microphone or a listener in a given location.

AMPERE – The basic unit of electrical current or rate of flow of electrons. A current of one ampere will flow when one volt is applied across a resistance of one ohm.

AMPLIFIER – An electronic device capable of increasing the magnitude of electrical signals. Amplifiers used to isolate, control, or process a signal do not always increase the level of the signal.

- **DISTRIBUTION AMPLIFIER** – An amplifier with at least one input and multiple outputs used to distribute a signal to a number of amplifiers or independent loads and designed so that changes in load do not appreciably affect other output circuits. May also serve as a buffer or as an isolation amplifier.
- **LINE AMPLIFIER** – An amplifier designed to supply audio to a transmission line at a specified level, usually between –10 and +24 dBm.
- **MONITOR AMPLIFIER** – A power amplifier used in conjunction with a loudspeaker for supervising or evaluating the audio quality of the output.
- **POWER AMPLIFIER** – (1) An amplifier intended to supply power to one or more loudspeakers. (2) The final amplifier in an audio chain, designed to increase the power of a signal, as opposed to a voltage amplifier.
- **VOLTAGE AMPLIFIER** – Designed primarily to increase the voltage of a signal. It is not designed to increase the actual power of the signal.

AMPLITUDE – The level or strength of an acoustic or electrical signal. Often used instead of "level" or "volume."

ARTICULATION – The clear, distinct production of speech sounds. In sound-reinforcement, "articulation" often refers to a quantitative measurement of the intelligibility of human speech.

ARTICULATION LOSS – An estimated measurement, shown as a percent, of consonants that have been masked or lost as a result of reverberation or extraneous noise. In a typical sound-reinforcement system, an articulation loss of up to 10 percent is considered acceptable.

ATMOSPHERIC ABSORPTION – The absorption of sound energy by the air in excess of the loss due to inverse square law. Energy losses that are due to atmospheric absorption increase with frequency and vary inversely with relative humidity.

ATTENUATION – Any decrease in the strength of an electrical or acoustic signal during its transmission from one point to another, usually expressed in decibels.

ATTENUATOR – A device for reducing the voltage or power of a signal. In audio, usually a resistive network for reducing the amplitude of an electrical signal without otherwise changing it.

AUDIO SPECTRUM – The complete range of frequencies audible to the human ear, considered to be from 20 to 20,000 hertz.

AUDIO SPECTRUM ANALYZER – An instrument that, through a swept filter or series of filters covering the audio spectrum, is capable of graphically displaying the frequency response of a device or a system.—See also “Real-Time Analyzer.”

AUTOTRANSFORMER – A single-winding transformer with taps on the winding to achieve various ratios of transformation. An autotransformer does not provide DC isolation between its input and output circuits.

AWG – Abbreviation for “American wire gauge,” an industry standard for specifying wire diameter. The greater the AWG number, the smaller the wire diameter.

AXIS – An imaginary centerline projected directly outward and forward from a transducer, such as a loudspeaker or microphone. “On axis” represents the 0° point on both horizontal and vertical polar patterns.

BALANCED – Electrically alike or symmetrical with respect to a common reference point or ground. Specifically, both signal leads in a balanced line are at equal potential with respect to ground, usually because of the grounded center tap of the transformer winding to which they are connected. Although technically different, “balanced” is sometimes substituted for the term “floating.”—See also “Floating.”

BAND – Any range of frequencies falling between two specified limits, such as an octave band.

BANDPASS – The number of hertz between the upper and lower limits of the spectrum that a filter allows to pass with minimum attenuation. Bandpass is usually expressed as the frequencies between the upper and lower half-power (–3 dB) points of a filter.

BANDWIDTH (Audio) – (1) The range of frequencies within the limits of a bandpass filter, usually measured between half-power points. (See also “Bandpass.”) (2) The range of useful frequencies over which an amplifier will respond and provide useful output.

BASS – The very low frequencies of the human voice or music, defined in music as all tones below middle C (261.63 Hz). In audio, the term normally includes frequencies below 500 hertz.

BEAM – Energy that radiates from a source and is concentrated in one direction.

BIAMPLIFICATION – A full-range audio system employing a low-level crossover to divide low and high frequencies into two separate audio signals so that they may be amplified and reproduced separately.

BOOM – The transverse portion of a microphone stand that can be adjusted to different angles.

BRIDGING – A method of obtaining a signal from an audio circuit without loading or taking appreciable power from that circuit. The input impedance of the bridging device is at least ten times the impedance of the signal source.

BRIDGING TRANSFORMER – See “Transformer.”

BRIGHT – An expression usually describing a large amount of high-frequency energy in a sound system.

BUFFER – (1) To isolate one circuit from another so as to avoid undesirable interaction between them while permitting the desired signal to flow through in the proper direction. (2) A device, such as an amplifier, that performs the above action.

CABLE – A combination of conductors insulated from one another, usually within a common protective sheath, and arranged so that the conductors may be used separately or in groups.

CALIBRATION – The adjustment of an instrument to make it conform to a certain standard of accuracy.

CAPACITIVE – Having to do with capacitance or the characteristics of a capacitor.

CAPACITOR – A device consisting basically of two conductive surfaces separated by an insulating material such as air, paper, or plastic. A capacitor stores electrical energy, blocks the flow of direct current, and permits the flow of alternating current to a degree dependent on the capacitance and frequency.

CARDIOID MICROPHONE – See “Microphone.”

CENTER FREQUENCY – The frequency where the greatest amount of boost or attenuation occurs in each filter section of an equalizer or a notch filter.

CERAMIC MICROPHONE – See “Microphone.”

CHASSIS – The metal case or frame on which an electronic device is mounted or the entire piece of equipment when removed from its enclosure.

CHOKE – An inductive device used to impede the flow of alternating current or pulsating direct current. In audio, chokes are often used to impede radio frequencies while allowing audio frequencies to pass; also used in DC power supplies to impede AC line ripple while allowing DC to pass.

CIRCUIT BREAKER – A device that, without damage to itself, automatically interrupts electrical flow in a circuit when the current flow exceeds a predetermined value.

CLIPPING – Severe audio distortion characterized by the flattening of the waveform at an amplifier’s output because the amplifier is being operated beyond its capacity. Clipping can also be caused by an excessively high input signal to an amplifier.

CLIPPING LEVEL – The signal level at which clipping begins to occur.

COMPONENT – An essential functional part of a system. (1) The individual basic parts used in building electronic equipment, such as resistors and capacitors. (2) A complete electronic device, such as a mixer or an equalizer, designed to perform a specific function in a sound system.

COMPRESSION DRIVER – A high-frequency dynamic transducer designed to be coupled to a horn to increase amplification and improve overall efficiency.

COMPRESSOR – An amplifier that automatically regulates its gain in order to reduce the dynamic range of program material. Normally, a compressor decreases gain only when the input signal exceeds a given level or threshold.—See also "Limiter."

CONDENSER MICROPHONE – See "Microphone."

CONTINUITY – A continuous path for the flow of current in an electrical circuit.

CORE – The magnetic material placed within a coil or transformer to intensify the magnetic field. In transformer design, the larger the core, the greater the inductance of windings and, thus, the greater the low-frequency efficiency of the transformer.

CRITICAL DISTANCE (D_C) – The distance from a sound source where the direct (inverse square) sound level equals the general reverberant level.

CROSSOVER FREQUENCY – The frequency point at which a division is made between high and low frequencies for separate amplification and/or reproduction.

CROSSTALK – Interference caused by stray coupling of energy from one circuit to another, as between pairs in a cable.

CURRENT – The movement of electrons through a wire or another conductor. Measured in amperes.

CURVE – A graphic representation of a range of measurements compared to time, frequency, and so forth. Usually related to the audio spectrum.

CYCLE – The change in an alternating wave—whether electrical, physical, or acoustic—from zero to a positive peak, then back through zero to a negative peak, and back to zero again.

CYCLES PER SECOND (cps) – The term refers to the frequency of a sound or another regular waveform. It has been replaced by the term "hertz."

DECAY – The gradual decrease of a quantity, as in the steady reduction in intensity of the reverberant field in a room after the source of sound energy is shut off.

DECIBEL (dB) – A unit for the measurement of ratios of sound level, power, voltage, and other quantities. Although the decibel is 1/10 of a bel, "decibel" is used exclusively instead of "bel." The term defines only a ratio between two levels of power. To express a specific quantity, the decibel must be referenced to some actual value, as described below.

- **dBm** – An electrical measurement of power, where 0 dBm is equal to one milliwatt (0.001 watt), which is equivalent to 0.775 volt across 600 ohms. Power levels in dBm are thus perceived as dB above (+) or below (–) one milliwatt.
- **dB-SPL** – A measure of acoustic sound pressure level. The lowest sound level the average youth can hear is 0 dB-SPL, which is referenced to 0.0002 dyne/cm^2. One dB-SPL is the smallest change in sound level detectable by the average person.
- **dBu** — A voltage level where 0 dBu is always referenced to 0.775 volt without regard for circuit or load impedance.
- **dBv** – Also a measurement of voltage, but where 0 dBv is equal to 0.775 volt. This measurement is now obsolete.
- **dBV** – An electrical measurement of voltage, where 0 dBV is equal to one volt.

DEDICATED LINE – A line or circuit specifically assigned to some special use or purpose.

DIGITAL DELAY – An electronic device utilizing digital techniques to delay an audio signal for a predetermined length of time, usually less than 250 milliseconds. Used in sound reinforcement to delay one signal with respect to another, thus compensating for the transit time of sound in air.

DIGITAL SIGNAL PROCESSOR (DSP) – An electronic device that typically converts analog audio into digital, allows for manipulation of the digital signal, and then converts the signal back into analog audio. DSP devices may offer gain adjustment, matrix mixing, compression, delay, equalization, feedback cancellation, and many other processing duties formerly handled by individual electronic units.

DIRECT CURRENT (DC) – A current, whether constant or varying, flowing in only one direction in an electric circuit. It may be continuous, or it may be interrupted—as in pulsed DC.

DIRECT FIELD – See "Fields."

DIRECTIVITY FACTOR (Q) – In loudspeakers, the ratio of sound pressure produced at a given distance on axis compared to what it would be if the device were an omnidirectional transducer of the same efficiency. Thus, the directivity factor is a measure of the ability of a loudspeaker to project sound in a forward direction.

DIRECTIVITY INDEX (Di) – Expressed in decibels, a means of describing the directivity of a loudspeaker in terms of forward gain compared to an omnidirectional radiator. Di is mathematically related to the directivity factor (Q) of a transducer, as shown by the formula: $Di = 10 \log Q$.

DISPERSION – The manner in which sound radiates from a loudspeaker; the coverage angle of a loudspeaker at specified frequencies.

DISTORTION – Unwanted changes in the waveform of a signal.

DISTRIBUTED SYSTEM – A sound system using many loudspeakers in various locations so as to cover the audience listening area. Since the acoustic path from loudspeaker to listener is relatively short, the overall level can be kept relatively low.

DISTRIBUTION AMPLIFIER – See "Amplifier."

DRIVER – (1) A term applied to any sound-reproducing device, but particularly to a transducer used to supply sound energy to a horn-type loudspeaker. (2) Any electronic circuit or device that supplies input to another circuit.

DYNAMIC MICROPHONE – See "Microphone."

DYNAMIC RANGE – (1) The difference in decibels between the maximum and minimum levels acceptable in a system or system component, as limited by inherent noise and maximum power-handling capacity of the equipment. (2) The span of volume between the loudest and softest sounds in program material.

EARTH – The British term used for "ground."

EFFICIENCY – Usually expressed as a percent, the ratio of the total useful output power of a device compared to its total input power. In loudspeakers, it is the useful sound energy produced divided by the electrical power supplied. Efficiency is not the same as sensitivity.—See also "Sensitivity."

EQUALIZATION – Adjustment of the frequency response of an audio circuit or system to achieve a flat or other desired response. Also applied to the adjustment of the overall system response, including the acoustics of the room.

EQUALIZER (EQ) – An audio device that can amplify (boost) and/or attenuate (cut) portions of the audio spectrum as desired.

- **ACTIVE EQUALIZER** (or Filter) – A frequency-selective device that has its own internal amplification for altering the frequency content of an audio signal, as opposed to a passive equalizer that only uses resistors, capacitors, or inductors.
- **PASSIVE EQUALIZER** (or Filter) – A frequency-selective device that does not utilize an amplifier or other powered electronics. Such devices usually have considerable insertion loss and require accurate impedance matching.

FADER – A potentiometer for controlling the signal level of one of the input or output channels of a mixer or console.

FAR FIELD – See "Fields."

FEEDBACK, ACOUSTIC – Unwanted interaction (oscillation) between the output and input of a sound system. Feedback occurs when the system gain has been increased to the point that sound from the loudspeakers reaches the microphone in phase and in amplitude at least equal to the talker's voice.

FIELDS

- **DIRECT FIELD** – In an enclosed space, the area in front of a loudspeaker where sound is composed primarily of direct energy from the loudspeaker and where inverse square law relationships are still apparent and measurable.
- **FAR FIELD** – An area distant enough from the sound source so that the sound pressure level decreases by 6 dB for each doubling of the distance from the source, according to inverse square law.
- **FREE FIELD** – An acoustic area where the effects of boundaries are negligible, such as an open field or an anechoic chamber, and where sound behavior is controlled predominantly by inverse square law.
- **NEAR FIELD** – The acoustic field very close to a loudspeaker where acoustic aberrations are likely to occur as a result of incomplete combining of sound from the various loudspeaker elements, usually considered to include a distance at least two times the major face dimension of the loudspeaker.
- **REVERBERANT FIELD** – The area in an enclosed space where sound is composed primarily of reverberation rather than direct energy from a loudspeaker. The imaginary boundary between the direct and reverberant fields is known as the critical distance (D_C).

FILTER – An electronic network that offers relatively little opposition to certain frequencies while blocking or attenuating others.

FLAT FREQUENCY RESPONSE – An expression describing an audio system or circuit that will pass audio frequencies with negligible variation over the frequency band of interest, which includes the entire audio spectrum from 20 hertz to 20 kilohertz.

FLOATING (Circuit) – An input or output circuit that is isolated from a ground by the use of transformer coupling. Many so-called balanced circuits, such as balanced microphone circuits, are actually floating circuits. In many circumstances, floating circuits are safer and are often preferred over true balanced circuits.—See also "Balanced."

FLUTTER ECHO – A rapid succession of echoes due to reflections between two parallel surfaces.

FREE FIELD – See "Fields."

FREQUENCY – The number of complete cycles per second of voltage, sound waves, or an alternating current, expressed in hertz.

FREQUENCY RANGE – The frequencies that a system is able to transmit without attenuating the signal more than an arbitrary amount—thus the range of frequencies an amplifier can reproduce. The phrase often refers to all the frequencies that can be passed regardless of whether or not they conform to a certain degree of flatness, thus differing from frequency response.

FREQUENCY RESPONSE – A measure of how uniformly a transducer or a system reproduces the audio frequency spectrum fed to it. The term has little meaning unless a tolerance, such as "±2 dB," is specified.—See also "Flat Frequency Response."

FUNDAMENTAL – The principal component of a sound wave—that is, the component with the lowest frequency or greatest amplitude. Also, the basic pitch of a musical note.

GAIN – The ratio indicating the increase in the strength of an electrical signal due to the action of an amplifier, usually expressed in decibels or as a multiplier of amplification.

GROUND – (1) *Earth:* A physical electrical connection to the earth. A metal rod driven into the earth or a cold-water pipe can provide such connection. Can also be termed "an earth ground." (2) *Electrical:* The zero voltage reference point in an electrical system or an electronic device. (3) *System:* The common point at earth ground potential to which the chassis and shields of various components are connected to minimize hum and noise.

GROUND LOOP – A condition where two or more paths to ground exist and form a loop, or circuit, and where stray voltages may appear in the circuit, causing hum or noise.

HARMONICS – Integer multiples of a fundamental frequency. The second harmonic is twice the fundamental frequency, the third harmonic three times the frequency, and so forth.

HEADROOM – A safety margin, usually expressed in decibels, between the normal operating level and the maximum, or clipping, level.

HERTZ (Hz) – The unit of frequency measurement. One hertz equals one cycle per second. One kilohertz equals 1,000 cycles per second, and so forth.

HIGH FREQUENCY – In audio, the treble, or higher-pitched, elements of voice or music. In sound reinforcement, however, it often refers to frequencies 5 kilohertz and above.

HIGH IMPEDANCE – Usually refers to circuit impedance above 2,000 ohms. A high-impedance input does not draw appreciable power from a low-impedance source feeding it but responds mainly to the voltage produced by the source.

HIGH-PASS FILTER – A filter that restricts the passage of signals below a certain frequency without appreciably affecting those above that frequency.

HORN – A flared, or funnel-shaped, enclosure that helps to control the dispersion of sound and to couple the driver acoustically to the air.

HOT – (1) Connected, live, energized. Often refers to any ungrounded terminal or conductor, such as the ungrounded side of an unbalanced line or circuit. (2) An expression used to describe an electrical or acoustic level that is higher than preferred or required.

HOUSE CURVE – The overall frequency response of a sound-reinforcement system operating in a room or an enclosure before any equalization is done.

HOUSE SYSTEM – The existing sound system in a facility.

HUM – A low-pitched droning noise frequently made up of several harmonically related frequencies that results from an inadvertent pickup of energy from an AC line due to improper shielding, faulty grounding, or inadequate filtering of the sound system power supply.

IMPEDANCE (Z) – The total opposition, including resistance and reactance, that a circuit presents to the flow of an alternating current at a given frequency. Measured in ohms.

- **INPUT IMPEDANCE** – The impedance “seen” by a signal source connected to the input of a device.
- **LOAD IMPEDANCE** – The total impedance presented to a signal source by the device(s) connected to it.
- **OUTPUT IMPEDANCE** – The impedance of a signal source, such as the output of an amplifier. In modern equipment, the term often refers to the nominal impedance in which the unit is

designed to operate. This may be many times higher than the actual source impedance of the unit.

IMPEDANCE BRIDGE – A device for measuring the combined resistance and reactance (i.e., the impedance) of a component, circuit, or line.

IMPEDANCE MATCHING – The technique of connecting a load to its source so that the impedances of the source and of the load are approximately equal. Important in audio power circuits to permit the most efficient transfer of power, as from amplifier to loudspeaker load. Exact impedance matching is seldom required in the interconnecting of low-level devices as long as the load impedance is greater than the source impedance.

INDUCTANCE – The electrical property of a coil or winding that opposes any change in current. Measured in henrys.

INPUT IMPEDANCE – See “Impedance.”

INSERTION LOSS – The power loss, usually expressed in decibels, resulting from the insertion of an electronic device, typically a passive type, into the signal path.

INSTABILITY – A tendency toward regeneration or oscillation.

INTELLIGIBILITY – The capability of being clearly understood.—See also “Articulation.”

INTENSITY – The relative amplitude, or strength, of energy, whether electrical or acoustic.

INTERCONNECTION – The physical wiring between electrical units and systems.

INVERSE SQUARE LAW – In a free field, the sound pressure level varies inversely as the square of the change in distance from the source. Thus, a doubling of the distance decreases acoustic energy to one quarter, an SPL loss of 6 decibels.

ISOLATION – Electrical or acoustic separation between two units or locations. The absence of signal leakage from one channel, or signal path, to another in a given component. Also, the absence of sound leakage in adjacent acoustic environments.

ISOLATION TRANSFORMER – See “Transformer.”

KILO – One thousand (10^3) times what follows, as in kilowatt (1,000 watts).

KILOHERTZ (kHz) – 1,000 hertz, or 1,000 cycles per second.

LEVEL – (1) A term loosely used to describe the amplitude of an electrical signal or a sound. (2) More precisely, the magnitude of a signal or a sound compared to a fixed reference. It is commonly expressed in dBm, dB-SPL, and so forth.

- **LINE LEVEL** – The standard nominal operating level for most recording and sound-reinforcement systems, where "0 VU" generally corresponds to +4 dBm, or 1.228 rms volts in a 600-ohm circuit.
- **MICROPHONE LEVEL** – The signal level approximating the nominal output level of a microphone, which is generally about 0.001 to 0.003 rms volt for a low-impedance dynamic microphone. In the presence of very loud sounds, however, most microphones are capable of producing considerably higher levels.

LEVEL INDICATOR – A device that visually displays the level of an audio signal as a means of controlling or supervising the optimum amount of signal in a given system.—See also "VU Meter."

LIMITER – A variable gain device in which the amplitude of the output is automatically prevented from exceeding a predetermined level. Similar to a compressor with a 10:1 or greater compression ratio.—See also "Compressor."

LIMITING DISTANCE – The maximum distance a listener can be from a loudspeaker in a given environment and still hear with acceptable intelligibility.

LINE AMPLIFIER – See "Amplifier."

LINEAR – Having an output or a response that varies in direct proportion to the input. A linear amplifier is one in which the output signal is always an amplified replica of the input signal.

LINE ARRAY – Typically, a grouping of loudspeakers coupled together so as to produce a relatively narrow vertical coverage pattern.

LINE INPUT TRANSFORMER – See "Transformer."

LINE LEVEL – See "Level."

LINE LOSS – The total of all losses in a transmission line, usually expressed in decibels.

LINE OUTPUT TRANSFORMER – See "Transformer."

LINE RADIATOR – A loudspeaker system that comprises two or more driving elements mounted in a line, commonly called a sound column. When the speakers are mounted vertically, this configuration produces a narrow vertical coverage angle but a broad horizontal dispersion angle similar to a single cone-type loudspeaker.

LISTENER FATIGUE – Auditory and/or mental exhaustion resulting from excessive sound levels, distortion, reverberation, or other factors that make hearing or understanding difficult.

LOAD – A circuit or a device that receives power. The amount of power drawn by such a device.

LOAD IMPEDANCE – See "Impedance."

LOSS – A decrease in power suffered by a signal as it is transmitted from one point to another, usually expressed in decibels. Loss is energy dissipated without accomplishing useful work.

LOUDNESS – The subjective perception of sound intensity by the human ear. Perceived loudness varies according to frequency, timbre, ambient noise, and hearing acuity, in addition to the actual level of sound.

LOUDSPEAKER – An electroacoustic transducer that converts audio frequency electrical energy into audible sound waves. "Loudspeaker" and "speaker" are often used interchangeably.

LOW FREQUENCY – The bass, or lower-pitched, tones of voice or music. In music, this refers to the frequencies below middle C (261.63 Hz). In sound systems, however, low frequency generally refers to the energy below a given crossover frequency, such as 500 or 800 hertz.

LOW IMPEDANCE – When used in reference to microphones and microphone inputs, the term refers to a characteristic impedance of 50 to 600 ohms. Low-impedance microphone circuits are characterized by a greater resistance to hum, RFI, and other types of interference.

MATRIX MIXER – Usually a function found on a DSP, this allows any input to be routed to any output.

MAXIMUM POWER RATING – The highest amount of electrical power that a loudspeaker, an amplifier, or another device can safely tolerate.

MEGA – One million (10^6) times what follows, as in megohm (one million ohms).

MICRO – One millionth (10^{-6}) part of what follows, as in microfarad (one millionth of a farad).

MICROPHONE – An acoustic transducer that converts sound waves into corresponding electrical signals.

- **CARDIOID MICROPHONE** – A directional microphone with an on-axis response shaped like a cardioid. It has a uniform response over 180 degrees in front and minimum response in back.
- **CERAMIC MICROPHONE** – Utilizes a crystalline substance, such as quartz or some other piezoelectric ceramic element, to convert sound to electrical energy. Also called a crystal microphone. While ceramic microphones have a relatively high output level, their sound quality is generally considered inadequate for sound-reinforcement applications.

- **CONDENSER MICROPHONE** – A microphone where the diaphragm is one plate of a variable capacitor (condenser). Condenser microphones require either phantom power or an external power supply to operate.
- **DYNAMIC MICROPHONE** – A microphone that converts sound waves to electrical energy by means of a moving coil vibrating in a magnetic field. Because they are relatively rugged and capable of smooth response over the usable spectrum, they are commonly used for sound-reinforcement work.
- **UNIDIRECTIONAL MICROPHONE** – A cardioid microphone.

MICROPHONE INPUT TRANSFORMER – See "Transformer."

MICROPHONE LEVEL – See "Level."

MILLI – One thousandth (10^{-3}) part of what follows, as a millisecond (one thousandth of a second) or milliwatt (one thousandth of a watt).

MISMATCH – A condition where distortion occurs and/or optimum power transfer is not realized because the source and load impedances do not correspond. A mismatch is often intentional in order to achieve desired signal attenuation.

MIXER (Audio) – A device that combines two or more separate input signals in the desired proportion to achieve an output signal of the desired balance.

MONITOR AMPLIFIER – See "Amplifier."

MONOPHONIC (Mono) – Single-channel sound.

MUDDY – An informal expression referring to a lack of articulation or clarity in a sound system.

NEAR FIELD – See "Fields."

NOISE – (1) An unwanted signal, such as a buzz, hiss, or hum, appearing in a system. (2) Controlled random energy used for test purposes.

- **PINK NOISE** – Random noise with equal energy distribution per *octave band,* differing from white noise, which is described as random noise with equal energy per *hertz.* Pink noise is obtained by filtering random white noise. It is a practical test signal for many types of audio equipment and systems, as its level is fairly uniform over the frequency spectrum.
- **WHITE NOISE** – Random noise with equal energy per hertz. White noise is not generally used for test purposes in audio work.

NOMOGRAM – A graph representing the relationship between several variables, shown on a single plane.

NOTCH FILTER – An electronic circuit or device that sharply attenuates one or more very narrow bands of frequencies, generally used to aid in reducing acoustic feedback.

OCTAVE – The interval between any two tones having a frequency ratio of 2:1. In music, eight successive whole notes of the musical scale. For example, A above middle C is 440 hertz; A an octave higher is 880 hertz.

OCTAVE BAND – A band of frequencies, the upper limit of which is double the frequency of the lower limit.

OHMMETER – An instrument to measure the DC resistance of a circuit in ohms.

OHM'S LAW – The group of mathematical formulas that describe the relationship between resistance (R) (or impedance [Z]) in ohms, voltage (E), current (I) in amperes, and power (P) in watts.

OMNIDIRECTIONAL – Displaying equal sensitivity or dispersion in all directions. An omnidirectional microphone is essentially nondirectional.

OSCILLOSCOPE – An instrument that displays the waveform of an audio signal. It is useful for analyzing the complex waveforms produced by various electronic devices for distortion, frequency response, and power-handling capability.

OUT OF PHASE – Not in line with respect to time (when referring to audio signals). Also used to describe the condition of circuits that are reversed in polarity with respect to one another. Electrical or acoustic signals that are 180 degrees out of phase will cancel each other when combined.

OUTPUT – (1) The useful energy delivered by an amplifier or another device. (2) The terminals of such a device where this energy is available.

OUTPUT IMPEDANCE – See "Impedance."

OVERDRIVE LEVEL – The level at which the input signal exceeds the capability of a device to handle it. This is the level at which input clipping may occur.

OVERLOAD – The condition marked by distortion or clipping that occurs when more power is demanded from a device than it is capable of delivering; often caused by connecting a load impedance of lower value than an amplifier or another device is designed to handle. Sustained overload may result in component damage.

OVERTONES – Components of a complex sound having a pitch higher than that of the fundamental frequency. Variations in overtones distinguish the timbre, or quality, of one musical instrument from another.

PAD – A resistive network for reducing the amplitude of a signal without altering its character or introducing distortion. Pads are often used to achieve proper termination as well as proper impedance matching between units.

PARALLEL – An arrangement in which each element of a circuit is connected to the corresponding terminals of all other elements of the circuit. Thus, a given voltage is applied across all elements simultaneously. To be in phase, loudspeakers connected in parallel would have their positive terminals connected to one conductor of the feed line and negative terminals to the other. In a parallel circuit, current flows through two or more paths.

PASSBAND – The range of frequencies that can pass through a filter. The edges of a passband are usually defined by the points where frequency response drops 3 decibels below the level at the middle of the passband.

PASSIVE DEVICE – An audio component that does not amplify energy but that may control it in some way. Such devices consist of resistors, capacitors, inductors, and so forth, as opposed to active components that amplify and that require a power source other than the signal itself.

PASSIVE EQUALIZER (or Filter) – See "Equalizer."

PEAK POWER – Maximum instantaneous output capability. Generally, an amplifier is capable of delivering more power during a brief peak than it can on a continuous basis.

PEAK-TO-PEAK – When referring to an audio or AC waveform, the overall difference between the maximum positive and the maximum negative voltage swing. The peak-to-peak value will always be higher than the average, or root-mean-square (rms), value of a waveform.

PHASE – The relationship of two signals with respect to time. For example, if two sources produce similar signals that are always of the same polarity at the same instant, the sources are said to be in phase.

PHASE CANCELLATION – The reduction of a signal or a sound wave resulting from the combining of two electrical or acoustic signals of opposite polarity, i.e., 180 degrees out of phase.

PHASE REVERSAL – A 180-degree change in phase resulting, for example, from reversing the connections on a loudspeaker, a microphone, or another balanced line.

PICO – One-trillionth (10^{-12}) times what follows, as in picofarad (one trillionth of a farad).

PINK NOISE – See "Noise."

PITCH – The perceived frequency of a sound. Relative pitch is subjective, depending on level as well as actual frequency.

POINT-SOURCE CLUSTER – A single grouping of loudspeakers designed to cover a listening area from a single location.

POLARITY – The phase relationship of signal leads in an audio system. (See “Phase Cancellation.”) Also refers to the positive (+) and negative (–) terminals of a DC power supply or battery.

POLAR PATTERN – A graph showing the angular distribution of energy radiated by a sound source on a given plane.

POP FILTER – See “Windscreen.”

POWER – Electrical energy that accomplishes some kind of work, such as moving a loudspeaker cone, lighting a bulb, heating a resistor, or moving a meter. Measured in watts.

POWER AMPLIFIER – See “Amplifier.”

POWER BANDWIDTH – The frequency range over which an amplifier can produce its rated power output with less than a specified amount of distortion.

POWER-HANDLING CAPABILITY – The maximum power that a loudspeaker or other component can safely handle without damage or improper operation. Power-handling capacity of most components will vary according to frequency and length of time the signal is applied.

POWER TRANSFORMER – See “Transformer.”

PREAMP – An amplifier before the power amplifier used to raise the output of a low-level source, such as a microphone, to a level convenient for further processing or for driving a power amplifier.

PRIMARY – In transformers, the winding(s) to which power or a signal is applied.

PROXIMITY EFFECT – The phenomenon, noted in many cardioid microphones, that emphasizes low frequencies when the microphone is operated very close to a sound source. Some cardioid microphones are specially designed to minimize this effect.

Q – (1) A measure of the directivity of a loudspeaker. (See also “Directivity Factor.”) (2) A description of the slope or sharpness of a filter, or equalizer.

RACK – A vertical frame on which electronic equipment can be mounted, usually designed to accommodate standard 19-inch (48 cm) panels or mounting brackets.

RADIATION – The transmission of energy through space or through a material.

RADIO FREQUENCY (RF) – Any frequency used in the radiation of radio signals.

RADIO FREQUENCY INTERFERENCE (RFI) – The phenomenon that takes place when radio frequency energy, either noise or transmitted signals, is detected in an audio system, creating audible noise, buzz, or intelligible interference.

RATED POWER – The power-handling capability or power-output capacity of an amplifier or another device, as specified by the manufacturer.

REAL-TIME ANALYZER – An instrument for providing a graphic visual display of the total energy present over the entire frequency spectrum on a real-time, or instantaneous, basis, thus permitting all portions of the audio spectrum to be monitored simultaneously.

RE-ENTRANT HORN – A horn-type loudspeaker in which the sound path is folded back on itself coaxially to reduce its physical length while retaining a long, effective acoustic length.

REFLECTIONS – The effect of a sound wave bouncing off one or more surfaces at equal angles of incidence and in the same plane.

REFRACTION – The bending of an acoustic wave as it changes velocity when traveling from one medium into another or through a nonuniform medium.

RELEASE TIME – The time required for an automatic gain-control device, such as a compressor or limiter, to recover from changes in signal level.

RESISTANCE – The opposition to flow of electrical current. Measured in ohms.

RESISTOR – A circuit component made of material having an electrical resistance of a specified value. Rated in ohms and power dissipation capability (watts).

RESONANCE – The tendency of an electrical or mechanical system to vibrate, or oscillate, at a certain frequency.

RESPONSE – See "Frequency Response."

REVERBERANT FIELD – See "Fields."

REVERBERATION – A fusion of sound images caused by multiple reflections from walls, floor, and ceiling. Can be created artificially by electronic or mechanical devices. Not to be confused with an echo.

REVERBERATION TIME (RT_{60}) – The time in seconds that it takes for the field of reverberant acoustic energy in a room to decrease 60 decibels, or to one millionth of its original intensity, after the sound source is turned off. RT_{60} is directly proportional to the volume of the room and inversely proportional to the total absorption in the room.

RF CHOKE – A coil designed to impede the flow of radio frequency current while presenting comparatively little resistance to audio frequencies and direct current.

RIDE GAIN – To monitor continuously and adjust the audio gain of a sound system manually in order to maintain a consistent or desired program level.

rms (Root-Mean-Square) – The power-effective value of an alternating voltage or current, resulting in the same power dissipation in a given load resistance as a DC voltage or current of the same value.

ROLL-OFF – A gradual attenuation, or loss, beyond a given frequency.

ROOM CONSTANT – A term used to indicate the overall absorptive qualities of a room.

ROOM GAIN – The amount of sound energy conserved by reflection from the boundary surfaces in a room compared to losses due to inverse square law in free space.

SABIN – A unit of acoustic absorption; equal to one square foot (or one square meter for a metric sabin) of totally absorptive material. —See also “Absorption Unit.”

SATURATION – A condition where further increase in signal level cannot be accommodated without unacceptable distortion. In transformers, the level at which the core approaches complete magnetization and begins to create severe distortion, usually occurring first at the lowest frequencies.

SECONDARY – In transformers, the winding(s) from which power or a signal is taken.

SENSITIVITY – (1) The minimum input signal required by a device to produce the specified output at a specified distortion figure. (2) The output of a microphone, usually specified in dBV, when it is exposed to a specified sound level (usually 74 or 94 dB-SPL). (3) The sound pressure level (dB-SPL) produced by a loudspeaker at a specified distance, usually 3.28 feet (1 m), when a one-watt input signal is applied to it.

SERIES – A circuit where the elements are connected end to end so that there is only one path in which current flows. In order to maintain proper phase relationship, loudspeakers connected in series must have the negative (–) terminal of the first loudspeaker connected to the positive (+) terminal of the next, and so on.

SHIELD – A metallic device or covering designed to protect a circuit, transmission line, or transformer from stray voltages or currents induced by electric and/or magnetic fields. At audio frequencies, electrostatic shielding from RF-type noise may be provided by a screen, wire braid, foil, or aluminum sheet. Magnetic shielding at audio frequencies is provided by materials of high magnetic permeability such as iron, steel, or mu-metal.

SIBILANCE – Excessive emphasis of the high-frequency elements of speech, such as *s* and *sh* sounds, resulting in a sharp or harsh sound quality.

SIGNAL-TO-NOISE RATIO (S/N) – The difference, usually expressed in decibels, between signal level and the level of unwanted noise. Signal-to-noise rating refers to the ratio between a specified operating level and the inherent noise level of the device in question.

SINE WAVE – A pure, single-frequency tone. As displayed on an oscilloscope, any deviation from the pure sine waveform indicates the presence of some form of distortion.

SINGING – An informal term describing the undesirable ringing sound that occurs when a sound system reaches the point of acoustic feedback.

SNAKE – A cable that runs from the stage of a live performance to the main mixing console. The snake typically contains one shielded pair of wires for each stage microphone.

SOUND COLUMN – See “Line Radiator.”

SOUND-LEVEL METER (SLM) – An instrument for measuring acoustic noise and sound levels, usually calibrated in dB-SPL.

SOUND PRESSURE LEVEL (SPL) – The measurement of acoustic energy, usually expressed in dB-SPL. The level considered to be the threshold of human hearing is referenced to 0 dB-SPL.

SOUND REINFORCEMENT – The acoustic amplification of a live program for a generally stationary audience. Sometimes referred to as “PA” or “public address,” although such terms also include systems serving a mobile public, as in supermarkets, department stores, and air and bus terminals.

SOUND WAVES – A series of variations in pressure propagated in the air or other elastic material.

SPECTRUM – See “Audio Spectrum.”

SQUARE WAVE – An electronically produced signal that, when displayed graphically on an oscilloscope, appears in the shape of a square or a rectangle. It consists of a fundamental frequency plus harmonics, making it well suited for analyzing the frequency response of amplifiers and other audio devices.

SUM – To calculate the sum of; total.

SWG – Abbreviation for "standard wire gauge," an industry standard for specifying wire diameter. The greater the SWG number, the smaller the wire diameter.

SYSTEM GROUND – The main ground point for a large sound-reinforcement system, used in lieu of several ground points in order to prevent ground loops and possible resultant interference.—See also "Ground Loop."

TAP – A connection to an intermediate point of a coil, or transformer winding, usually for selecting a ratio of transformation.—See also "Autotransformer."

TERMINATION – A load, usually resistive, connected to a line or a device in order to match its circuit impedance properly.

THRESHOLD – The point, or level, at which some action begins to occur, such as the threshold of feedback.

TIMBRE – The characteristic tonal quality of a voice or a musical instrument that allows it to be distinguished from other voices or instruments, resulting from variations in the number and amplitude of harmonics or overtones produced by the voice or instrument.

TONE CONTROL – A simple form of equalizer for altering the frequency response of an amplifier or a system. Often referred to as bass or treble control, it usually affects a very broad portion of the frequency spectrum.

TOTAL HARMONIC DISTORTION (THD) – The total percentage of undesirable harmonic content added to the overall output signal by imperfections in an amplifier or another device.

TRANSDUCER – A device for converting one form of energy into another form. For example, a microphone converts acoustic energy into electrical energy, or a loudspeaker converts electrical energy into acoustic energy.

TRANSFORMER – An electrical device that converts variations of current in a primary circuit into variations of voltage and current in a secondary circuit by means of electromagnetic induction. Transformers operate only with alternating current and are not intended to alter the frequency response of a circuit. Transformers are frequently used for impedance matching, as well as to provide an increase or a decrease in audio and other AC voltages. Most types of transformers are designed for specific purposes.

- **BRIDGING TRANSFORMER** – A transformer used to obtain a signal from a circuit without significantly loading it, having a primary impedance of 10,000 ohms or higher.—See also "Bridging."
- **ISOLATION TRANSFORMER** – A transformer designed to operate at line level, usually at a 1:1 impedance ratio with

negligible insertion loss. Used to establish a "floating" (balanced) line, thus isolating grounds to prevent unwanted noise due to ground loops.

- **LINE INPUT OR OUTPUT TRANSFORMER** – A transformer that usually operates at line level but that often provides a 1:2 or 1:4 voltage step-up ratio. Also provides DC and ground isolation.
- **MICROPHONE INPUT TRANSFORMER** – A transformer that provides impedance matching and radio frequency rejection for microphone levels. These transformers are sometimes built into the input section of high-quality mixers.
- **POWER TRANSFORMER** – A transformer used in the power supply of electronic equipment to convert AC line voltage to a more suitable voltage for electronic circuitry. The adjusted voltage is then rectified and filtered to provide the appropriate DC voltage required.

TRANSISTOR – An active semiconductor device capable of performing amplification and rectification. Because transistors control current without having moving parts, heated filaments, or vacuum gaps, they are referred to as solid-state devices.

TREBLE – The higher audio frequency range. In music, the frequencies above middle C (261.63 Hz), but in sound reinforcement, those from about 3,000 hertz upward.

ULTRASONIC – Having a frequency above the range of normal human hearing, or greater than 20 kilohertz.

UNBALANCED LINE – A line in which one of the two circuit conductors is connected to a ground. This grounded conductor often serves as a shield.

UNIDIRECTIONAL MICROPHONE – See "Microphone."

UNITY GAIN – When the output level of an amplifier or active circuit is equal to the input level, the result is unity gain. This is a typical condition of signal-processing devices, such as equalizers or compressors, operating at line level.—See also "Gain."

VOICE COIL – The coil of wire attached to the diaphragm of a loudspeaker or driver unit. When an audio signal is applied, the coil functions as an electromagnet that interacts with the permanent magnetic field already in the speaker's frame to produce acoustic vibrations of the cone or diaphragm.

VOLT – The basic unit of electromotive force, commonly abbreviated EMF, in all electrical and electronic circuits.

VOLTAGE AMPLIFIER – See "Amplifier."

VOLUME – A loosely used term describing the level of an audio signal or the intensity of a sound.

VU METER – A meter designed to measure audio level in volume units, having suitable ballistics to provide good indication of the average level of rapidly fluctuating signals, as in speech or music. VU meters are generally calibrated so that “0 VU” equals +4 dBm, but they may also be switchable to other standards.

WATT (W) – A unit of measure for electrical or acoustic power.

WAVELENGTH, ACOUSTIC – The distance a sound wave travels to complete one cycle. Wavelength equals the velocity of sound divided by the frequency.

WEIGHTED SCALES – The scales on a sound-level meter—designated “A,” “B,” and “C”—that correspond approximately to the frequency response of the human ear at different sound levels.

- The A scale is less sensitive to low-frequency sound and approximates the human ear at low sound levels.
- The B scale is more nearly linear, corresponding to moderate levels. (The B scale is no longer in widely accepted use.)
- The C scale is essentially linear, representing the approximate response of the human ear at high sound levels.

WHITE NOISE – See “Noise.”

WINDINGS – The coils of wire on a transformer or an inductor.

WINDSCREEN – A microphone cover of open-celled foam rubber or polyurethane that minimizes breath and wind noise. Also called a pop filter.

XL-TYPE CONNECTOR – Any of several varieties of three-pin latching-type audio connectors commonly used for microphone and line-level connections in professional sound systems. Sometimes called Cannon connectors after the original manufacturer.

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