Electronic Sound Sytems Design

Equipment, Application, Specification, and Installation

Neil Thompson Shade

1997 Theodore John Schultz Grant for Advancement of Teaching and Research in Architectural Acoustics

The Robert Bradford Newman Student Award Fund
Lincoln, Massachusetts

June 2001

"Sound is what happens when air gets pushed."

— Richard C. Heyser in 1982

"When we include a piece of the universe with brick and mortar and wooden walls and call it an auditorium, that is when we architects get into all sorts of trouble."

— Architect Rudoolph Markgraf in 1911

"It is not my fault that acoustics and I can never come to an understanding. I gave myself great pains to master this bizarre science, but after fifteen years' labor, I found myself hardly in advance of where I stood on the first day...I had read diligently in my books, and conferred industriously with philosophers - nowhere did I find a positive rule of action to guide me; on the contrary nothing but contradictory statements."

—Architect Charles Garnier in 1880

Chapter

1

Introduction and Overview

"In order that the amplified and reproduced sound give an illusion of reality, the amplifying equipment must be free from distortion; it must be free from distracting or annoying noises; and it must reproduce the sound at a loudness level which is adequate for distinct and comfortable hearing—and at a level which is comparable with the loudness of sound which we customarily hear. If the equipment possess these specified characteristics, the sound which is impressed upon the microphone will be reproduced by the loud speaker with a fidelity that will make it indistinguishable from the original sound. Unfortunately, the existing commercial equipment does not adequately possess these characteristics."

Knudsen, Chapter 14, *Architectural Acoustics*, John Wiley & Sons, New York, NY (1932).

Copyright Information

First Edition
First Printing - 2001

The author and The Robert Bradford Newman Student Award Fund assume no responsibility for errors, omissions, or damage resulting from the use of the information contained herein. This book is published with the understanding that the author and The Robert Bradford Newman Student Award Fund are supplying information but are not attempting to render professional engineering services to the reader. If the reader is uncertain of his or her capabilities, the assistance of a professional acoustician should be retained. Information on appropriate professionals can be obtained through the National Council of Acoustical Consultants, 66 Morris Avenue, Suite 1A, Springfield, NJ 07081 (Tel: 973.564.5859; web: www.ncac.com).

Terms and products that are known to be trademarks or service marks are noted in the text. All trademarks and service marks are property of their respective owners. The author and The Robert Bradford Newman Student Award Fund can not be responsible to the accuracy of credited trademarks or service marks and the omission of uncredited trademarks or service marks.

Acknowledgments

The preparation of this book has involved many individuals for whom the author is truly appreciative. Recognition is given to those affiliated with The Robert Bradford Newman Student Award Fund, other contributors rendering services beyond the author's capabilities, colleagues and manufacturers providing photographs and equipment samples, and special individuals whose encouragement and support during the numerous writing periods were always a welcome inspiration.

Partial financial support for this book was provided by the 1997 Theodore John Schultz Grant for Advancement of Teaching and Research in Architectural Acoustics, sponsored by The Robert Bradford Newman Student Award Fund, Lincoln, MA. The author is grateful for being selected and acknowledges funding to prepare this book. Particular appreciation is extended to Mary Shaw Newman, Carl Rosenberg, and William Cavanaugh.

Book design is by Betsy Berlin of Miss Pixel Graphics. Layout and desktop publishing were provided by Betsy Berlin of Miss Pixel Graphics and Lorraine Frankfurt of lafproductions. Equipment photographs were taken by Paul Gillis Photography. Other photographs are credited where appropriate. Single line art illustrations were prepared by Robin Alleyne and Sam French, RA. Loudspeaker computer modeling simulation was performed by John Nicholas. Proofreading and overall review were by my students Damien Moody and Stephanie Renaud. Finally, thanks to Bill the UPS guy for delivery and pick-up of the 100 plus manufacturer product sample boxes.

For photographs and equipment samples, the author would like to thank: Michael Akrep (Polar Focus, Inc.); Aphex Systems; ARTEC Consultants, Inc; Atlas/Soundolier, Inc.; John Cardone (Cardone Solomon Associates); Paul Corraine (Convergent Technologies Design Group); Perry D'Angelo (S.K. McDonald, Inc); Marcel Daugherty (OmniMount Systems); Carl Dorwaldt (Renkus-Heinz); Duke Ducoff (White Instruments, Inc.); Roger DuNaier (Performance Devices/Duran Audio); Kathi Evans (Neutrik USA); Mark Fitch (Renkus-Heinz); Mark Gander (JBL Professional); Martin Glasband (Equi=Tech Corporation); Brian Griswald (TOA Electronics, Inc.); Yvonne Ho (Phonic Ear, Inc.); Paul Jernigan (Peavey Electronics, Inc.); John Kowalski (Systems Wireless, Ltd.); John Lowell (Lowell Manufacturing Company); Michael McCook (New Frontier Electronics, Inc.); Jay Mitchell (Frazier Loudspeakers); Roy Pass (Richard Pass Associates, Inc.); Michael Pettersen (Shure Brothers, Inc.); Lynn Potter (LynTec); Janice Sandri (FSR, Inc.); Richard Shaeer (Sound Sales, Inc.); Matt Swysgood (Kinetics Noise Control); Steve Taub (Taub Sales); Rick Waller (Shure Brothers, Inc.); Ewart "Red" Wetherill (Shen, Milsom & Wilke + Paoletti); and Tom Young (Electro-Acoustic Design Services).

The author is indebted to certain individuals for personal and professional inspiration. The following individuals have contributed to my knowledge and professional development: Russell Campbell, Ph.D., of Boise State University whose enthusiasm in teaching physics was a constant inspiration; Jerry P. Christoff, of

Veneklasen Associates who taught me the practice of acoustical consulting; Jose C. Ortega, of Veneklasen Associates who taught me field and laboratory acoustical measurement techniques; Romeo Segnan, Ph.D., of American University, who enabled me to tailor my academic studies to suit my unique interests and to later offer a teaching position in the Audio Technology Department at American University; the late Paul S. Veneklasen, whose hands-on approach through experimentation and prototyping served as a model for problem solving; and Richard V. Waterhouse, Ph.D., for teaching me basic architectural acoustics. Finally, Victoria Vestrich, my wife, and my two children Nicolette and Nathan, deserve special thanks for their steadfast love.

Preface

The demand for sound system performance is growing, no doubt by the public's experience with advanced business, computer, concert, home entertainment, motion picture, and sporting event media systems. Spaces should have sound systems which meet listener and end user expectations, are aesthetically integrated within the architectural design, and achieve a level of technical performance equal to other building systems.

Training in university architecture programs generally excludes sound systems. While sound system design is beyond the scope of an architect's responsibility, the practitioner should understand the fundamentals to accommodate the recommendations by the project acoustician, sound system designer, electrical engineer, sound system contractor, and others.

The idea for this book came about from my frustration with the available books and their suitability as a university-level teaching tool for those with little exposure to sound systems design. Sound system books tend to fall into two categories: (1) those that emphasize engineering electro-acoustics (lots of math and little practical application) or (2) those describing operation techniques (useful for live sound operators but lacking for the design professional). Neither provides concise and relevant information needed to accommodate sound system requirements in the architectural design process. This book is intended to provide architecture faculty, students, and others with a resource that describes the basics of audio equipment and sound systems, design principles and performance objectives, architectural programming requirements, specification criteria, and installation practices to achieve the above stated objectives. Information is presented both as elementary concepts, providing a general overview, and in-depth suitable for detailed engineering design. Architectural faculty and students will find useful concepts to integrate sound systems within building design. Engineering faculty and students will appreciate the detailed functional descriptions of equipment component items and their application to sound systems design.

The author assumes the reader has previous exposure to basic architectural acoustics and a mathematical proficiency at the elementary college level. An understanding of room acoustics is necessary to properly design a sound system. Selected room acoustical concepts relevant to sound systems are covered in the first chapter for those requiring a brief review. Equations throughout the book provide computational solutions to different sound system design fundamentals.

The book topics are arranged in a logical sequence as occurs when designing a sound system. Many topics covered in this book are not addressed in other books, including: (1) assistive listening systems; (2) the six major sound system types; (3) design team planning responsibilities; (4) system documentation; and (5) architectural, electrical, and mechanical system infrastructure. The book material is too long for use in less than a three credit hour semester course. Instructors may wish to use portions of this book based on their particular course emphasis: (1) sound systems overview and architectural installation factors (Chapters 1, 2, and 7); (2) design team planning and system specification (Chapters 1, 2, and 6); and (3) detailed design, engineering, and testing at the equipment component and system levels (Chapters 3, 4, and 5).

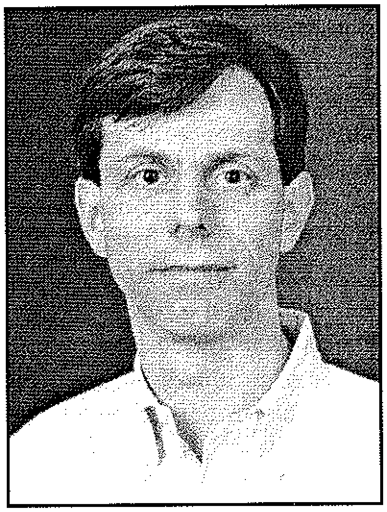
Chapter 1 introduces the functions of sound systems, basic room acoustical considerations, design responsibilities, and performance criteria. Chapter 2 covers basic sound system types and applications (voice reinforcement, music reinforcement, paging, communications, emergency announcement, language interpretation, sound masking, teleconferencing, assistive listening, audio for video, cinema, and room acoustical enhancement systems). Chapter 3 describes sound system equipment including microphones, line level sources, mixing and signal processing devices, computer and digital signal processing, power amplifiers, loudspeakers, assistive listening devices, and production intercoms. Chapter 4 addresses sound system design and installation practices both at the system level (calculations of gain, sound level coverage, and speech intelligibility), and at the equipment component level (installation of microphones, system electronics, loudspeakers, and equipment racks), concluding with system configurations to avoid. Chapter 5 examines requirements for testing and adjusting installed sound systems. Chapter 6 reviews sound system documentation for the design, bidding, and installation phases. Chapter 7 provides information on sound system support spaces (equipment rooms, control rooms, and mixing positions), equipment installation (loudspeaker and microphone locations and rigging), electrical systems (power loads, conduit, grounding, cables and connectors, and codes), and HVAC systems (equipment heat loads). Additional technical topics are included at the end of each chapter to provide a more detailed understanding of selected concepts covered in the text. Appendix A provides a survey checklist to identify different sound systems in buildings. Appendix B lists references and other sources of technical information on sound systems design. Appendix C is a glossary of the italicized terms used in the text.

The author would appreciate receiving comments on the book, including topics to cover in future editions, and being notified of any errors, inconsistencies, and problems which the reader has encountered. Every attempt has been made to check the technical validity and typography of the book contents, however some errors may have occurred.

Neil Thompson Shade
Acoustical Design Collaborative, Ltd
Falls Church, VA USA
email: neil@akustx.com

June 2001

About the Author



Neil Thompson Shade is an acoustician with 20 years experience in consulting, project management, software development, and teaching. He has served as a Principal Consultant on over 800 projects for most building types including 45 auditoria, 35 studios, and 50 worship houses.

Neil is President and Principal Consultant of Acoustical Design Collaborative, Ltd, an acoustical consulting practice emphasizing the design of critical listening spaces (auditoria, broadcast/recording studios, court rooms, drama theaters, lecture halls, music rehearsal rooms, and worship houses). Prior to forming Acoustical Design Collaborative, Ltd in 1991, Neil was a Project Manager for Wyle Laboratories in Arlington, VA. After graduating from college he joined Veneklasen Associates/Western Electro-Acoustics Laboratory in Santa Monica, CA as a staff acoustician. While attending college, Neil was employed full-time in the naval architecture industry as a draftsman and mechanical systems designer.

Neil received a B.S. degree cum laude in Audio Technology/Acoustics from American University in Washington, DC. His principal teachers were Scott Parker, Ph.D., Romeo Segnan, Ph.D., and Richard V. Waterhouse, Ph.D. Subsequent education has included professional design seminars sponsored by Concert Hall Research Group (Tanglewood '99 Institute), Harvard University Graduate School of Design (Theater and Auditorium Design), Syn-Aud-Con (Sound Systems Design), University of California (HVAC Systems Design), and University of Wisconsin (Vibration in Microelectronics Facilities), and numerous manufacturer-sponsored training seminars.

Neil is currently coordinator of the M.S. degree program in Acoustics at Johns Hopkins University/Peabody Conservatory of Music in Baltimore, MD. He teaches courses in Acoustical Measurements, Architectural Acoustics, Computer Modeling, Electro-Acoustics, Musical Acoustics, Noise Control, Psychoacoustics, Physical Acoustics, and Sound Systems Design. Between 1988 and 2000 he was an Adjunct Faculty Member at American University where he taught classes in Basic Architectural Acoustics, Advanced Architectural Acoustics, Audio Technology, and Sound Systems Design. Between 1987 and 1991 Neil was a Guest Lecturer at the School of Architecture at the University of Maryland, College Park, MD. Since 1993 he has been a Guest Lecturer for the Washington, D.C., Baltimore, and Hampton Roads AIA Chapters where he conducts review sessions to architects preparing for their architectural license examinations. Annually, since 1990, Neil conducts a design course in studio acoustics for audio engineers in developing countries under the auspices of the Voice of America (VOA).

Neil has presented and written 15 papers and reviewed three books for architectural and acoustics publications. He is the author of the room acoustics design program "Sabine".

Neil's professional affiliations include the Acoustical Society of America (ASA), Audio Engineering Society (AES), Institute of Noise Control Engineers (INCE), National Council of Acoustical Consultants (NCAC), and National Systems Contractors Association (NSCA).

When he is not involved with acoustics-related activities, Neil enjoys raising his two children, attending classical music concerts, playing viola, and collecting recorded music.

1

- When Sound Systems are Required
- Room Size
- Room Acoustical Properties
- Ambient Noise Levels
- Program, Artistic, or Signal Transmission Requirements
- Planning and Design Considerations
- General Performance Objectives
- Chapter Summary
- Technical Notes

When the eminent acoustician Vernon Knudsen wrote the introductory statement, the use of sound systems in public assembly spaces was just beginning. Twenty years earlier Lee de Forest patented the three-element "Audion" vacuum tube, making electronic sound amplification possible, the public was first exposed to sound amplification during the 1920 presidential conventions, and in 1927 the first "talkie" motion picture (Al Jolson's *The Jazz Singer)* appeared. For its time, the technological achievement of electronic sound systems, culminating in amplified dialog and sound effects for motion pictures, has been compared equal to the efforts of putting man on the moon some 40 years later. Basic research between 1912 and 1932 led to the development of the fundamental sound system building blocks (*cardioid microphone*, coaxial loudspeaker, compression driver, *condenser microphone*, cone loudspeaker, electronic crossover, feedback amplifier, folded horn loudspeaker, multi-cell horn loudspeaker, and *stereophonic* sound) which are still in use today, albeit with refined technologies.

This chapter will provide the reader with background information on sound systems including use and application, basic room acoustics, planning and design considerations, and general performance objectives.

1.1 When Sound Systems Are Required

A sound system is necessary when unamplified sound levels or acoustical conditions are not adequate to provide individuals with good listening conditions. Sound systems are recommended in all public assembly spaces such as amphitheaters, arenas, auditoria, conference rooms, courtrooms, houses of worship, lecture halls, legislative chambers, and stadia, particularly if these facilities are to be used by inexperienced talkers.

Factors that may determine the need for a sound system are described below.

1. **Room Size:** The source sound level at the listener (receiver) will vary as a function of the room size and the distance between the source and receiver. Unamplified sound levels may not be sufficiently loud at all locations in the room if one or more of the following conditions exist: (1) the audience capacity exceeds 350; (2) the room volume is larger than 50,000 ft3; (3) the distance between source and farthest receiver exceeds 50 ft (indoors) or 25 ft (outdoors); or (4) the receiver is located behind or beyond 70° horizontally from the source.

2. **Room Acoustical Properties:** The room acoustical properties may not be appropriate for all program types that occur in the space. The room *reverberation time* may be too long (exceeding 1.6 s), *early reflections* from room surfaces may not be strong enough to increase unamplified source sound levels, sound reflections from remote room surfaces may result in echoes at the receiver location, or the *reverberant sound level* may be too high.

3. **Ambient Noise Levels:** The ambient noise levels may be excessive, greater than *Noise Criterion* (*NC*) -40 indoors and 55 *deciBels A-weighted* [dB(A)] outdoors, making listening conditions difficult due to *sound masking* effects.

4. **Program, Artistic, or Signal Transmission Requirements:** Specific needs for playback of prerecorded media, *archival recording*, special performance or artistic sound effects, transmission of signals to remote locations, and amplification for hearing-challenged individuals may be necessary.

1.2 Room Size

The room size has a direct correlation on the source sound level at the receiver's location. Sound levels will be less in large rooms or as the distance between the source and receiver increases. The suggested limits, before using sound reinforcement, of an audience capacity greater than 350 and a room volume in excess of 50,000 ft^3 covers most public assembly spaces. In these spaces the audience and the seats provide the majority of the room sound *absorption*. Thus, assuming the room volume remains constant, larger seating areas will have greater sound absorption, which will result in lower reverberant sound levels in the space. Typically, for each doubling of the audience seating area, the reverberant sound level will decrease approximately 3 dB(A). To make up for the loss of reverberant sound level requires doubling the performance ensemble size or doubling the number of amplified loudspeakers in the room.

One way to estimate the amount of sound absorptive material in a space is to use the *room constant* (***R***) which gives the effective absorption in terms of an equivalent area of totally absorptive material. The value of **R** can be calculated using the following equation:

$$\mathbf{R} = \frac{S\bar{\alpha}}{1-\bar{\alpha}} \tag{1.1}$$

where,

R is the room constant, ft^2

S is the room surface area, ft^2

$\bar{\alpha}$ is the average absorption coefficient of finish materials, dimensionless

(See Technical Notes, Section 1.A, at the end of this chapter, for additional information on the room constant.)

As the distance between the source and receiver increases, the *direct sound* level decreases. In its most simple form, a sound source can be approximated as a *point source* which radiates spherically from its origin. The direct sound level decreases 6 dB per doubling of distance from the source (*inverse-square loss*).

1.3 Room Acoustical Properties

Requirements for a sound system are directly related to the acoustical properties of the space. A sound system is an information transferal mechanism and is not a substitute for good room acoustical design. The quality of signal transmission between the source and listener can be significantly improved by a sound system only when it is matched to the room acoustical properties.

Room reverberation is the prolongation of sound, due to a multiplicity of reflections from the room surfaces. It can be viewed in terms of duration (time) or magnitude (level). The reverberation time ($\mathbf{T_{60}}$) is the time in s for the sound to decay 60 dB from its original intensity once the sound source has stopped radiating. A variant, the *early decay time* (**EDT**), is the decay over the first 10 dB extrapolated to 60 dB and provides a more accurate assessment of the perceived reverberation in a space. The premise with **EDT** is the listener perceives only the first part of the reverberant decay and the latter reverberant decay is masked by the "running" program material. The magnitude of the reverberant level expressed in dB quantifies the reverberant sound "loudness." Larger rooms, or ones with little acoustical absorption, will have a longer reverberation time and a greater reverberant sound level than smaller or more acoustically absorptive rooms.

Acoustical requirements for speech and music are in opposition, with music requiring greater reverberation than speech, both in duration and magnitude. Reverberation degrades *speech intelligibility* but enhances the blending of musical sounds. Table

1-1 provides design T_{60} target values for different space types based on the average of the 500 and 1,000 hertz (Hz) octave frequency band T_{60} values.

TABLE 1-1. Reverberation Times for Indoor Spaces

Space Type	Average T_{60}	T_{60} Range (Low - High)
Churches (Contemporary Worship)	1.3	0.9 - 1.6
Churches (Liturgical Worship)	3.0	2.0 - 4.0
Cinemas	0.9	0.5 - 1.3
College Classrooms	0.7	0.4 - 1.0
Conference Rooms	0.7	0.4 - 0.9
Courtrooms and Civic Facilities	0.8	0.6 - 1.1
Drama Theaters	1.0	0.8 - 1.2
Elementary School Auditoria	1.1	0.9 - 1.4
High School Auditoria	1.6	1.3 - 1.9
Hotel Ballrooms	1.3	1.0 - 1.6
Lecture Halls	0.9	0.7 - 1.2
Multi-Purpose Auditoria	1.7	1.4 - 1.9
Opera Houses	1.6	1.3 - 1.8
Recital Halls	1.6	1.4 - 1.7
Sports Arenas and Gymnasia	2.5	2.0 - 3.0
Symphony Halls	2.0	1.8 - 2.2

The reverberation time can be calculated knowing the room volume, the surface areas, and the room finish material sound absorption coefficients. The equation which is used most frequently and has application to a wide variety of rooms is the Sabine equation, first developed by Wallace Clement Sabine in 1898. The value of T_{60} using Sabine's procedure can be calculated using the following equation:

$$T_{60} = \frac{0.049V}{A} \tag{1.2}$$

where,

T_{60} is the reverberation time, s

V is the room volume, ft^3

A is the sound absorption of room finish materials, sabins

The value of **A** can be calculated using the following equation:

$$\mathbf{A} = \Sigma \mathbf{S}_i \alpha_i \tag{1.3}$$

where,

A	is as above
$\mathbf{S}_i$	is the surface area of each individual room finish material, ft^2
α_i	is the sound absorption coefficient of each individual room finish material, dimensionless

Normally, $\mathbf{T}_{60}$ and **A** are calculated in six octave frequency bands (125, 250, 500, 1,000, 2,000, and 4,000 Hz).

(See Technical Notes, Section 1.B, at the end of this chapter, for additional information on reverberation time equations.)

The rate of normal conversational speech articulation is three to five syllables per second. Thus, it can be seen that room reverberation causes overlapping of speech syllables as words do not fully decay to inaudibility before successive words are spoken. This overlapping of speech syllables degrades speech intelligibility. Fortunately, there are sufficient contextual cues in sentence structure enabling speech to be reasonably understood in spaces having a reverberation time up to approximately 1.6 s. Individuals with age-related high-frequency hearing loss (*presbycusis*), children less than 12 years old, or individuals who are not fluent in the country's native language require a lower reverberation time, on the order of less than 0.6 s, to realize good speech intelligibility.

Sound reflections from room surfaces which arrive at the listener's location can interfere with the direct sound from the source resulting in frequency selective cancellation, image shift, or perceived echoes. Relative to the direct sound, high amplitude secondary reflections of the sound within 2 dB, or time delayed greater than 35 ms, will impair speech intelligibility. Figure 1-1 provides general guidelines on the subjective audibility of reflected sound and its implications to speech intelligibility.

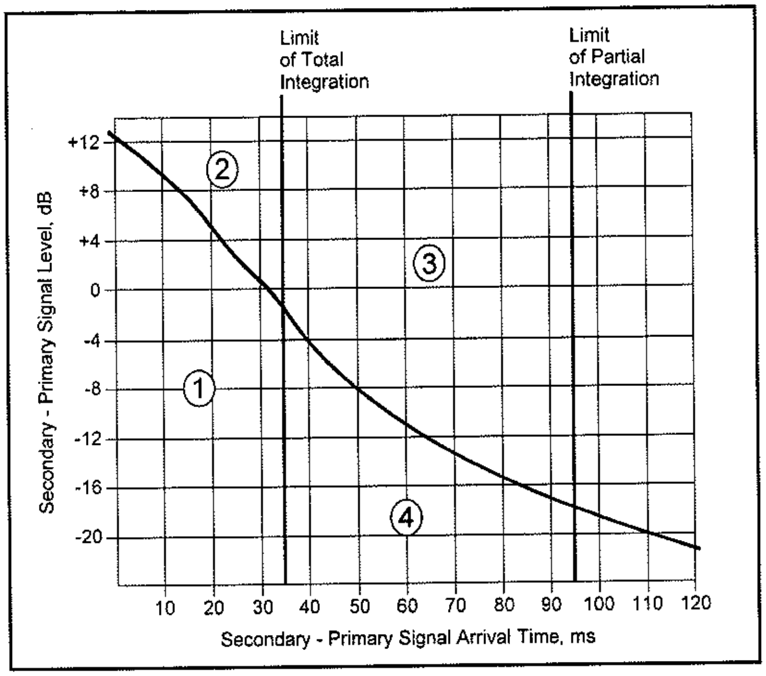


FIGURE 1-1. Integration of sound and speech intelligibility as a function of the delay and level of the secondary source compared to the primary source. Zone 1: full integration, intelligibility increases; Zone 2: full integration, intelligibility increases but image shifts; Zone 3: delayed sound heard as echoes, intelligibility impaired; and Zone 4: delayed reflected sound adds to direct sound, intelligibility increases. Data after Mapp.

Indoors, the listener receives a combination of direct sound from the source and reflected (reverberant) sound from the room surfaces. The direct sound level decreases with distance as outdoors, however beyond the *critical distance* (**D_C**) of the room, the reverberant sound dominates. At **D_C** the direct and reverberant sound are equal in magnitude. Listeners within **D_C**, close to the sound source, hear more direct than reverberant sound. When remote from the sound source, listeners are beyond **D_C**, and hear more reverberant than direct sound. The value of **D_C** can be calculated using the following equation:

$$\mathbf{D_C = 0.14\sqrt{QR}} \tag{1.4}$$

where,

D_C is the critical distance, ft

Q is the source *directivity factor* at a given frequency, dimensionless

R is as above

The direct sound level should be greater than the reverberant sound level, i.e., the *direct-to-reverberant* (**D/R**) ratio must be high to achieve good speech intelligibility. The directional properties of actual sound sources are more complex than the simplified point source model. One primary objective in sound system design is to select loudspeakers with directional properties that radiate sound only to the audience, while minimizing radiation to the room surfaces, thus controlling the reverberant sound level. Manufacturers have developed a wide variety of loudspeaker types with different directional properties to achieve this goal.

1.4 Ambient Noise Levels

Ambient noise at the receiver will affect the perceived sound source loudness. The indoor ambient noise is often controlled by the HVAC system, although audience noise or sound transmitted into the room from other locations, may also be present. Outdoors, the primary noise sources are aircraft, vehicles, wind, and insects.

The source sound level and the ambient noise level can be considered a *signal-to-noise* **(S/N)** ratio relationship where the signal (source program) competes with ambient noise at the listener's location. Depending on the frequency content of the noise, sound masking of speech consonants can result, making program comprehension difficult. Clearly, increasing the source sound level or decreasing the ambient noise level will improve the **S/N** ratio, thus raising speech intelligibility.

Indoor noise is most commonly evaluated using the NC level metric which provides a single composite number rating of noise in the octave frequency bands between 63 and 8,000 Hz. The space NC level value is compared to accepted guidelines for various space types. Noise levels which exceed NC-40 are judged to be moderately loud and a sound system is often required to provide an adequate signal level to the listeners. Table 1-2 summarizes recommended maximum ambient noise levels for different space types exclusive of audience and activity noise.

TABLE 1-2. Maximum Ambient Noise Levels (NC) for Indoor Spaces

Space Type	NC Range
Churches (Contemporary Worship)	25 - 30
Churches (Liturgical Worship)	20 - 25
Cinemas	30 - 35
College Classrooms	30 - 35
Conference Rooms	25 - 30
Courtrooms and Civic Facilities	25 - 30
Drama Theaters	20 - 25
Elementary and High School Auditoria	25 - 30
Hotel Ballrooms	35 - 40
Lecture Halls, no sound amplification	25 - 30
Lecture Halls, sound amplification	30 - 35
Multi-Purpose Auditoria	20 - 25
Opera Houses	15 - 20
Recital Halls	20 - 25
Sports Arenas and Gymnasia	45 - 50
Symphony Halls	< 15

Outdoor noise levels are quantified using *A-weighted* sound levels [dB(A)]. Typical long-term outdoor noise levels range between 45 and 70 dB(A) for suburban and urban locations not directly adjacent to major noise sources such as transportation or industrial facilities. Table 1-3 summarizes acceptable maximum ambient noise levels for different outdoor facilities exclusive of audience and activity noise.

TABLE 1-3. Maximum Ambient Noise Levels [dB(A)] for Outdoor Facilities

Space Type	dB(A) Range
Amphitheaters	< 55
Courtyards	55 - 65
Race Tracks	70 - 80
Stadia	65 - 75
Swimming Pools	60 - 70
Tennis Courts	55 - 65
Theme Parks	60 - 70

The need for a sound reinforcement system can be determined from knowledge of the source sound power level, room size, acoustical properties, distance between source and receiver, and the ambient noise levels. An example is worked out below for both indoor and outdoor conditions.

(See Technical Notes, Section 1.C, at the end of this chapter, for additional information on computing sound levels.)

1.4.1 Indoor Condition

A large college classroom has dimensions of 50 ft long, 50 ft wide, and 20 ft high ($\mathbf{V}$ = 50,000 ft^3) with an audience capacity of 350 (α = 0.80 and $\mathbf{S}$ = 3,000 ft^2), and minimal sound absorption from the other room finish materials (α = 0.20). The ambient noise level is NC-40. The sound power level for an unamplified talker using a "loud" vocal effort is approximately 79 dB(A).

Using equation 1.G, the maximum indoor distance between talker and listener ($\mathbf{r}$ = 50 ft), and the directivity factor of the human voice ($\mathbf{Q}$ = 2) yields a direct sound level of 47.4 dB(A). The direct sound level is only 0.4 dB(A) above the equivalent NC-40 noise level [47 dB(A)].

Using equations 1.3 and 1.I, the above room dimensions, acoustical properties, and vocal effort yields a reverberant sound level of 59.6 dB(A).

The total sound level at the listener due to the direct and reverberant sound levels will be 59.9 dB(A) and is controlled by the reverberant sound level. The total level is 12.9 dB(A) above the maximum ambient noise level. Good design practice would provide a source sound level at the listener a minimum of 25 dB(A) above the ambient noise level. In this example a 72 dB(A) amplified sound level would be required [47+25 dB(A)]. Since the unamplified voice level is 12.1 dB(A) less than the ambient noise plus 25 dB(A) level, the use of a sound reinforcement system is recommended to improve listening conditions.

(See Technical Notes, Section 1.D, at the end of this chapter, for additional information on maximum sound system levels for indoor spaces.)

1.4.2 Outdoor Condition

An outdoor courtyard is to have a small performance stage. The maximum distance between talker and listener is 35 ft. The ambient noise level is 55 dB(A). At the limiting distance of 25 ft, beyond which a sound reinforcement system is necessary, the direct sound level will be 54.6 dB(A) using equation 1.G and a loud vocal effort. Note that outdoors there will be little reflected sound and hence no reverberant sound to increase the total sound level at the receiver. The unamplified sound level at the receiver will be 0.4 dB(A) less than the ambient noise level and 25.4 dB(A) less than the recommended ambient noise plus 25 dB(A) level. Therefore, a sound reinforcement system is recommended to improve listening conditions.

Combining sound levels from multiple sources, such as the direct and reverberant sound levels above, is based on logarithmic, not arithmetic, addition. Table 1-4 can be used to determine the combined sound level from two sources, such as a loudspeaker and talker's direct sound level arriving at a listener.

TABLE 1-4. Addition of Sound Levels

Difference in Sound Levels, dB	Value to Add to Higher Sound Level, dB
0 to 1	3
2 to 3	2
4 to 8	1
> 9	0

When more than two levels are to be combined, the sum of the first two sound levels should be determined using Table 1-4. This value is then combined with the third sound level to arrive at a new combined sound level. The technique is repeated for each remaining sound level to arrive at the final sound level.

1.5 Program, Artistic, or Signal Transmission Requirements

Sound systems are often needed to satisfy special program, artistic, or signal transmission requirements which the listeners may not be aware of.

Playing back and recording programs are often required in educational, governmental, public assembly, music, and theatrical events. Prerecorded sound effects are often played back as part of dramatic and musical productions. Recording of proceedings for archival purposes is common in civic and legislative sessions.

Sound systems, in particular sound reinforcement, may not be desired for certain program types, due to the nature of the performance, artistic and audience expectations, and other factors. Solo instrumental and vocal recitals, chamber and orchestral music, choir, and small scale drama normally do not use sound reinforcement when performed indoors, but may make use of recording, special sound effects, and assistive listening systems. When performed outdoors, these programs almost always use sound reinforcement due to the lack of acoustically reflecting surfaces, higher ambient noise levels, and often larger audience seating areas.

Transmission of proceedings via press feeds, remote broadcast, satellites, and teleconferencing are common as part of civic, educational, and sports events. These audio signals can be distributed to other remote sound systems for real time playback or recorded for archival purposes. Individuals with impaired hearing can use a fixed or portable *assistive listening system* (ALS) to improve their localized listening conditions to satisfy the requirements of the Americans with Disabilities Act (ADA).

1.6 Planning and Design Considerations

Sound systems require planning to meet user requirements, integrate the equipment in the room, coordinate details with other design professionals, and review the contractor's installation. Individuals and their interaction in the design of sound systems are summarized below.

1. **User:** Will have responsibility to describe the desired sound system functional and performance features, requirements for operational control, needed tie-ins with other building systems, available architectural and electrical provisions, restrictions for incorporating the sound system into the space, financial budget for system design and procurement, and future system expansion plans.

2. **Acoustician or Sound System Designer:** Will have responsibility to design and specify the sound system to suit the user's requirements, provide guidance to other design professionals on integration of the sound system with their specific areas of responsibility, develop equipment budget, assist the electrical and sound system contractors with installation details, and approve the electrical and sound system contractor's work.

3. **Architect:** Will be concerned with the type, placement, support, and integration of the loudspeakers in the space, location of audio control and equipment rooms, location of audio/video receptacle plates, and project budget.

4. **Electrical Engineer:** Will be concerned with the electrical power circuits, conduit, junction and back box requirements, technical and safety grounding, and possible interface with fire and emergency alarm systems.

5. **Mechanical Engineer:** Will be concerned with the *sensible heat* load produced by the sound system equipment so that adequate cooling can be provided.

6. **Structural Engineer:** Will be concerned with the placement and support of the loudspeakers in the space.

7. **Theatrical and Lighting Consultants:** Will be concerned with how the sound system equipment, particularly loudspeakers, impacts rigging equipment, draperies, catwalks, and stage lighting positions.

8. **Equipment Manufacturers:** Will have responsibility to provide standard and customized equipment as specified by the designers, provide assistance to the designers and sound system contractor, and develop specialized systems and software for control of the sound systems.

9. **Sound System Contractor:** Will have responsibility to procure and install sound system equipment as specified, perform equipment adjustments and calibrations, prepare equipment operation/ maintenance manuals, train users in the system operation, and provide warranty service.

10. **Electrical Contractor:** Will have responsibility to procure and install electrical power circuits, conduit, junction boxes, pull boxes, back boxes and other related building infrastructure needed for the sound systems, and may in some cases, assist the sound system contractor in pulling cable and installing ceiling loudspeaker back boxes.

The program plan outlined below is suggested to ensure that all design aspects and installation of a sound system are covered by the responsible parties. The work items in the program plan are subdivided into the standard architectural design phases: (1) Schematic Design; (2) Design Development; (3) Construction Documents; and (4) Construction Administration. While thorough in scope, not all work items listed in the program plan may be required, particularly if a simple sound system is needed, or if modifications to an existing sound system are to be provided.

The purpose of the Schematic Design phase is to meet with the client and users, evaluate room acoustical conditions, develop general system design concepts, establish preliminary financial budget, and prepare a concepts report for client and user approval. The scope of work for the Schematic Design phase is described below.

1. **Interviews:** Conduct interviews to determine the range of program types that require sound systems, needed equipment operational features and capabilities, whether there will be a designated sound system operator, and the financial budget for the sound systems.

2. **Evaluate Room Acoustical Conditions:** Determine room reverberation time and level, presence of strong late-arriving room reflections, and ambient noise levels either through acoustical measurements, if an existing space, or through design calculations, if a new space.

3. **Determine Architectural and Electrical Interface:** Evaluate requirements for audio control and equipment rooms, special

architectural considerations to accommodate sound systems, conduit routing, electrical power services, and local Code requirements.

4. **Develop System Design Concept:** Using information from 1., 2., and 3., above develop general sound system design concepts which provide operational features, electro-acoustical performance, and integrates the equipment within the space considering both architectural aesthetics and electrical service requirements. Particular attention should be paid to the loudspeaker design concept and the location of the loudspeakers in the room.

5. **Develop Cost Estimate:** Prepare a preliminary cost estimate for the sound system equipment, installation, modification to architectural elements, and electrical services related to the sound system.

6. **Prepare Concepts Report:** Summarize the results of user interviews, room acoustical evaluation, describe proposed sound system design concepts, necessary architectural and electrical modifications, and estimated installation costs in a narrative report using layman's language. Obtain written approval of the report from the client and users prior to proceeding with further design effort.

The purpose of the Design Development phase is to refine the sound system concepts into a more comprehensive design establishing electro-acoustical performance, final selection of equipment, determining electrical infrastructure for the sound systems, calculating equipment heat loads, sizing of equipment rooms, and resolving architectural issues related to the sound systems. The scope of work for the Design Development phase is described below.

1. **System Performance Parameters:** Determine required system and equipment electro-acoustic performance parameters to include *gain-before-feedback*, acceptable distortion, maximum sound levels, S/N ratio, *frequency response* characteristics, signal delay requirements, and speech intelligibility performance.

2. **Select Equipment:** Sound system equipment should be selected which satisfies anticipated program types, user operational features and capabilities, electro-acoustic performance criteria, and financial budget.

3. **Finalize Loudspeaker System Concept:** Based on an evaluation of the room acoustical properties, finalize selection of the loudspeaker type(s) and location in the room for distributing sound to the audience.

4. **Perform Calculations:** Using standardized electro-acoustic calculations, evaluate sound system gain-before-feedback, direct and reverberant sound levels, and speech intelligibility for the selected loudspeaker type(s). Perform acoustical modeling of loudspeaker

coverage and speech intelligibility to determine optimum loudspeaker placement and aiming.

5. **Electrical System:** Determine requirements for conduit, back boxes, junction boxes, pull boxes, electrical power, and audio/video receptacle plate locations. Develop sound system grounding requirements. Provide marked-up drawings with the above information to the project electrical engineer for incorporation into the electrical system drawings.

6. **HVAC System:** Determine the sensible heat load produced by the sound system equipment. Provide heat loads to the project mechanical engineer to determine requirements for supplemental cooling.

7. **Architectural Requirements:** Determine size and location of support spaces for the sound system to include equipment room(s), separate audio control room, or a location in the audience seating area for the sound system operator and main mixing console. Resolve architectural issues relating to installation of loudspeakers. Provide marked-up drawings with the above information to the architect for incorporation into the architectural drawings.

8. **Structural Requirements:** Determine hanging and support of loudspeakers from building structural members. Provide loudspeaker aiming angles, dimensions, weights, and center of gravity information to structural engineer to determine requirements for building structural member reinforcement and support of loudspeakers.

9. **Revised Cost Estimate:** Develop a revised cost estimate which more accurately reflects the anticipated installation costs for the sound systems.

The purpose of the Construction Documents phase is to prepare drawings and specifications describing the sound systems which are to be included in the project bid documents. The scope of work for the Construction Documents phase is described below.

1. **Drawings:** Prepare sound system drawings including block and signal flow diagrams, technical grounding, installation details of loudspeakers, equipment rack layouts, audio device receptacle plates, and other drawings and schedules as necessary to show the requirements for the installed sound systems. Show related sound system infrastructure items on architectural, electrical, mechanical, and structural drawings.

2. **Specifications:** Prepare standard three part sound system specifications outlining conditions of contract, functional description of the sound system, overall system performance, description of

individual equipment component items, installation practices, equipment adjustment, commissioning test procedures, and warranty information. Include related sound system infrastructure items in architectural, electrical, mechanical, and structural specifications.

3. **Quality Control:** Perform a general quality control review of architectural, electrical, mechanical, and structural drawings and specifications to verify that items relating to the sound systems have been included in these documents. Cross check sound system drawings and specifications to correct errors and omissions.

The purpose of the Construction Administration phase is to assist in the bidding, review shop drawings, interpret the construction documents, periodically review the sound system installation, witness systems acceptance testing, provide training to users, make final site inspection(s) for approving the contractor's work, close-out of the contract, and to provide follow-up services to the users. The scope of work for the Construction Administration phase is described below.

1. **Bidding:** Attend pre-bid conferences, evaluate electrical and sound system contractor proposed substitutions, bid prices, and make recommendation for contract award.

2. **Shop Drawings and Construction Documents:** Review electrical and sound system contractor shop drawings and submittals, respond to contractor inquiries, provide field coordination services, and aid in interpreting the contract documents should a dispute arise.

3. **Inspection and Acceptance Testing:** Perform periodic site inspections to review equipment and installation practices. Witness system acceptance testing on the installed sound systems to verify compliance with specifications and system design criteria.

4. **Training:** Review operation/maintenance manuals prepared by the sound system contractor, and instruct users in sound system operation and maintenance.

5. **Installation Approval and Project Close-Out:** Upon completion of the installation, commissioning tests, and user training perform a final site inspection to determine compliance with the project drawings and specifications. Notify the electrical and sound system contractors of any sound system or other work items requiring correction or replacement.

6. **Follow-Up Services:** Provide a follow-up site inspection approximately six months after system installation to review user operational concerns, make adjustments to the system, and provide recommendations for future system enhancements.

1.7 General Performance Objectives

A sound system should augment the natural sound produced by the source and provide a degree of naturalness so the listeners are not acutely aware of its presence. Exceptions include popular music performance, contemporary worship, and musical theater productions where the sound system is an integral part of the "show." Factors which determine the overall quality of sound system reproduction are described below.

1. **Loudness:** The sound system should be loud enough to be heard over the ambient noise levels and provide a high **S/N** ratio. In general, excellent **S/N** ratio conditions will be achieved if the sound system provides 25 dB(A) greater sound level than the ambient noise level. In reasonably quiet ambient noise levels [less than NC-30 or 38 dB(A)] a minimum total sound level of 63 dB(A) is adequate to provide good listening conditions. Higher ambient noise levels, [NC-45 or 52 dB(A)] require minimum sound levels of 77 dB(A) for the same listening conditions. Special music performance and sports facility sound systems should provide the capability to reproduce sound levels up to 105 dB(A). Sound levels above this threshold are not recommended due to potential hearing damage to performers, system operators, and audience members.

   Note that the above sound levels and those in Table 1-A are for steady state (continuous) conditions. Both speech and music are dynamic signals and the sound level amplitude continuously varies. This requires the sound system to provide short-term sound levels (*headroom*) of 6 to 12 dB(A), with 10 to 15 dB(A) preferred, greater than the steady state sound levels to reproduce dynamic program signals with minimum distortion.

2. ***Direct-to-Reverberant Ratio*:** A sufficiently high **D/R** ratio is required for good speech intelligibility, with a minimum value of -4 dB necessary in the 500 to 4,000 Hz octave frequency bands. The **D/R** ratio depends on the loudspeaker directivity factor (***Q***), the number of loudspeakers (**N** factor), the distance between the loudspeaker and receiver ($\mathbf{D_2}$ factor), and the room $\mathbf{T_{60}}$. Achieving a high **D/R** ratio below 500 Hz is difficult due to the omnidirectional radiation pattern of most loudspeakers at low-to-mid-frequencies. The **N** factor should be kept low, preferably no greater than 3, and the $\mathbf{D_2}$ factor should be limited to 50 ft.

3. **Intelligibility:** The amplified sound quality should provide good simulation of the unamplified signal. A high degree of intelligibility is required for sound systems which amplify speech. Speech intelligibility is a function of frequency. Figure 1-2 provides

information on the percent contribution to speech intelligibility as a function of frequency.

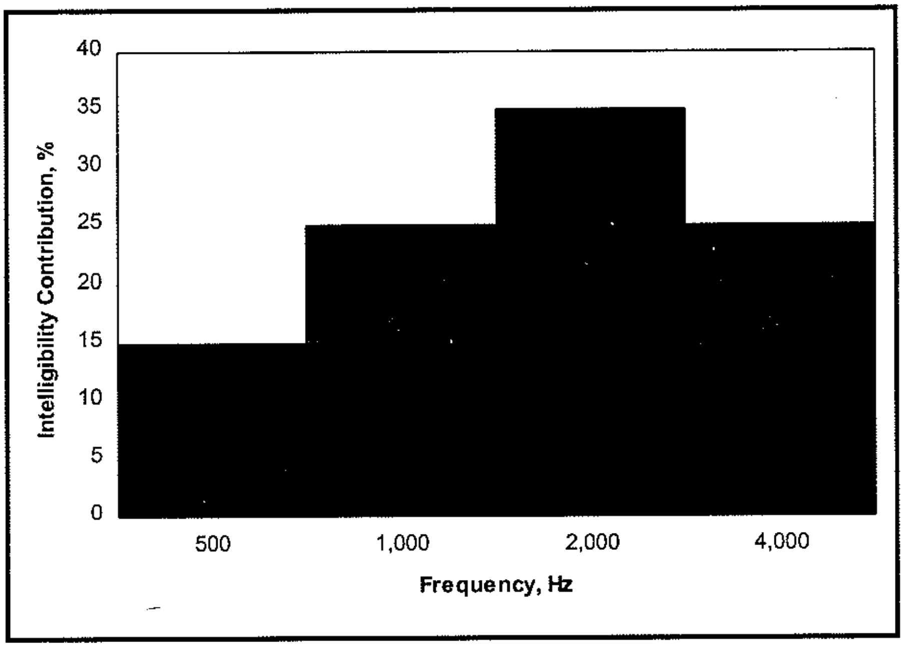


FIGURE 1-2. Percent contribution to speech intelligibility as a function of octave frequency between 500 and 4,000 Hz. Note the predominance of the 2,000 Hz octave frequency band which contributes 35 percent of speech intelligibility. Little speech intelligibility is contributed at below 500 Hz and above 4,000 Hz.

Speech intelligibility, in terms of the *Percent Articulation Loss of Consonants* (**%AL$_{CONS}$**), provides a measure of the degradation of speech consonants due to reverberation, sound reflections, and the **D/R** ratio. The **%AL$_{CONS}$** assumes the **S/N** ratio is at least 25 dB. The metric calculates the 2,000 Hz octave frequency band percent loss of consonants, with 0 percent indicating no loss of consonants (perfect intelligibility) and 15 percent indicating the maximum limit of acceptability (poor intelligibility). A good rule-of-thumb is to design for 5 to 10 percent **AL$_{CONS}$**. Achieving less than 5 percent **AL$_{CONS}$** is difficult unless the room is non-reverberant, the distance between loudspeaker and listener is small, or very directional loudspeakers are used.

Another speech intelligibility rating is based on the *Speech Transmission Index* (***STI***) and its abbreviated version, the *RApid Speech Transmission Index* (***RASTI***). This measurement evaluates the degradation in speech amplitude modulation patterns by reverberation, sound reflections, and ambient noise. The **STI** metric calculates the modulation patterns in the 125 to 8,000 Hz octave frequency bands, using voice modulation frequencies between 0.63 and 12.5 Hz. The **RASTI** technique is similar, but is restricted to the 500 and 2,000 Hz octave frequency bands. A **RASTI** score of 1 would indicate no loss of modulation patterns (perfect intelligibility) and 0 would indicate complete loss of modulation patterns (zero speech intelligibility).

Other speech intelligibility metrics exist and all of the rating schemes are highly correlated with one another. The **%AL$_{CONS}$** metric seems to yield the most accurate results for sound systems when compared to subjective speech intelligibility evaluations.

Subjective speech intelligibility will depend upon the hearing acuity and language development skills of the listener. Individuals with reduced hearing acuity or poor language development skills will require higher sound system speech intelligibility performance than the general population. Note that sound systems with very high speech intelligibility performance may still not be satisfactory for all individuals. In such cases the only solution is for those individuals to use an ALS.

Good sound system design practice would provide the minimum objective speech intelligibility performance values listed below.

General Population: %AL$_{CONS}$ (10 percent or less) or **RASTI** (0.52 or greater), resulting in subjectively "fair" speech intelligibility conditions.

Reduced Hearing Acuity/Reduced Language Skills Population: %AL$_{CONS}$ (5 percent or less), **RASTI** (0.65 or greater), resulting in subjectively "good" speech intelligibility conditions.

(See Technical Notes, Section 1.E, at the end of this chapter, for additional information on **%AL$_{CONS}$** *and* **RASTI***.)*

4. **Directional Realism:** Subjective directional realism is a function of both auditory and visual cues. The sound from the loudspeakers should appear to come from the original source location, even if the loudspeakers are remote from the source. This is achieved when the unamplified source sound arrives at the receiver first, followed by the amplified sound within approximately 35 ms. Loudspeakers should not be placed behind the listeners, as the audible cue will be from behind while the visual cue will be from the front. The split visual and hearing perception cues will be confusing.

5. **Frequency Response:** The sound system frequency response should be similar to the sound source. Speech reinforcement systems have lesser requirements than do music reinforcement systems. As a minimum, a frequency response between 300 and 4,000 Hz is required for low-quality speech reinforcement systems, but a response between

150 and 8,000 Hz is recommended for improved quality reproduction. Selective frequency boosting in the 2,000 to 4,000 Hz octave frequency bands can improve speech intelligibility. Music systems require a minimum frequency response between 80 and 10,000 Hz for realistic reproduction but an extended response between 25 and 15,000 Hz may be required for certain types of music, such as symphonic or electronically synthesized compositions.

6. **Distortion:** Various types of distortion (*harmonic, intermodulation,* and *slewing rate*) are generated by sound system equipment, particularly when equipment is operated at conditions approaching or beyond intended design limits. Fortunately, the human hearing mechanism is reasonably tolerant of most distortions. Low-to-moderate levels of harmonic distortion, generally less than 3 percent at the full power output of the equipment, are not objectionable to most listeners. Subjective loudness will increase with harmonic distortion. Odd-order harmonic distortion and all forms of intermodulation distortion are subjectively more audible. Harmonic distortion at lower frequencies can result in excess power generated at higher frequency harmonics which can be passed on to the individual *drivers* in a loudspeaker system, resulting in damage. Note that *transducers*, such as loudspeakers and microphones, produce significantly greater distortion than electrical components such as amplifiers, mixers, and signal processors.

7. ***Dynamic Range*:** The range of unamplified sound levels between the softest and loudest signals may approach 45 dB for speech and 90 dB for certain types of music. This requires the sound system be able to reproduce high sound levels without distortion and low sound levels without *self-noise*. Contrary to intuition, most amplified popular music exhibits less dynamic range than unamplified classical-type music. Figure 1-3 provides information on sound levels and dynamic range for different music types.

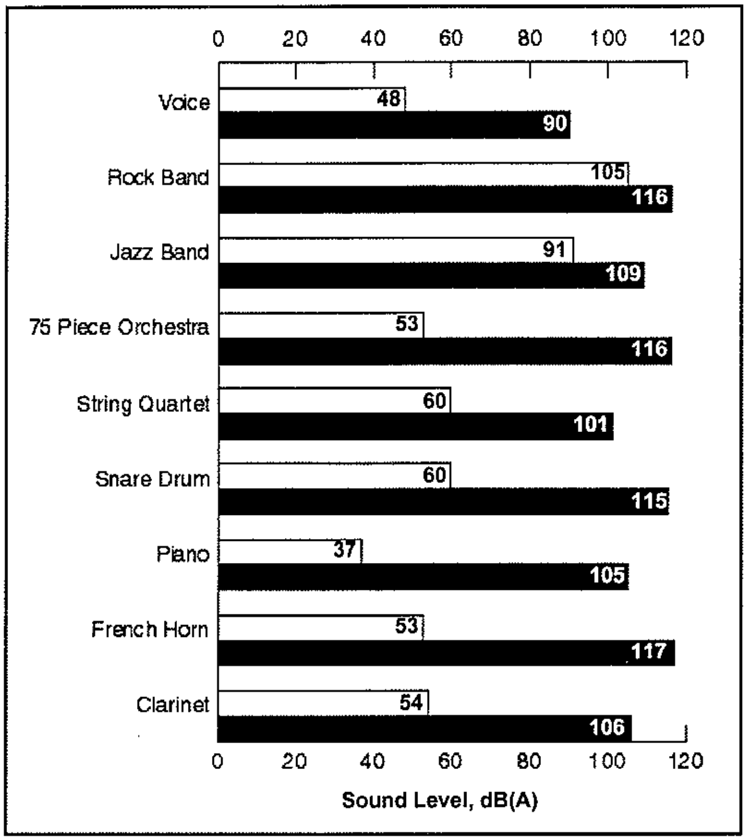


FIGURE 1-3. Typical sound levels [dB(A)], low-to-high, at 10 ft away for common music sources. The source dynamic range is the difference between the high and low sound levels.

8. **Spurious Noise:** The sound system electrical equipment should not produce excessive inherent self-noise, be susceptible to pick-up of electro-magnetic noise, or result in audible grounding "hum" or "buzzing." Loudspeakers should be properly installed to prevent rattles or secondary radiation from the building structure, lights, or ductwork when played at high sound levels. The self-noise of the sound system electrical equipment should be a minimum of 60 dB below the reproduced signal level.

9. **Feedback:** *Feedback* is the regeneration of sound between a microphone and a loudspeaker. It occurs when the loudspeaker sound reaching the microphone is greater in level than the source sound reaching the microphone. Feedback can be of either a low-level type, resulting in an audible "hollow" quality to the reproduced sound, or a high-level type, resulting in loud "howling" or "squealing." Feedback can result in damage to equipment and potential hearing damage. The sound system *gain* is limited by feedback. Typical causes of feedback include: (1) too great a distance between the microphone and the talker; (2) too small a distance between the microphone and loudspeaker; (3) a microphone in the radiation pattern of the

loudspeakers; (4) poor loudspeaker directional control; (5) uneven microphone frequency response characteristics; or (6) room sound reflections picked-up by the microphone.

1.8 Chapter Summary

This chapter has provided the reader with information on use and application of sound systems, basic room acoustic fundamentals that affect sound system performance, requirements for planning and design, and sound system performance criteria.

Sound systems are widely used in a variety of buildings. The expectation of "good acoustics" makes sound system use mandatory in almost all public assembly spaces. The sound system must be designed specifically for the space to optimize electro-acoustic performance, maintain aesthetics, and integrate with architectural and electrical systems. The sound system performance will depend upon the type of program(s) the system is to serve and the acoustical characteristics of the space. Studies have developed a range of objective and subjective performance criteria which are used as benchmarks to evaluate sound systems. Careful design and installation coordination with architects, engineers, and contractors is necessary for a successful sound system.

The next chapter will cover the major types of sound systems encountered by the architect and sound systems designer, along with the general application and design requirements for these systems.

1.9 Technical Notes

1.A Modification of the Room Constant

Acoustician George Augspurger has proposed a modified room constant (**R′**) accounting for the directional properties of a loudspeaker when aimed at the audience seating area. Here, the primary sound reflection will be much less, due to the highly sound absorptive audience, compared to the general case of an *omnidirectional* source radiating sound in the same room, which is the basis of equation 1.1. Since less acoustical energy is available for subsequent sound reflections when a directional loudspeaker is aimed at an audience, the overall level of reverberant sound will be less than in the general case. When directional loudspeakers are used, the Augspurger-modified room constant equation will yield more accurate results. The value of **R′** can be calculated using the following equation:

$$\mathbf{R'} = \frac{S\bar{\alpha}}{1 - \alpha_a} \qquad \textbf{(1.A)}$$

where,

R′ is the modified room constant, ft^2

S is the room surface area, ft^2

$\bar{\alpha}$ is the average absorption coefficient of room finish materials, dimensionless

α_a is the absorption coefficient of audience seating area, dimensionless

1.B Limitations of the Sabine and Other Reverberation Time Equations

While the Sabine reverberation time equation is commonly used due to its mathematical simplicity, the calculated results may not have good agreement with reverberation time measurements in the completed room. This can be attributed to the rooms which Sabine evaluated and the underlying assumptions in which the equation was developed. These include:

1. Statistically diffuse reverberant field.
2. Moderately long reverberation time, greater than approximately 1.6 s.
3. Low value of **A** with absorption evenly distributed throughout the room.
4. Room volume greater than 2,500 ft^3.
5. "Regularly" shaped rooms in which both the length and width dimensions are less than five times the height.

Since 1930, other acousticians have developed modifications to the Sabine equation in order to overcome some of the limitations when the above room conditions are not satisfied. The two most commonly used equations are those developed by Eyring and Fitzroy.

Eyring recognized that the Sabine equation can compute a finite reverberation time for a room in which all of the surfaces are totally absorptive and his modified equation accounts for this anomaly. The Eyring reverberation time equation has been developed based on the following assumptions:

1. Moderately short reverberation time, less than approximately 1.6 s.
2. High value of **A** with absorption evenly distributed throughout the room.
3. Room volume less than 2,500 ft^3.

The $\mathbf{T_{60}}$ value from Eyring's procedure can be calculated using the following equation:

$$\mathbf{T_{60E}} = \frac{\mathbf{0.049V}}{-\mathbf{S}\ln(\mathbf{1}-\bar{\alpha})} \quad \textbf{(1.B)}$$

where,

$\mathbf{T_{60E}}$ is the Eyring reverberation time, s
V is the room volume, ft^3
S is the room surface area, ft^2
$\bar{\alpha}$ is the average absorption coefficient of room finish materials, dimensionless

An additional complication with the Eyring equation is the need to convert the sound absorption coefficients, normally derived from reverberation room measurements using the Sabine equation, into Eyring sound absorption coefficients. The value of $\bar{\alpha}$ can be converted into the Eyring$\bar{\alpha}$ using the following equation:

$$\bar{\alpha}_e = \mathbf{1} - \mathbf{e}^{-\bar{\alpha}} \quad \textbf{(1.C)}$$

where,

$\bar{\alpha}_e$ is the Erying average absorption coefficient of room finish materials, dimensionless
$\bar{\alpha}$ is as above

Fitzroy recognized differences in calculated and measured reverberation times when the acoustical finish materials were not evenly distributed in the room. This is often the case in rooms in which the ceiling and floor surfaces are highly absorptive and the walls have little absorption. In such a room the axial room modes between the floor and ceiling surfaces will dissipate faster than the axial room modes between the two opposite pair of walls. The value of T_{60} from Fitzroy's procedure can be calculated using the following equation:

$$T_{60F} = \frac{0.049V}{S^2}\left[\frac{x^2}{A_x} + \frac{y^2}{A_y} + \frac{z^2}{A_z}\right] \quad \text{(1.D)}$$

where,

T_{60F}	is the Fitzroy reverberation time, s
V	is as above
S	is as above
x	is total area of parallel room surfaces x direction, ft²
y	is total area of parallel room surfaces y direction, ft²
z	is total area of parallel room surfaces z direction, ft²
A_x	is total absorption of parallel room surfaces x direction, sabins
A_y	is total absorption of parallel room surfaces y direction, sabins
A_z	is total absorption of parallel room surfaces z direction, sabins

Often the measured room reverberation time will be longer than the calculated reverberation time using either the Sabine or Eyring equations. In these cases, better agreement between measured and calculated values can be obtained using the Fitzroy equation.

1.C Inverse Square Law, Direct Sound Level, and Reflected Sound Level

Each time the distance from a point source is doubled, the area of the spherically radiating sound is quadrupled resulting in one-fourth the sound intensity (**I**) and a 6 dB decrease in level. Sound intensity is a measure of the *acoustic power* (**W**) in acoustic *watts* which flows through a given area. For an omnidirectional (spherical) source, the value of **I** can be calculated using the following equation:

$$I = 10.76\left[\frac{W}{(4\pi r^2)}\right] \quad \text{(1.E)}$$

where,

I	is the sound intensity, W/ft²
W	is the source sound power, acoustic watts re 10^{-12} watt
r	is the distance from the source, ft
10.76	is a constant

In equation 1.E, **I** is inversely proportional to the square of **r**, hence the "inverse-square" relationship.

The sound level (**L**) at a specified distance can be computed using the inverse square relationship. To do this requires the concept of a reference sound level ($\mathbf{L_{REF}}$) at a reference distance ($\mathbf{r_{REF}}$) from which **L** can be determined. The distance loss is independent of frequency and occurs where *free field* conditions exist. These conditions arise outdoors in the absence of reflecting surfaces and indoors for the direct sound path. The value of **L** can be calculated using the following equation:

$$\mathbf{L = L_{REF} - 20\log_{10}\left[\frac{r}{r_{REF}}\right]} \tag{1.F}$$

where,

$\mathbf{L}$	is the sound level at distance $\mathbf{r}$, dB
$\mathbf{L_{REF}}$	is the reference sound level at distance $\mathbf{r_{REF}}$, dB
$\mathbf{r}$	is the distance at which the sound level is to be calculated, ft
$\mathbf{r_{REF}}$	is the reference distance, ft

The direct and reverberant sound levels can be computed for a given indoor distance. The direct sound level depends on **W**, **r**, and **Q**, while reverberant sound level depends only on **A**. The value of the direct sound level ($\mathbf{L_D}$) can be calculated using the following equation:

$$\mathbf{L_D = L_W - 20\log_{10}(r) + 10\log_{10}(Q) - 0.6} \tag{1.G}$$

where,

$\mathbf{L_D}$	is the direct sound level at distance $\mathbf{r}$, dB
$\mathbf{L_W}$	is the source sound power level, dB
$\mathbf{r}$	is as above
$\mathbf{Q}$	is the source directivity factor, dimensionless
$\mathbf{0.6}$	is a constant

Equation 1.G has introduced the term $\mathbf{L_W}$, which is a ratio of the sound power of a source to a reference sound power. The value of $\mathbf{L_W}$ can be calculated using the following equation:

$$\mathbf{L_W = 10\log_{10}\left[\frac{W}{W_{REF}}\right]} \tag{1.H}$$

where,

$\mathbf{L_W}$	is as above
$\mathbf{W}$	is as above
$\mathbf{W_{REF}}$	is the reference sound power, 10^{-12} acoustic watts

The value of the reverberant sound level (L_R) can be calculated using the following equation:

$$L_R = L_W - 10\log_{10}(A) + 16.3 \qquad \textbf{(1.I)}$$

where,

L_R is the reverberant sound level, dB
L_W is as above
A is the sound absorption of room finish materials, sabins
16.3 is a constant

1.D Indoor Sound System Levels

Maximum continuous sound levels for sound systems are based on the space type, ambient noise levels, source sound levels, and audience expectations for amplified sound levels. Table 1-A provides guidelines on maximum continuous sound levels for different space types, exclusive of headroom factors.

TABLE 1-A. Maximum Sound System Levels [dB(A)] for Indoor Spaces

Space Type	dB(A) Range
Churches (Contemporary Worship)	100 - 105
Churches (Liturgical Worship)	75 - 80
Cinema	95 - 100
College Classrooms	70 - 75
Conference Rooms	70 - 75
Courtrooms and Civic Facilities	70 - 75
Drama Theaters	80 - 85
Elementary and High School Auditoria	85 - 90
Hotel Ballrooms	80 - 85
Lecture Halls	75 - 80
Multi-Purpose Auditoria (Music)	100 - 105
Multi-Purpose Auditoria (Speech)	80 - 85
Opera Houses	90 - 95
Sports Arenas and Gymnasia	100 - 105

1.E Relationship Between RASTI and AL_{CONS}

Various speech intelligibility metrics are highly correlated since they often are derived from common acoustical conditions. In 1986, Farrel Becker developed a mathematical relationship between **%AL_{CONS}** and **RASTI** scores based on a series of objective and subjective speech intelligibility tests. Table 1-B provides a summary of these scores and the perceived speech intelligibility quality.

TABLE 1-B. Relationship Between Subjective Intelligibility, RASTI, and %AL_{CONS}

Subjective Intelligibility	RASTI	%AL_{CONS}
BAD	0.20	57.7
	0.22	51.8
	0.24	46.5
	0.26	41.7
	0.28	37.4
	0.30	33.6
	0.32	30.1
	0.34	27.0
POOR	0.36	24.2
	0.38	21.8
	0.40	19.5
	0.42	17.5
	0.44	15.7
	0.46	14.1
	0.48	12.7
	0.50	11.4
FAIR	0.52	10.2
	0.54	9.1
	0.56	8.2
	0.58	7.4
	0.60	6.6
	0.62	5.9
	0.64	5.3

Subjective Intelligibility	RASTI	%AL_{CONS}
GOOD	0.66	4.8
	0.68	4.3
	0.70	3.8
	0.72	3.4
	0.74	3.1
	0.76	2.8
	0.78	2.5
	0.80	2.2
	0.82	2.0
	0.84	1.8
	0.86	1.6
EXCELLENT	0.88	1.4
	0.90	1.3
	0.92	1.2
	0.94	1.0
	0.96	0.9
	0.98	0.8
	1.0	0.0

Chapter 2

Application and Classification

"Sound systems is a term used to designate all manner of sound reproducing systems as follows: public address, sound reinforcing, electric megaphone, intercommunicating, announce, paging, and call systems. In general, most of these systems involve microphones, amplifiers, and loudspeakers arranged in different ways to accomplish the desired results."

Olson, Chapter 13, *Acoustical Engineering*, Van Nostrand Company, New York, NY (1957).

2

- Sound Reinforcement Systems
- Sound Distribution Systems
- Sound Reproduction Systems
- Sound System Applications
- Chapter Summary
- Technical Notes

Sound systems can be designed to serve a wide variety of functions, program types, and spaces. They can fulfill a single purpose, such as voice reinforcement for a house of worship, or be multi-purpose and provide voice reinforcement, archival recording, evidence playback, language translation, audio for video, and speech privacy sound masking for a courtroom.

It is convenient to group sound systems into three general categories: (1) sound reinforcement; (2) sound distribution; and (3) sound reproduction. These categories can be further subdivided based on specific sound system functional features. Sound reinforcement systems are used for amplification of the spoken word and music during presentations and performances. Sound distribution systems provide paging, communication, emergency announcement, language translation, teleconferencing, press feeds, and assistive listening capabilities. Sound reproduction systems record or playback audio media in video and cinema installations, distribute background music, provide speech privacy sound masking, and enhance room acoustical properties.

This chapter will provide the reader with an overview of the functional and operational requirements of these various sound systems.

2.1 Sound Reinforcement Systems

Sound reinforcement systems amplify the sound level of speech, singing, or music produced by a talker or performer located in the same space as the audience. These systems can be subdivided into voice and music reinforcement systems.

The basic components of a sound reinforcement system comprise microphones (to convert acoustical signals to electrical signals), signal processing equipment (to mix different signal sources, alter frequency characteristics, or provide signal distribution and routing), amplifiers (to increase the signal level), and loudspeakers (to convert electrical signals to acoustical signals).

2.1.1 Voice Reinforcement Systems

Voice reinforcement systems amplify the spoken word so the message can be heard at distances beyond which unamplified speech would be inadequate for the listener. These systems are used in virtually all public assembly spaces and are usually permanently installed in the space. High quality equipment and installation practices, particularly in selecting and locating the loudspeakers relative to the talker and audience locations, are essential to achieve good performance characteristics from voice reinforcement systems.

2.1.2 Music Reinforcement Systems

Music reinforcement systems function similar to voice reinforcement systems but usually provide greater sound levels and wider frequency response characteristics. These systems are restricted to auditoria, amphitheaters, arenas, worship houses, and similar spaces and may be either permanently installed or serve as a temporary "touring" system used by performers specifically for their show. High quality equipment, installation practices, and operator control are required to achieve good performance characteristics from music reinforcement systems.

2.2 Sound Distribution Systems

Sound distribution systems enable speech, music, or public warning signals to be distributed to listeners in the same location as the source or to spaces remote from the point of signal origination. These systems can be subdivided into paging, communication, emergency announcement, language translation, teleconferencing, press feeds, and assistive listening systems.

The basic components of a sound distribution system are similar to the sound reinforcement system except signal sources can comprise microphones or prerecorded audio media.

2.2.1 Paging, Communication, and Emergency Announcement Systems

Paging, communication, and emergency announcement systems provide distribution of voice or alarm signals to a defined area. While used primarily for voice announcements, paging and communication systems can also provide background music distribution. Emergency announcement systems, such as fire alarm systems, are required by Codes to be a dedicated system and to use equipment certified by Canadian Standards Association, (CSA), Factory Mutual (FM), or Underwriters Laboratories (UL). These systems must also be capable of operation from battery back-up or emergency electrical generators.

Equipment performance requirements are generally less stringent for paging, communication, and emergency announcement systems due to the limited frequency response characteristics of the signal. The sound level output should be a minimum of 15 dBA above the ambient noise levels or as directed by applicable Codes. Installation requirements for these systems, such as number of loudspeakers, loudspeaker spacing, or system redundancy, may be very stringent and are specified in the Codes.

2.2.2 Language Translation Systems

Language translation systems are used by an interpreter to translate foreign languages for listeners. These systems are commonly used in civic buildings, convention centers, international governmental buildings, judicial facilities, and museums. The foreign language signal is picked-up by a microphone and routed to an interpreter wearing a headset. The translated signal from the interpreter's microphone is amplified and distributed to the attendee who wears a headset to hear the translated message. The equipment comprising these systems has been specially designed for this function and is of a very high quality.

2.2.3 Teleconferencing Systems

Teleconferencing systems enable individuals at remote locations to simultaneously meet via electronic audio and video formats. These systems use microphones and video cameras to pick-up the acoustic and video signals, which are encoded, broadcast over telephone, microwave, or satellite systems, decoded at the receiving end, and reproduced via amplified loudspeakers and a video monitor, all in real time. Teleconferencing is commonly used for "distance learning" in educational and medical institutions, to hold corporate business meetings, and in remote judicial proceedings. The equipment used for teleconferencing can be self-contained on a portable roll-about equipment cart or installed as part of a permanent large scale system in a dedicated teleconferencing room.

2.2.4 Press Feed Systems

Press feed systems can distribute audio signals to a dedicated audio receptacle jack in the room, a dedicated press room, or to an outside panel where a remote broadcasting truck can park and access the audio signal outputs. A composite mix of the audio signals from the sound system is routed a common output point which may terminate in a "mult box" having numerous different audio connectors capable of providing different signal levels. The press members can plug in tape recorders or audio signal distribution equipment and be able to match the audio signal level to their equipment. Most civic, judicial, legislative, and sports facilities have provisions for a press feed system.

2.2.5 Assistive Listening Systems

Assistive listening systems provide localized sound reinforcement to listeners who have hearing difficulty due to reduced hearing acuity, poor language development skills, or interference from room acoustical conditions. These systems are commonly used as part of voice and music reinforcement systems. The electrical output from a dedicated microphone or the sound system signal mixer is routed to a transmitter. *Frequency modulation* (*FM*), *infrared* (*IR*), or *induction loop* (*IL*) transmitters radiate the modulated audio signal which is picked-up by a receiver. A small headset worn by the listener is connected to the receiver. Current requirements mandated by the ADA require assistive listening systems in all places of public assembly seating more than 50 occupants or in facilities using a sound amplification system. Exempt from these requirements are houses of worship and private facilities not open to the public. Assistive listening systems can be permanently installed in the space or can be a portable type, used on an as-needed basis.

2.3 Sound Reproduction Systems

Sound reproduction systems provide audio signals from a variety of media formats with the intent of either realistic signal reproduction or synthesized effects enhancement. These systems can be subdivided into audio media recording and playback, audio for video, background music, cinema sound, speech privacy sound masking, and room acoustical enhancement systems.

The basic components of a sound reproduction system are similar to the sound reinforcement system except the signal sources generally comprise prerecorded audio media. Audio media recording and audio for video systems often have microphone inputs, while room acoustical enhancement systems always use microphones.

2.3.1 Audio Media Recording and Playback Systems

Audio media recording and playback systems provide for amplification of sources such as audio tape, compact disc (CD), or mini-disc (MD). These systems can function as an element of a larger sound system or as a stand alone system. Recording systems use dedicated microphones or an electrical output from the sound system signal mixer to record the program to cassette tape, digital audio tape (DAT), or MD. Playback systems amplify sound from signal storage media such as cassette tape, DAT, CD, digital versatile disc (DVD), MD, message repeaters, tape carts, or from a distant origin, such as radio or TV transmissions. The quality of the equipment used in these systems varies depending on application. Voice and music systems have high quality components and other systems, such as airport message announcement systems, have lesser quality components.

2.3.2 Audio for Video Systems

Audio for video systems provide the means to record or play an audio signal for use with a video program. Videotape reproducers have audio inputs to receive microphone or *line level* signals to record the audio portion of the program. DVD players and videotape reproducers have audio outputs to route the signal to the sound system to permit the audience to hear the recorded portion of the video program. The equipment used for these systems is of high quality and normally is part of a permanently installed system.

2.3.3 Background Music Systems

Background music systems provide distribution of prerecorded or programmed music to low-level distributed loudspeakers. The music program signal can be played back from audio tape, CD, and MD sources or received as part of a commercial subscription service such as MUZAK®. The subscription service provides audio input via the telephone line, or more increasingly, through cable or satellite link-ups. The equipment used for these systems is of low quality and normally is part of a permanently installed system.

2.3.4 Cinema Sound Systems

Cinema sound systems playback dialog, music, and sound effects which have been encoded on the film soundtrack. The basic system uses five channels with loudspeakers located behind the screen at the front of the theater. Newer cinema sound formats use "surround" and sound effects loudspeakers located on the side and rear walls to supplement the front loudspeakers. There are a variety of proprietary signal enhancement systems used in cinema sound systems to provide extended dynamic range, wider frequency response, surround sound channels, and other special effects. Some currently used systems include DTS™ by Digital Theater Systems, SRD/Dolby Digital™ by Dolby Laboratories, and SDDS/Sony Dynamic Digital Sound™ by Sony Cinema Products Corporation. Cinema sound systems are permanently installed using high quality equipment approved by the proprietary cinema sound system franchiser.

(See Technical Notes, Section 2.A, at the end of this chapter, for additional information on cinema sound.)

2.3.5 Speech Privacy Sound Masking Systems

Speech privacy sound masking systems radiate *pink noise* which is adjusted in level and frequency content to make speech less intelligible to the listener. These systems are commonly used in open office environments in combination with partial height screens around workstations and liberal use of sound absorbing materials. Another application is in courtrooms where it may be desired to increase the speech privacy during conversations at the judge's bench. The key to successful implementation with sound masking systems is to have an even low-level sound distribution throughout the space so the occupants are not aware of the masking sound. This requires using a sufficient number of loudspeakers which are properly adjusted in both frequency and level so the listeners are not consciously aware of the masking sound. The exception to this is in courtroom sound masking systems where the pink noise is at least 10dB(A) above the room ambient noise level.

2.3.6 Room Acoustical Enhancement Systems

Room acoustical enhancement systems ("electronic architecture") modify the natural room acoustics using specialized electro-acoustic techniques. These systems can increase the room reverberation time and level, provide early reflections to simulate nearby sound reflecting surfaces, improve speech intelligibility, and enhance the blending of musical sounds. An electronic architecture system comprises several microphones located above and forward of the stage to pick-up the sound field. The electronic signals are sent to a routing matrix to subdivide and distribute the separate signals to different frequency equalizers, delay lines, and reverberation generators. The processed signals are distributed to numerous power amplifiers and loudspeakers located throughout the room which radiate the acoustically modified signal.

Electronic architecture systems can be installed in both new and existing construction. They are often used in large rooms, where extensive architectural modifications to improve acoustical conditions might be cost-prohibitive, or where it is necessary to provide variable acoustical conditions to suit different program types. These systems have a long development history with each successive system improving on prior technology. All of the earlier systems suffered from audible coloration, particularly at high gain settings, until the advent of signal generation by computer technology. The currently available systems impart a very convincing acoustic signature and can be unhesitatingly recommended. These systems are always custom designed for the particular space and use very high quality equipment properly adjusted to make their use transparent to both the performers and audience.

(See Technical Notes, Section 2.B, at the end of this chapter, for additional information on room acoustical enhancement systems.)

2.4 Sound System Applications

Sound systems having single or multiple functions may be required for different types of spaces. Table 2-1 summarizes the different sound systems described above and their common applications in a range of buildings. These different systems may operate as stand alone systems or be interconnected with each other.

TABLE 2-1. Sound Systems Application Matrix

Building and Room Type	Sound Reinforcement		Sound Distribution						Sound Reproduction					
	Music Programs	Voice Programs	Assistive Listening	Emergency Announcement	Language Translation	Paging/Communication	Press Feed	Teleconferencing	Audio Media Recording/Playback	Audio for Video	Background Music	Cinema Sound	Room Acoustic Enhancement	Speech Privacy Sound Masking
CIVIC														
Council Chambers		X	X	X	(X)	(X)	X	(X)	(X)	(X)				(X)
Courtrooms		X	X	X	(X)	(X)	X	(X)	X	(X)				(X)
Hospitals		(X)	X	X		X		(X)	(X)	(X)	(X)			
EDUCATION														
Classrooms		(X)	(X)	X	(X)	(X)		(X)	(X)	(X)		(X)		
Lecture Halls	(X)	X	X	X	(X)	(X)		(X)	(X)	(X)		(X)		
Museums	(X)	(X)	X	X	(X)	(X)		(X)	(X)	(X)	(X)	(X)		
ENTERTAINMENT														
Clubs	(X)	(X)	(X)	(X)					(X)	(X)	(X)			
Hotels	(X)	(X)	X	X	(X)	X		(X)	(X)	(X)	(X)			
Restaurants	(X)			(X)		(X)			(X)		X			(X)
PUBLIC ASSEMBLY														
Arenas	(X)	X	X	X		X	X		X	(X)	(X)			
Auditoria	(X)	X	X	X	(X)	X			X	(X)		(X)	(X)	
Conference Centers		X	X	X	(X)	X	(X)	X	(X)	(X)	(X)	(X)		
Exhibition Halls	(X)	X	X	X	(X)	X	(X)	(X)	(X)	(X)	(X)	(X)		
Sports Facilities	(X)	X	X	X		X	X		(X)	(X)	(X)			
Worship Houses	(X)	X	X	X					X	(X)	(X)	(X)	(X)	
TRANSPORTATION FACILTIES														
Airports		X	X	X	(X)	X					(X)			
Rail Stations		X	X	X	(X)	X					(X)			
OTHER														
Industrial Facilities			X	X		X		(X)		(X)	(X)			
Offices				X		X		X	(X)	(X)	(X)			(X)
Retail Stores		(X)	X	(X)		(X)				(X)	X			

X = System normally required. (X) = System may be required based on building function.

2.5 Chapter Summary

This chapter has provided the reader with an overview of the three general sound system categories (sound reinforcement, sound distribution, and sound reproduction). Within these broad groupings are a variety of sound system types, each fulfilling a specific or unique group of functions.

Buildings have varying sound system needs, which can range from simple to complex, depending upon the space activities and user requirements. Many sound systems comprise several smaller subsystems, which can be interconnected to serve a common space, or operate independently to serve other spaces. Equipment for different sound systems vary, but all commonly have an input source, signal processing, amplification, and output source.

The next chapter will cover the different types of electronic and electro-acoustical components used in sound systems, including their application, design, and operation.

2.6 Technical Notes

2.A Cinema Sound Formats

Multi-channel surround sound will be the future of audio playback. Already this format is available in selected commercial broadcast, cable, and motion picture releases. The technology has made considerable advances since the failed quadraphonic system formats of the early-to-mid 1970s. Several competing formats are currently in use.

The undisputed leaders in developing multi-channel surround formats are the various cinema production companies. The very first multi-channel system was developed by RCA in 1940, in cooperation with Walt Disney Productions and conductor Leopold Stokowski, for the movie "Fantasia." When it premiered on 13 November 1940 at the Broadway Theater in New York, New York audiences experienced the state-of-the art sound system for its day. The stereophonic (two-channel) system was called "Fantasound" and used a unique system of 90 loudspeakers to surround the audience. The complexity of the sound and movie projection systems, along with the need for each movie house to be individually set-up, limited the distribution of films with Fantasound.

Dolby™ Stereo by Dolby Laboratories is the most common surround sound format. Movies and major television shows are released in this audio format. The system uses the Dolby™ process of motion picture encoding to create a two-channel release that can be decoded into four channels with the appropriate hardware. Should no decoder hardware be available, the program audio is played back in conventional stereophonic or *monophonic* (single-channel) audio. When decoded into four channels, the audio is routed to right, center, left, and monophonic surround channels. One prime advantage of the Dolby™ Stereo format is only one film print needs to be released since monophonic, stereophonic, and four-channel audio formats are encoded on the print, ensuring compatibility with a wide variety of playback systems.

SR-D/Dolby™ Digital is gaining popularity as a surround sound format for both cinema and DVD releases. The format differs from Dolby™ Stereo using 5.1 channels configured as left, center, right, left surround, right surround, and low-frequency (subwoofer) channels. Five channels of full-bandwidth audio and one channel of limited low-frequency bandwidth audio are used. The low-frequency channel reproduces frequencies below 80 to 120 Hz, depending upon the specifics of the processor. The digital audio signals are optically stored on the film print or within the DVD. Future high definition television (HDTV) will use the 5.1-channel audio format.

SDDS™ by Sony Cinema Products Corporation is a new surround sound format using eight channels of audio comprising left, left center, center, right, right center, left surround, right surround, and low-frequency (subwoofer) channels. The system can be considered to be a 7.1-channel audio format with seven channels of full-bandwidth audio and one channel of limited low-frequency bandwidth. A dedicated SDDS™ decoder is necessary to realize this audio format. The decoder also has the capabilities of playing the 7.1-channel format into a 5.1-channel or smaller audio format system for compatibility between different systems.

DTS™ by Digital Theater Systems is a digital audio 5.1-channel surround system similar to SR-D/Dolby™ Digital but the audio program is stored on a separate laserdisc which is time code linked with the film print.

2.B Development of Room Acoustical Enhancement Systems

The first system type, "Ambiophony", was developed by the Philips Company of the Netherlands in 1953. Microphones picked-up the source signal which was routed to a series of tape loops and signal delays. The decaying signal feedback loops were amplified and distributed to multiple loudspeakers in the auditorium. Reverberation time and level were increased, but due to the limitations of the equipment technology, the sound quality was generally poor. One system was installed in the La Scala Opera House in Milan, Italy and used for many years.

"Assisted Resonance" (AR), was developed in the early 1960s by the late Peter Parkin of the Building Research Station in the United Kingdom. The purpose of the system was to increase the low-frequency reverberation time and level in London's Royal Festival Hall, which was unacceptably low due to underestimating the audience sound absorption properties during the auditorium design. The system worked by amplifying the resonant modes in the auditorium using a series of *Helmholtz resonators* each tuned to a 3 Hz bandwidth. The frequency response of the system was limited between 40 and 1,000 Hz and used 320 channels. The major drawbacks of the system include lack of high frequency reverberation and audible coloration since each channel operates near its feedback threshold to achieve a modest increase in reverberant sound. The AR system was later commercialized by AIRO Systems in England and distributed until the late 1970s.

The "Multi-Channel Reverberation System" (MCRS) was developed by the Philips Company in the mid 1960s as a replacement for their "Ambiophony" system. Like the AR system, it uses an acoustic feedback loop, but unlike the AR system, it is a wide-bandwidth system. Due to the extremely low gain of each channel, up to 100 channels are required for a 3 dB gain increase and a doubling of reverberation time. Other drawbacks include amplifying the existing room reverberation, which can magnify existing acoustical problems, and inadvertent pick-up and amplification of background noise, because of the broadband system response.

Work by the late Paul S. Veneklasen in the mid 1960s resulted in his "Stage-to-Hall Transfer" and "Auditorium Synthesis" systems. The former system placed two microphones in the reverberant stage house above the orchestra enclosure, the outputs were amplified, and routed to two non-directional loudspeakers acoustically coupled to the upper main auditorium volume. The "Auditorium Synthesis" system used a series of microphones which picked-up stage sound and routed it to signal delay lines and a small reverberation chamber containing a loudspeaker and microphones. The **D/R** ratio and lateral (envelopment) sound could be adjusted before being amplified and routed to separate loudspeakers in the auditorium.

In the early 1970s the "Electronic Reflected Energy System" (ERES) was developed by Christopher Jaffe. The system used microphones placed in the ceiling above the stage which were routed to a three-channel signal delay. One delay output was routed to loudspeakers placed in the forestage canopy to reproduce early reflections for acoustical "intimacy". The other two outputs were routed to reverberant field loudspeakers placed in the upper main auditorium volume. The loudspeakers were bandwidth limited between 30 and 250 Hz to provide a sense of acoustical "warmth."

During the mid 1980s Peter Barnett of AMS Acoustics in England developed the "Reverberation on Demand System" (RODS). The primary advantage with this system was it did not use a feedback loop to generate the reverberation, resulting in less audible coloration. Working with AMS Acoustics, Wade Bray of the acoustical consulting firm Jaffe Holden Scarborough, developed a technique using random modulation of delay times in the signal delay lines to further improve the naturalness of the reverberant sound. Both the ERES, for early sound enhancement, and RODS, for reverberant sound enhancement, were used in several auditorium and church installations.

In 1990, the US manufacturer Lexicon developed the "Lexicon Acoustic Reverberance and Enhancement System" (LARES) which uses time variation in generating reverberation. This process decorrelates the signal paths, resulting in a more natural sounding reverberation with significantly higher gain settings. Microphones pick-up the sound which is routed to a bank of matrix mixing reverberation generators employing the time variant principle. The outputs are routed to wall and ceiling loudspeakers, and each listener receives sound from several loudspeakers, furthering the sound field enhancement.

The most recently developed system is the "System for Improved Acoustic Performance" (SIAP) by SIAP, B.V. of the Netherlands, introduced in the early 1990s. The SIAP system reinforces the missing reflective and reverberant components while retaining the overall acoustical characteristics of the auditorium through use of discrete reflection generation rather than acoustic feedback techniques. This permits the sound level, early reflections, and reverberation time to be independently adjustable. Like the other systems, microphones pick-up the acoustic signal which is digitally processed to achieve the desired acoustical characteristic, amplified, and routed to wall and ceiling loudspeakers. The loudspeakers comprise both hemispherically and direct radiating types which can be adjusted to provide different amounts of early and late acoustical energy. The SIAP system can provide up to 448 decorrelated reflection patterns with up to 26.5 dB of gain.

Chapter 3

Components and Equipment

"The sound man must be intimately familiar with the operating principles and available types of microphones, amplifiers, modulating devices, loudspeakers, recorders, and reproducers. His success depends upon his ability to properly handle these various tools and to direct their uses in the best way to secure the result which is desired for the final product."

Miller, Chapter 1, *Motion Picture Sound Engineering*, Van Nostrand Company, New York, NY (1938).

3

A sound system consists of a variety of different electrical, electro-mechanical, and electro-acoustical components which are interconnected together to perform a particular function. In its most basic form, a sound system comprises a source component (e.g., CD player, microphone, or tape reproducer), a signal processing component (e.g., frequency equalizer or mixer), an amplifier (e.g., preamplifier or power amplifier), and a transmitting component (e.g., antenna, headphone, or loudspeaker). Upon closer inspection some complicated sound systems are a series of interconnected subsystems, each fulfilling a specific function as part of the larger system.

This chapter will provide the reader with an overview of the functional and operational characteristics of the basic component elements which make up sound systems.

3.1 Microphones

A microphone is a transducer which converts sound waves into an alternating current (AC) voltage corresponding to the frequency and magnitude of the source. Microphones can be classified by type, transducer element, or *polar pattern*.

3.1.1 Microphone Types

The major microphone types include: (1) thin profile lectern; (2) handheld; (3) boundary layer; (4) lavaliere; (5) headset; (6) overhead; (7) shotgun; and (8) wireless.

The thin profile lectern microphone has a small *condenser* capsule element which is mounted at the end of a long narrow flexible tube. The major functional advantages with this microphone are the ability to position it close to the talker and its discrete unobtrusive profile. Figure 3-1 shows a thin profile lectern microphone.

The handheld microphone has a larger microphone capsule, which is often a *dynamic* element, with an integral windscreen mounted on a shaft. These microphones have been specially designed to withstand rough handling while providing good electro-acoustical response under a wide variety of conditions. Figure 3-1 shows a handheld microphone.

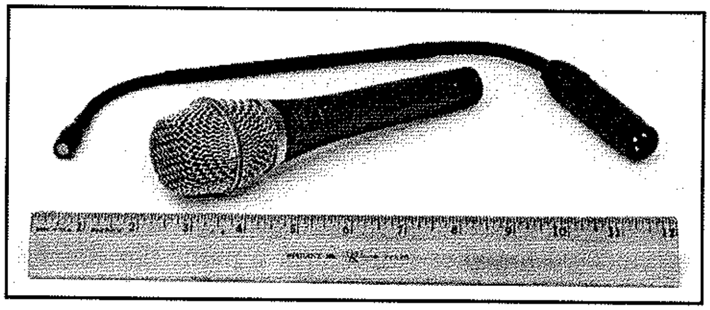


FIGURE 3-1. Thin profile lectern microphone (Shure MX412 top) and handheld microphone (Shure Beta 87A bottom). Products Courtesy of Shure Brothers, Inc.

The boundary layer microphone has a small condenser microphone capsule located close to a flat mounting plate. These microphones are intended to be set on a large boundary surface, such as a ceiling, table, or wall, to help extend the low-frequency response of the microphone. The proximity of the microphone capsule to the boundary surface has the advantage of minimizing interference effects between the direct and reflected sound arriving at the capsule. These microphones have a 6 dB increase in output level due to coherent summation of the direct and reflected sound. Boundary layer microphones are often used for picking-up multiple sources, such as at a conference table, and are used as "foot" microphones at the front of a stage. Figure 3-2 shows two types of boundary layer microphones.

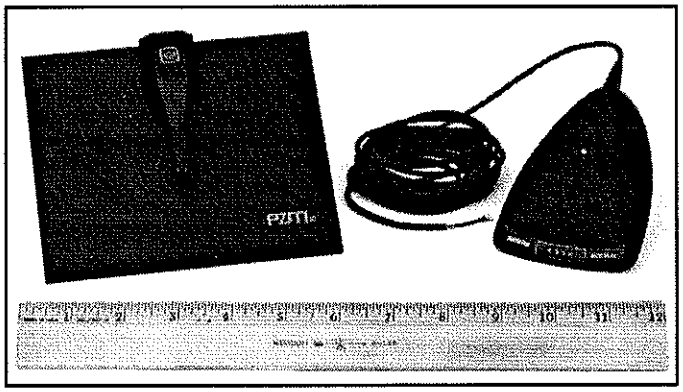


FIGURE 3-2. Boundary layer microphones (Crown PZM 30D left and Shure MX392/C right). Products courtesy of Crown International, Inc. and Shure Brothers, Inc.

A lavaliere microphone has a miniature condenser microphone capsule installed on a mounting clip which is worn on the clothing of the talker. The principle advantages with this microphone are its unobtrusive size and freedom of placement, with locations on the torso and head (for stage actors) being the most common. Figure 3-3 shows a lavaliere microphone.

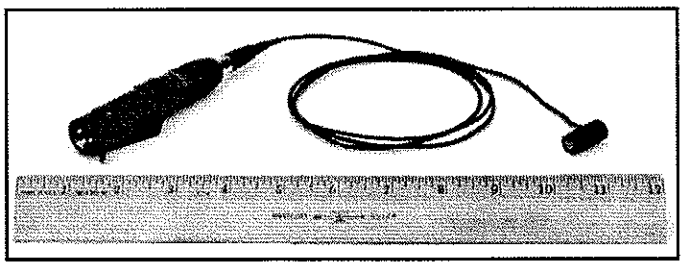


FIGURE 3-3. Lavaliere microphone with preamplifier (Shure MX185). Product courtesy of Shure Brothers, Inc.

A variant of the lavaliere microphone is the headset microphone which has a small miniature condenser microphone capsule at the end of a flexible tube mounted on a small headset. The flexible tube permits locating the microphone close to the mouth. These microphones are used by popular music performers or others in high noise environments where the maximum voice level must be picked-up to increase the sound system gain-before-feedback.

The overhead microphone has a small condenser element microphone capsule mounted on the end of a long thin cable. An integral retaining wire or flexible tube at the cable end closest to the microphone element enables aiming the microphone at the sound source. The small size of the overhead microphone has the advantage of minimizing its visual appearance when hung from the ceiling. These microphones are available in different colors to blend in with the room decor. Common uses of overhead microphones are for pick-up of church choirs and on-stage actors. Figure 3-4 shows an overhead microphone.

The shotgun type microphone has a small dynamic or condenser microphone capsule mounted on a 12 to 18 in long tube with slots along the back of its length. The open slots permit sound to enter the microphone body out-of-phase with the sound arriving at the front of the microphone. The front and rearward travelling sound waves combine to increase the microphone *sensitivity* to frontally-arriving sound. These microphones have a very narrow pick-up pattern and are used at locations where it is not possible to position a conventional microphone close to the source. Figure 3-4 shows a shotgun microphone.

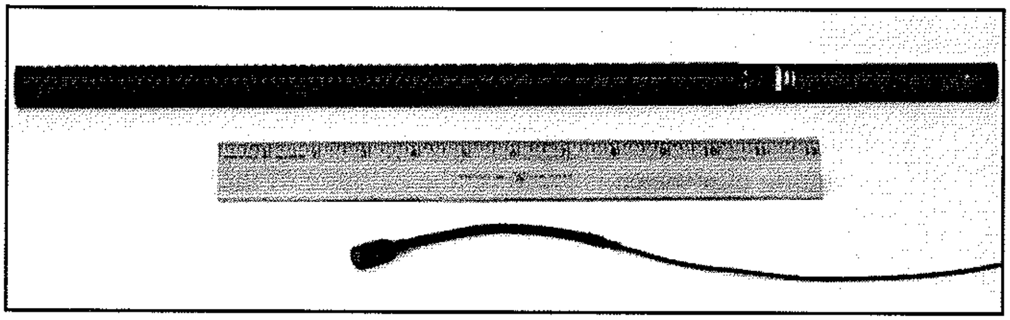


FIGURE 3-4. Shotgun microphone (Shure SM89 top) and overhead microphone (Shure MX202B/C bottom). Products courtesy of Shure Brothers, Inc.

A wireless microphone comprises a condenser microphone capsule, *preamplifier*, transmitter with antenna, and receiver with antenna(e). The transmitter and receiver are tuned to a common *carrier frequency*. The output from the microphone capsule is boosted by the preamplifier and routed to the transmitter and antenna, located on the microphone shaft for handheld microphones, and for lavaliere microphones, within a separate belt pack. The audio signal is frequency modulated by the transmitter, similar to an FM radio broadcast, and radiated from the antenna. The radiated signal is picked-up by a receiver with an antenna and demodulated. Improved reception can be realized with a true diversity type system which uses two separate receiver channels and antennae. The diversity principle compares the strength of the two received signals and uses the stronger signal, resulting in a reduction in noise and *radio frequency interference* (*RFI*). Figure 3-5 shows a true diversity wireless microphone system.

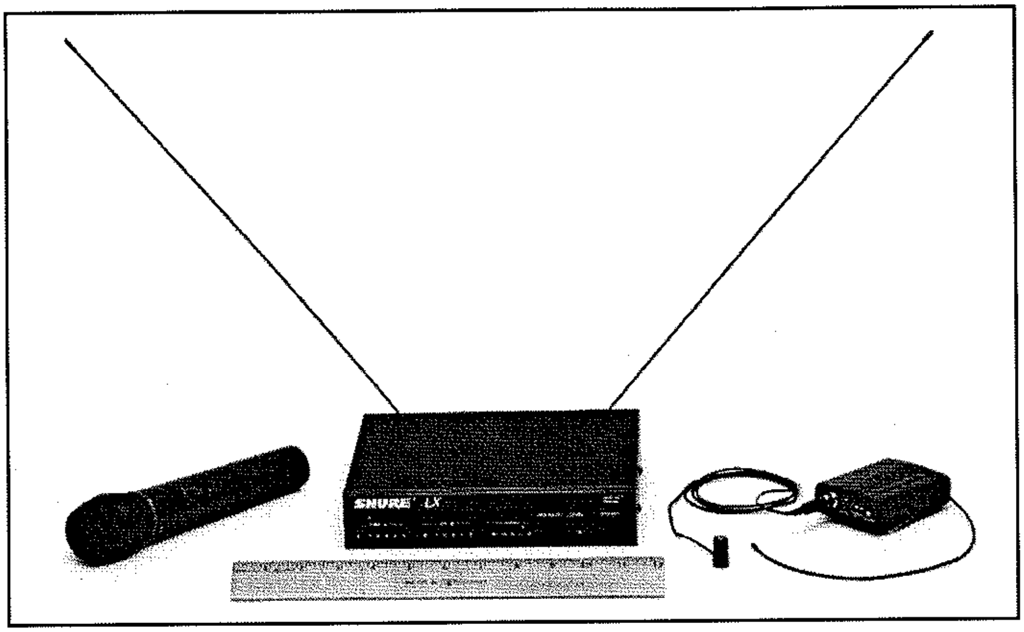


FIGURE 3-5. Wireless microphone system with handheld microphone transmitter (Shure LX2/87 left), diversity receiver (Shure LX4 center), and lavaliere microphone (Shure WL185) with belt pack transmitter (LX1 right). Products courtesy of Shure Brothers, Inc.

Wireless microphone systems can operate either on the *very high-frequency* (*VHF*) or the *ultra high-frequency* (*UHF*) bands, with the VHF bands being more common. The Federal Communications Commission (FCC) has assigned frequency bandwidths for both VHF and UHF wireless microphone systems. For VHF systems the 49 to 108 MHz (low band) and 169 to 216 MHz (high band) frequencies are used. Both are limited to a transmitter power output of 50 mW. The UHF systems are assigned the 450 to 806 MHz (low band) and 900 to 952 MHz (high band) frequencies, both limited to a transmitter power output of 250 mW. Only a portion of the FCC-assigned bandwidths are used for wireless microphone transmission due to competition from other radio services. The 174 to 216 MHz bandwidth is used for VHF systems, while the 774 to 862 MHz bandwidth is used for UHF systems.

Multiple channel VHF and UHF wireless microphone systems are common, particularly in large productions such as "Broadway" musicals. Each performer is assigned a separate channel with a unique carrier frequency to prevent interference. Up to 50 channels can be simultaneously broadcast, but this requires a very sophisticated multi-channel system to correctly receive and demodulate the signal channels.

The principle advantage with a wireless microphone is the mobility it gives a user since there is no microphone cable. One major drawback with wireless microphones is they should not be used where security or privacy is of concern since the signal radiates approximately 300 ft and can pass through typical building construction. Another potential drawback is the susceptibility to the pick-up of adjacent radio signals and other RFI which can be amplified and transmitted along with the desired audio signal.

Table 3-1 summarizes common applications for the different microphones described above.

TABLE 3-1. Microphone Application Matrix

Space Type	Microphone Type							
	Thin Profile Lectern	Handheld	Boundary Layer	Lavaliere	Headset	Overhead	Shotgun	Wireless
Amphitheaters	(X)	X	(X)	(X)	(X)	(X)		X
Churches	X	(X)	(X)	(X)		X		X
College Classrooms	X	(X)		(X)		(X)	(X)	(X)
Conference Rooms	X	(X)	(X)	X		(X)	(X)	(X)
Courtrooms and Civic Facilities	X	(X)	X	(X)		(X)	(X)	
Drama Theaters		(X)	X	X		(X)	(X)	X
Elementary and High School Auditoria	X	X	(X)	(X)		(X)		(X)
Hotel Ballrooms	X	X		(X)	(X)			(X)
Lecture Halls	X	(X)		(X)		(X)		(X)
Multi-Purpose Auditoria	X	X	X	X		X	(X)	X
Opera Houses	(X)	(X)	X	X		X	(X)	X
Race Tracks	X	X		(X)	X			(X)
Recital Halls	(X)	(X)	(X)	(X)		(X)	(X)	(X)
Sports Arenas and Gymnasia	X	X		X	X	(X)	(X)	X
Stadia	X	X		X	X			X
Swimming Pools	X	X		X	X			X
Symphony Halls	(X)	(X)	(X)	(X)		(X)	(X)	(X)
Tennis Courts	X	X		(X)	(X)		(X)	X
Theme Parks	X	X	(X)	X	X	(X)	(X)	X

X = Normally required. (X) = May be required based on sound system function.

3.1.2 Microphone Transducer Elements

Microphones use dynamic, *ribbon*, or condenser transducer elements to convert sound waves into an analogous electrical signal. Figure 3-6 shows the major parts of a microphone.

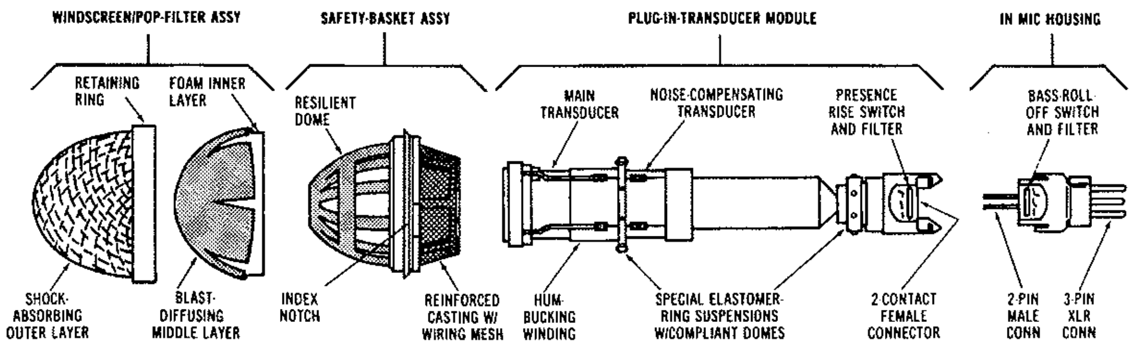


FIGURE 3-6. Sectional drawing of a dynamic cardioid microphone showing details of the major component parts. Drawing courtesy of AKG Acoustics US.

The dynamic microphone uses a thin Mylar™, or similar plasticized diaphragm, which vibrates in response to the sound waves. This in turn moves a small coil of wire suspended between two magnetic poles creating an output voltage. The transduction principle of a dynamic microphone operates opposite to that of a dynamic loudspeaker. Advantages of dynamic microphones include their durability, less sensitivity to temperature and humidity extremes, and simplicity of operation.

The ribbon microphone is similar to the dynamic microphone except a metalized ribbon is used in place of the plasticized diaphragm. While the ribbon microphone is often considered the best microphone type where natural voice reproduction is required, it is not frequently used for sound reinforcement due to the fragile nature of the ribbon element.

The condenser (capacitor) microphone uses an electrically polarized movable diaphragm which vibrates in response to the sound waves. The spacing between the movable diaphragm and an oppositely charged fixed backplate varies, which changes the capacitance between the two elements. A condenser microphone needs a source of power to provide the polarization voltage between the movable diaphragm and fixed backplate. This is supplied as *phantom power* from a separate external power source or, more commonly, from the signal mixer the microphone is plugged into. Typical phantom power varies from 12 to 48 V (*volts*) but has little current. The phantom power is delivered to the microphone via the microphone cable, with the two signal conductors handling the positive direct current (DC) voltage and the cable shield conductor handling the negative return. Another type of condenser microphone uses a permanently charged *electret* element. These microphones are pre-polarized during manufacturing and do not require an external source of polarization voltage. Advantages of condenser microphones include their greater sensitivity, extended frequency response, and small size. One major disadvantage is their less rugged construction compared to dynamic microphones, so care must be exercised in their use.

3.1.3 Microphone Directional Patterns

The major directional patterns of microphones include: (1) omnidirectional; (2) unidirectional (*cardioid*, *supercardioid*, and *hypercardioid*); and (3) *bi-directional.*

An *omnidirectional microphone* responds equally to sound arriving at the microphone from all directions. One feature of an omnidirectional microphone is its smoother off-axis frequency response compared to other microphone directional patterns. A smooth frequency response, with an absence of peaks, helps to improve the sound system gain-before-feedback. The smoother frequency response characteristic can be an advantage particularly if the talker or sound source is not directly "on-mic" or it is desired to pick-up the room acoustic signature.

A cardioid microphone is most sensitive to sound arriving on-axis perpendicular to the front and is less sensitive to sound arriving from the rear. Cardioid microphones are almost universally used in sound reinforcement because these microphones have the capability of rejecting unwanted sound and reverberation, which may help in certain installations to increase the sound system gain-before-feedback. However, this capability diminishes below approximately 500 Hz, where the microphone pick-up becomes more omnidirectional. One particular disadvantage with cardioid microphones is their greater sensitivity to low-frequency sound as the microphone is brought closer to the sound source (*proximity effect*). Often, this build-up of low-frequency sound can degrade the naturalness of the voice, resulting in a "chesty" or "boomy" characteristic. Positioning the microphone remote from the source will decrease its low-frequency output, making the voice sound unnatural, resulting in a "thin" or "nasal" characteristic. Thus, positioning a cardioid microphone relative to the sound source to achieve a balanced natural sound is more demanding than an omnidirectional microphone which does not exhibit the proximity effect.

The supercardioid and hypercardioid microphones share the advantages and disadvantages of the cardioid microphone, but exhibit improved on-axis pick-up performance at greater distances from the source and reduced sensitivity to pick-up of off-axis sound.

The bi-directional microphone is most sensitive to sound arriving on-axis perpendicular to the front and rear, while rejecting sound arriving from the sides. This results in a "figure-of-eight" pick-up pattern, which may be advantageous when talkers are on opposite sides of a table, as in an interview.

(See Technical Notes, Sections 3.A and 3.B, at the end of this chapter, for additional information on microphone polar patterns.)

3.1.4 Microphone Interconnection

Microphones are connected in a three conductor *balanced* configuration, which uses two signal conductors (one for positive (+) signal and the other for negative (-) signal) and one ground (shield) conductor. The audio signal is carried on the two opposite *polarity* signal conductors and are combined at the input to the signal mixer where one of the signal conductors inverted. This method results in greater signal strength and a reduction in noise pick-up. The increased signal level arises since one of the opposite polarity signal conductors is inverted (180° polarity reversal) where it combines in-phase with the other signal conductor resulting in a 6 dB increase. The noise reduction arises when the one signal conductor is inverted, resulting in the noise signals being out-of-phase with each other where they cancel at the signal mixer input. The most common audio connector used for balanced microphone interconnection is the three pin XLR. Other audio components connected in a balanced configuration use a ¼ in phone plug with a tip-ring-sleeve arrangement.

(See Technical Notes, Section 3.C, at the end of this chapter, for additional information on balanced and unbalanced connectors.)

The output *impedance* of microphones is relatively low, on the order of less than 300 Ω (*ohms*), to minimize loading on the interconnected signal mixer.

3.2 Line Level Sources

Line level sources include audio formats such as magnetic tape, audio and optical discs (CD, DVD, and MD), videotape, telephonic/teleconferencing devices, and radio. These sources are classified as recorded audio (magnetic tape, audio discs, optical discs, and videotape) or *real time* audio (telephonic/teleconferencing devices and radio). The output from these sources is distributed to other sound system components such as signal mixers and amplifiers for reproduction or playback.

3.2.1 Magnetic Tape Formats

Magnetic tape provides a means to record and playback audio programs. A variety of tape formats have evolved over the years to satisfy user requirements and technical advances in the medium. The two principal methods of recording and playback are the *analog* and digital formats. Within these formats are a variety of magnetic recording media including: (1) open reel; (2) cassette; (3) DAT; and (4) tape carts.

Open reel tape recorders use reels of open tape, typically ¼ in wide on 7 in diameter spools, to record and playback audio program material. The recorder is a large unit which is stand-mounted. Open reel tape recorders can operate in analog or digital formats. The primary advantages with the open reel tape format are ease of editing and superior audio performance characteristics, particularly if a digital format is used.

Cassette tape recorders use a self-contained tape cartridge to record and playback audio program material in an analog format. Originally developed for dictation use only, the cassette medium has found wide use for high quality audio applications due to proprietary noise reduction circuitry (Dolby B™, Dolby C™, and Dolby S™) to reduce the level of tape "hiss" and self-noise inherent in the recording/playback process. The primary advantages with the cassette tape format are ease of operation and low cost.

DAT recorders are similar to cassette recorders except they record and playback in a digital format, using a smaller cassette-type tape cartridge. The primary advantages with the DAT format are the ease of operation and superior audio performance characteristics compared to analog cassette tape. One disadvantage with the medium is relatively high cost for the tape cartridges.

Tape carts are playback only devices used for prerecorded messages and have applications ranging from retail stores to transportation facility voice announcement systems. These devices can be analog or digital, however the latter format is becoming increasingly popular. A variant is the computer controlled message repeater which provides complete messages from separately recorded and stored words.

3.2.1.1 Basics of Magnetic Recording

Magnetic recording can be performed using analog or digital recording techniques. Analog recording captures the entire audio signal while digital recording captures a series of "snapshots" to approximate the audio signal with high accuracy. The basic components of an analog magnetic tape recorder include: (1) magnetic tape; (2) transport mechanism; (3) tape heads; and (4) equalization circuitry. A digital magnetic tape recorder is similar, but also has analog-to-digital (A/D) and digital-to-analog (D/A) circuitry.

3.2.1.1.1 Analog Recording and Playback

Recording an audio program starts with some form of magnetic tape which stores and permits playback of the recorded audio signal. Magnetic tape consists of a polyester layer to which a magnetic oxide surface coating is bonded to. Within the unrecorded magnetic oxide surface coating is a random orientation of small permanent magnets, called domains. The random orientation of the domains results in an average signal output of zero due to the north and south poles of the domains canceling each other out. However, when magnetized by the audio signal, the north and south poles of the domains are reoriented to correspond to the audio signal with an equivalent output level.

The tape transport mechanism has a series of motors, a capstan with a pinch roller, and guides to move the tape across the magnetic heads. Good quality tape transports will have three motors, each for the supply tape reel, capstan, and take-up tape reel. The tape transport provides tension on the tape which helps maintain contact across the magnetic heads.

Tape recorders have specialized magnetic heads to perform erase, recording, and playback functions. The magnetic heads are transducers and have analogous operating principles to a dynamic microphone. The erase head removes any trace of a previous recording that might be stored on the tape. The recording head imprints the audio signal onto the magnetic tape. The playback head works in a reverse fashion to the record head to recover the audio signal off the magnetic tape. Good quality tape recorders will have three separate heads, while lesser quality machines will have two heads to provide erase and combined recording/playback functions.

The recording head has magnetic pole pieces separated by a very small gap. Coils of wire are wrapped around the pole pieces through which the audio signal (current) to be recorded flows. A magnetic force is generated which flows through the pole pieces and across the recording head gap. The signal flows from one pole piece onto the magnetic tape and back to the opposite pole piece. As the magnetic tape is moved across the recording head, the audio signal is recorded onto the tape. The playback head retrieves the recorded signal across its head gap which causes a magnetic field to be generated in the pole pieces. A current equal to the recorded signal is developed and flows through the coils of wire wrapped around the pole pieces which can then be amplified.

The final signal processing in a tape recorder is some form of equalization to help linearize the recording and playback process. The magnetic energy applied to the tape by the recording head is greater than that retained on the tape itself. Portions of the signal on the tape may be too strong, resulting in saturation distortion while other portions may not be strong enough to reorient the domains. Application of a *bias current* to the recording head helps in the recording process to reduce distortion and increase recorded signal levels. The playback head circuits have a 6 dB/octave output boost which requires a complementary 6 dB/octave response cut during the playback process in order to achieve flat frequency response output.

3.2.1.1.2 Digital Recording and Playback

Digital recording and playback are electro-mechanically similar to analog recording and playback. They are electricially different in that the analog audio signal is first converted into a digital signal prior to storage on magnetic tape and upon playback the digital signal is converted back to an analog signal. The analog signal, as is our decimal number system, uses the base^{10} system. The digital conversion process encodes the audio signal through use of the binary (base^{2}) number system which provides a fast and efficient means of manipulating audio signal data. The base^{2} number system uses a series of "1" (on) and "0" (off) signals to represent binary words which approximate the analog audio signal. Upon playback the system converts the base^{2} digital audio signal data back to base^{10}.

The basic digital audio recording chain includes: (1) low-pass filter; (2) sample-and-hold circuit; (3) A/D converter; (4) signal coding and error correction circuitry; and (5) D/A converter. The analog audio signal first passes through a low-pass filter to remove frequencies which are above one-half the *Nyquist frequency* prior to being *sampled.* A sample-and-hold circuit retains and measures the analog signal level so the A/D converter can process the audio signal. The A/D converter samples the audio signal and assigns an equivalent DC voltage to the nearest amplitude step of the sampled waveform through a process known as *quantization*. After quantization a digital word is generated by the A/D converter to represent the sampled audio signal level. Additional manipulation of the digital words provide data coding, modulation, and error correction. The data coding, typically pulse code modulation (PCM), allows for the most efficient storage of the digital audio signal on the magnetic tape. Due to the extremely high density of the digitally encoded signals, errors due to imperfections in the magnetic tape medium need to be corrected. The most common forms of error correction are data redundancy, which uses parity bits and check codes, and interleaving, which scatters the digital data across the digital bit stream.

Digital signal playback is essentially the reverse of the recording process. The first step restores the digital signal back to its original modulated binary state. The data are then reconverted back to the non-error corrected state and then restored into raw PCM data. The data then pass through a D/A circuit to restore the digital words back to an equivalent analog voltage. A sample-and-hold circuit retains the digital signal so the D/A converter can determine the appropriate voltage level. Finally the signal passes through a low-pass filter to smooth out any non-linear steps created in the conversion process, resulting in a smooth analog audio signal.

3.2.2 Audio and Optical Disc Formats

Audio discs include CD and MD formats while the optical disc format is the DVD, which can also playback audio. The CD and DVD formats are primarily used for playback but the MD format can record and playback. Special versions of the CD and DVD formats can be used for recording, but are not normally found in sound system applications.

3.2.2.1 CD Playback

The CD, developed jointly by Philips Company and Sony Corporation in the early 1980s, was the first commercially available digital audio format. It consists of a prerecorded 4¾ in diameter aluminized disc which rotates at a variable speed. The aluminized disc contains a series of microscopic "pits" encoded in a 16 bit 44.1 kHz sampled digital audio signal format. The pattern of the recorded pits on the disc is equivalent to the original analog signal. The CD storage capacity is 680 Mbyte and playback times up to 74 minutes for stereophonic and 148 minutes for monophonic programs are possible.

The pits are read by a small laser which focuses on the aluminized disc layer reflecting light back to a photo detector. The distance between the disc surface and the laser is continuously adjusted by a servomotor system to provide the best retrieval of the digitized pits. The output signal from the photo detector is digital and it must be converted back into analog by a D/A converter and associated circuits.

3.2.2.2 DVD Playback

The DVD, developed in the late 1990s, is a digital optical storage medium identical in appearance to a CD but is able to hold 15 times more data. The electro-optical playback mechanism of the DVD is very similar to the CD but has a transfer rate 20 times faster. The storage capacity is about 8.5 Gbyte and program material playback times up to 4 hours in length are possible. Audio data is stored in a 24 bit 98 kHz digital format permitting 5.1 channel surround sound playback.

The DVD can be used for playback of prerecorded audio and video images when used with an appropriate player or a computer. The DVD, with its enhanced storage capacity and better audio performance, will eventually become the format of choice and replace both CD and videotape.

3.2.3 MD Recording and Playback

The MD is a digital audio recording and playback format developed by Sony Corporation in the mid 1990s as a competing audio format for cassette tape. The recordable MD is slightly smaller than a 3½ in floppy disk. Recording times are up to 74 minutes for stereophonic and 148 minutes for monophonic programs are possible.

The recording and playback is a digital magneto-optical process. A laser heats up the disc surface causing demagnetization. The recorder applies a magnetic field that remagnetizes the disc particles to etch a 16 bit 44.1 kHz digital audio signal onto the disc. During playback the laser refocuses on the disc surface and registers changes in polarization to retrieve the recorded signal.

The MD format has the capacity to store approximately 140 Mbyte of data, which is a fairly small storage capacity for audio program material. A process called Adaptive TRansform Acoustic Coding (ATRAC) is used to improve disc storage density. The ATRAC process divides the digital audio signal into 512 frequency sub-bands, with smaller frequency sub-bands at lower frequencies. A "lossy" data omission and compression algorithm, based on *psycho-acoustical* sound masking and hearing threshold criteria, is applied to the audio signal which enables approximately 80 percent of the actual audio signal to be deleted in the recording process. Critical listening studies have shown the MD format with newer generations of ATRAC circuitry to be audibly similar to CDs, although not identical.

3.2.4 Videotape Recording and Playback

Videotape, developed in the mid 1970s, can record and playback prerecorded audio signals. The standardized videotape format is the ½ in video high speed (VHS) cassette. Recording and playback is through a video cassette recorder (VCR). Both single-channel monophonic and two-channel stereophonic formats are available, with the latter being the most common for prerecorded media.

Recording an audio signal on the same magnetic tape used to store the video images is an analog process. It involves compressing the audio signal to reduce the dynamic range and then modulating the audio signal with a carrier frequency prior to storage. The signal is expanded upon playback to restore its original dynamic range.

3.2.5 Telephone/Teleconferencing Devices

Transmission of audio information can occur through the telephone or through a dedicated teleconferencing system. For good communication at both the send and receive ends, it is essential that each of these systems provide *full duplex* audio.

3.2.5.1 Telephone System

Local telephone service uses a two wire system between each send or receive location and the central telephone substation. A four wire system is used within telephones with two wires each dedicated for sending and receiving the telephonic signals. At the central telephone office, the telephonic signals are routed to other central telephone offices via the four wire system using land lines, microwave, or satellite transmission. Thus, the telephone system has completely separate send and receive paths.

Connecting a telephone to an audio system requires using a two-to-four wire digital telephone hybrid between the telephone output and the audio system input. The telephone hybrid isolates the telephone send and receive lines and prevents feedback between the two telephonic transmission paths. Various manufacturers make telephone hybrid and other interfacing devices.

3.2.5.2 Teleconferencing System

In its most simple form, teleconferencing can be accomplished via a speaker phone. Improved audio transmission and reception can be realized with a complete teleconferencing system installed at each send and receive location. The teleconferencing is referred to as point-to-point when only two communication links are established. Multiple communication links are referred to as multi-point.

Each teleconferencing system uses dedicated microphone(s) to pickup the talkers, line level audio sources, a signal mixer, and signal processing equipment to modify the audio signal before being routed to a *codec*. The codec encodes the signal which is then routed to a telephone land line, microwave transmitter, or more commonly, a satellite transmission system. At the receive end a complementary codec decodes the signal and routes it to the audio system. Teleconferencing systems have the capability of integrating graphics and video with the audio signal when suitable projectors and cameras are used.

3.2.6 Radio

Radio broadcasts, primarily FM, cable-type radio services, or MUZAK® can be used as an input to a sound system. Several manufacturers produce professional-grade FM radio receivers which can be connected to a signal mixer. The cable type radio is a pay-for-service system which requires installation of the service provider's proprietary reception decoding box.

3.2.7 Line Level Source Interconnection

Line level sources are usually connected in a balanced configuration, although *unbalanced* connections are common, particularly with consumer-grade audio equipment. Unbalanced lines use one signal conductor and one ground conductor. The audio signal is carried on the signal conductor only. The main disadvantage with unbalanced connections is the reduced immunity to noise pick-up. The most common audio connectors used for unbalanced interconnection are the ¼ in phone plug and the phono (RCA) plug, both with a tip-sleeve arrangement.

The output impedance of line sources is relatively high, on the order of greater than 10,000 Ω, to minimize loading at the inputs of interconnected equipment.

Special signal converters are available which provide signal level and balancing functions to enable interconnection of equipment having different signal levels and wiring formats, as occurs when using consumer-grade and professional-grade equipment.

3.3 Signal Mixers

Signal mixers combine the electrical outputs of microphones, line level sources, or combinations of these devices, adjust the relative level between the sources, and route the electrical signals to other sound system equipment.

Signal mixers are commonly used in multiple input source applications where they activate microphones being used, mute or turn down the level of unused microphones, turn on and off line level sources, increase or decrease gain between channels, and increase or decrease the output level to improve gain-before-feedback characteristics.

Programs such as music, drama, or panel discussions may use numerous microphones which can complicate the performance requirements of the signal mixer. The sound system designer or operator must carefully select the number of active microphones based on the number of sound sources and their proximity to the microphones. Using too few microphones can result in decreased sound levels and reduced intelligibility for sources remote from the nearest microphone, while sources close to the microphones can be overly loud by comparison. Decreasing the distance between the microphone and the sound source by one-half will result in 6 dB greater signal level at the microphone. On the other hand, using too many microphones may result in the pick-up of room noise and reverberation, reducing the sound system gain-before-feedback. Eight open microphones will result in the pick-up of 9 dB more room noise and reverberant signal, which will be perceived approximately twice as loud, than if only one microphone were used.

Each doubling of the number of open microphones will decrease the sound system gain-before-feedback by 3 dB. Thus, as the number of open microphones is increased, the overall output level of the sound system must be lowered to maintain the same relative freedom from feedback. The goal is to use the fewest number of microphones, consistent with program functions. Doing so will result in improved intelligibility, a louder signal, and less likelihood of the sound system going into feedback.

A byproduct of multiple microphones at different distances to a common sound source is the potential for *comb filtering* effects when the microphones are operated at the same gain level. The time delayed sound can be picked-up by the different microphones and when mixed will result in a subjectively "hollow" audible quality, due to the phase interference between the signals.

(See Technical Notes, Section 3.D, at the end of this chapter, for additional information on comb filter effects.)

3.3.1 Classification of Signal Mixers

Signal mixers are broadly classified as being either manual or automatic in their operation. The term "signal mixer" is a generic designation for a mixer, console, mixing console, or automatic mixer. A final variant is the powered mixer which combines a manual mixer with a power amplifier onto a single chassis.

Manual mixers are sometimes referred to as mixers, consoles, mixing consoles, or mixing desks. These terms are often used interchangeably, but each has specific component features. A mixer is a small unit which has 12 or fewer inputs and can be placed on a desktop or permanently installed in an equipment rack. A console is a shortened abbreviation for mixing console. A mixing console or mixing desk has 16 or more inputs and is a large free standing unit intended for placement on a desktop. An automatic mixer is a small unit with eight or fewer inputs intended for permanent installation in an equipment rack. These mixers usually have a "link" circuit so that several units can be interconnected when more than eight inputs are required. Powered mixers have between two and 12 inputs and are intended for permanent installation in an equipment rack for portable desktop use.

Manual mixers are used primarily in music and theatrical applications where operator flexibility and artistic control of the complex dynamic and tonal aspects of speech and music signals are of paramount importance. Automatic mixers are used primarily in boardrooms, classrooms, and similar spaces where speech is the primary sound source and user control of the audio signal is not necessary. The use of an automatic microphone mixer should be considered mandatory when four or more microphones are to be used and no sound system operator is to be present. Powered mixers are used mostly in low-budget installations such as small assembly spaces where speech and line level sources are the primary signal inputs and minimal user control of the audio signal is required.

3.3.2 Manual Mixers

Manual mixers require hands-on control by an operator to select signal source inputs, turn on/off microphones and line level sources, adjust relative levels between different signal sources, tailor frequency characteristics, adjust the final sound system output level, and route audio signals to other sound system equipment.

In their most basic form, manual mixers include signal inputs and preamplifiers, a program mixing *buss*, a master output section, and metering. Other features common on more complex manual mixers include microphone phantom powering, frequency equalization, an auxiliary buss, input returns/output sends from the auxiliary buss, and monitoring/*foldback* capabilities.

3.3.2.1 Signal Flow Through a Manual Mixer

Manual mixers are probably the most complicated piece of equipment in a sound system. The number of knobs, buttons, and switches on a manual mixer can be truly daunting, but many of their functions are identical for the different audio signal channels, which helps to simplify learning. Figures 3-7 and 3-8 show small and large manual mixers.

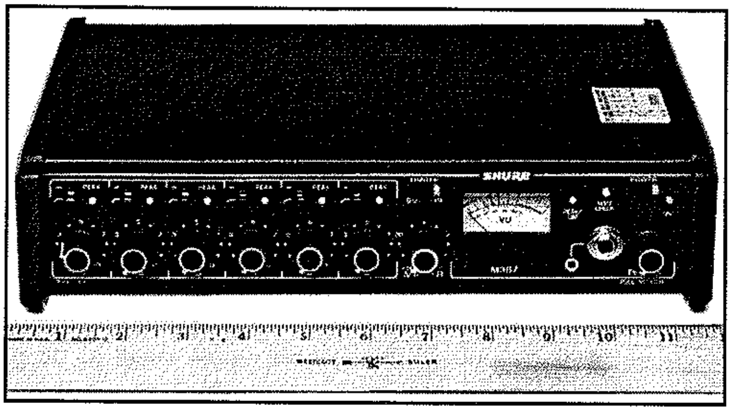


FIGURE 3-7. Six-channel manual mixer (Shure M367). Product courtesy of Shure Brothers, Inc.

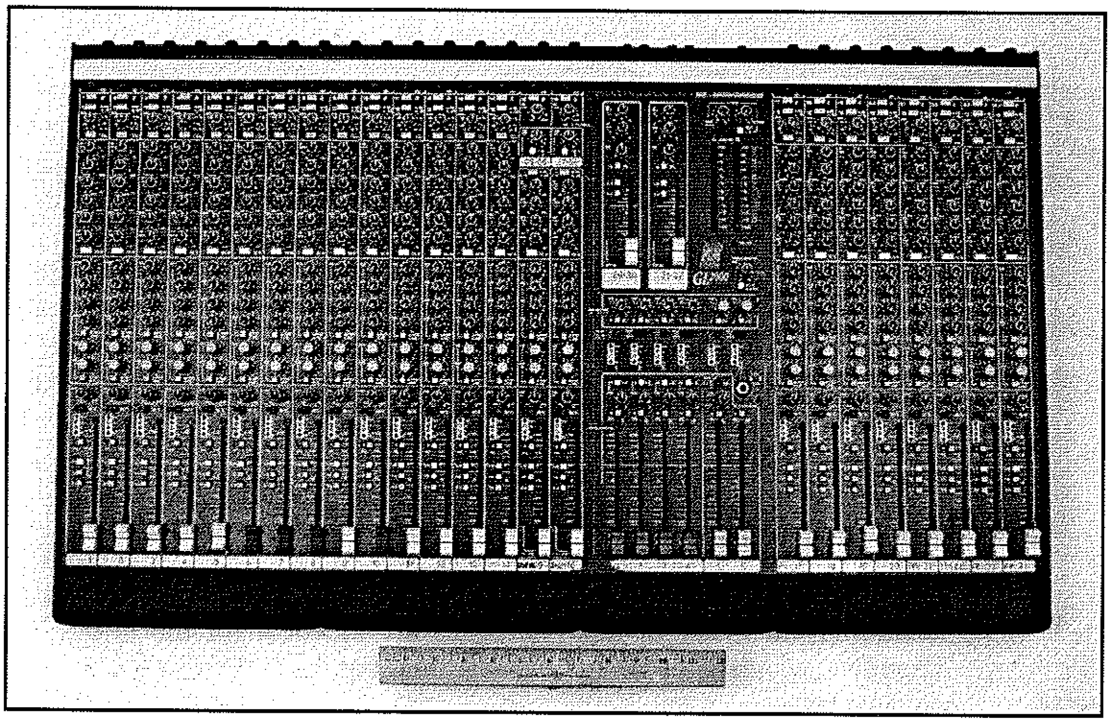


FIGURE 3-8. 24-channel mixing console (Allen & Heath GL2000). Product courtesy of Allen & Heath.

Each audio signal is carried via a cable connected to a dedicated channel input at the back of the manual mixer. Phantom powering for microphones is carried from the manual mixer by the cable back to the microphone. High-level signals, which might overload the preamplifier or other mixer amplification stages, can be attenuated by as much as 20 dB with an input *pad*. Input signal levels are monitored by meters or light emitting diodes (LEDs) to indicate their strength. Microphone level signals are amplified approximately 100,000 times by a preamplifier to increase the signal magnitude and minimize susceptibility to noise. Line level signals, which are higher in level than microphone level signals, bypass the preamplifier. The signal is then routed to an optional frequency equalizer, a *fader* for adjusting relative levels between the input signals, a line amplifier, and to a *pan pot* for positioning the signal to create a sense of auditory spatial impression.

From the pan pot of each channel, the signals are routed to the program mixing buss where they are combined into a composite audio signal. The mixed signal can be routed to the auxiliary buss, for further signal processing, or to the master output section. The master output section has summing amplifiers, faders, and line amplifiers to adjust the final output level routed to the sound system amplifiers. The signal output magnitude is monitored with meters or LEDs. Signals from the auxiliary buss can be routed to output sends, for special effects signal processing, and back to the auxiliary buss via input returns. Other possible routing from the auxiliary buss can be to a tape recorder, assistive listening system, stage monitoring/foldback systems, or other specialized subsystems. Monitoring of the output send and input return levels is done with meters or LEDs.

When describing manual mixers it is common to make reference to the number of input channels and the number of output channels. Sometimes different output channels are grouped together into a common mixing buss output. Thus, a 16 x 8 manual mixer would have 16 input channels and eight mixing buss outputs. Two of these outputs might be assigned to the master output section and six assigned as special effects sends/returns. More complicated manual mixers may have *submix* capabilities. For example, the eight busses could be further mixed into a two-channel stereo mix. Such a manual mixer would be described as a 16 x 8 x 2 device.

Another common feature on manual mixers is the way signal attenuation is handled by the auxiliary busses. Post-fader type auxiliary busses are designed to be affected by the channel fader. This arrangement is often used for signal processing effects loops, where the processed signal level can be adjusted to suit the desired artistic intent. Pre-fader type auxiliary busses are not affected by the channel fader. This arrangement is useful in monitoring/foldback.

3.3.3 Automatic Mixers

Automatic mixers, once adjusted and calibrated, will function without operator control. These devices will smoothly switch on/off microphones and line level sources, increase or decrease input signal levels to maintain a constant sound output regardless of the talker's vocal effort, adjust the final output level, and route auxiliary audio signals to other sound system equipment. Some devices have the capability to provide priority muting of channels, *mix-minus* matrixing, and last microphone on features. Automatically activating microphones and adjusting the system gain to account for the number of open microphones are the two most important functions that automatic mixers provide.

The basic components of an automatic mixer includes signal inputs and preamplifiers, phantom powering, a program mixing buss, automatic threshold and *gating* circuits, a master output section, and metering. Other features common on more complex automatic mixers include frequency equalization, compression/limiting, auxiliary inputs/outputs, and provisions for computer control. Special logic control circuits are available on some automatic mixers to provide indication of microphone status, loudspeaker attenuation, and selecting priority channels. One particularly useful feature is an adjustable hold time circuit which keeps the channel momentarily open when the sound level decreases to avoid gating the channel completely off which reduces the possibility of losing spoken words or syllables.

Various operational philosophies and circuit designs have evolved to activate microphones, control the method of raising the input channel gain, dynamic gating and envelope action of the input channels, and the method for gain sharing based on the number of open microphones. In the most basic of automatic mixers, microphones are gated on when the sound level at the microphone exceeds a reference threshold level set within the automatic mixer. As the sound level falls below the threshold, the microphone is shut off. Automatic mixers use fixed, adjustable, and automatically adjustable gating circuits. Some automatic mixers use special directional-sensing microphones which gate microphones on based upon sound reception from a desired angular direction.

Use of automatic mixers tends to be more successful in relatively non-reverberant spaces since reverberant sound can negatively impact operation of some automatic mixers. Directional microphones are recommended to be used with automatic mixers to minimize false triggering from off-axis sources and to reduce the pick-up of audience noise incident at the rear of the microphone.

(See Technical Notes, Section 3.E, at the end of this chapter, for additional information on automatic microphone mixing circuits.)

3.3.3.1 Signal Flow Through an Automatic Mixer

The basic signal flow concepts in manual mixers are applicable to automatic mixers and major differences will be described below. Variations in signal flow through automatic mixers will vary among different manufacturer's products. Figure 3-9 shows two automatic mixers.

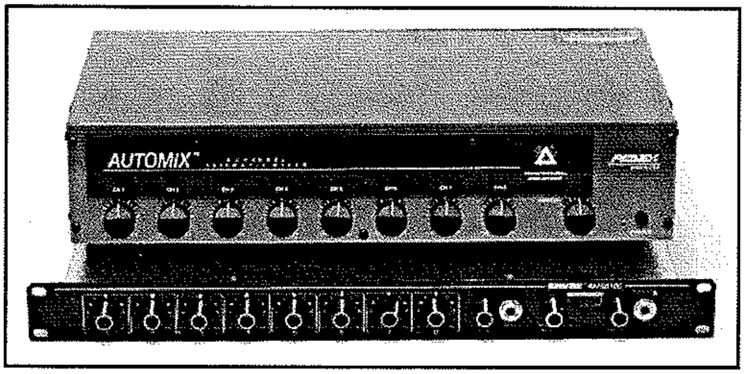


FIGURE 3-9. Automatic microphone mixers (Peavey Automix™ top and Shure SCM810 bottom). Products courtesy of Peavey Electronics Corporation and Shure Brothers, Inc.

The signal output from the preamplifier of each input channel is routed to an optional high-pass filter and frequency equalizer. From here the signal is routed to adjustable gating and level control circuitry used to turn on/off microphone and line level sources. The outputs from each channel are routed to the program mixing buss and then to a linking network which receives the output from other automatic mixers. These signals are summed and routed to an optional compressor/limiter. The composite signal is routed to both the master output and auxiliary output sections. Monitoring of signal levels is through front panel-mounted LEDs.

3.3.4 Powered Mixers

Powered mixers contain separate mixer, signal processing, and power amplifier sections on a single chassis. Most are intended for permanent installation in equipment racks, although desktop units are available. The mixing section uses a manual mixer with up to 12 input channels for both microphone and line level interconnection. The signal processing section contains limited frequency equalization, often only bass and treble tone controls, although some units have built-in six band octave frequency equalizers. The power amplifier section has ratings between 35 to 150 electrical watts (W) per channel. Powered mixers are a good choice for sound systems having a restricted budget and minimal performance requirements. Figures 3-10 and 3-11 show powered mixers.

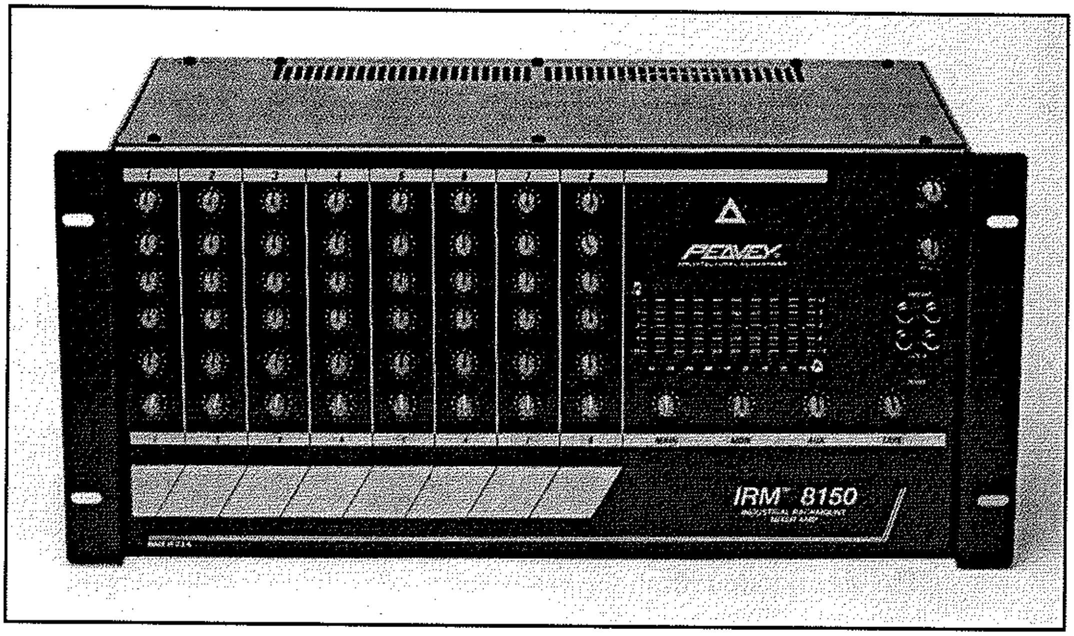


FIGURE 3-10. Eight-channel 150 watt/channel powered mixer (Peavey IRM™ 8150). Product courtesy of Peavey Electronics Corporation.

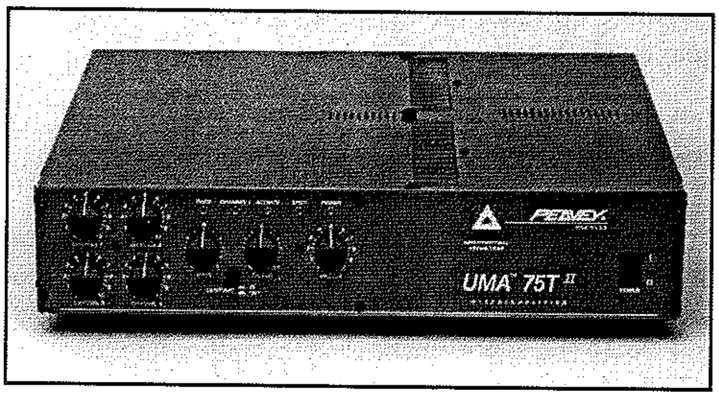


FIGURE 3-11. Four-channel 75 watt/channel powered mixer (Peavey UMA™ 75TII).Product courtesy of Peavey Electronics Corporation.

3.4 Audio Distribution, Buffer, and Summing Amplifiers

Audio distribution, buffer, and summing amplifiers provide a variety of signal routing, mixing, and level conversion functions. These devices can distribute a common audio signal to other sound system components, combine different audio signals into a composite signal, match consumer (-10 dBV) signal levels to professional (+4 dBu) signal levels (and vice versa), and convert unbalanced signals to balanced signals (and vice versa).

The distinction between audio distribution, buffer, and summing amplifiers is not sharply delineated among the manufacturers. A product by one manufacturer may be functionally identical to another manufacturer's product even though both have different descriptive names. When selecting equipment it is best to check product literature to determine equipment functional features rather than relying solely on descriptive product names.

Strictly speaking, an audio distribution amplifier routes signals to different locations, a buffer amplifier provides unity voltage gain and is intended to isolate audio components from interconnection loading effects, and a summing amplifier mixes several inputs into one or more outputs. Some common applications of these devices are described below.

1. An audio distribution amplifier can be used at the output of a signal mixer or crossover to distribute a common audio signal to different sound system components.

2. A summing amplifier at the input of a signal mixer can convert the left and right outputs of a consumer tape recorder to +4 dBu and sum the two channels into a single channel output.

3. A buffer amplifier can be inserted between two components to isolate different input and output impedances.

4. An audio distribution or summing amplifier can be used in a basic sound system where the expense of a signal mixer might not be warranted, but it is still necessary to route or combine different audio signals.

Figure 3-12 shows different audio distribution, buffer, and summing amplifiers.

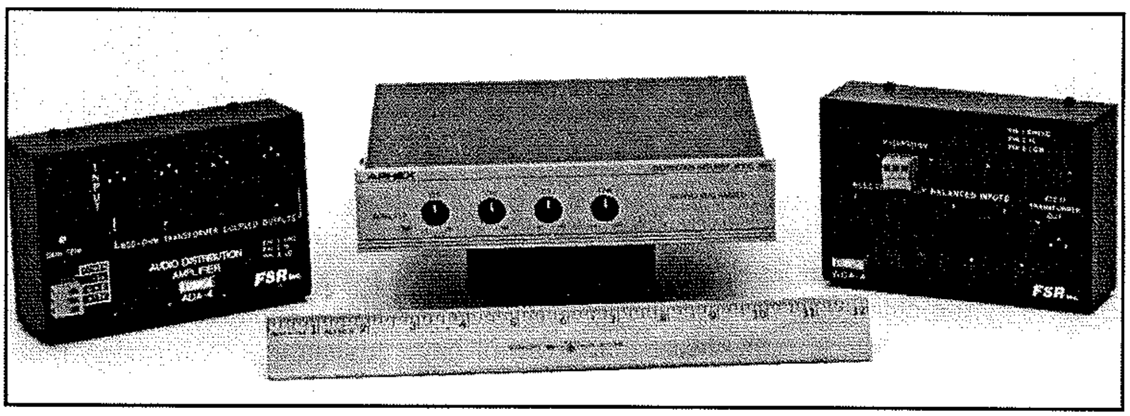


FIGURE 3-12. Audio distribution, buffer, and summing amplifiers (FSR ADA-4 left, Aphex Model 120A center, and FSR WCA-4 right). Products courtesy of Aphex Systems and FSR, Inc.

3.5 Signal Processing Equipment

Signal processing equipment modifies the electrical audio signal in various ways to provide compression/limiting or expansion of dynamics, adjustment of frequency characteristics, time delay, and subdivision and routing of signals having different frequency bandwidths. Some signal processing components provide only one function. Others combine several signal processing functions, by using on-board *digital signal processing* (*DSP*) chips, while the latest trend is to implement multiple signal processing functions through computer controlled DSP systems.

3.5.1 Dynamics Processors

Dynamics processors are signal processing devices that affect the *dynamic range* of the signal and include compressors, limiters, and expanders. Compressors and limiters decrease the signal dynamic range and expanders increase the signal dynamic range. Dynamics processors can provide one or multiple dynamic range enhancement signal processing functions. Figure 3-13 shows a combination compressor/leveler/limiter.

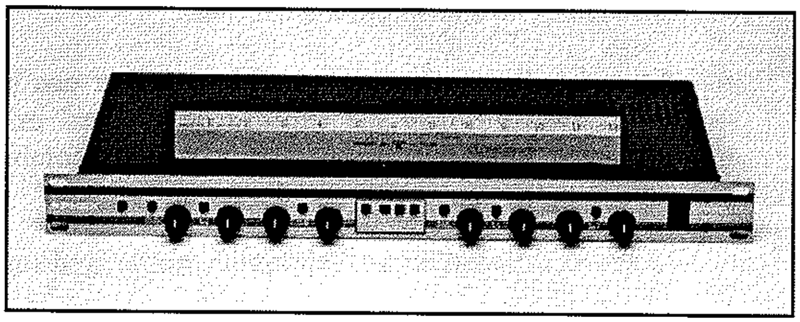


FIGURE 3-13. Compressor/leveler/limiter (Aphex Compellor®). Product courtesy of Aphex Systems.

3.5.1.1 Compressors

A compressor gradually reduces the signal output level above a threshold point as the signal input level increases. These devices work by monitoring the *root-mean-square* (*RMS*) signal level, which corresponds to the perceived signal loudness. The ratio of the change in the signal output level to the signal input level (input:output) is the compression ratio. Most compressors have adjustable compression ratios between 1.5:1 and 4:1. For example, a 3:1 compression ratio would result in an input signal level increase of 3 dB having a 1 dB increase in output level. The audible effect of compression is to make the loud portions of the program quieter.

Compressors can be used to limit the overall sound system output level or to selectively reduce the output of certain input signals. The former application would include reducing the dynamic range of the signal so a sound system can produce higher sound levels in a noisy environment before overload distortion becomes audible. The latter application would include reducing the output of a vocalist's microphone, who might be 5 to 15 dB higher in level, relative to accompanying vocalists or instrumentalists.

Adjustment of compressors involves setting the *attack time* (to reduce the level) and the *release time* (to restore the level) at the threshold point based on the anticipated signal content. Fast attack times (less than 20 ms) and slower release times (2 to 3 s) sound the most natural. Figure 3-14 shows typical compression and limiting response characteristics.

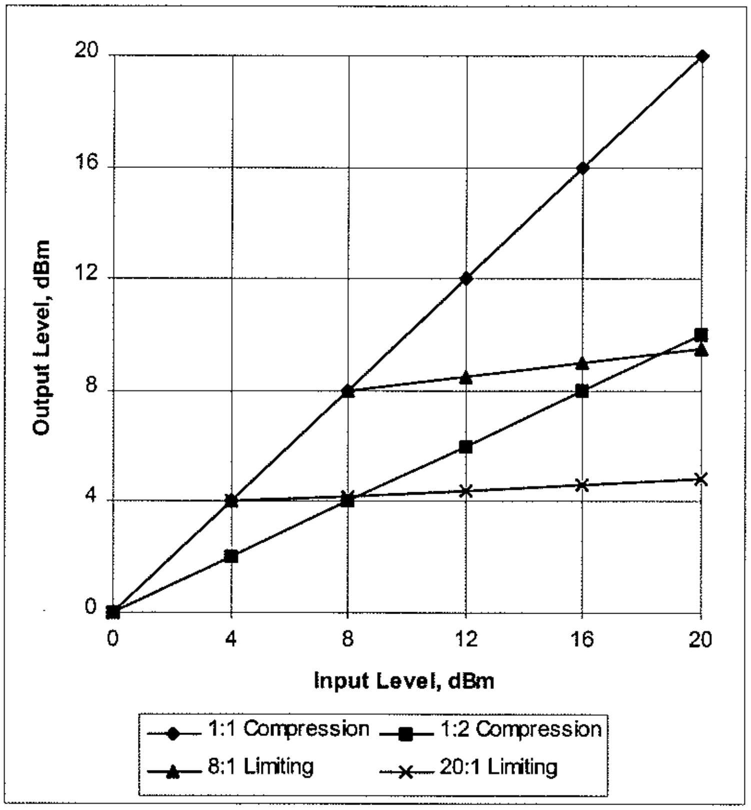


FIGURE 3-14. Typical compression and limiting response curves showing relationship between input and output levels.

3.5.1.2 Limiters

Limiters are similar to compressors with the exception that they use higher compression ratios, between 5:1 and 20:1, and respond to the signal peaks. The purpose of a limiter is to prevent the signal from exceeding a predefined threshold level, above which the signal output can not increase. In this regard, this device can be referred to as a leveling amplifier.

Limiters are usually adjusted to have faster attack and release times than compressors since the intent is to prevent short-term momentary or transient signal overload.

3.5.1.3 Expanders

Expanders can be considered to be a compressor operating in reverse and are used to increase the dynamic range of a low-level signal by reducing the signal level below the threshold point. Most expanders have gentle expansion rates between 2:1 and 4:1. A frequent use of an expander is for noise control. The audible effect of expansion is to make the quiet portions of the program quieter. Figure 3-15 shows typical expansion response characteristics.

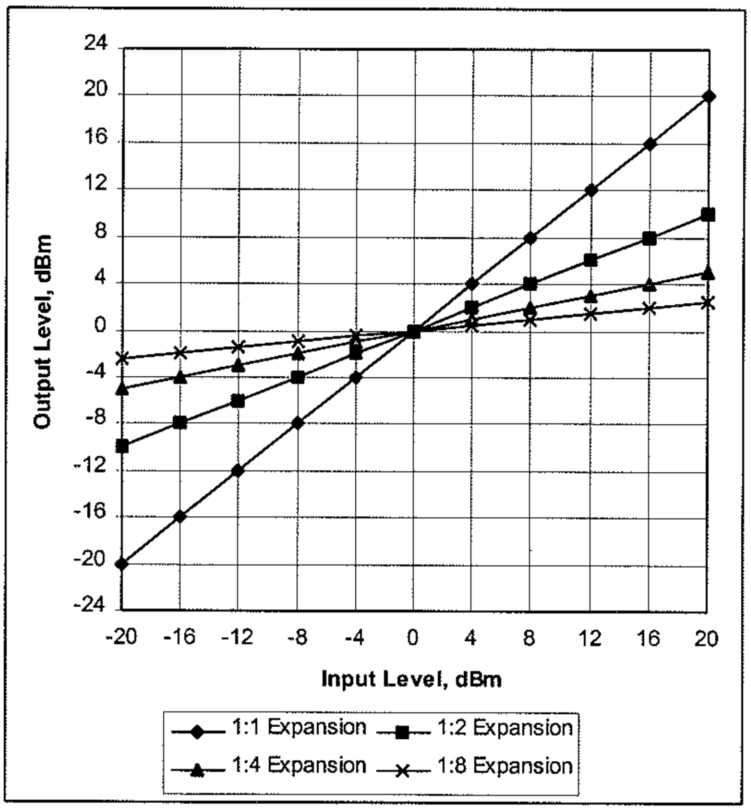


FIGURE 3-15. Typical expansion response curves showing relationship between input and output levels.

A special type of expander, with greater expansion ratios, is a noise gate. This device prevents unwanted noises, as can occur with an open microphone, from entering the sound system by reducing the output to zero, once the signal has dropped below the threshold level.

3.5.2 Frequency Equalizers

Frequency equalizers adjust the relative balance between low-, mid-, and high-frequency portions of the audio signal by boosting or attenuating the signal amplitude. An electrical filter, or more commonly, a series of electrical filters are used to construct a frequency equalizer. The filter design uses minimum phase networks so the *phase shift* associated with the amplitude correction does not result in excessive all-pass *group delay*.

As noted above, signal mixers often have limited frequency equalization capabilities, however separate frequency equalizers can provide greater control of sound system frequency characteristics. Frequency equalizers are classified as broadband or narrowband devices. Broadband equalizers are used to smooth out the sound system *power response*. Narrowband equalizers are used to adjust specific frequencies. Frequency equalizers can be further subdivided into shelving, graphic, parametric, and notch types. Shelving equalizers are broadband devices. Notch equalizers are narrowband devices. Graphic and parametric equalizers can have both broadband and narrowband frequency equalization characteristics. Regardless of the type of frequency equalizer used, careful adjustment through precise electro-acoustical measurements is necessary to optimize full performance capabilities.

Frequency equalizers are commonly mistaken as a panacea for improving all sound system performance problems. The user must be able to distinguish between what these devices can and can not do. Common applications and limitations of frequency equalizers are described below.

1. Frequency equalizers are most useful in modifying the loudspeaker direct sound output. Doing so will provide a more evenly balanced sound in the audience seating area. Modifying the direct sound will affect the spectral content of room surface sound reflections which will to a lesser degree affect the reverberant sound.

   Nonuniform loudspeaker coverage, poor loudspeaker-to-room interface, or room standing waves can not be corrected through frequency equalization.

2. Minor variations in mid- to high-frequency response due to the effects of surface boundary absorption near the loudspeaker can be corrected.

   Major modifications to the room reverberant sound, so-called "tuning" a room, are not possible with a frequency equalizer since the room acoustical properties include both time and frequency domain phenomena. Frequency equalizers can not correct for time domain phenomena. For example, a notch in the frequency response of a loudspeaker caused by the direct sound combining with reflected sound can not be corrected with a frequency equalizer.

3. Minor deficiencies in loudspeaker frequency response can be corrected and the loudspeaker frequency range can be moderately extended or flattened. Horn loudspeakers start to roll-off between 3,000 and 4,000 Hz and cone loudspeakers start to roll-off between 5,000 and 6,000 Hz. Special equalization networks have been developed to compensate for directivity characteristics of certain loudspeakers, such as *constant directivity horns*, which have a relatively constant directivity factor versus frequency characteristic, requiring boosting above approximately 3,000 Hz for flat power response.

   Frequency equalizers can not provide major modification to loudspeaker frequency response, overly correct for poorly designed devices, or compensate for loudspeakers operating beyond their intended performance limits.

4. Maximizing gain-before-feedback using narrowband frequency equalizers can improve sound system performance by attenuating feedback frequency *ring modes*. The narrowband filters used for feedback control are based on the *transfer function* between the relative position between the microphone and loudspeaker.

   Making a sound system free of feedback using narrowband equalization is not possible due to the unstable nature of feedback frequencies caused by changes in position between the microphone and loudspeaker and the varying signal spectral content. Newer automatic feedback controllers, which automatically detect feedback frequencies and insert a narrowband filter at that frequency, have improved ability to dynamically control feedback.

5. The overlapping of the frequency filter bandwidth *skirts* makes precise frequency equalization adjustment difficult. Equalization in one frequency band will affect the response in adjacent lower and higher frequency bands. This characteristic is more apparent with equalizers having wider frequency filter bandwidth skirts. Perfect equalization is not possible and compromises in the resultant equalization characteristic must be accepted.

(See Technical Notes, Section 3.F, at the end of this chapter, for additional information on frequency equalization practices.)

3.5.2.1 Shelving Equalizers

Shelving equalizers provide basic tone control functions at low- and high-frequencies with up to 12 dB of boost or cut. These devices provide little control at mid-range frequencies between approximately 250 and 2,500 Hz. The *knee-point* can be either fixed or adjustable. The fixed low-frequency knee-point is typically around 100 Hz. The fixed high-frequency knee-point is typically around 10,000 Hz. Adjustable shelving filter knee-points are variable over a range of several octaves.

A high-pass filter provides attenuation of low-frequencies while permitting higher frequencies to "pass" through the equalization network. Similarly, low-pass filters provide attenuation of high-frequencies permitting low-frequencies to "pass" through the equalization network. Figure 3-16 shows typical shelving equalizer response characteristics.

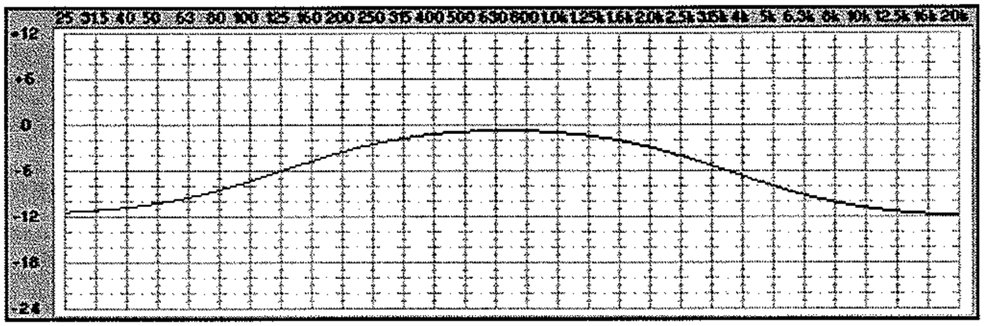


FIGURE 3-16. Typical shelving equalizer curves from Shure DFR11EQ showing relationship between frequency and cut characteristics. The low-cut shelving starts at 200 Hz and the hi-cut shelving starts at 2,000 Hz. Both low- and high-frequency responses roll-off at 12 dB/octave. Note relatively smooth frequency response between 200 and 2,000 Hz. Data after Shure Brothers, Inc.

3.5.2.2 Graphic Equalizers

The most common frequency equalizers are one-third octave band and two-third octave band graphic equalizers. These devices provide up to ±12 dB boost and cut of signals, with some devices providing a cut function only. The frequency equalization filters are on International Standards Organization (ISO) band center frequencies. One-third octave band equalizers have 15 to 31 fixed frequency bands between 31.5 and 16,000 Hz. Two-third octave band equalizers have 8 to 16 fixed frequency bands between 31.5 and 16,000 Hz. Many graphic equalizers also include fixed or adjustable low-pass and high-pass shelving filters. Figure 3-17 shows two graphic equalizers.

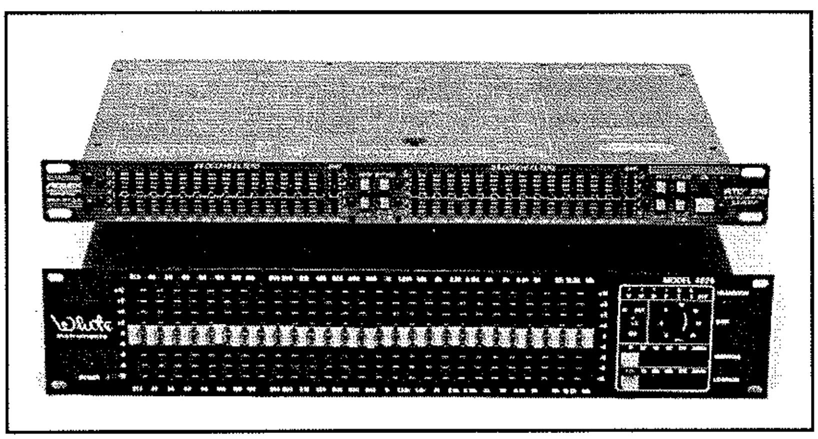


FIGURE 3-17. One-third octave band graphic frequency equalizers 15-band two-channel type (Peavey RTD™ 215 top) and 28-band single-channel type (White 4828 bottom). Products courtesy of Peavey Electronics Corporation and White Instruments.

One-third octave band graphic frequency equalizers are flexible enough to provide precise frequency adjustment to tailor frequency response to the desired equalization characteristic. Two-third octave band graphic equalizers are too broad in their response characteristics for precise frequency control but are useful for adjusting the overall power response and tonal characteristics of the sound system.

Both graphic frequency equalizer types are available with fixed or variable filter bandwidth *Q* characteristics. The latter type varies the bandwidth based on the amount of boost or cut. Narrower bandwidths, characteristic of greater boost or cut settings and higher **Q**, resulting in sharper frequency response curves.

Another characteristic of graphic frequency equalizers is how the adjacent band filters interact. Equalizers with combining filters have a smoother transition to adjacent frequency bands which interact to produce a smoother resultant response, compared to equalizers with non-combining filters. Figure 3-18 shows typical graphic equalizer response characteristics.

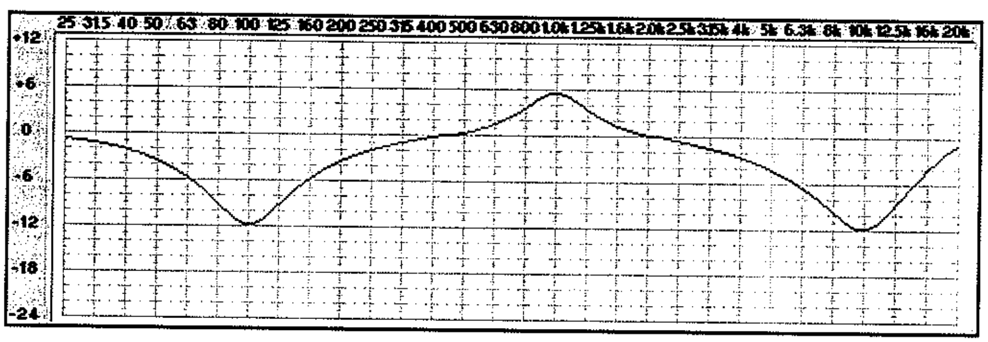


FIGURE 3-18. Typical one-third octave band graphic frequency equalizer curves from Shure DFR11EQ showing relationship between frequency and boost and cut characteristics. The frequency cuts are at 100 and 10,000 Hz with 12 dB attenuation at each frequency. The frequency boost is at 1,000 Hz with 6 dB increase. Note overall interaction of different frequency bands. Data after Shure Brothers, Inc.

Finally, some graphic frequency equalizers are available as cut-only devices. In many sound reinforcement applications these devices are preferred since they introduce less noise by cutting frequency bands outside the desired program frequency range, thus increasing the S/N ratio. Some sound system designers feel it is better to cut frequency peaks than to boost adjacent frequency dips to equal the frequency peaks, since the boosting may potentially cause signal overload and excess phase anomolies.

3.5.2.3 Parametric Equalizers

Parametric equalizers are "tunable" devices having adjustable center frequencies, variable *passband* **Q**, and controllable boost and cut features. Typically, three to five separate filters are included to provide equalization for a particular frequency range. Parametric equalizers can provide more precise control of problem frequency regions than graphic equalizers. Broadband boost or cut over several octaves or narrowband boost or cut over a fraction of an octave can be achieved with a parametric equalizer.

Since both the frequency and **Q** are adjustable, a minimum amount of boost or cut can be used, resulting in less interaction with adjacent frequency bands. When adjusted to a low **Q** setting, the parametric equalizer provides broadband frequency tailoring. Higher values of **Q** are useful for notch filtering. Figure 3-19 shows typical parametric equalizer response characteristics.

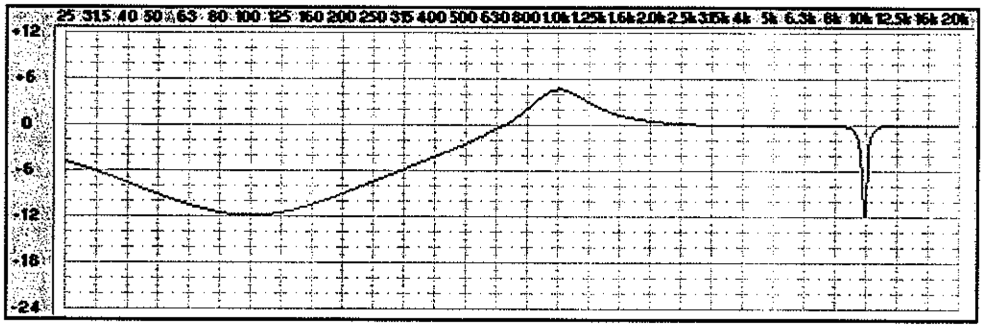


FIGURE 3-19. Typical parametric equalizer curves from Shure DFR11EQ showing relationship between frequency and boost and cut characteristics. The frequency cuts are at 100 Hz with 12 dB attenuation over a 2 octave bandwidth and 10,000 Hz with 12 dB attenuation over a 1/40 octave bandwidth. The frequency boost is at 1,000 Hz with 6 dB increase over a 2/3 octave bandwidth. Note less interaction of different frequency bands compared to one-third octave graphic equalizer. Data after Shure Brothers, Inc.

Common design practice is to use both graphic and parametric equalizers for optimizing sound system performance. The graphic equalizer is used to adjust the overall power response characteristic and the parametric equalizer is used to provide precise adjustment of selected frequency regions, either for electro-acoustical control or artistic effect.

3.5.2.4 Narrowband Equalizers

In the 1950s the late acoustician Paul Boner developed the concept of frequency selective notching to improve sound system gain-before-feedback by using tuned narrowband filters inserted between the signal mixer output and the power amplifier input. This concept was refined by Don Davis in 1967, then at Altec-Lansing, where the "Acousta-Voicing" equalizer was developed.

The bandwidth of these filters is extremely narrow, on the order of less than 5 Hz, and their attenuation characteristics can approach 50 to 60 dB. Narrow band equalizers are cut-only devices and can be separate adjustable filter sets or hardwired circuits specific to the sound system ring modes. Figure 3-20 shows typical narrowband equalizer response characteristics.

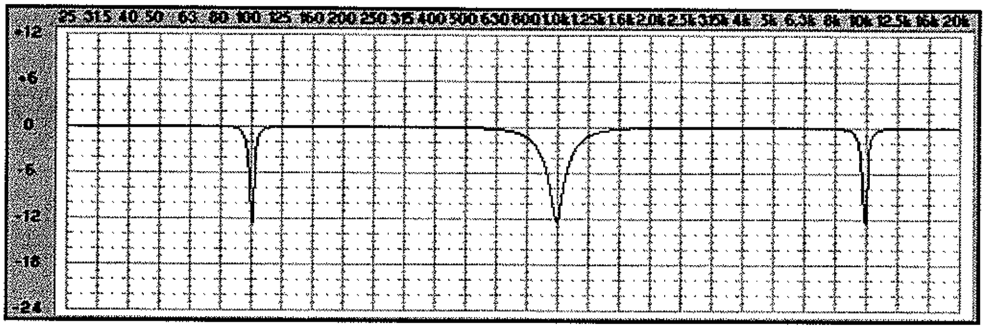


FIGURE 3-20. Typical narrowband equalizer curves from Shure DFR11EQ showing relationship between frequency and cut characteristics. The frequency cuts are at 100 Hz with 12 dB attenuation and high Q filter characteristics, 1,000 Hz with 12 dB attenuation with low Q filter characteristics, and 10,000 Hz with 12 dB attenuation with high Q filter characteristics. Note the wider frequency bandwidth attenuation characteristics at 1,000 Hz compared to 100 and 10,000 Hz. Data after Shure Brothers, Inc.

The disadvantage with these equalization methods is the feedback frequencies are not adjustable in real time. If the transfer function between microphone and loudspeaker changes, as occurs when the microphone moves, the fixed narrowband equalizer may not be tuned to the new ring mode frequency. Current practice is to now use DSP-controlled devices which detect, attenuate, and adjust automatically to new feedback frequencies as the transfer function changes.

3.5.2.5 Minimum User-Adjustable Equalizers

One recent trend in frequency equalizers is to provide devices which have a minimum, or no, user-adjustable controls. The main purpose of this is to prevent unauthorized adjustment of the equalization response and the necessary future service call from the sound system contractor to correct the misguided error. The frequency equalizers are typically one-third octave graphic types and have memories where different settings can be stored and recalled later by the user to suit varying

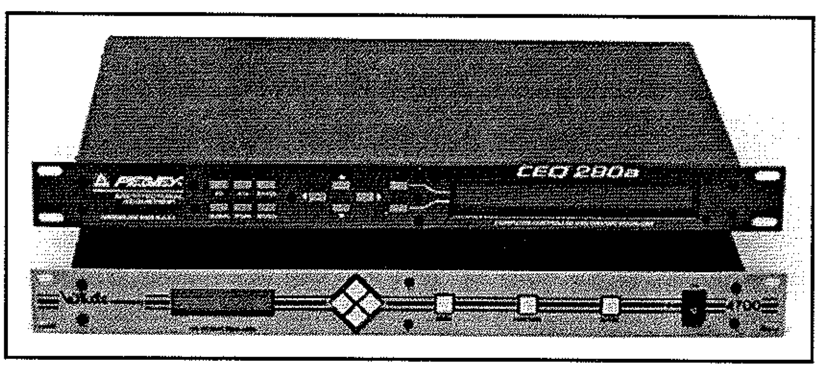


FIGURE 3-21. Minimum user-adjustable one-third octave band frequency equalizers 28-band single-channel type (Peavey CEQ™ 280a top and White 4700 bottom). Products courtesy of Peavey Electronics Corporation and White Instruments.

program requirements. The initial set-up is performed by the sound system contractor using precision electro-acoustic measurement equipment to adjust the equalizer to the desired response settings. Recall of predefined system equalization is by pushbutton control on the frequency equalizer front panel. Figure 3-21 shows two minimum user-adjustable equalizers.

3.5.3 Signal Delay Lines

Signal delay lines are audio components that time delay a signal relative to a reference signal. These devices take advantage of two primary psycho-acoustic phenomena: (1) the incident direction of sound arriving first at the listener is perceived more strongly while short term delayed secondary sounds are suppressed (Law of the First Wavefront) and (2) secondary sounds, even if higher in level, arriving within approximately 30 ms after the primary sound are integrated by the ear/brain and only one apparent louder signal is perceived (Haas/precedence effect).

(See Technical Notes, Section 3.G, at the end of this chapter, for additional information on the Haas/precedence effect.)

Signal delay lines operate in the digital domain. The incoming audio signal in converted from its analog format to a digital format which is electronically stored as 16 to 20 bit word lengths, retrieved to affect the desired delay time, and finally converted back to analog format before being routed to other sound system components. The delay time is user-adjustable by front panel controls or through a computer-controlled software interface. Delay times can be short, between several microseconds and 30 ms, or long, between several milliseconds and 1.5 s. Signal delay lines have one or two inputs and three or four outputs which are useful for setting different delay times for multiple loudspeaker zones, common in large spaces. Figure 3-22 shows a signal delay line.

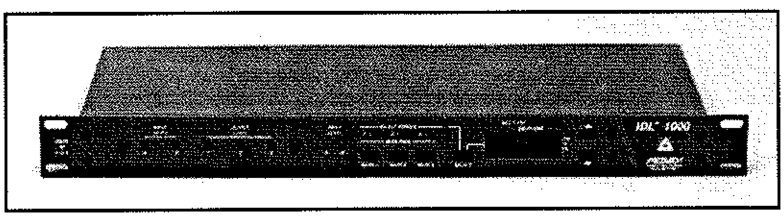


FIGURE 3-22. Signal delay line (Peavey IDL™ 1000). Product courtesy of Peavey Electronics Corporation.

Improving the naturalness of amplified sound and correcting certain audio signal problems are the primary uses for signal delay lines. Common applications of signal delay lines are described below.

1. The perception of auditory realism can be improved by "pulling" the loudspeaker signal towards the talker's position, thus localizing the auditory image with the visual cues from the talker.

2. Artificial echoes from widely spaced loudspeakers whose signals arrive at the listener more than 30 ms apart can be eliminated.

3. Synchronizing the electrical signal input to loudspeakers covering the same audience seating area can eliminate uneven frequency response comb filtering due to selective phase cancellation of loudspeaker signals arriving at slightly different times.

4. Correcting signal alignment problems at the crossover frequency in multi-way loudspeaker systems where the low-frequency driver reproduces frequencies above the crossover point, and the high-frequency driver reproduces frequencies below the crossover point, resulting in comb filtering due to comon signals which arrive at the same location at slightly different times.

5. Electronic "aiming" of a line source array (column) system, by varying the timing of electrical signals to each driver, can alter the "shape" of the column, thus changing its directional pattern characteristics.

Most loudspeaker configurations can benefit from signal delay and their application is discussed below.

1. Ceiling distributed full-range loudspeakers are frequently used to provide sound coverage under an auditorium balcony or church transept where a central cluster system can not provide coverage due to acoustical shielding by an architectural element. In this application the delay line is used to delay the signal to the ceiling loudspeakers so the perceived sound arrival is from the stage or chancel.

2. Distributed column or small full-range loudspeakers are often used in long narrow spaces, such as a cathedral or central atrium. Several delay zones, corresponding to greater distances from the primary sound source are commonly used. Each delay zone is provided with a separate signal delay setting to pull the auditory image towards the primary sound source.

3. Remote loudspeakers in stadium, amphitheater, and other outdoor sound systems are used in delay towers for providing coverage to remote seating areas. Due to the great distances between the primary loudspeakers and the delay towers there exists the potential for an

artificial echo to occur at the remote seating locations. Signal delay is used to eliminate this echo problem.

4. Large conference rooms or hotel ballrooms may require several delay zones to progressively delay ceiling distributed full-range loudspeakers to distant seats to pull the auditory image back to the lectern or head table.

(See Technical Notes, Section 3.H, at the end of this chapter, for additional information on the psycho-acoustic basis of signal delay.)

3.5.4 Crossovers

Crossovers are electronic devices which comprise resistors, capacitors, and inductors in a tuned circuit(s) which divide the audio signal into separate frequency limited passbands. The passband outputs are routed to other audio components or directly to the individual drivers within a loudspeaker enclosure. Crossovers are used to tailor the frequency spectrum of the signal routed to specific drivers. They also can protect drivers by preventing low-frequency signals from being routed to high-frequency drivers. Crossovers are classified by their signal level, *order*, or the number of passbands.

3.5.4.1 Signal Level Characteristics

Crossovers are designed to handle either *high-level* or *low-level* signals.

High-level crossovers are *passive* high-current devices which are used after the output of the power amplifier and are commonly installed within loudspeaker enclosures and on coaxial loudspeaker frames. The crossover network subdivides and routes the audio signals to the separate drivers. Moderate power handling capabilities and the need for specific tuning to accommodate the differences in *sensitivity* between the individual drivers are necessary in the design of high-level crossovers. As such, these devices are loudspeaker system specific and often require modification when different drivers are used. The advantages of high-level crossovers include simplicity of use, since the crossover in integral with the loudspeaker system, and lower cost.

Low-level crossovers are *active* low-current devices which are used before the input to the power amplifiers and are common in *bi-amplified* or *tri-amplified* applications. These devices provide a wide range of flexibility to adjust the *crossover frequency*, passband attenuation characteristics, and output level. Some low-level crossovers provide minimal time delay or phase adjustment for signal alignment through the crossover region. Advantages of low-level crossovers include preserving the amplifier *damping factor*, maintaining coupling of the amplifier to the loudspeaker, lower distortion, optimizing driver response characteristics, and flexibility in sound system design.

3.5.4.2 Order Characteristics

The number of *poles* a filter possesses is the filter order. It is characterized by the slope in dB/octave of the filter response over the *stopband* and is equal to 6 times the order. Thus, a first order filter has one pole with a total slope of 6 dB/octave, a second order filter has two poles with a total slope of 12 dB/octave, et cetera. Normally, the highest order filter used in crossovers is the fourth order. Higher order (18 and 24 dB/octave) filters have greater phase shift and more audible effects compared to lower order (6 and 12 dB/octave) filters. However, lower order filters may not have sufficient attenuation below the cutoff, or crossover frequency, to block the passage of undesired signals.

3.5.4.3 Number of Passbands

The frequency at which the signal is divided is referred to as the crossover frequency. Thus, a 500 Hz two-way crossover would split the signal at 500 Hz into two passbands, one below 500 Hz and one above 500 Hz. At the crossover frequency the upper and lower passbands are 3 dB down, so when the signals combine, the signal is restored to unity level at the crossover frequency. Figure shows typical crossover response characteristics.

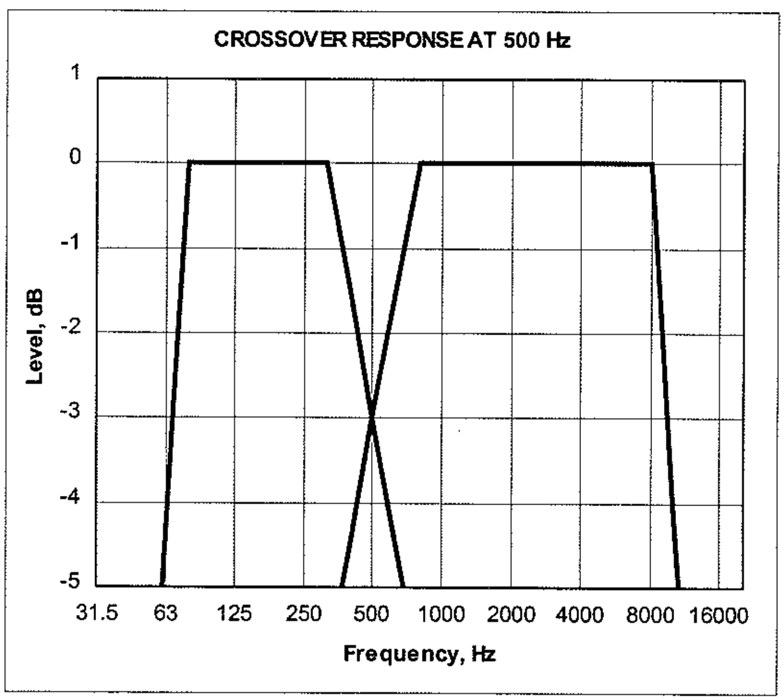


FIGURE 3-23. Typical crossover frequency curves showing relationship between low- (left) and high-pass (right) section characteristics.

Crossovers are available in two-, three-, and four-way configurations. Two-way crossovers commonly split the signal at 500 or 800 Hz. Three-way crossovers commonly split the signal at 500 and 1,500 or 3,000 Hz. Four-way crossovers commonly split the signal at 500, 2,000, and 5,000 Hz.

3.5.5 Multi-Purpose Signal Processing Devices

Recent developments with DSP technology have made it possible to combine multiple signal processing functions into one component. These devices are classified either as loudspeaker signal processors, which are specific to a particular manufacturer's loudspeaker model, or general purpose signal processors which can be used in any sound system. The primary advantage with the latter is the signal processing flexibility is contained in a small package, typically less than three *rack spaces* high, replacing several dedicated signal processing components. This can be an advantage where equipment rack space is at a premium. The multi-purpose signal processing devices are often less expensive than the multiple dedicated components they often replace. Figures 3-24 and 3-25 show different multi-purpose signal processing devices.

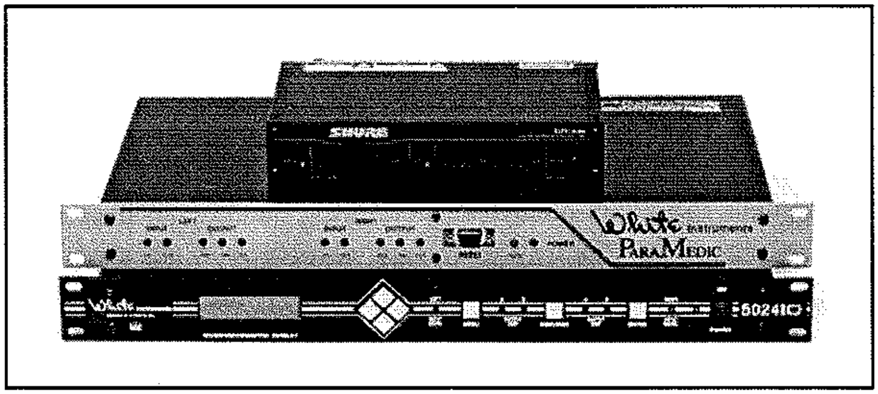


FIGURE 3-24. Multi-purpose signal processing devices (Shure DFR11EQ top, White Instruments ParaMedic middle, and White Instruments 5024IQ bottom). Products courtesy of Shure Brothers, Inc. and White Instruments.

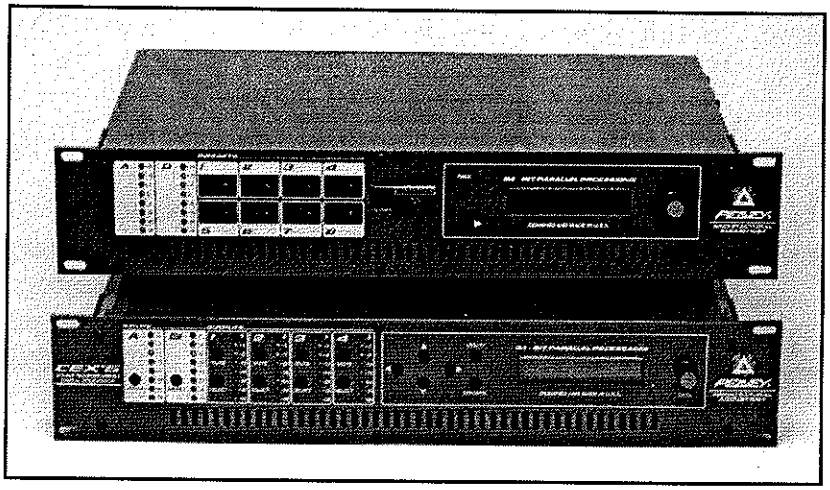


FIGURE 3-25. Multi-purpose signal processing devices (Peavey X-Frame™, top and Peavey CEX™5, bottom). Products courtesy of Peavey Electronics Corporation.

3.6 Monitoring Sound Systems

It is necessary to monitor the sound system to check signal levels, verify that overload or distortion are not present, and confirm that signals are routed to the correct locations. Visual meters and loudspeakers are commonly used to monitor the signals. Computer control of monitoring functions is becoming more common.

3.6.1 Meters

Visual monitoring of audio signals is most commonly done with *volume unit* (*VU*) meters. The advantages with the VU meter are its response provides an approximate average of the signal level and corresponds closely to human hearing characteristics. The main disadvantage with the VU meter is it is not fast enough to respond to short-term transient signals which may momentarily overload the audio equipment. Often LED "ramp" meters are used to monitor transient peak signal levels which have varying dynamic characteristics. These meters can provide a better indication of signals which might overload audio equipment. Peak responding LEDs are frequently used in

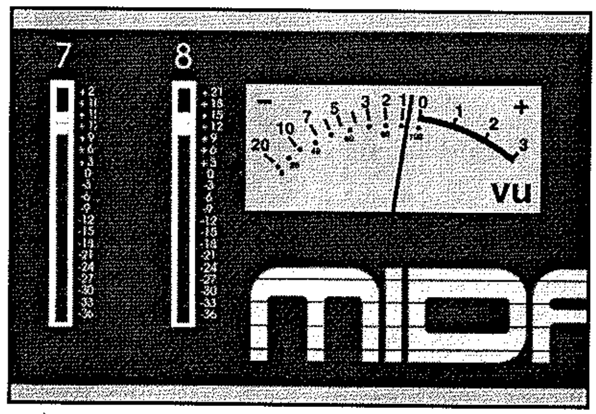


FIGURE 3-26. Typical peak-reading LED ramp (left) and average-reading VU meters (right). Drawing courtesy of Mark IV Pro Audio Group.

combination with VU meters to monitor transient signals of at least 10 ms duration to assess both average signal levels and high-level short duration signals. Figure 3-26 shows standard VU meters and peak responding LED level indicators.

3.6.2 Monitor Loudspeakers

Loudspeakers are used to audibly monitor audio equipment signals. Monitoring can be achieved with a loudspeaker connected across the sound system output, prior to the main sound system power amplifiers, or by using a switchable loudspeaker that can be connected across different sound system components. General purpose monitor loudspeakers are small full-range devices less than 8 in diameter which are mounted to a standard 19 in rack mount panel installed on the front of the equipment rack. High quality monitor loudspeakers using separate low-frequency and high-frequency drivers are often mounted above the mixing console in an audio control room.

3.7 Computer Control of Sound Systems

Computer control of sound systems is becoming common for communication between components, monitoring equipment operational status, adjusting system parameters from remote locations, and providing customized signal mixing, processing, and routing functions. These systems can take the form of individual components executing standardized communication system protocols operating through network control systems to complete DSP-based computerized sound systems. While still in their infancy, computer controlled sound systems will be the basis of future sound system design. Like anything digital, this is a fast evolving field and whether the communication standards, network control systems, or computerized sound systems described below survive into the future remains open.

(See Technical Notes, Section 3.I, at the end of this chapter, for additional information on DSP theory.)

3.7.1 Communication Standards

Electronic industry trade association and manufacturer-derived communication standards have been developed to permit communication and control of audio equipment from different manufacturers and within a specific manufacturer's product line. Communication standards can be subdivided into communication protocols and network control systems.

3.7.1.1 Communication Protocols

Communication protocols are sanctioned universal standards developed by industry trade associations to permit compatibility between different equipment. Some commonly used communication protocols include musical instrument digital interface (MIDI), RS-232, RS-422, RS-485, and Firewire®.

3.7.1.1.1 MIDI

MIDI is a serial communications protocol developed in the early 1980s as a means to control electronic musical instruments and interconnected signal processing equipment. Some signal mixers, signal processing equipment, and tape recorders are MIDI compatible. The data transfer rate of MIDI is restricted to 31.25 *kbaud*. This speed is sufficient for musical instrument control but often is not fast enough for real time sound system control and special provisions may need to be made.

Equipment having MIDI control capabilities are provided with dedicated ports (MIDI IN, MIDI THRU, and MIDI OUT). The specific port type(s) will depend if the equipment generates or receives MIDI commands. Connection between equipment is made with five pin *DIN* plugs. Cable length for MIDI control is limited to 50 ft.

3.7.1.1.2 RS-232, RS-422, and RS-485

The RS-232, RS-422, and RS-485 communication protocols have been standardized by the Electronic Industries Alliance (EIA) and the Institute of Electrical and Electronic Engineers (IEEE) for communication between computers and other digital equipment.

RS-232 control is found on computers and uses an unbalanced configuration for serial data transfer having a maximum speed of 19.2 kbaud. Equipment which is RS-232 compatible has 25 or 9 pin D subminiature connectors. Cable lengths for RS-232 control are limited to 15 ft for high data transfer rates. Longer cables can be used, but the data transfer rate is considerably slower.

RS-422 is similar to RS-232 except it uses a balanced configuration and the data transfer rate is faster, between 19.2 kbaud and 2 Mbaud in the standard mode and up to 10 Mbaud in the fast synchronous mode. Equipment which is RS-422 compatible has 9 pin D subminiature connectors. Cable lengths up to 4,000 ft can be used at a 19.2 kBaud transfer rate, but faster transfer rates require shorter cables. The AES has adopted a modified version of RS-422 as standard PA-422 for control of audio system equipment, which restricts the data transfer rate to 19.2 kbaud.

RS-485 a balanced configuration similar to RS-422 except the data transfer rate is faster, up to 10 Mbaud. A variety of connectors, including, 9 pin D subminiature, phoenix, and RJ-45, are used with RS-485 compatible equipment. Cable lengths up to 4,000 ft can be used without degradation of signal transmission characteristics. The most common cable type used is CAT-5 data cable, which is an industry standard cable for telecommunication systems.

3.7.1.1.3 Firewire®

Firewire® is a serial communications protocol developed by Apple Computer in the mid 1980s for their computers and accessories. It has since been standardized as

IEEE 1394 and is now an open communication protocol that has been adopted by most personal computer system hardware and operating system manufacturers. The protocol is compatible with Dolby Pro Logic™, Inc., Dolby AC-3™, DVD, digital broadcasting services, electronic musical instruments, and other digital formats. A system network can be created with over 64,000 interconnected devices and 833 channels when the highest data transfer rate is used.

Two primary modes of data transfer, asynchronous and isosynchronous, are implemented. Twenty percent of the signal bandwidth is dedicated to asynchronous transmission, which is used for less critical data, such as control signals. The remaining 80 percent of the signal bandwidth is dedicated to isosynchronous transmission which is used for critical data, such as synchronization of audio and video signals.

Three data transfer rates are supported by IEEE 1394: (1) 98.3 Mbaud; (2) 196.6 Mbaud; and (3) 393.2 Mbaud, referred respectively as S100, S200, and S400. Equipment which is IEEE 1394 compatible uses a specialized six pin connector, with four pins for carrying data and two pins for carrying power. Cable lengths are limited to 15 ft for 24 *AWG* cable, 45 ft for 22 AWG cable, and 330 ft for fiber optical cable. Longer cables can be used, but slower data transfer rates result.

3.7.1.2 Network Control Systems

Network control systems have been developed by specific equipment manufacturers to permit control of their remotely located sound system equipment. Often a manufacturer's proprietary network control system is adopted by other manufactures and becomes an industry-wide de-facto standard.

3.7.1.2.1 CobraNet™

CobraNet™ was developed by Peak Audio in 1996 as a control system for sound and video systems. It operates over a standard 100 Base-T Ethernet network to communicate between equipment via CAT-5 data cable or fiber optic cable. The key to the CobraNet™ system is that it converts the sporadic real time nature of the Ethernet network into a reliable real time transmission medium, which is important when synchronizing different audio and video channels from different sources.

The system is capable of carrying digital audio, video, and control signals with up to 64 data channels using 20 bit digital word lengths sampled at a 48 kHz resolution in an uncompressed format on a single CAT-5 data cable. CobraNet™ supports RS-232, RS-422, and RS-485 communication protocols. It has been licensed to over 30 major audio equipment manufacturers and is becoming an industry standard for communication between equipment.

3.7.1.2.2 Crest NexSys®

NexSys® was developed by Crest Audio in 1991 to provide communication and control via computer interface between Crest power amplifiers and MIDI controlled devices. The basic system tracks sound system performance and executes statistical functions to determine whether the power amplifiers are operating in a safe and optimal manner. Advanced functions can provide automated event scheduling, frequency equalization, digital signal processing, and automatic sequential switching control of power amplifiers.

System control between NexSys® and peripheral equipment is carried out by interface cards using RS-485 communication protocol. One controller is located between the computer and the power amplifiers and a second controller is located between the output of the power amplifiers and the input to the loudspeakers. Twisted pair cable is used to deliver the communication signals. Interconnected equipment can be separated up to 4,000 ft.

3.7.1.2.3 Crown IQ® System

Crown developed the IQ® System in 1987 to provide computer communication and control of Crown power amplifiers. The system functions have been expanded to include control of signal mixers, distribution/routing amplifiers, and frequency equalizers. Other manufacturers have developed power amplifiers and signal processing equipment which are IQ® System compatible. Some functions available with the IQ® System include monitoring of power amplifier operating status, signal mixing and routing, frequency equalization, and ambient noise level sensing.

Communication between equipment and the computer is via RS-232 or RS-422 protocol distributed in a current loop operating at 38.4 kbaud. Equipment is connected in series using 26 AWG twisted pair cable with a cable length up to 4,000 ft. Cable connection to the equipment is by four pin barrier strips or five pin DIN connectors.

3.7.2 Computerized Sound Systems

The latest development in sound systems is to combine DSP signal processing and computer technology to create user customizable sound systems. These systems handle all signal processing and routing functions from the microphone output to the power amplifier input. Connecting microphones, power amplifiers, and loudspeakers completes the sound system.

3.7.2.1 Peavey Media Matrix® System

Peavey Electronics developed the Media Matrix® system in 1993 as a stand alone computerized audio processing system. The Media Matrix® system is a combination of software, developed by Peak Audio, and audio components, developed by Peavey Electronics. The basic audio components include a computerized processing frame, DSP cards, and combination A/D and D/A converters. Different size Media Matrix® systems have been developed to suit a variety of sound system design requirements. Control of remote amplifiers and other audio equipment from the Media Matrix® system is through RS-485 communication protocol. Visual monitoring of the system is by a standard computer CRT. Figure 3-27 (above) shows a Peavy Media Matrix® miniframe system

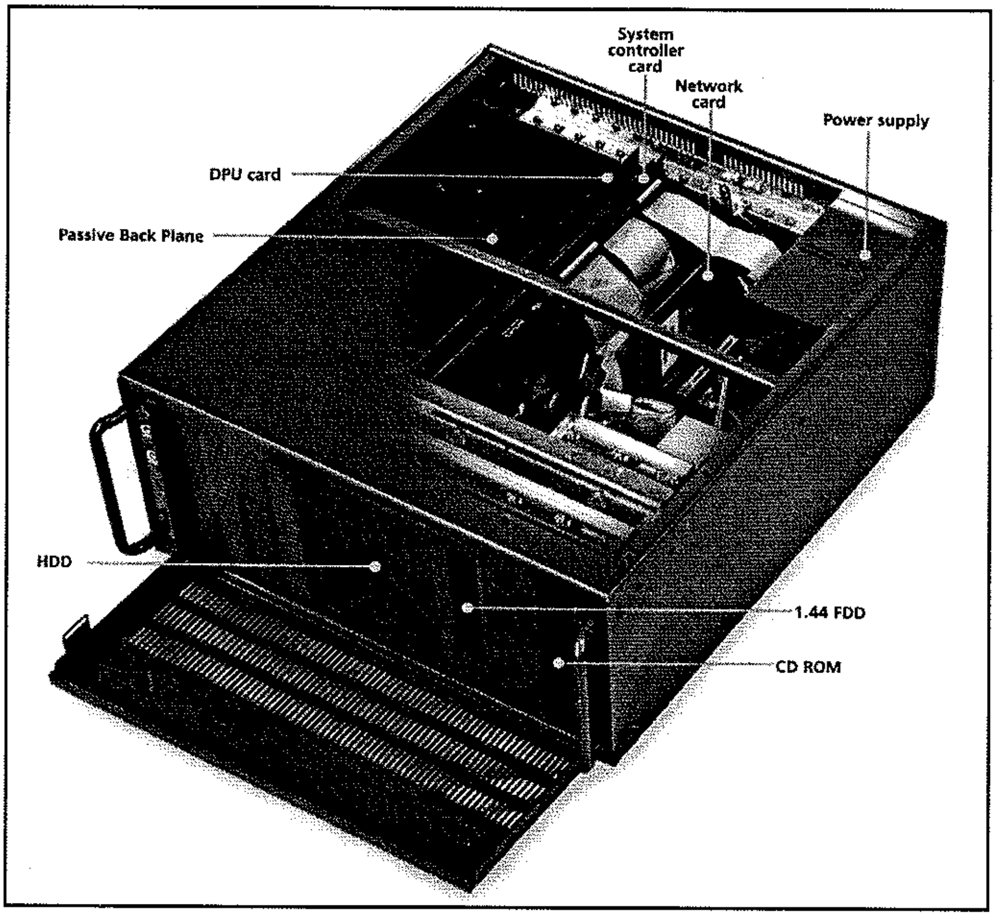


FIGURE 3-27. Computerized sound system (Peavey MediaMatrix® Miniframe 760nt™). Photo courtesy of Peavey Electronics Corporation.

To create a sound system using Media Matrix®, the designer develops a functional *block diagram* within the computerized processing frame from a library of predefined audio function software modules. Some available audio function modules include: (1) signal mixing; (2) compression, limiting, expansion, and gating; (3) frequency equalization; (4) signal delay; (5) gain control; (6) crossover and frequency subdivision; (7) signal distribution and routing; and (8) monitoring and control. Additionally, the designer can create custom audio function modules. Each separate DSP card within the computerized processing frame can handle up to 32 inputs and 32 outputs. A total of 256 channels of 24 bit digital audio can be included in a sound system design.

Once the sound system block diagram is completed, the sound system design is compiled by the computer within the processing frame to create a working system file which is loaded into the system. DSP cards within the computerized processing frame execute the audio system functions in real time. The system has the flexibility to run without operator control or to permit manual control by an operator, similar to a standard sound system.

The audio signal inputs from microphones and line level devices are connected to a preamplifier with output to an A/D converter. The digital signal output from the A/D converter is routed to the computerized processing frame. Output from the processing frame is routed to a D/A converter which converts the digital signal back to an analog signal which can be connected to power amplifiers or other audio equipment.

3.7.3 Remote System Controllers

Remote system controllers provide a means to select equipment operational parameters and activate other sound system related functions, such as video presentation systems, projection screens, and light dimmers. These systems are commonly installed in boardrooms, conference rooms, classrooms, and similar spaces where a non-technical user needs to control the sound system.

The remote system controller comprises a modular control frame and a touch screen panel. The modular control frame is located in the equipment rack. The touch screen panel is typically installed in lecterns, moderator/chairman panels, court clerk desks, or on the room object wall. Communication between the touch screen, control frame, and the sound system is through RS-232, RS-422, or PA-422 protocols. Control of the sound system and other related equipment is made at the touch screen which sends signals to the modular control frame to activate the selected equipment. Customized programming by the product manufacturer or the sound system contractor is necessary to implement the specific equipment control functions. Figure 3-28 shows a small remote system controller.

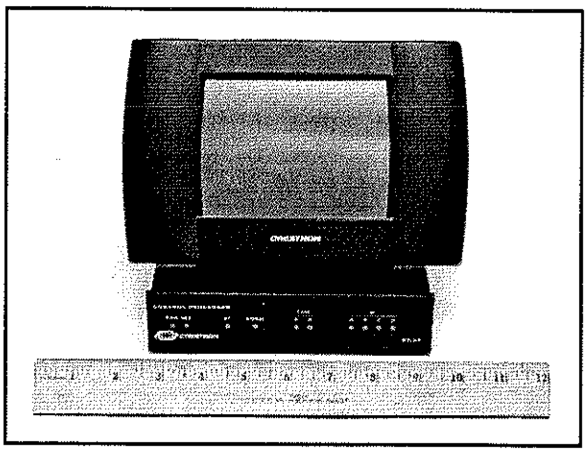


FIGURE 3-28. Remote system controller comprising touch screen (Creston ST 1550C top) and system controller (Creston ST-CP bottom). Products courtesy of Creston Electronics, Inc.

3.8 Power Amplifiers

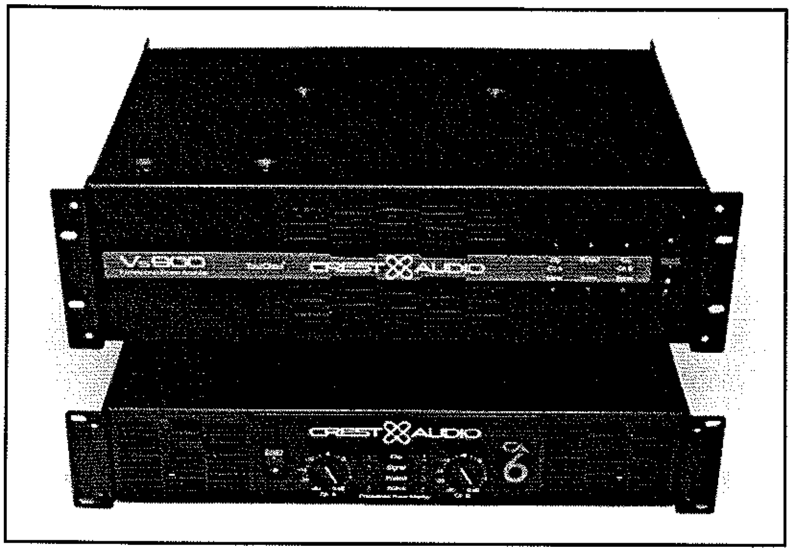


FIGURE 3-29. Power amplifiers (Crest Audio Vs900 top and Crest Audio CA6 bottom). Products courtesy of Crest Audio, Inc.

Power amplifiers are electronic devices which increase the voltage gain of a line level signal to a high level (loudspeaker) signal. They are the last active component in the sound system signal path and are placed before the loudspeakers. Power amplifiers are available as a stand alone component intended for high quality sound systems or integrated with other components, such as preamplifiers, frequency equalizers, or signal mixers to comprise an integrated package intended for modest sound systems. Figure 3-29 shows two power amplifiers.

The electronic circuitry of a power amplifier comprises an input stage, driver stage, output stage, and a power supply. The input stage increases the amplifier voltage gain. An input attenuator (volume control) is part of the input stage and adjusts the output level of the power amplifier. The driver stage controls the output devices (power transistors) and splits the signal into positive and negative components to drive the power transistors. The output stage connects the positive and negative portions of the power supply to the loudspeaker load and controls the output in response to the applied audio signal. The power supply converts the 120 VAC from an electrical power receptacle into a DC voltage used by the power amplifier circuitry.

Power amplifiers are classified by the number of output channels (single, dual, or multi-channel), the output impedance/voltage configuration (low impedance/variable voltage or high impedance/constant voltage), and the power output characteristics.

(See Technical Notes, Section 3.J, at the end of this chapter, for additional information on different amplifier output stage classes.)

3.8.1 Number of Output Channels

Different power amplifier output channel configurations are available and can comprise single-channel (monophonic), two-channel (stereophonic), and multi-channel (three-, four-, six-, eight-, or 16-channels). Most dual and multi-channel power amplifiers share a common power supply, however higher powered amplifiers may use separate power supplies for each channel. Some multi-channel power amplifiers are based on a modular format with separate power amplifier channel "cards" which plug into a common frame/power supply to create customized configurations. Single chassis two-channel power amplifiers are the most commonly used amplifier configuration in sound systems.

3.8.2 Impedance/Voltage Characteristics

Power amplifiers are broadly classified as either low impedance/variable voltage or high impedance/constant voltage devices.

3.8.2.1 Low Impedance/Variable Voltage Amplifiers

Low impedance/variable voltage power amplifiers drive loudspeakers having nominal input impedances of 4, 8, or 16 Ω. The output voltage (**V**) developed by the power amplifier when connected across a loudspeaker varies as a function of the input signal level, loudspeaker impedance (**Z**), and the amplifier power supply.

Power amplifiers are rated by the amount of electrical power (**P**) in watts delivered to the connected load impedance (**Z**). The **P** of a low impedance/variable voltage power amplifier due to **Z** can be calculated using the following equation:

$$\mathbf{P} = \frac{\mathbf{V}^2}{\mathbf{Z}} \tag{3.1}$$

where,

P is the power output of the amplifier, watts
V is the voltage drop across the loudspeaker, volts
Z is the nominal impedance of the loudspeaker, ohms

Advantages of low impedance/variable voltage power amplifiers include potential for high sound levels and flexibility in powering specific loudspeakers. Low impedance/variable voltage power amplifiers are used where high quality sound reproduction is required.

3.8.2.2 High Impedance/Constant Voltage Amplifiers

High impedance/constant voltage amplifiers are used where lesser quality sound reproduction and lower sound levels are acceptable. A constant voltage power amplifier, usually operating at 70.7 V (although 25 and 100 V systems are common), is connected to distributed loudspeakers normally installed in the ceiling. The loudspeakers are connected by "step-down" transformers in a parallel configuration across the power amplifier output. The loudspeaker transformer matches the load impedance of each loudspeaker to the power amplifier. The *primary* tap of the loudspeaker transformer is adjusted to draw a specified amount of power from the output of the amplifier. The *secondary* tap of the loudspeaker transformer is adjusted to match the loudspeaker impedance.

The amplifier power output (**P**) is regulated enabling it to maintain a constant voltage regardless of the load impedance (**Z**). It acts as a voltage source which does not change providing there is adequate current. Proper power amplifier loading occurs when the sum of the total power delivered to the individual loudspeaker transformers does not exceed the amplifier power output rating. The power amplifier output can provide a constant voltage by one of three means: (1) external step-up transformer; (2) internal step-up transformer; or (3) high voltage transformerless output section. The first two methods increase the power amplifier output voltage to the system operating voltage, typically 70.7 V.

When properly loaded, all of the power from the amplifier will be delivered to the loudspeaker line. The **P** of a high impedance/constant voltage power amplifier due to **Z** can be calculated using the following equation:

$$\mathbf{P} = \frac{\mathbf{5000}}{\mathbf{Z}} \tag{3.2}$$

where,

P is as above

Z is the impedance of the fully loaded loudspeaker line, ohms

5000 is constant for 70.7 *volt* lines

Advantages of high impedance/constant voltage power amplifiers include ease of installation, flexibility in adding or removing loudspeakers, and use of smaller gauge loudspeaker cable. High impedance/constant voltage power amplifiers are used where lower quality sound reproduction is required.

3.8.3 Power Output Characteristics

Low impedance/variable voltage power amplifiers have typical outputs between 60 and 800 W per channel, with devices in the 100 to 200 W per channel range finding the majority of application in sound systems.

High impedance/constant impedance power amplifiers have outputs between 100 to 250 W per channel, but the output delivered to the loudspeaker itself is controlled by the primary tap setting on the loudspeaker transformer.

3.8.3.1 Amplifier Channel Bridging

Some dual and multi-channel power amplifiers have the capability to have their output channels combined, or *bridged*, into single or multiple channels of greater power output. In this configuration the output channels are operated in series with a resultant doubling of the amplifier output voltage. For a given loudspeaker impedance, the power from the amplifier will be four times that of a single amplifier. The load impedance seen by the amplifier when operated in a bridged mode should not drop below 8 Ω if the full amplifier power output is to be realized.

3.8.3.2 Power Amplifier Output and Loudspeaker Damage

The power output from an amplifier will vary depending on the load impedance the amplifier is driving. Note that while a loudspeaker has a nominal impedance rating, such as 8 Ω across its bandwidth, the actual impedance will vary as a function of frequency. The impedances at some frequencies may be one-fifth the nominally rated loudspeaker impedance. As the load impedance decreases, the power output from the amplifier increases. At very low impedances, typically below 2 Ω, the amplifier may become unstable resulting in damage to the amplifier output transistors.

Most loudspeakers are robust enough to handle short duration signal transients up to 10 times their rated long-term power handling capabilities, assuming the power amplifier signal does not contain significant distortion. Thus, it is acceptable to use power amplifiers having short-term power output capabilities (headroom) which exceed the long term loudspeaker handling characteristics. Headroom is necessary to reproduce speech and music signal peaks without audible distortion since these signal peaks can exceed their long-term average levels by up to 12 dB. An amplifier of low power output which is *clipping* and operates for a long duration is potentially more damaging to a high-frequency driver than an amplifier of high power output with little distortion which operates for short durations throughout peak signal transients. The distortion in the first case results in added higher-frequency harmonic signals causing thermal stress on the driver voice coil windings or can melt the voice coil in the magnetic gap.

A common misconception relates to the power amplifier output and the corresponding increase in perceived loudness. When the power amplifier output is doubled (say, from 50 to 100 W) the loudspeaker sound level increases by 3 dB, resulting a just noticeable change in perceived loudness. In order for the loudspeaker sound level to be subjectively perceived to be twice as loud requires the power amplifier output to be increased by 10 times (say, from 50 to 500 W).

3.8.4 Other Considerations

Power amplifiers produce the greatest sensible heat load of all sound system equipment. Operating a power amplifier beyond its thermal limits can cause the unit to automatically shut off for self protection, or worse, result in permanent damage to the output transistors. Amplifiers require natural convection or forced fan cooling to reduce their operating temperature. The latter can be provided in the form of an integral fan on the amplifier chassis or as a supplemental fan installed in the equipment rack. Sound systems which use numerous amplifiers may have dedicated amplifier rooms which are provided with cooling from the building HVAC system.

Some newer power amplifier designs use switching power supplies which have heat loads approximately one-third of conventional amplifier power sources.

Remote control of amplifiers not in close proximity to the sound system operator is often necessary to troubleshoot system installation, adjust operating parameters, or turn on/off amplifiers. There are several manufacturer-derived systems which provide this type of control.

The magnetic field emitted by the large power transformers in power amplifiers can be picked-up by the low-level electrical circuits in the signal processing equipment. Note that equipment which carries the "CE" label emit less electronic and magnetic radiation and will likely be less susceptible to electromagnetic interference (*EMI*) problems.

3.9 Loudspeakers

A loudspeaker is a transducer which converts an AC voltage corresponding to the input signal magnitude and frequency into sound waves. Loudspeakers can be designed to reproduce a wide frequency bandwidth (full-range) or a restricted frequency bandwidth. Full-range drivers reproduce frequencies between 50 to 15,000 Hz and beyond. Restricted bandwidth drivers include *woofers* (reproduce low-frequencies below 500 Hz), mid-range units (reproduce frequencies between 200 and 4,000 Hz), and *tweeters* (reproduce high-frequencies above 1,000 Hz). These frequency limits are approximate for the three general driver categories and manufacturer's specific products can vary from these generalized characteristics.

The three basic driver types include: (1) dynamic cone direct radiator; (2) combination horn/compression driver; and (3) coaxial. Dynamic cone loudspeakers can be used for full-range, low-frequency, mid-range, or high-frequency reproduction. Horn/compression driver combinations are used to reproduce mid-range and high-frequency signals. Coaxial loudspeakers are used for full-range, low-frequency, or high-frequency reproduction.

Cone drivers can be installed in sealed infinite baffle, ported, horn-loaded, or bandpass enclosures. Compression drivers are installed on a horn which provides the necessary acoustical loading and directional sound control. Horn types include: (1) exponential; (2) multi-cell; (3) radial; (4) bi-radial (constant directivity); (5) defined coverage; and (6) Complex Conic™. Coaxial loudspeakers are available in cone, horn, and combination horn/cone configurations. The latest practice in loudspeaker design uses DSP technology to provide loudspeaker-specific signal processing to optimize performance.

(See Technical Notes, Section 3.K, at the end of this chapter, for additional information on terminology for drivers and loudspeakers.)

3.9.1 Dynamic Cone Direct Radiator Loudspeaker

A dynamic cone direct radiator loudspeaker is a moving coil driver which comprises a magnet with a round air gap and a voice coil centered in the air gap. The voice coil moves inside the air gap and is connected to a cone. The cone motion is caused by the magnetic force created by an audio signal current in the coil which reacts with the stationary magnetic field emitted by the fixed magnet.

Cone drivers are available in a variety of sizes between 4 and 18 in diameter. The larger drivers have improved low-frequency response, higher sound power output, and can handle greater amplifier power. Smaller drivers have improved high-frequency response, lower sound power output, and restricted power handling capabilities. Low-frequency drivers are 12, 15, and 18 in diameter. Mid-range drivers are 6, 8, 10, and 12 in diameter. High-frequency drivers are 1, 2, 3, and 4 in diameter. Full-range drivers are 4, 6½, and 8 in diameter. The specific size and frequency range will vary between different manufacturer's products. Figure 3-30 shows cone drivers of different sizes.

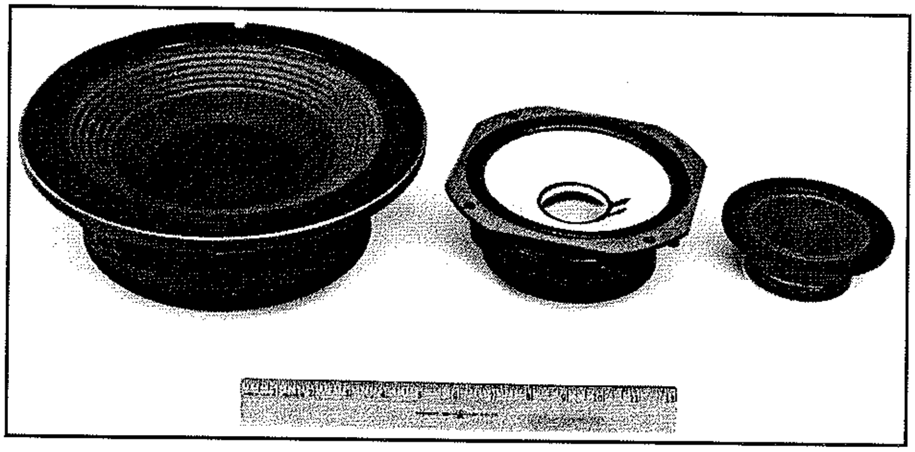


FIGURE 3-30. Dynamic cone drivers of various sizes: 12 in diameter low frequency type (JBL 2206H/J left), 8 in diameter full-range type (JBL LE8T-H center), and 4 in diameter full-range type (Altec Lansing 405-8H right). Products courtesy of JBL, Inc. and Mark IV Audio Group.

The *resonant frequency* is an important characteristic of a dynamic cone driver. At the resonant frequency the cone will vibrate at its maximum with a minimum of signal input. *Damping* is usually added to the driver to control its output at resonance. Below the resonant frequency, the driver output rapidly falls off. Above the resonant frequency the response is relatively uniform for several octaves. These limits essentially define the driver passband.

An enclosure is necessary to prevent the rearward radiating acoustical energy of the cone, which is out of phase with the forward radiating acoustical energy, from combining and reducing the driver low-frequency response.

The cone driver efficiency is very low, with values between 0.25 and 3 percent being typical. Approximately 99.75 to 97 percent of the electrical power input is lost as heat in the transduction process and not converted into acoustical ouput. The efficiency can be raised by installing the driver in an enclosure or mounting it to a horn.

(See Technical Notes, Section 3.L, at the end of this chapter, for additional information on the history of the development of the dynamic cone driver.)

3.9.1.1 Basic Construction

The dynamic cone driver comprises several distinct elements and subassemblies: (1) rigid frame; (2) magnetic motor assembly; (3) cone diaphragm; (4) suspension system; and (5) dust cap. Each of these subassemblies has many different component parts. Figure 3-31 shows the major parts of the dynamic cone driver.

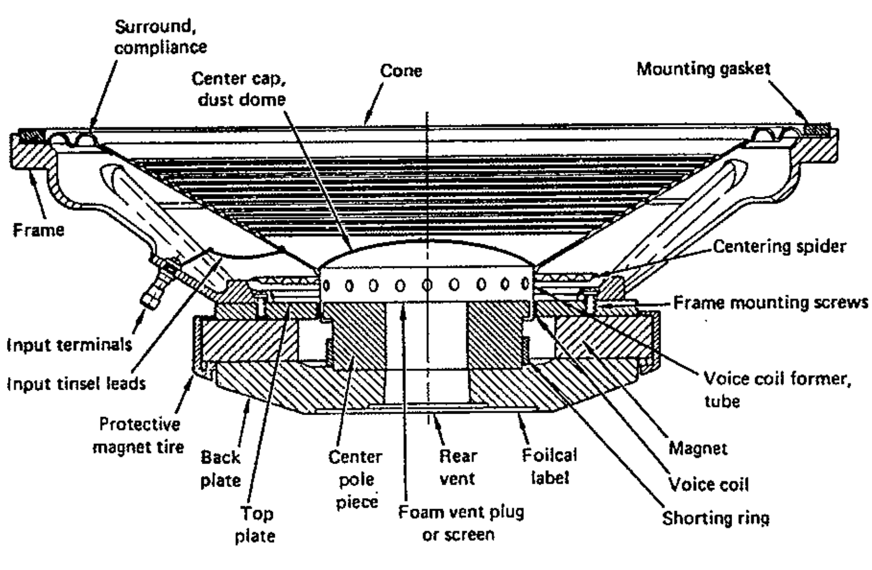


FIGURE 3-31. Sectional drawing of a typical dynamic cone driver showing details of the major component parts. Drawing courtesy of JBL, Inc.

3.9.1.1.1 Frame

The outer body of the driver comprises a die-cast or stamped metal frame. The purpose of the frame is to hold the driver components together and to provide a means of mounting to an enclosure. Die-cast assemblies are mechanically stiffer and more robust than stamped metal assemblies, which can bend if improperly installed in an enclosure, resulting in misalignment of the driver components.

3.9.1.1.2 Magnetic Motor Assembly

The magnetic motor assembly, consisting of a fixed magnet and a coil of wire called the voice coil, delivers a controlled force to the cone diaphragm making it vibrate. A high strength magnet with a uniform magnetic field surrounds the voice coil. A small air gap lies between the magnet and the voice coil. An electro-magnet is created when an alternating electrical current (**I**), corresponding to the audio signal, passes through the voice coil of length (ℓ). The electro-magnet generates an alternating magnetic field which is proportional to **I**. The generated magnetic field interacts with the steady state magnetic field radiated by the permanent magnet. A mechanical force (**F**) is developed on the voice coil by the interaction of the poles of the two magnetic fields. The alternating electrical current direction in the voice coil determines whether the voice coil moves inward or outward from the magnetic gap. When **I** increases, the magnetic field strength radiated by the electro-magnet also increases, which interacts more strongly with the permanent magnet. This results in larger cone excursions which produce greater sound levels. The value of **F** can be calculated using the following equation:

$$\mathbf{F} = \mathbf{BI}\ell \tag{3.3}$$

where,

F	is the generated mechanical force, newtons
B	is the magnetic flux density, tesla
I	is the current flowing through the voice coil, amperes
ℓ	is the voice coil length within magnetic gap, meters

The fixed magnet is constructed from permanently magnetized materials including alnico (a mixture of aluminum, nickel, and cobalt), ferrite ceramics, or rare earth materials such as neodymium or samarium. Of the three materials, ferrite ceramics are the most commonly used due to low cost and engineering properties. Alnico is seldom used due to the expense of cobalt, although it was the material of choice between 1950 and 1975. Rare earth magnetic materials are finding increasing application in high efficiency designs.

The voice coil is a length of wire which is wound around a circular-shaped former. One end of the voice coil wires attach to the cone. The other end of the wires attach to terminals mounted on the frame and are connected to the crossover or directly to the power amplifier through a separate cable. High quality drivers use flat ribbon-shaped aluminum wire rather than circular copper wire. The former offers advantages in lower mass, greater packing density, higher rigidity, and flexibility in adjusting the voice coil impedance (**Z**).

The magnetic motor assembly is based on one of three designs: (1) underhung (short) voice coil; (2) voice coil and top plate of equal length; and (3) overhung (long) voice coil. The underhung and overhung voice coil designs are preferred for "long throw" cone displacement, necessary to reproduce low-frequencies or generate high sound levels. Of the two, the underhung voice coil design is more expensive to manufacture due to the need for a larger magnet. The equal length voice coil design is used for drivers requiring a small displacement, such as mid-range and tweeter units. Figure 3-32 shows the three voice coil types.

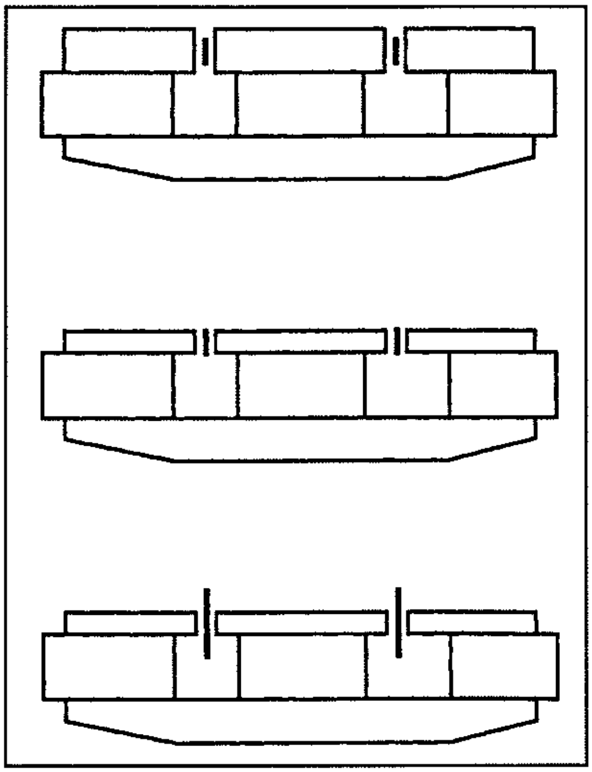


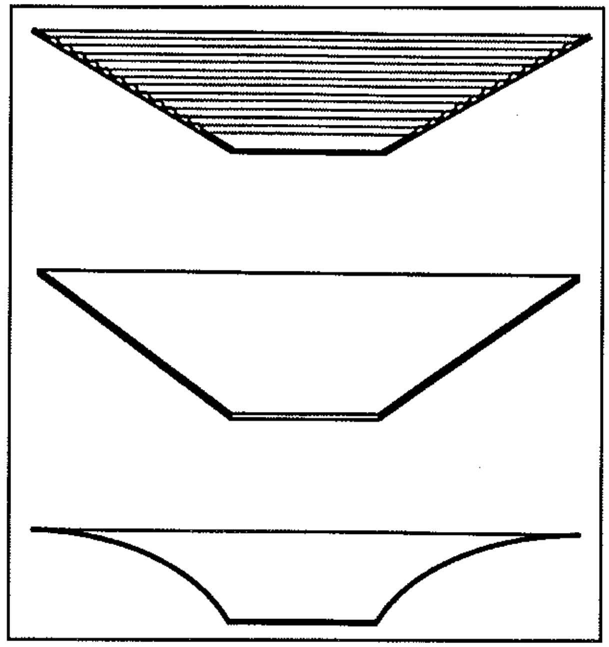
FIGURE 3-32. Sectional drawing of three magnetic motor assemblies: underhung (short) voice coil (top), voice coil and top plate of equal length (middle), and overhung (long) voice coil (bottom). Drawing courtesy of JBL, Inc. and John M. Eargle.

3.9.1.1.3 Cone Diaphragm

The cone diaphragm is the driver component which vibrates and generates acoustical energy. The cone is driven from its center inner edge, resulting in wave propagation towards the outer edge, where it is reflected back towards the center. Its radiating surface couples the voice coil motion to the surrounding air and acts as an impedance matching device. Ideally the cone will move with "pistonic" behavior resulting in uniform frequency response. However, most cone diaphragm materials do not exhibit pistonic behavior during large excursions, causing acoustical "break-up" of the cone surface. Distortion increases and uneven response, particularly at higher frequencies, results. In addition, vibrational modes can occur on the cone surface due to a complex interaction of the cone size and flexibility, radiating frequency, and the incident and reflected sound waves on the cone surface. These variables can further degrade high-frequency linearity. The cone profile shape and material selection can improve the pistonic behavior and extend the high-frequency response linearity.

The three principal cone shapes include: (1) straight side with ribs; (2) deep straight side; and (3) curvilinear. The addition of corrugated concentric ribs to the cone increases the stiffness and extends the high frequency response with little added weight. Increasing the cone depth stiffens the cone, making it suitable for high level application, but has the disadvantage of reduced high-frequency output. The curvilinear cone, while not as stiff as a straight-sided cone, has advantages of extended high-frequency response and greater sound level output. Figure 3-33 shows the three cone types.

FIGURE 3-33. Sectional drawing of three cone geometries: straight side with ribs (top), deep straight side (middle), and curvilinear (bottom). Drawing courtesy of JBL, Inc. and John M. Eargle.



Cone size is a factor in determining the driver frequency range and *coverage angles*. A large cone has a lower resonant frequency than a small cone resulting in more extended low-frequency response. As frequency is lowered, and the wavelength increases, the driver will radiate in a more omnidirectional pattern (larger coverage angle). Omnidirectional radiation results when the wavelength is greater than approximately four times the cone diameter. Small cones have the potential for more extended high-frequency response than large cones. As frequency increases, and the wavelength decreases, the driver will radiate in a more unidirectional pattern (smaller coverage angle). Unidirectional radiation results when the wavelength is approximately equal to or smaller than the cone diameter, with directivity increasing as the cone size decreases. The upper frequency limit of the driver is inversely proportional to the cone diameter. For instance, 18 in and 4 in diameter drivers will have upper frequency limits of approximately 500 and 2,200 Hz, respectively.

Different materials, such as felted paper, fiberglass, Kevlar™, plastic, aluminum, and titanium are used to make cones. Felted paper has the advantage of low cost, ease of manufacture, and smooth frequency response characteristics. Its disadvantages include limited high-frequency response, less durability, and susceptibility to humidity, which can alter its performance characteristics. Fiberglass, Kevlar™, and plastic cones have the advantage of stiffness which results in more pistonic motion over a wider frequency range. Additionally, these materials offer greater damping, durability, and humidity resistance. Aluminum or titanium alloy cones provide the stiffest assembly with excellent durability, but with the disadvantage of little internal damping. As a result, a strong fundamental resonant mode beyond 18,000 Hz, occurs. Fortunately, this mode is at such a high frequency as to be inaudible.

3.9.1.1.4 Suspension Systems

The driver has inner and outer suspension systems to align the voice coil former and the cone. The purpose of the suspension systems is to provide a restoring force to the movement of the voice coil and cone. Additionally, they serve to provide lateral stability to restrict side-to-side movement. This is particularly important for the voice coil where it is centered in a magnetic gap on the order of 0.010 in!

The inner suspension assembly, called the "spider", attaches the exterior of the voice coil former to the metal frame. A pleated design of treated fabric is commonly used to suppress resonance modes and permit the passage of air due to the backward cone motion.

The outer suspension system, called the "surround", attaches the cone perimeter to the frame. A secondary role of the surround is to suppress reflected traveling waves on the cone surface which can arrive out-of-phase with the indent wave. The surround has three primary geometries: (1) half-roll; (2) double half-roll; and (3) pleated accordion. The half-roll surround is usually made from polyurethane foam, or in more expensive designs, of synthetic rubber. Its geometry and materials results in a very compliant system. A disadvantage with polyurethane foam is it can disintegrate over time ("foam rot") due to atmospheric contaminants and humidity. The double half-roll and pleated accordion type surrounds are more rugged and are suitable for high excursion drivers. These surrounds are made of treated fabric and are not as compliant as the half-roll design.

3.9.1.1.5 Dust Cap

The dust cap is a hemispherical-shaped piece of paper, plastic, or metal attached to the voice coil former end. Its purpose is to prevent dirt and dust from entering into the tiny magnetic gap which could interfere with the moving voice coil. The dust cap has a secondary role to further stiffen the moving voice coil assembly. Some driver designs use the dust cap to extend the on-axis high-frequency response. At higher frequencies, the more massive cone decouples from the voice coil, and the dust cap becomes an effective radiator because of its smaller size.

The "whizzer" cone is a refinement on the dust cap and is commonly used on inexpensive full-range coaxial loudspeakers to extend the high-frequency response. A separate highly resonant small cone is directly attached to the voice coil former which decouples the short wavelengths reproduced by the larger cone. The lower mass and smaller moving piston size of the whizzer cone, compared to the larger cone, extends the high-frequency output and provides a larger coverage angle.

3.9.2 Low-Frequency Enclosures

Cone drivers are normally installed in an enclosure which acts as a baffle extending the low-frequency response and increasing the driver efficiency. Trade-offs between bass extension, efficiency, and enclosure size are part of the design process. Increasing the enclosure size improves bass extension and efficiency, but large enclosures can pose problems with room installation. Various designs have evolved to extend low-frequency response using relatively small enclosures. The primary enclosures include: (1) sealed; (2) ported; (3) ported horn-loaded; and (4) bandpass designs.

3.9.2.1 Sealed Enclosure

The simplest enclosure is the sealed box with the driver mounted on the front baffle. The sealed enclosure prevents rearward radiating acoustical energy from combining with forward radiating acoustical energy due to motion of the cone. The rearward radiating acoustical energy is absorbed by fiberglass damping material installed within the enclosure. In order to perform optimally, sealed enclosures must not have acoustical leaks. Sealed enclosures have limited sound level output and low-frequency extension. Typical driver response in a 2 ft^3 volume enclosure is approximately 70 Hz at the -3 dB down points. Due to its relative inefficiency compared to the ported enclosure, a larger power amplifier is required to develop similar sound level output.

The most common sealed enclosure is based on the "acoustic suspension" design principle developed by Harry Olson of RCA and refined in 1954 by Edgar Villchur of Acoustic Research. Here a small diameter heavyweight cone with a flexible surround, capable of large excursions, is suspended in the enclosure. The driver resonant frequency rises when mounted in the enclosure due to less air compliance, compared to the free air condition. The sealed air within the enclosure acts as a linear spring and provides a restoring force to return the cone to its equilibrium position. When the *tuning ratio* is greater than four the enclosure compliance dominates. The fiberglass damping material extends the system linearity, low-frequency response, and attenuates standing waves within the enclosure. The sealed enclosure system is characterized by a 12 dB/octave attenuation below the fundamental system resonance.

3.9.2.2 Tuned Ported Enclosure

A more complex enclosure is the tuned ported ("bass reflex") design with the driver mounted on the front baffle and single or multiple port(s) located on the front or back of the enclosure. The port(s) can be of any rigid material, such as wood, cardboard, or PVC plastic, in round, rectangular, or square shapes. The tuned ported enclosure carefully controls the interaction of the cone's rearward and forward radiating acoustical energy. Over a narrow bandwidth, the rearward radiating acoustical energy emanates from the port, and combines in-phase with the front radiating acoustical energy from the cone. The low-frequency response below the driver free air resonant frequency is extended through superposition of sound from the cone and port. Tuned ported enclosures have greater sound level output and low-frequency extension than simple sealed enclosures. Typical driver response in a 4 ft^3 volume enclosure is approximately 60 Hz at the -3 dB down points. Advantages of tuned ported enclosures include smaller physical size, less driver distortion since the lowest frequencies are handled by the port, and driver protection from excessive cone excursion at high sound levels. Figure 3-34 shows a ported low-frequency enclosure.

FIGURE 3-34. Ported low-frequency enclosure with 12 in driver (Renkus-Heinz CELF12-1). Photo courtesy of Renkus-Heinz, Inc.

The tuned ported enclosure behaves as a simple Helmholtz resonator. The enclosure has a ported opening of a certain diameter and length which is tuned to extend the system low-frequency response. The air in the length of the port acts as an acoustical mass. The enclosure is tuned to the lowest design frequency to be reproduced, which can result in tunings between 20 and 45 Hz. The tuning is achieved by adjusting the port diameter and length. Increasing port area (all other parameters equal) raises the resonant frequency. Increasing the port length (all other parameters equal) lowers the resonant frequency. Thus, to maintain the tuned resonant frequency requires increasing both the port area and the length.

Matching the driver free air resonant frequency and the tuned enclosure resonant frequency determines the overall system response. At the enclosure resonant frequency, the cone excursion is minimal and the *volume velocity* of air through the port increases, resulting in significant sound level output through the port. As the enclosure resonant frequency approaches the driver free air resonant frequency, the output from the port diminishes and becomes negligible. It is common for the audio signal to be electronically high-pass filtered below the port resonant frequency to prevent driver over-excursion, which would result in excess distortion. Figure 3-35 shows typical ported loudspeaker response characteristics.

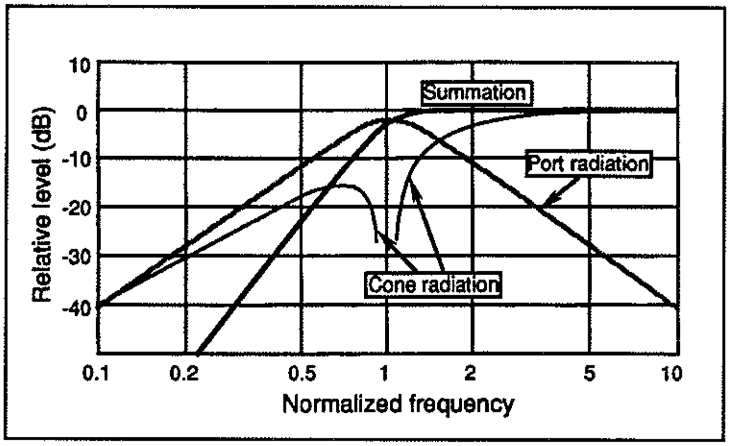


FIGURE 3-35. Typical ported low-frequency loudspeaker response characteristics showing the relationship between cone radiation, port radiation, and the summation of the two responses. Data after of JBL, Inc. and John M. Eargle.

The location of the port(s) is not critical and can be at the enclosure back assuming there is no obstruction from nearby surfaces. Turbulence-generated noise can be significant due to the high air velocities passing through the port. Flared port configurations, port damping, or avoiding port geometries with areas less than approximately one-third the driver diameter can reduce port noise. Tuned ported enclosures use little internal damping material compared to sealed enclosures. Excessive damping material will have the effect of reducing the low-frequency output from the port. The tuned ported enclosure system is characterized by a 24 dB/octave attenuation below the fundamental system resonant frequency.

(See Technical Notes, Sections 3.M and 3.N, at the end of this chapter, for additional information on port enclosure tuning and Thiele-Small parameters.)

3.9.2.3 Ported Horn-Loaded Enclosure

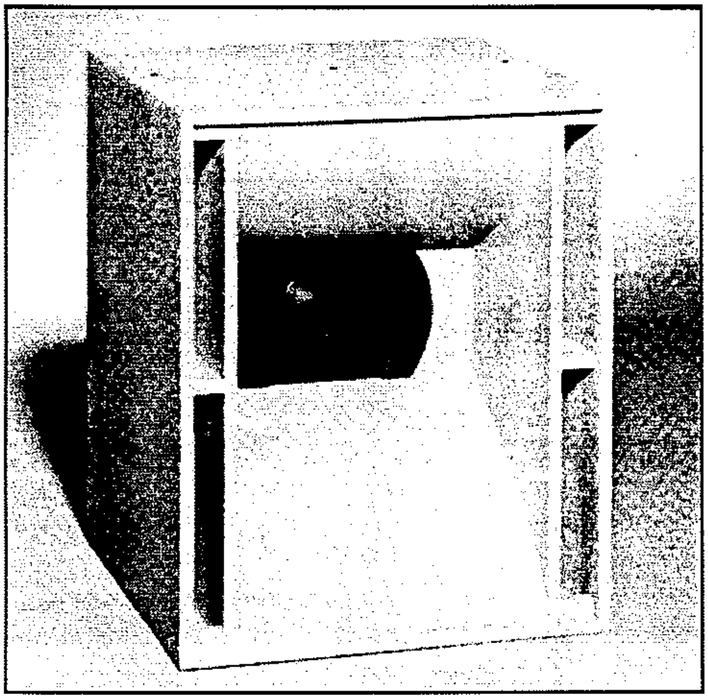


FIGURE 3-36. Ported horn-loaded low-frequency enclosure with 16 in driver (Altec Lansing 816VI). Photo courtesy of Mark IV Audio Group.

Ported horn-loaded loudspeakers combine the advantages of increased low-frequency output due to the tuned ports with improved directivity control due to the horn. The ports are tuned in the 40 to 60 Hz region and the front-mounted driver radiates into the horn throat. For a single driver arrangement, the horn loading is effective to approximately 200 Hz. When two drivers are used, the horn loading is lowered to approximately 150 Hz. Figure 3-36 shows a ported horn-loaded enclosure.

The low-frequency acoustical output from the ports is similar to the tuned ported enclosure discussed above. The acoustical output increases over the mid-bass region compared to the tuned ported enclosure, making these systems appropriate for high output speech applications. Typical driver response in a 10 ft^3 volume enclosure is approximately 55 Hz at the -3 dB down points. Due to its relative efficiency compared to the ported enclosure, a smaller power amplifier is required to develop a similar sound level output.

3.9.2.4 Bandpass Enclosure

Bandpass enclosures are characterized by the driver placed within a chamber internal to the loudspeaker enclosure so the cone does not radiate directly into the surrounding air. Sound emanates from one or more ports connected to the enclosure chambers. The enclosure passband is determined by the low-pass and high-pass chamber tunings within the enclosure. The primary advantage with these systems is the loudspeaker response can be tuned to the desired passband which eliminates the need for low- or high-pass frequency equalization. These systems are not frequently used due to design complexities and construction costs. Figure 3-37 shows two bandpass enclosures.

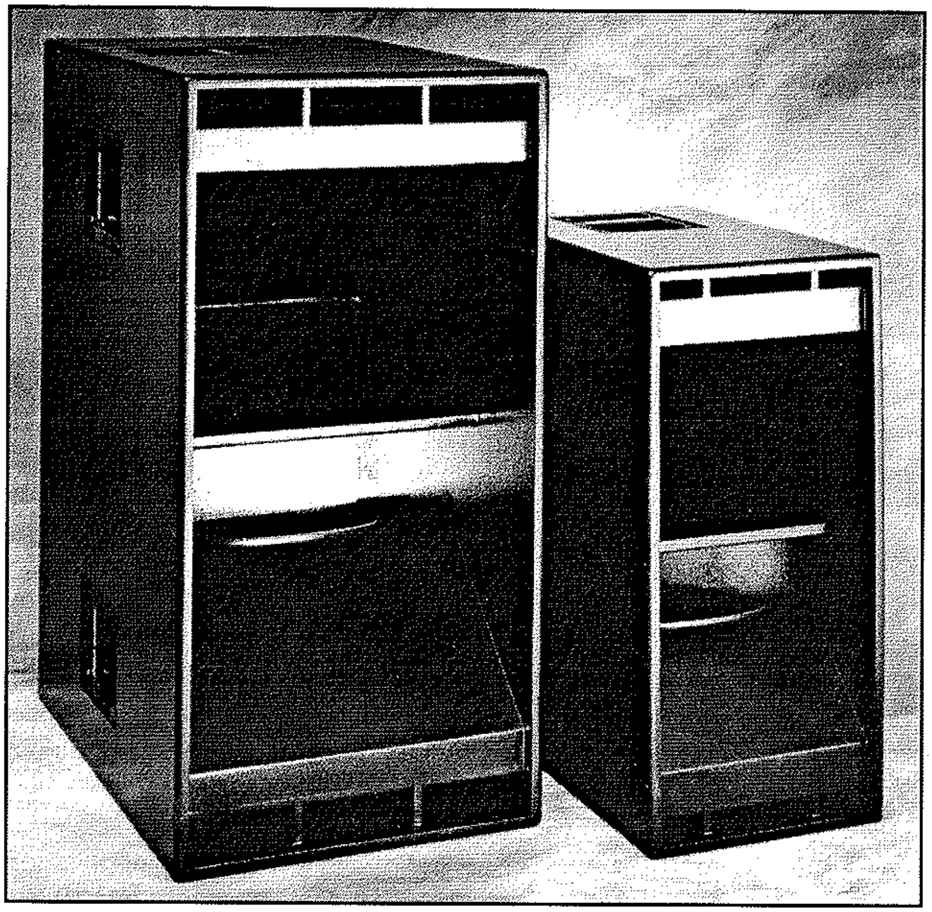


FIGURE 3-37. Bandpass low-frequency enclosure with two 15 in drivers (Renkus-Heinz BPS15-2 left) and two 12 in drivers (Renkus-Heinz BPS12-2 right). Photo courtesy of Renkus-Heinz.

The three principle bandpass designs include: (1) sealed-series enclosure; (2) ported-series enclosure; and (3) parallel enclosure. The sealed-series design places the driver in a sealed infinite baffle rear chamber radiating into a forward chamber with a port at the front radiating into the surrounding air. The ported-series design is similar to the sealed-series design, except the rear chamber is ported, enabling sound to radiate into the front chamber by the driver and the port. The parallel design is similar to the ported-series design except the rear chamber is ported to the exterior. The latter two enclosure designs have been developed and patented by Bose Corporation.

3.9.3 Compression Drivers

A compression driver is a special type of transducer which shares many of the same design features as the dynamic cone driver. Compression drivers were developed in the early 1930s by Bell Telephone Laboratories and Western Electric to provide greater acoustical output than dynamic cone drivers. Their first application was in the burgeoning movie industry where it was necessary to fill large theaters with sound from low-powered (often less than 10 W) vacuum tube power amplifiers. A horn is attached to the end of the compression driver and provides both acoustical impedance matching and directional sound control. The efficiency of a compression driver varies between 20 and 40 percent when matched to a proper acoustical load, making them suitable for sound reinforcement applications. The basic design principles of the compression driver remain essentially unchanged from 60 years ago, although advances in materials science have resulted in more robust devices with greater power handling, extended frequency response, and lower distortion. Figure 3-38 shows compression drivers of different sizes.

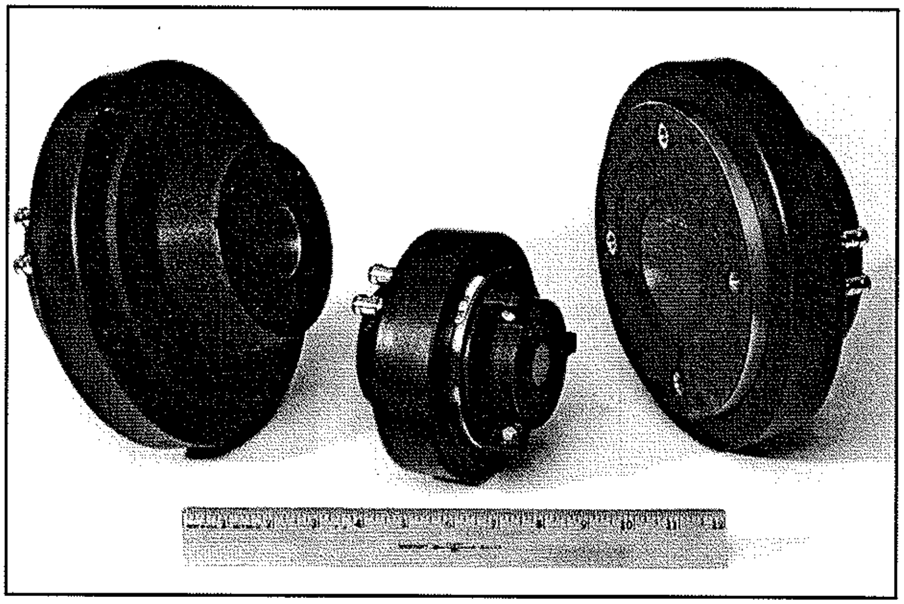


FIGURE 3-38. Compression drivers of various sizes: 4 in diaphragm/2 in exit size diameter full-range general purpose type (JBL 2446J left), 1¾ in diaphragm/1 in exit size diameter full-range general purpose type (JBL 2426H center), and 4 in diaphragm/3 in exit size diameter mid-range type (JBL 2490H right). Products courtesy of JBL, Inc.

Compression drivers can be grouped into three categories: (1) general purpose types with a nominal frequency response between 500 and 15,000 Hz, available with 500 or 800 Hz low-frequency cutoff limits; (2) mid-range types with frequency response between 200 and 4,000 Hz; and (3) very high-frequency types with frequency response between 3,000 and 20,000 Hz. These frequency limits are approximate and different manufacturer's products can vary from these generalized characteristics.

3.9.3.1 Basic Construction

The design and construction of a compression driver is a marvel of electro-acoustical engineering due to the precise manufacturing tolerances involved. For example, the gap separating the voice coil and the magnet and the gap separating the diaphragm and the *phasing plug,* have tolerances on the order of 0.002 in. Even with these minute manufacturing tolerances the motor assembly can move freely and generate extremely high sound levels. Figure 3-39 shows the major ports of the compression driver.

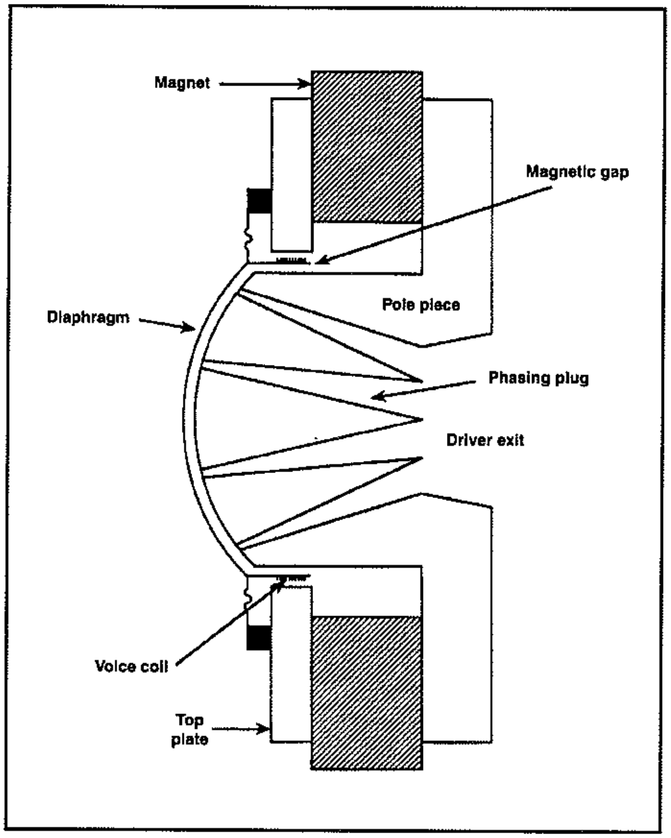


FIGURE 3-39. Sectional drawing of a typical compression driver showing details of the major component parts. Drawing courtesy of JBL, Inc.

The major difference between a compression driver and a standard dynamic cone driver is the use of a phasing plug. The basic construction of general purpose, mid-range, and very high-frequency compression drivers are similar, but each is optimized for its intended bandpass characteristics. Described below are the major design features applicable to these drivers.

(See Technical Notes, Section 3.O, at the end of this chapter, for additional information on equations for evaluating compression driver performance.)

3.9.3.1.1 Diaphragm

Sound radiates from the compression driver diaphragm, similar to the cone on a conventional dynamic driver. Diaphragms are constructed from a variety of materials including aluminum, beryllium, Mylar™, phenolic-coated linen, and titanium. Regardless of the material, the diaphragm must provide pistonic motion, exhibit extended high-frequency response, and resist damage from physical over excursion, and excessive amplifier power. A voice coil assembly suspended in a magnetic gap is terminated with two wires attached to the diaphragm which radiates into the phasing plug and into the horn throat.

The diaphragm of a compression driver is extremely lightweight. A 1¾ in diaphragm and voice coil weighs as little as 1½ g and a 4 in diaphragm and voice coil weighs approximately 3½ g. Compression drivers with larger diaphragms typically have greater sound level output and higher power handling capabilities, but high-frequency extension is relatively independent of diaphragm size.

Compression drivers are often classified by the diaphragm size. Standard diaphragm sizes include: (1) small, corresponding to 1, 1½, 1¾, and 2 in diameter; (2) medium, corresponding to 3 and 3½ in diameter; and (3) large, corresponding to 4 and 6¾ in diameter. The diaphragm size is larger than the phasing plug exit size which matches the internal diameter of the horn throat. Table 3-2 provides information on diaphragm and phasing plug sizes.

TABLE 3-2. Typical Diaphragm and Phasing Plug Exit Sizes

Diaphragm Diameter Size		Phasing Plug Exit Size, in
Size Classification	Physical Size, in	
Small	1, 1¼, and 1¾	1
	2	1 or 1⅜
Medium	3 and 3½	2
Large	4	1½, 2, or 3
	6¾	4

3.9.3.1.2 Phasing Plug

The compression driver diaphragm does not radiate directly into air. A phasing plug is placed in front of the diaphragm and sound passes through a series of annular or tangential slit openings in the phasing plug before entering the horn throat. The slit openings provide varying path lengths for sound from different diaphragm regions to pass through. This results in sound arriving in-phase at the exit of the phasing plug and reduces the likelihood of phase cancellation at the entrance to the horn throat.

Most compression drivers have a slit area which is approximately 10 percent of the diaphragm area. The compression driver efficiency is related to the ratio of the slit-to-diaphragm areas, known as the loading factor. The restricted openings in the phasing plug results in high pressure developed in the air space between the diaphragm and the phasing plug. A pressure-to-volume velocity transformation of 10-to-1 occurs between the diaphragm and the phasing plug exit, which is capable of driving a horn to high sound levels.

3.9.3.1.3 High-Frequency Response

Compression drivers are designed to reproduce mid- to high-frequencies. All drivers begin to roll-off their response above the *mass break-point frequency* ($\mathbf{f_{HM}}$), which for most compression drivers is approximately 3,500 Hz. Above this frequency, the driver high-frequency response decreases at 6 dB/octave. Figure 3-40 shows typical mass break-point response characteristics.

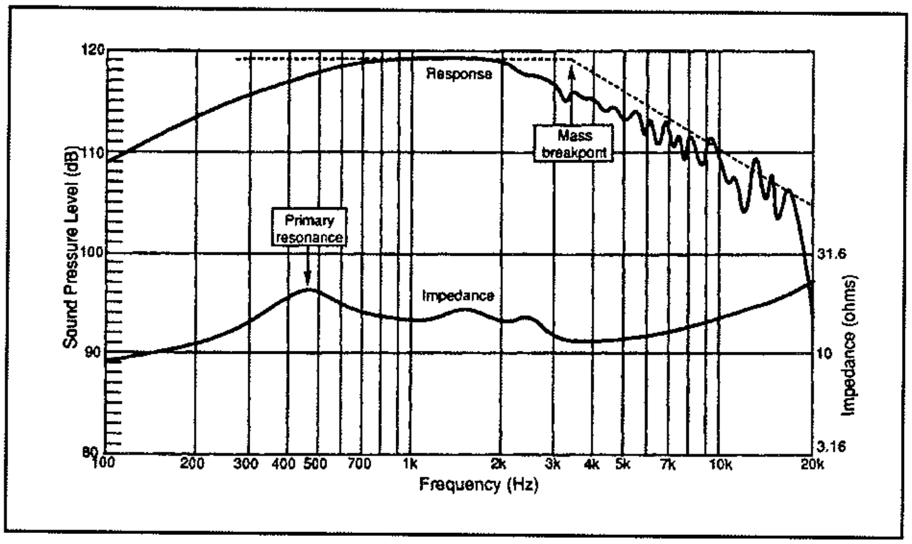


FIGURE 3-40. Typical compression driver frequency response showing high-frequency roll-off above the mass break-point frequency. Data after JBL, Inc. and John M. Eargle.

Certain standardized sound system frequency equalization practices may approximate the high-frequency roll-off above the mass break-point frequency. In such cases, the mass break-point frequency roll-off is not a critical consideration in the sound system response and usually can be ignored. Speech reinforcement systems are more tolerant of the high-frequency roll-off above the mass break-point frequency. Often with music programs, which require extended high-frequency response, it becomes necessary to boost the high-frequency response above the mass break-point frequency, or supplement the sound system with very high-frequency drivers, to achieve the necessary frequency equalization characteristic.

The compression driver can be modeled as an equivalent electrical circuit. This model equates three reactive elements in the driver which controls the high-frequency response: (1) shunt capacitance; (2) voice coil inductance; and (3) the air volume between the diaphragm and the phasing plug. The shunt capacitance determines the mass break-point frequency. The voice coil inductance can result in another high-frequency break-point, but some manufacturers compensate for this through techniques in voice coil manufacturing. The air volume, if too large, can cause yet another high-frequency break-point, but it occurs at a very high-frequency and can be ignored.

The mass break-point frequency is essentially the same for all compression driver diaphragm sizes. Large diaphragm compression drivers achieve this by using a larger magnet and a longer voice coil, providing a greater driving force, which offsets the larger diaphragm mass and higher voice coil resistance.

One way to compensate for high-frequency roll-off above the mass break-point frequency is to take advantage of secondary resonances developed on the diaphragm and the surround. Secondary resonances on the surround result in greater motion by the surround than the diaphragm, causing a peak in the frequency response, followed by a rapid high-frequency roll-off. Secondary resonances also occur due to break-up modes on the diaphragm. It is possible to control the secondary resonances, and extend high-frequency capabilities, but this results in a reduced output level with a slightly rolled-off response. Finally, the choice of surround material and design can help extend flat frequency response above approximately 8,000 Hz.

The compression driver low-frequency response is controlled by the fundamental resonant frequency of the diaphragm. For most compression drivers this occurs between 500 and 800 Hz. Many horns can provide loading below the diaphragm fundamental resonant frequency. Damage to the compression driver can occur when coupled to a horn with a lower cut-off frequency than the compression driver.

3.9.3.1.4 Efficiency

The maximum theoretical efficiency of a compression driver operating in its mid-band frequency range below the mass break-point frequency is 50 percent. This will only occur when the voice coil resistance ($\mathbf{R_E}$) and the effective radiation resistance ($\mathbf{R_{ET}}$) of the compression driver are equal. Rarely does this idealized condition occur and most practical compression drivers have efficiencies in the 25 to 30 percent range, with some devices approaching 40 percent efficiency. The efficiency differences are primarily due to eddy current losses in the driver top plate and the magnetic pole piece. Even when operating in the lower efficiency range, a compression driver is at least five times more efficient than the most efficient dynamic cone driver.

3.9.3.1.5 Distortion

Compression drivers generate predominately harmonic distortion which can be very audible and is a characteristic of "horn sound." The air in the gap between the diaphragm and the phasing plug is thermodynamically altered due to extremely high acoustical pressures which exist between these two elements and is the primary cause of the distortion. Moving diaphragm or motor unit non-linearities from excessive input signal level are a negligible contributor to distortion. The design of newer large diaphragm compression drivers having exit geometries matched to the throat of a rapid flare rate horn significantly reduces distortion.

The harmonic distortion byproducts are primarily second and third harmonics of the fundamental resonant frequency. Both occur over the majority of the compression driver operating bandwidth and increase with frequency and/or greater input drive level. With a constant drive level, the second harmonic distortion will double per doubling of frequency and the third harmonic distortion will increase as the square of the frequency. The third harmonic distortion is most noticeable above 5,000 Hz and is a cause of "harsh" sounds produced by some compression driver and horn combinations. The larger diaphragm compression drivers tend to have less harmonic distortion than the smaller diaphragm devices. They are also able to handle greater power due to a larger voice coil and have higher heat sinking capacity. Thus, larger diaphragm compression drivers are more suitable for high-level sound reinforcement applications.

Another distortion related to compression driver non-linear behavior is intermodulation distortion. This type of distortion results in upper and lower frequency sidebands. Generally, the upper sidebands are sufficiently high in frequency and are not audible, but the lower sidebands can fall into the mid-range bandwidth where they are very audible.

3.9.3.1.6 Power Handling and Driver Protection

The power handling capacity of compression drivers is considerably less than most dynamic cone drivers. In practice, the higher efficiency of the compression driver offsets the reduced power handling capability and achieving adequate sound level is generally not a problem. The high-frequency power handling is controlled by the voice coil heat dissipation, referred to as the thermal power rating. The low-frequency power handling is controlled by the diaphragm displacement at the fundamental resonant frequency. Damage can occur when the diaphragm contacts the phasing plug. The long-term power handling capability of compression drivers is rated at the crossover frequency and the crossover slope attenuation characteristics. Some manufactures provide data on short-term peak power levels which correspond to speech and music dynamic signal transients.

Compression drivers should be protected to prevent inadvertent operational damage. Common sources of damage include: (1) low-frequency transients as amplifiers are turned on and off, or from an accidently dropped microphone; (2) harmonic distortion; and (3) excessive amplifier power. Any one of these can damage the voice coil, by exceeding the thermal power rating, or the diaphragm, through contact with the phasing plug.

Compression drivers which are part of a multi-way loudspeaker system with an internal passive crossover generally do not require protection. Compression drivers used with an active crossover in bi-amplified or tri-amplified systems should be protected. The most common form of protection is a "blocking" capacitor in series between the power amplifier output and the compression driver input. The capacitor rating is selected to provide a reactance at one-half the crossover frequency which equals the driver nominal impedance. Both quantities are measured in ohms. Table

3-3 provides values of capacitors for different cutoff frequencies, below which protection is provided, and loudspeaker impedances.

TABLE 3-3. Capacitor Values, Loudspeaker Cutoff Frequency, and Loudspeaker Impedance

Capacitor Value, μFarad	Cutoff Frequency 4 Ω Loudspeaker	Cutoff Frequency 8 Ω Loudspeaker
72.0	1,000	500
52.0	1,500	750
20.0	4,000	2,000
16.5	5,000	2,500
13.5	6,000	3,000
12.0	7,000	3,500
8.0	10,000	5,000
7.0	N/A	6,000
6.0	N/A	7,000
4.0	N/A	10,000

Some loudspeaker manufacturers have produced dedicated electronic controllers which provide driver protection, usually by automatically limiting the signal level to the driver, once a certain threshold level is exceeded.

3.9.3.2 Mid-Range Compression Drivers

Special mid-range compression drivers have been developed to reproduce the frequency range between 200 and 4,000 Hz. The majority of acoustical energy in speech and music is contained in the 200 to 2,000 Hz bandwidth, which is one decade wide. A conventional two-way loudspeaker system comprising separate low- and high-frequency drivers can have difficulty reproducing the critical mid-frequency range. The low-frequency driver rolls-off between 500 and 1,000 Hz and the high-frequency driver can have inadequate power handling to reproduce the 500 to 1,500 Hz range with satisfactory loudness. Thus, mid-range drivers are often used as part of three-way loudspeaker systems comprising separate low-frequency, mid-range, and high-frequency drivers. Figure 3-41 shows a large mid-range compression driver.

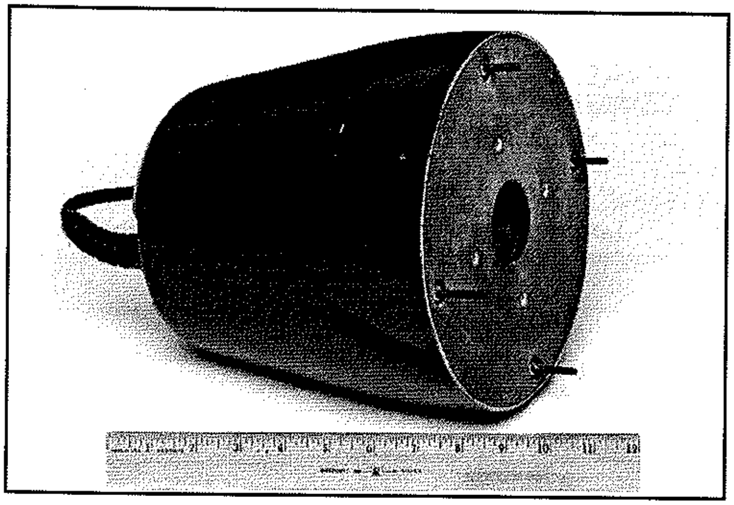


FIGURE 3-41. Mid- to high-frequency range compression driver. (Renkus-Heinz CDT-1). Product courtesy of Renkus-Heinz.

The mid-range driver has a lower cutoff frequency than the general purpose compression driver, requiring a larger diaphragm. Diaphragm sizes up to 6¾ in diameter are used in some mid-range drivers, with a phasing plug slit area approximately 25 percent of the diaphragm area. The phasing plug geometry results in a pressure-to-volume velocity transformation of 4-to-1 helping to reduce distortion due the high intensity sound levels in the horn throat.

Other design features of mid-range drivers include larger voice coils and magnets enabling the drivers to handle greater power amplifier input. Typical efficiencies are in the 25 to 30 percent range, with some devices having efficiencies in excess of 40 percent.

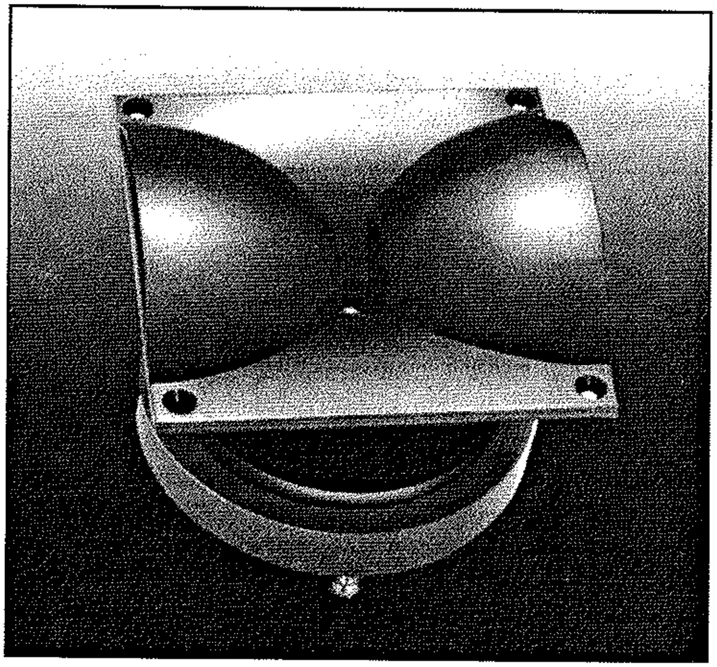


FIGURE 3-42. Very high-frequency ring radiator driver (JBL 2404H). Photo courtesy of JBL, Inc.

3.9.3.3 Very High-Frequency Compression Drivers

A problem with general purpose compression drivers is the high-frequency response rolls-off rapidly above the mass break-point frequency. This can be compensated through electronic frequency equalization, but has disadvantages in that the driver power derating has to be accounted for. Additionally, excess phase distortion can result with frequency equalization. When extended high-frequency response is required, the compression driver can naturally roll-off above the mass break-point frequency, where it is crossed over to a very high-frequency compression driver called a ring radiator. The ring radiator can extend the frequency response in the 3,000 to 20,000 Hz range. The extended high-frequency response is generally achieved only on-axis and the response diminishes rapidly beyond about 30° off-axis unless multiple units are used. Figure 3-42 shows a ring radiator.

The construction of a ring radiator combines a compression driver and a small horn into a single integrated unit. The diaphragm is approximately 1¾ in diameter and is anchored both at the middle and the outer perimeter. A torus (ring-shaped) radiating element results from the double clamping and acts as a double tuned system. The diaphragm outer perimeter has greater mass and a lower resonant frequency than the inner diaphragm area, which weighs less, and has a higher resonant frequency. The combined resonant frequencies generate sound over a two octave wide passband. The efficiency of ring radiators is low compared to other compression driver types, with efficiency ratings between 5 and 10 percent. The nominal power handling can be up to 40 W. Where greater sound levels are required, multiple units can be used, and when horizontally splayed, extends the high-frequency coverage area. Sound levels will increase 3 dB for each numerical doubling of ring radiators. Different horn configurations are available including exponential, bi-radial, and diffraction types having nominal coverage angles of 40° horizontal by 40° vertical, 100° horizontal by 25° vertical, and 100° horizontal by 100° vertical.

3.9.4 Horns

Horns have been used for thousands of years as communication devices and are found on many musical instruments. Their first electro-acoustic application was in the early telephone and phonograph recorders of the 1870s developed by Alexander Graham Bell and others. Horn loudspeakers as we know them today evolved from research at Bell Telephone Laboratories, General Electric Company, RCA, and Westinghouse Electric in the late 1920s and early 1930s paralleling the development of the compression driver.

Examining the history and application of horn loudspeakers suggests evolutionary steps in 20 year increments. The first commercial horn applications were in the 1910s when simple conical horns were used on acoustic phonograph record players. In the 1930s advances in horn technology produced the multi-cellular horn with greater sound levels for use in movie theaters. Twenty years later, radial horns were developed to provide wider sound coverage and lower distortion for use in recording and broadcast facilities. In the 1970s constant directivity horns evolved for the touring and installed sound industries where very high sound levels and even coverage were necessary. The 1990s has seen advances in horn technology taking advantage of new materials and manufacturing techniques coupled with DSP signal processing to achieve greater fidelity and sound control over larger audience seating areas.

In its simplest form a horn is a smoothly tapered tube-like element with a varying sectional area. It is small at one end, which connects to a compression driver, and larger at the opposite end which terminates in a radiating medium, usually air. The horn comprises four basic elements: (1) throat flange to mount the compression driver; (2) flared throat to couple sound from the compression driver; (3) flared horn to act as a waveguide to radiate sound in a directional pattern; and (4) mouth to match the horn impedance to the radiating medium. The horn can be viewed as an acoustical transformer which matches the high mechanical impedance of the diaphragm to the low acoustical impedance of the air. High pressure and low volume velocity are present at the horn throat, which is transformed to low pressure and high volume velocity at the horn mouth. The horn improves the coupling of the compression driver to the radiating medium, thus increasing its efficiency.

Horns are broadly classified as *near-throw* and *far-throw* devices. The names suggest the degree to which the horn can project sound to a given location. Horizontal and vertical *coverage angles* define the geometry of the sound beam projected by the horn. Near-throw horns are physically smaller than far-throw horns. Larger horns are required to project the sound beam within its defined coverage angle to a distant location. The projected horn coverage angles onto the audience will be different than the nominal horn coverage angles. Tilting the horn downward at the audience, and the spherically radiating waves projected onto the flat audience plane, cause an apparent distortion of the nominal horn coverage angles.

A horn operates over a given frequency bandwidth determined by the cutoff frequency. Below the cutoff frequency the horn does not provide an acoustical load to the driver diaphragm. The lower the cutoff frequency, the larger the horn must be. Most large mid- and high-frequency horns have a low-frequency cutoff around 300 Hz. Above the horn low-frequency cutoff, the wavelength of the reproduced sound is smaller than the horn mouth dimensions, resulting in directional sound control. The horn mouth dimensions determine the upper and lower frequency limits of directional control. As the frequency increases, and the wavelengths become much smaller than the horn dimensions, directional control is lost and high-frequency "beaming" results. At low-frequencies, where the wavelengths are larger than the horn dimensions, the sound diffracts around the mouth and radiates in a more omnidirectional pattern. For example, a small format horn with mouth dimensions of 6 in high by 12 in wide will exhibit directional control down to 2,260 Hz (vertical plane) and 1,130 Hz (horizontal plane). A large format horn with mouth dimensions of 36 in high by 36 in wide will exhibit directional control down to 375 Hz in both planes. As can be seen, large horn sizes are necessary for low-frequency directional control. Low-frequency directional sound control can make or break a sound reinforcement system since it affects gain-before-feedback, the **D/R** ratio, and speech intelligibility.

Horns are constructed from a variety of materials. The early horns were fabricated from various metals, which have undesirable resonant frequencies, unless treated with a damping compound. Wood was commonly used as a substitute due to its non-resonant nature. However, wood is time consuming and difficult to fabricate for complex horn geometries. In the 1950s fiberglass composites were developed and adapted for use in horns. This material is ideal since it is non-resonant, can be easily molded, and is lightweight. Today, most horns are fabricated from fiberglass or plastic composites.

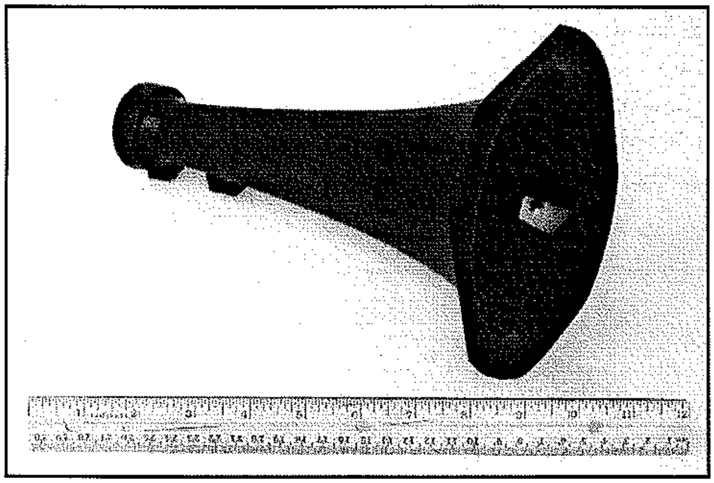


FIGURE 3-43. Exponential horn with 45° horizontal by 45° vertical coverage angle (Selenium HL14-25). Product courtesy of Selenium Loudspeakers USA.

3.9.4.1 Exponential Horns

Maxfield and Harrision of Bell Telephone Laboratories developed the generalized theory of exponential horns in 1925. The principles of exponential horns have served as a basis for the development of other horn geometries. Exponential horns exhibit poor directional control which narrows as frequency increases resulting in on-axis beaming. Pure exponential horns are not commonly used today, although the geometry is used as a starting point to shape some of the different horn geometrics discussed below. Figure 3-43 shows a small exponential horn.

The simplest horn shape is an exponentially expanding section of infinite length. The exponential flare rate of the horn is necessary to optimize compression driver acoustical loading. The flare constant (**m**) determines the horn shape as a function of the distance from the throat. For an exponential horn, **m** can be calculated using the following equation:

$$\mathbf{m} = \frac{4\pi \mathbf{f_c}}{\mathbf{c}} \tag{3.4}$$

where,

m	is the horn flare constant, meter^{-1}
4π	is constant
$\mathbf{f_c}$	is the horn cutoff frequency, Hz
c	is the speed of sound, 340 meter/s

The value of $\mathbf{f_C}$ should be one octave less than the lowest frequency the horn is to reproduce. Above $4\mathbf{f_C}$, the horn exhibits pistonic behavior. High-frequency horns are used in the range of 8 to $10\mathbf{f_C}$.

For an exponential horn, the radiation resistance ($\mathbf{R_{AT}}$) increases with frequency and is constant over most of the bandwidth while the radiation reactance ($\mathbf{X_{AT}}$) decreases with frequency. Below $\mathbf{f_C}$ the horn impedance (**Z**) will be controlled by $\mathbf{X_{AT}}$ and little acoustical power will be transmitted. At higher frequencies $\mathbf{X_{AT}}$ decreases and **Z** will be controlled by $\mathbf{R_{AT}}$. The loading will be resistance controlled when the ratio of $\mathbf{f/f_C}$ is essentially greater than or equal to two. Horns are designed to operate in the region where $\mathbf{R_{AT}}$ comprises the majority of the impedance characteristic.

The idealized curves of $\mathbf{R_{AT}}$ and $\mathbf{X_{AT}}$ for an infinite exponential horn will be smooth. However, all horns are a finite length. This departure from infinite length results in radiation resistance and reactance curves which are not smooth. The "ripples" in the response curves are due to reflections from the horn mouth back to the driver from the impedance mismatch with the radiating medium at the horn mouth. Figure 3-44 shows idealized and actual response characteristics for exponential horns.

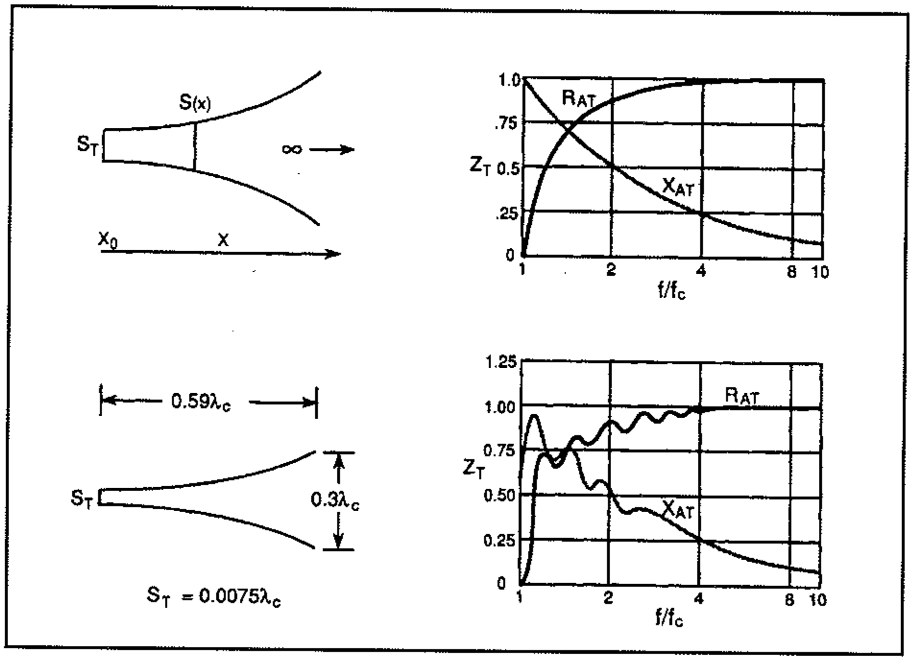


FIGURE 3-44. Typical radiation resistance and radiation reactance response characteristics for infinite (top) and finite (bottom) length exponential horns. Data after Leo L. Beranek.

The horn will exhibit a smooth radiation resistance characteristic and approximates infinite exponential geometry behavior when the horn mouth circumference dimension, divided by the wavelength of the lowest reproduced frequency, is greater than approximately three.

3.9.4.2 Multi-cell Horns

Multi-cell horns were first developed by Hannah and Slepian of Westinghouse Electric in the early 1920s and were refined by Wente and Thuras of Bell Telephone Laboratories in the early 1930s. This horn provides wider and more uniform horizontal and vertical coverage with less on-axis beaming than the exponential horn. Their first application was in movie theaters and later widely used for installed sound reinforcement systems. Figure 3-45 shows a multi-cell horn.

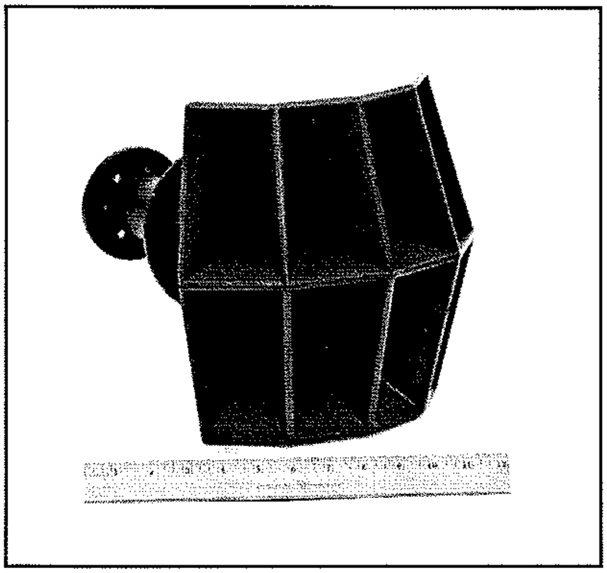


FIGURE 3-45. Wooden eight cell multi-cell horn with 100° horizontal by 40° vertical coverage angle designed and built by the late Paul S. Veneklasen. Product courtesy of the author.

Multi-cell horns comprise a grouping of small individual exponential horns with a nominal 25° horizontal and vertical coverage angles connected to a common throat and compression driver. The horn mouth is rectangular in shape with a large aspect ratio subdivided into 2, 6, 8, 10, 15, or 18 individual cells. The basic premise is that each cell controls sound radiation into a small area and the total number of cells defines the coverage area. Multi-cell horns are characterized by a wider horizontal coverage angle than vertical coverage angle. Typical coverage angles range from 35° horizontal by 25° vertical to 135° horizontal by 75° vertical. The low-frequency cutoff is 500 or 800 Hz.

Directional control for multi-cell horns above 1,000 Hz is determined by the number of cell multiples. A larger number of cells in the width or height dimension results in a wider coverage pattern. Between 500 and 1,000 Hz the coverage pattern narrows significantly as the number of cells increase. The narrowing is due to the individual cells acting as a single larger cell at the frequency having a wavelength equal to the width of the combined cell mouths. Multi-cell horns tend to exhibit *lobing* at frequencies starting around 4,000 Hz which becomes more severe as the frequency increases. The lobing is due to the individual cells decoupling from the group and approximating individual cells at higher frequencies. The lobing shape tends to follow the individual horn cell subdivision pattern. At lower frequencies directivity control diminishes and the multi-cell horn exhibits omnidirectional radiation.

One advantage multi-cell horns have over other horn types is the ability to modify the coverage pattern by plugging individual cells with fiberglass. This can be useful to reduce sound radiation off wall and ceiling surfaces which could give rise to echoes or decrease the **D/R** ratio. The fiberglass should not be rigid, otherwise sound would reflect back to the compression driver. In practice, fiberglass of nominal 1 to 3 lb/ft^3 density compressed approximately 25 percent of its original thickness is sufficiently absorptive. An attenuation of roughly 1 dB/in of fiberglass thickness can be achieved for the first 3 in; beyond this limit the attenuation is approximated by $2\log_{10}$ (depth of material). Approximately 9 dB of attenuation can be achieved without the sound reflecting back to the compression driver. Filling up to one-half the cells with fiberglass will not adversely affect the frequency response of the horn.

3.9.4.4 Constant Directivity Horns

Constant directivity (CD) horns, sometimes called constant or uniform coverage horns, were first developed by Don Keele of Electro-Voice in the early 1970s. Later refinements using a horizontal diffraction horn geometry were made by Cliff Henricksen and Mark Ureda of Altec Lansing in the late 1970s and then by Keele of JBL in early 1980s using a bi-radial horn geometry. CD horns have superceded multi-cell and radial horns in sound reinforcement applications due to their wider pattern control versus frequency characteristic, overall smoother and extended frequency response, greater sensitivity, and lower distortion from a more rapid horn flare rate. They have a distinct physical appearance identified by a wide rectangular single-cell mouth. Figures 3-47 and 3-48 show CD horns of different sizes.

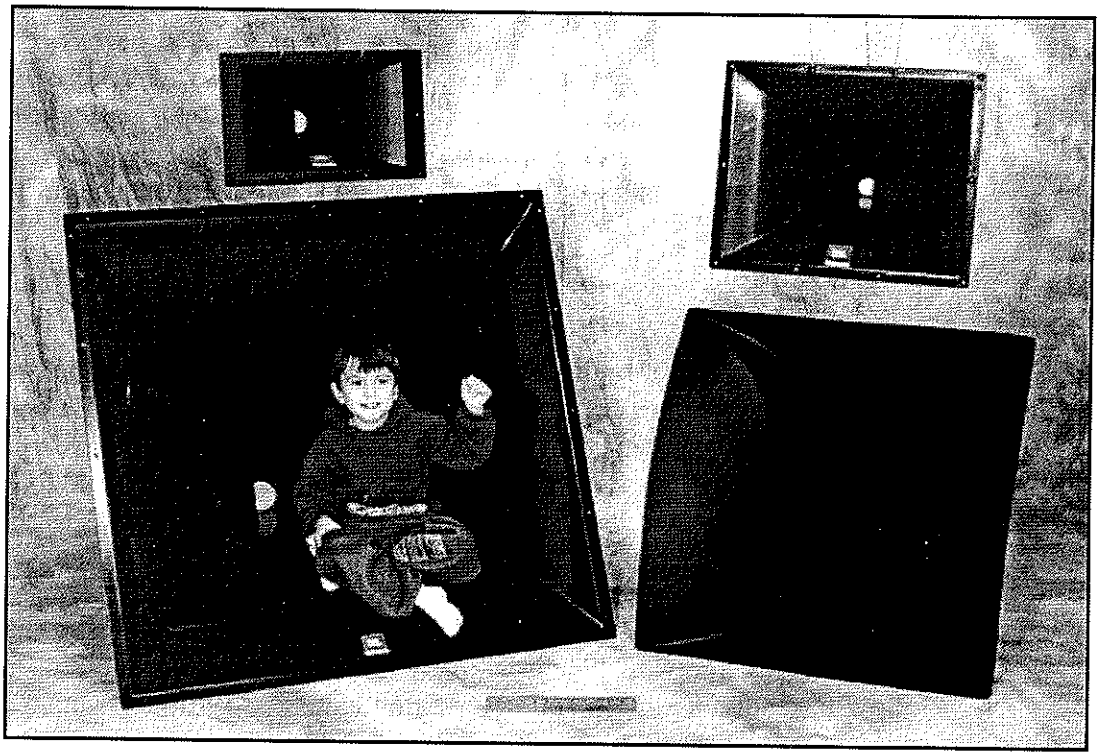


FIGURE 3-47. Frontal view of different size 60° horizontal by 40° vertical coverage angle constant directivity horns. Small format flat front bi-radial type (JBL 2385A top left), large format bi-radial (JBL 2393 bottom left), medium format bi-radial (JBL 2353 top right), and large format bi-radial constant directivity (JBL 2365A bottom right). Products courtesy of JBL, Inc. Nathan Vestrich-Shade (age 4) courtesy of author and Victoria A. Vestrich.

CD horns provide tight pattern control in both the horizontal and vertical planes characterized by even sound levels throughout the coverage area which then drop off abruptly at the pattern coverage edge. In contrast, the other horn types discussed above have good pattern control only in the horizontal plane which gradually rolls-off towards the pattern coverage edge. The design coverage angles of CD horns remain fairly constant throughout a frequency range between 800 to 13,000 Hz with horizontal pattern control maintained to roughly one octave below.

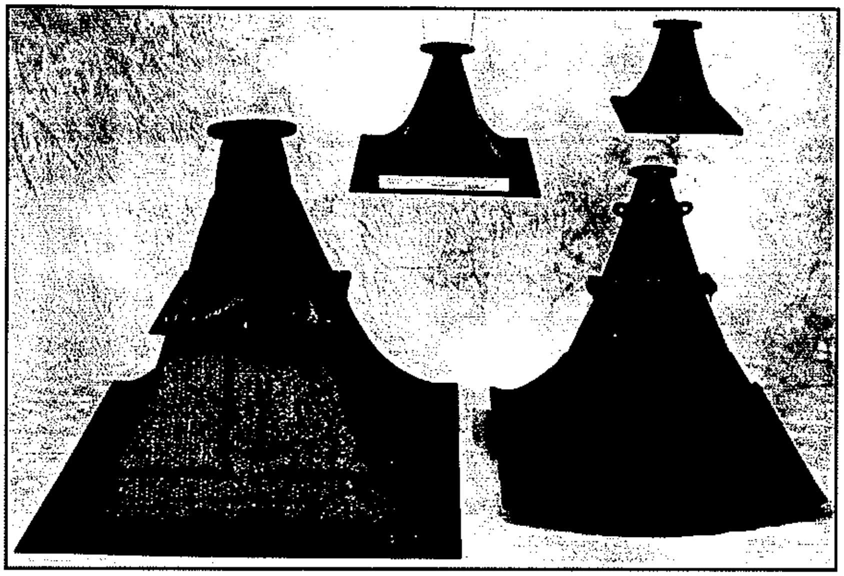


FIGURE 3-48. Side view of horns shown in Figure 3-47. Medium format bi-radial type (JBL 2353 top left), large format bi-radial (JBL 2393 bottom left), small format flat front bi-radial (JBL 2385A top right), and large format bi-radial constant directivity (JBL 2365A bottom right). Products courtesy of JBL, Inc.

The geometry of CD horns is quite complex and both flat and curved surfaces are used to provide the design coverage angles. Each of the three manufacturers noted above have developed their own unique horn geometries. The Electro-Voice CD horn uses a hyperbolic throat geometry coupled to a conical radial bell. The horn mouth was designed to correct for mid-range beaming by expanding the horn walls at a greater flare rate resulting in perimeter flanges at the horn mouth. The Altec-Lansing CD horn utilizes a vertical narrow angle diffraction horn, which transitions into a conical waveguide, terminating into a larger vertical horn mouth geometry with perimeter flanges to reduce mid-range beaming. The conical geometry provides driver loading only down to 800 Hz, which is adequate for two-way and voice loudspeaker systems. The JBL CD horn uses a radial horn geometry in both the horizontal and vertical planes without flanges at the horn mouth to reduce edge diffraction effects. In the horizontal plane, an exponentially expanding horn radiates into a small curved diffraction slot which transitions into a large flared bell with tapered sides to develop the design horizontal coverage angle. For the vertical plane, the horn shape is fairly straight except for a flaring at the mouth. The exponential horn section flare rate prior to the diffraction slot determines the vertical coverage angle. Typical coverage angles range from 40° horizontal by 20° vertical, 60° horizontal by 40° vertical, 90° horizontal by 40° vertical, and 120° horizontal by 40° vertical.

The high-frequency response of a CD horn will fall off above the compression driver mass break-point frequency. It is necessary therefore to use electronic frequency equalization to restore flat on- and off-axis response above approximately 3,500 Hz. Many electronic crossovers and signal processing devices have built-in CD horn compensation circuits. Alternately, separate electronic frequency equalizers can be used.

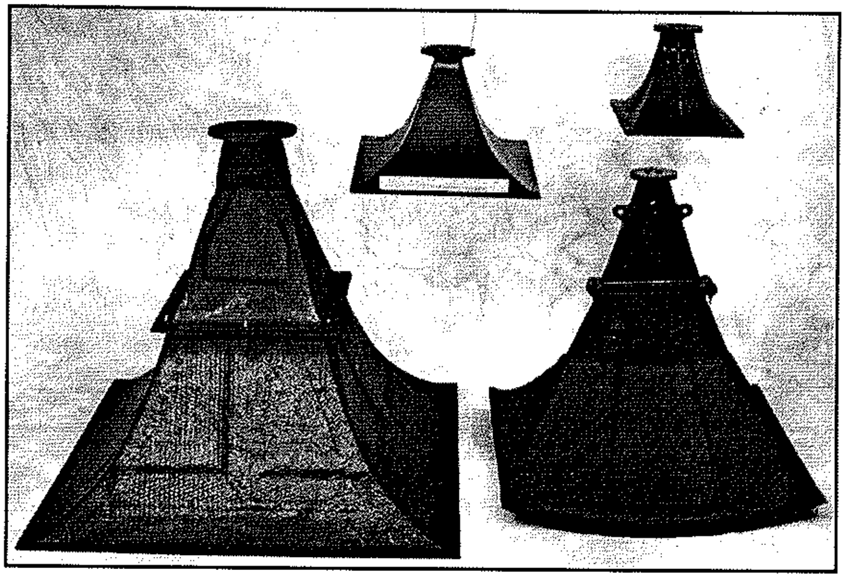


FIGURE 3-48. Side view of horns shown in Figure 3-47. Medium format bi-radial type (JBL 2353 top left), large format bi-radial (JBL 2393 bottom left), small format flat front bi-radial (JBL 2385A top right), and large format bi-radial constant directivity (JBL 2365A bottom right). Products courtesy of JBL, Inc.

The geometry of CD horns is quite complex and both flat and curved surfaces are used to provide the design coverage angles. Each of the three manufacturers noted above have developed their own unique horn geometries. The Electro-Voice CD horn uses a hyperbolic throat geometry coupled to a conical radial bell. The horn mouth was designed to correct for mid-range beaming by expanding the horn walls at a greater flare rate resulting in perimeter flanges at the horn mouth. The Altec-Lansing CD horn utilizes a vertical narrow angle diffraction horn, which transitions into a conical waveguide, terminating into a larger vertical horn mouth geometry with perimeter flanges to reduce mid-range beaming. The conical geometry provides driver loading only down to 800 Hz, which is adequate for two-way and voice loudspeaker systems. The JBL CD horn uses a radial horn geometry in both the horizontal and vertical planes without flanges at the horn mouth to reduce edge diffraction effects. In the horizontal plane, an exponentially expanding horn radiates into a small curved diffraction slot which transitions into a large flared bell with tapered sides to develop the design horizontal coverage angle. For the vertical plane, the horn shape is fairly straight except for a flaring at the mouth. The exponential horn section flare rate prior to the diffraction slot determines the vertical coverage angle. Typical coverage angles range from 40° horizontal by 20° vertical, 60° horizontal by 40° vertical, 90° horizontal by 40° vertical, and 120° horizontal by 40° vertical.

The high-frequency response of a CD horn will fall off above the compression driver mass break-point frequency. It is necessary therefore to use electronic frequency equalization to restore flat on- and off-axis response above approximately 3,500 Hz. Many electronic crossovers and signal processing devices have built-in CD horn compensation circuits. Alternately, separate electronic frequency equalizers can be used.

3.9.4.5 Defined Coverage Horns

Defined coverage horns, first developed by Don Keele of JBL in the mid 1980s and by Dave Gunness of Electro-Voice in the late 1980s, have an asymmetrical coverage pattern based on modified CD horns combining far-throw and near-throw horn sections. The coverage pattern in the horizontal plane can be symmetrical or asymmetrical, based on the horn design. The coverage pattern in the vertical plane is asymmetrical and is skewed front to back. These horns are intended to provide coverage for seating areas conforming to specific aspect ratios or geometries, such as rectangular or trapezoidal shapes. Figure 3-49 shows a defined coverage horn.

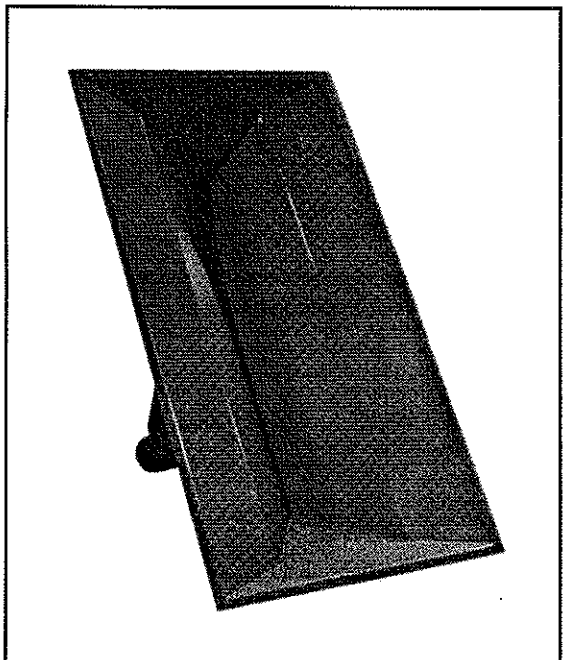


FIGURE 3-49. Asymmetrical defined coverage horn with 90° horizontal by 60° vertical near-throw and 60° horizontal by 60° vertical far-throw sections (Altec Lansing Vari-Intense™ VIR). Photo courtesy of Mark IV Audio Group.

The defined coverage horns provide a higher sound level, with a narrower horizontal coverage angle at the top of the vertical far-throw pattern, to cover distant seating locations. A lower sound level, with a wider horizontal coverage angle at the bottom of the vertical near-throw pattern, covers the nearby seating locations. These loudspeakers have output levels from the far-throw section between 6 and 9 dB greater than the near-throw section. Since the seats covered by the far-throw section may be two times farther away than the seats covered by the near-throw section, the direct sound levels for both will be approximately equal, due to greater inverse-square law loss at the distant seats. Typical coverage angles range from 60° horizontal by 60° vertical far-throw combined with 90° horizontal by 60° vertical near-throw to 38° horizontal by 25° vertical far-throw combined with 110° horizontal by 25° vertical near-throw.

Advantages with these horns include simplifying sound system design, since fewer horns and power amplifiers are required, reduced costs, and potential for higher **D/R** ratio in the room. A drawback is the horn positioning relative to the room geometry is more critical than conventional horns to achieve the intended coverage pattern. This may preclude their use in certain rooms.

3.9.4.6 Complex Conic™ and CoEntrant™ Horns

Complex Conic™ horns were developed in 1994 by Ralph Heinz of Renkus-Heinz and combine the advantages of oval and constant directivity horns. The horn is a complex oval shape and provides a broad elliptical coverage pattern over a wide frequency bandwidth. One advantage of this horn geometry is it eliminates the low-frequency directional 90° "*pattern flip*" common with rectangular CD horns. The smooth profile Complex Conic™ shape reduces reflections between the horn walls resulting in less harmonic distortion and a smoother frequency response. Typical coverage angles range from 60° horizontal by 40° vertical, 90° horizontal by 40° vertical, 90° horizontal by 60° vertical, and 120° horizontal by 60° vertical. Figure 3-50 shows a Complex Conic™ horn.

CoEntrant™ horns were developed by Ralph Heinz at Renkus-Heinz in 1996 and approximate a horn-within-a-horn design which shares some features of coaxial horns described below. The mid-range drivers are arranged symmetrically about the high-frequency section creating a coaxial source. A compound throat section is used with separate mid-range and high-frequency throat sections which combine into a common single throat horn. Different size horn apertures are used for the drivers to optimize performance. The high-frequency drivers have narrow apertures to ensure wide dispersion. The mid-range drivers have larger apertures to reduce throat distortion. The mid-range and high-frequency drivers are physically aligned resulting in a time coherent point source from the apex of the common horn, assuring good transition between mid- to high-frequencies. Typical coverage angles range from 60° horizontal by 30° vertical, 90° horizontal by 40° vertical, and 120° horizontal by 50° vertical.

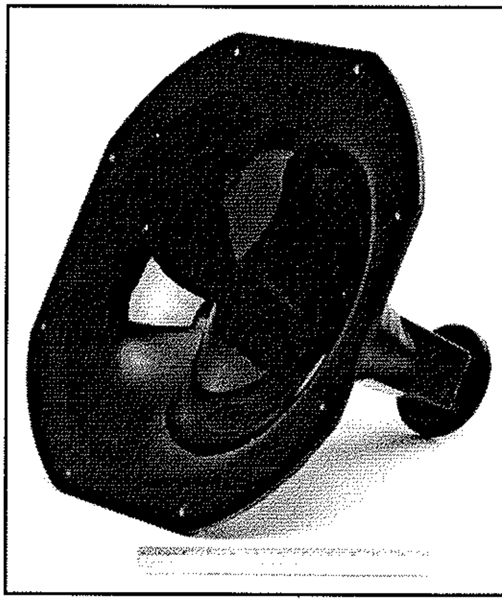


FIGURE 3-50. Complex Conic™ horn with 60° horizontal by 60° vertical coverage angle (Renkus-Heinz CDT500/66). Product courtesy of Renkus-Heinz.

(See Technical Notes, Section 3.P, at the end of this chapter, for additional information on horn directivity patterns.)

3.9.5 Paging Horns

Paging horns are complete with a compression driver and integral horn in a prepackaged assembly. They are often available with a line matching transformer for use in low-level constant voltage systems, but can be used as a high-level point source without the transformer. Construction of the horn is usually of metal, although plastics are increasingly used. Figure 3-51 shows a double reentrant paging horn.

The most common horn geometry is the reentrant horn, a type of folded horn having a longer path length between the compression driver and the horn mouth, than a standard horn. These systems have a restricted frequency response and frequency-dependent coverage angles due to the small compression driver and horn size. The useful frequency response of most paging horns is between 400 and 5,000 Hz, with the output falling off beyond these limits. The frequency response is not very smooth due to numerous system resonances. The horizontal and vertical coverage angles vary between 40° and 140° in the speech frequency range.

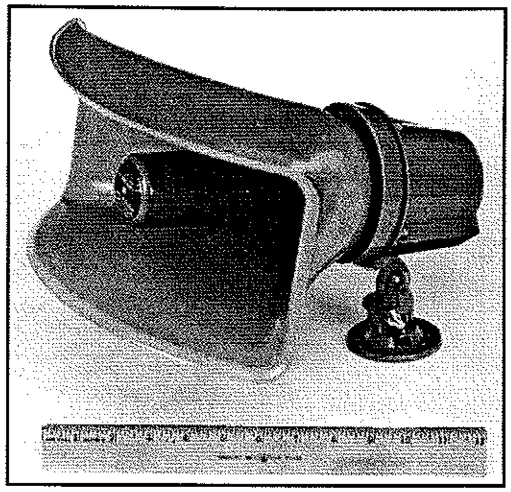


FIGURE 3-51. Double reentrant paging horn with 140° horizontal by 80° vertical coverage angle (Atlas/Soundolier APC-30T). Product courtesy of Atlas/Soundolier, Inc.

Paging horns are commonly used in industrial and transportation facilities where greater sensitivity, typically 10 dB higher than standard cone drivers, and rugged construction are advantageous. As the name implies these horns perform best for voice annunciation only.

3.9.6 Coaxial Loudspeakers

Coaxial loudspeakers were developed in the 1940s by Altec Lansing, Jensen, RCA, and Tannoy to provide a wide frequency range system in a single-chassis assembly. Notable loudspeakers were developed by James B. Lansing, the Model 604, for Altec Lansing in 1943 and by Rackham, the Dual Concentric, for Tannoy in 1947. These two loudspeakers established a basis of design for this loudspeaker type which has gradually been improved over the years.

The coaxial loudspeaker utilizes separate low- and high-frequency drivers, with the high-frequency driver mounted in the center of the low-frequency driver. The in-line drivers radiate from the same point as would a single driver. Some designs use a small format horn to reproduce the high frequencies. A crossover with steep slopes and signal delay to align the *acoustic centers* of the individual drivers can optimize the system frequency and time domain performance. Figure 3-52 shows the major parts of the coaxial loudspeaker.

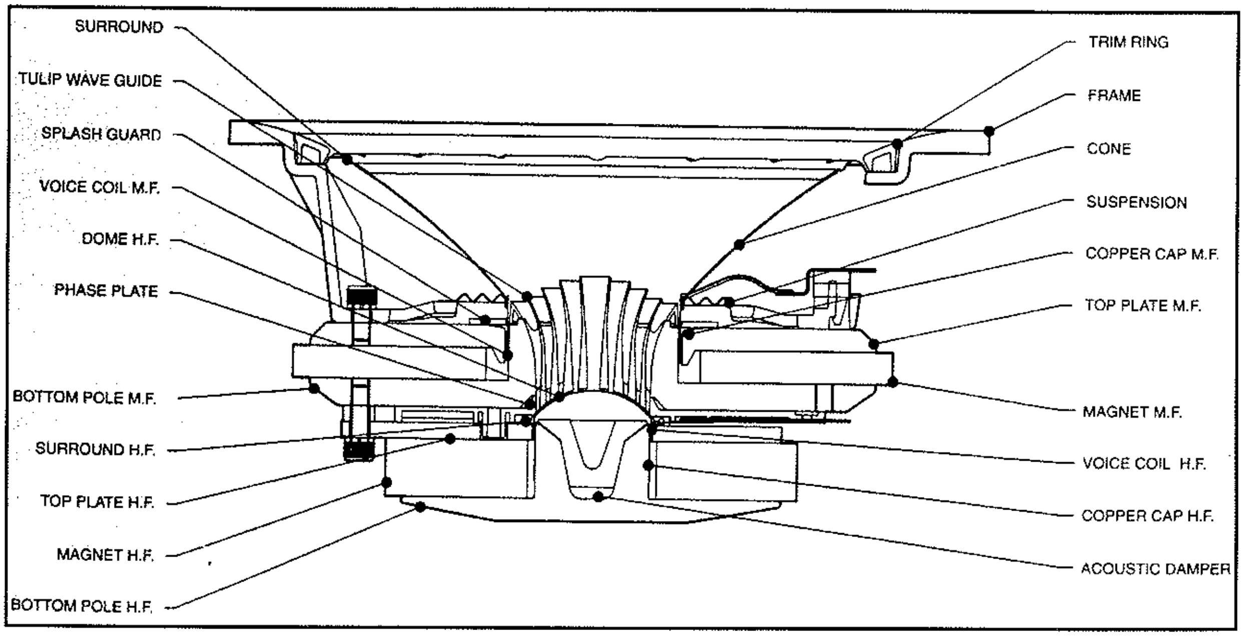


FIGURE 3-52. Sectional drawing of a typical coaxial driver showing details of the major component parts. (Tannoy Super Dual Concentric). Drawing courtesy of Tannoy.

Advantages of coaxial loudspeakers compared to loudspeakers with separately spaced drivers include: (1) wider high-frequency coverage with smoother off-axis response; (2) improved vertical coverage; (3) less diffractive effects; and (4) better time-domain behavior. Coaxial loudspeakers have low-to-medium **Q** values due to direct radiating tweeters and small format high-frequency horns. These loudspeakers work best when used as a single system in low reverberant spaces or when placed close to listeners. They are often inappropriate for a multiple loudspeaker central cluster systems or split systems because of the wide coverage overlap between the individual loudspeakers. Typical coverage angles range from 40° horizontal by 30° vertical, 50° horizontal by 40° vertical, 90° horizontal by 40° vertical, and 90° horizontal by 90° vertical.

Coaxial loudspeakers are classified into three broad categories: (1) separate high- and low-frequency cone drivers; (2) combination high-frequency horn and low-frequency cone driver; and (3) separate high- and low-frequency horn drivers. Figure 3-53 shows coaxial drivers of various various sizes and types.

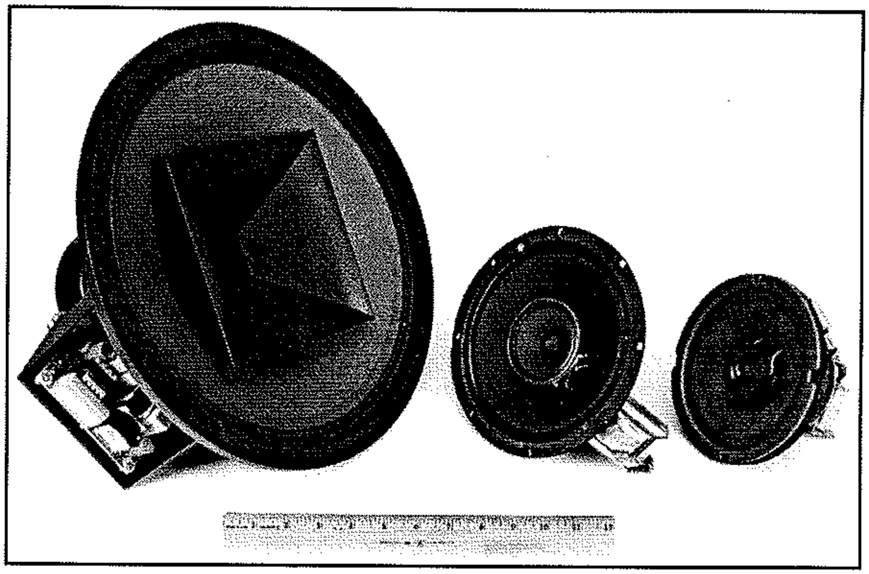


FIGURE 3-53. Coaxial drivers of various sizes and types: 15 in diameter cone low-frequency element with 90° horizontal by 90° vertical coverage angle small constant directivity horn (JBL 2155H left), 8 in diameter cone low-frequency element with 2½ in diameter cone high-frequency element (Altec Lansing 409-16T center), and 6½ in diameter cone low-frequency element with ¾ in diameter cone high-frequency element (Soundolier FA136T87 right). All drivers have integral crossovers and line matching transformers for 70.7 V operation. Products courtesy of JBL, Inc., Mark IV Audio Group, and Atlas/Soundolier, Inc.

3.9.6.1 Separate High- and Low-Frequency Cone Drivers

The simplest and least expensive coaxial loudspeaker uses a high-frequency cone tweeter placed in front of a low-frequency cone driver. The tweeter is mounted with a stud through the woofer pole piece or with a metal bracket across the woofer frame. The tweeter is a simple dome or miniature cone between ¾ and 2½ in diameter. The tweeter is crossed over at approximately 2,000 Hz to the low-frequency cone driver, which can be 6½, 8, 10, or 12 in diameter. Figure 3-54 shows two small coaxial loudspeakers using separate high- and low-frequency drivers.

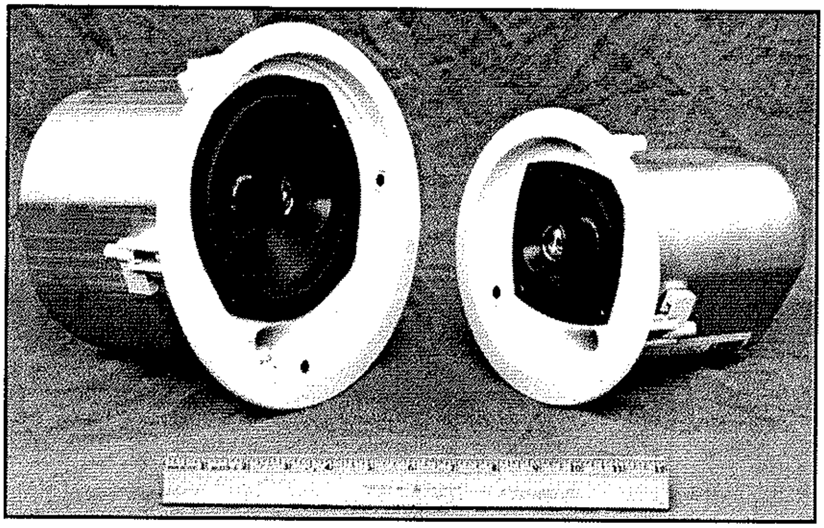


FIGURE 3-54. Coaxial loudspeakers using separate high- and low-frequency cone drivers of different sizes: 6 in diameter cone low-frequency driver (JBL Control® 26CT left and 4 in diameter cone low-frequency driver (JBL Control® 24CT right). Both have ¾ in diameter titanium high-frequency drivers. Loudspeaker enclosures have grilles removed. Products courtesy of JBL, Inc.

The overall frequency response is 80 to 15,000 Hz, but smooth on-axis frequency response is limited to around 12,000 Hz. The off-axis frequency response tends to fall off rapidly beyond 45° for frequencies above 2,000 Hz. Power handling can be up to 100 W when the loudspeaker is not used with a line matching transformer. Most low-cost coaxial loudspeakers are used in 70.7 V ceiling distributed systems and the line matching transformer controls the power delivered to each loudspeaker. The nominal coverage angle is 90° in both horizontal and vertical planes.

Disadvantages with low-cost coaxial loudspeakers include limited tweeter power handling and uneven frequency response in the crossover region. At frequencies just below the crossover frequency, the woofer becomes more directional, while the tweeter is more omnidirectional at frequencies just above the crossover frequency. The resulting summation of acoustic energies with different directivities in a common bandwidth results in poor frequency response about one-half octave above and below the crossover frequency.

Because of their limitations, coaxial loudspeakers using high- and low-frequency cone drivers are best suited to low-level ceiling distributed systems.

3.9.6.2 Combination High-Frequency Horn and Low-Frequency Cone Drivers

Combination high-frequency horn and low-frequency cone driver coaxial loudspeakers can be subdivided into systems having a separate front-mounted high-frequency horn driver and systems having a back-mounted high-frequency driver with the woofer cone functioning as the horn.

3.9.6.2.1 Front-Mounted High-Frequency Horn Driver

Coaxial loudspeakers with a front-mounted high-frequency horn have a compression driver located behind the central axis of the woofer magnet pole piece. The compression driver terminates into a small high-frequency horn frontally-mounted to the woofer. The horn throat passes through the center of the woofer assembly back to the compression driver phasing plug. One to 2 in compression drivers are commonly used which cross over between 1,500 and 3,500 Hz to a woofer of 10, 12, 15, or 18 in diameter. The higher sensitivity compression driver produces greater sound levels than the woofer. It is necessary to attenuate the horn output level within the crossover network to linearize the response of the two drivers. Figure 3-55 shows two medium size coaxial loudspeakers using a front-mounted high-frequency horn driver.

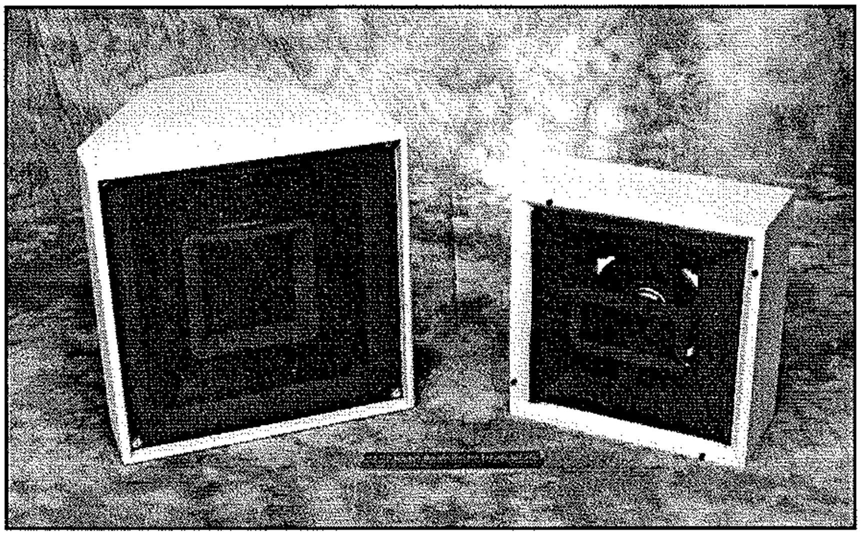


FIGURE 3-55. Medium size coaxial loudspeaker using one 1 in compression driver on a 50° horizontal by 40° vertical small constant directivity horn with 10 in low-frequency driver within a medium constant directivity horn (Frazier CAT™ 56 left) and medium size coaxial loudspeaker using one 1 in dome high frequency driver on a 90° horizontal by 90° vertical horn with an 8 in low-frequency driver within a conic horn (Frazier CAT™ 40 right). Note use of acoustical foam at edges of horns to minimize sound diffraction effects and angled edges at the sides for low-frequency directional control. Loudspeaker cabinets have grilles removed. Products courtesy of Frazier Loudspeakers.

The overall frequency response is between 60 to 17,000 Hz, with smooth on-axis frequency response extending to around 15,000 Hz. The off-axis frequency response tends to fall off beyond 45° for frequencies above 4,000 Hz. The power handling capacity of the horn coaxial loudspeaker system is greater than the cone coaxial system, with power ratings up to 250 W when the loudspeaker is not used with a line matching transformer. However, like any other compression driver, high-frequency power handling is limited by thermal and excursion limits of the motor assembly and diaphragm. The woofer is excursion limited since the high-frequency throat extends into the cone. Some loudspeakers are available with high-output line matching transformers for use in 70.7 V ceiling distributed systems. The nominal coverage angle is 90° in both horizontal and vertical planes.

The high-frequency horn provides directional control for several octaves above the crossover frequency. Directional control below 1,500 Hz is generally not feasible due to the small horn size. A larger horn can not be used since it will acoustically shadow mid-range sound radiating from the woofer. Above the crossover frequency the woofer becomes more directional and the horn can affect the radiation pattern in this frequency range.

Coaxial loudspeakers with a front-mounted high-frequency horn are commonly used for quality ceiling distributed systems requiring high sound levels.

3.9.6.2.2 Back-Mounted High-Frequency Driver with Cone Horn

The coaxial loudspeaker with a back-mounted high-frequency compression driver uses the woofer cone geometry as a horn extension to provide directional sound control. The driver and the horn throat are installed behind the woofer with the cone completing the high-frequency horn flare. Some designs use a tulip-shaped waveguide to couple the compression driver to the cone. The back-mounted coaxial driver alignment has the benefit of reducing diffraction interference due to the horn blocking the front of the cone, but can result in higher intermodulation distortion.

The British firm Tannoy is well known for this type of coaxial loudspeaker. Their products use 1 and 2 in compression drivers coupled to 10, 12, and 15 in woofers. A unique aspect of this loudspeaker type is the need for a minimum phase crossover network since the low- and high-frequency acoustic centers are in close physical alignment.

The profile, size, and depth of the woofer cone determines the directivity factor of the coaxial loudspeaker system across the frequency bandwidth. High-frequency coverage uniformity is controlled by the cone profile and depth. The cone profile, be it conical, exponential, or straight, has unique directional properties. A cone with an exponential flare will exhibit narrower coverage as frequency increases. A conical-shaped cone will approximate the rapid flare rate of a CD horn. Shallower cones tend to have a more rapid flare rate with a resulting reduction in second harmonic distortion, better high-frequency pattern control, and reduced high-frequency beaming. The low-frequency coverage directivity is controlled by the cone size. As with all cone drivers, the directivity factor increases as frequency increases for a given cone size.

Much of the material relating to frequency response and power handling discussed in the front-mounted horn coaxial loudspeaker section is applicable to this type of coaxial loudspeaker.

Advantages with the back-mounted high-frequency driver with cone horn include: (1) closer "point source" approximation; (2) compact size; (3) even, controlled dispersion in both horizontal and vertical planes on- and off-axis; and (4) smooth acoustical and electrical impedance characteristics. These loudspeakers find wide application in small clusters, high quality ceiling distributed systems, and control room monitoring. Figure 3-56 is representative of this coaxial loudspeaker type.

3.9.6.3 Separate High- and Low-Frequency Horn Drivers

Fully horn-loaded coaxial loudspeakers, using a horn-within-a-horn concept, provide better directional control at lower frequencies, have higher efficiency, and can handle greater power amplifier input than the other coaxial loudspeakers discussed above. An early successful loudspeaker of this type was developed by Bruce Howze at Community Light Sound in 1975. Figure 3-56 shows a coaxial loudspeaker with separate high- and low-frequency drivers.

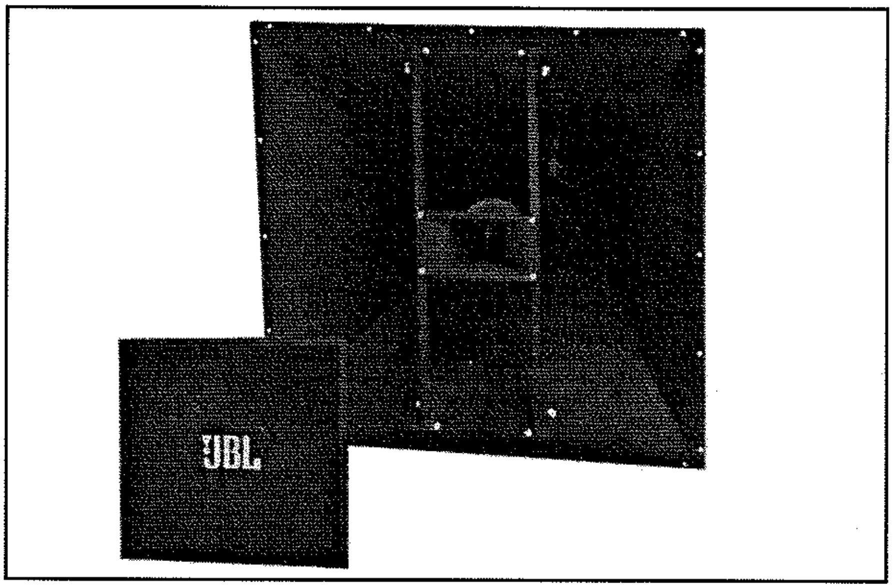


FIGURE 3-56. Two-way coaxial loudspeaker having a 90° horizontal by 50° vertical coverage angle using one 4 in compression driver on a front-mounted small format constant directivity horn within a large format constant directivity horn with a 4 in mid/low-frequency compression driver (JBL 2192). Nominal dimensions are 44 in wide by 44 in high by 34 in deep. Photo courtesy of JBL, Inc.

These systems are available in both two-way and three-way configurations. Two-way systems use 8, 12, or 15 in cone drivers installed in a large format horn, or a horn-loaded mid-range compression driver with a 3 to 4 in diameter diaphragm, to reproduce the frequency range between 200 and 2,000 Hz. A separate horn-loaded high-frequency compression driver with a 1 to 4 in diameter diaphragm is nested in front of the mid-range horn to cover the frequency range between 2,000 and 12,000 Hz. Three-way systems use dual 12 in, single 15 in, or single 18 in cone drivers to reproduce the frequency range below 600 Hz, a horn-loaded mid-range compression driver with a 2 to 3 in diameter diaphragm to reproduce the frequency range between 600 and 3,500 Hz, and a horn-loaded high-frequency compression driver with a 1 in diameter diaphragm to reproduce the frequency range between 3,500 to 15,000 Hz. A wide variety of coverage patterns are available for near-throw and far-throw applications, with CD horns commonly used for the high-frequency reproduction.

Large coaxial horn systems find wide application in high-level sound reinforcement such as in outdoor stadia and sports facilities.

3.9.7 Single Full-Range Multi-Way Loudspeakers

The first full-range multi-way loudspeaker system was developed by Rice and Kellogg of the General Electric Company in 1923. Their system used three horn loudspeakers, each covering a separate frequency range. The first commercial application of full-range multi-way systems was in the motion picture industry. A curled horn (to minimize space), similar in appearance to a forward-radiating French horn, was used for the mid- and high-frequencies and dynamic loudspeakers in a baffle were used for the low-frequencies. Later work by Volkmann of RCA in the early 1930s developed the phase-aligned "pyramid radial horn" multiple loudspeaker array.

Today, single full-range multi-way loudspeakers are commonly used for both speech and music programs in permanent installations and portable systems. Manufacturers provide a variety of loudspeakers having different frequency response characteristics, coverage angles, and power handling capabilities. Common configurations use a combination of low-frequency cone drivers, mid-range and high-frequency horns, and an internal crossover network all within a single enclosure. Two-way and three-way loudspeaker systems are the most popular, some of which can be bi- or tri-amplified. Figures 3-57, 3-58, and 3-59 show single full-range multi-way loudspeakers of various sizes and types.

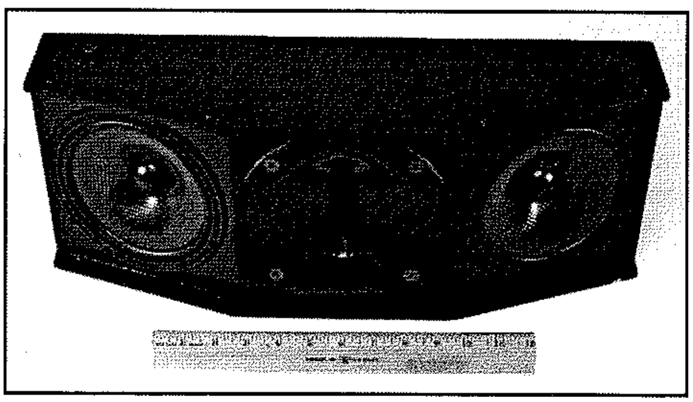


FIGURE 3-57. Small size full-range two-way loudspeaker using two 6½ in low-frequency drivers, one 1 in compression driver on Complex Conic™ horn with 150° horizontal by 60° vertical coverage angle (Renkus-Heinz SR62H). Loudspeaker is intended for horizontal placement under balconies. Loudspeaker cabinet has grille removed. Product courtesy of Renkus-Heinz.

FIGURE 3-58. Medium size full-range two-way loudspeaker using two 8 in low-frequency drivers, one 1 in compression driver on Complex Conic™ horn with 90° horizontal by 40° vertical coverage angle (Renkus-Heinz TRC82 left) and three-way loudspeaker using one 12 in low-frequency driver, one 8 in mid-range driver, and one 1 in compression driver on Complex Conic™ and CoEntrant™ horn with 90° horizontal by 40° vertical coverage angle (Renkus-Heinz SR6/9 right). Loudspeaker cabinets have grilles removed. Products courtesy of Renkus-Heinz.

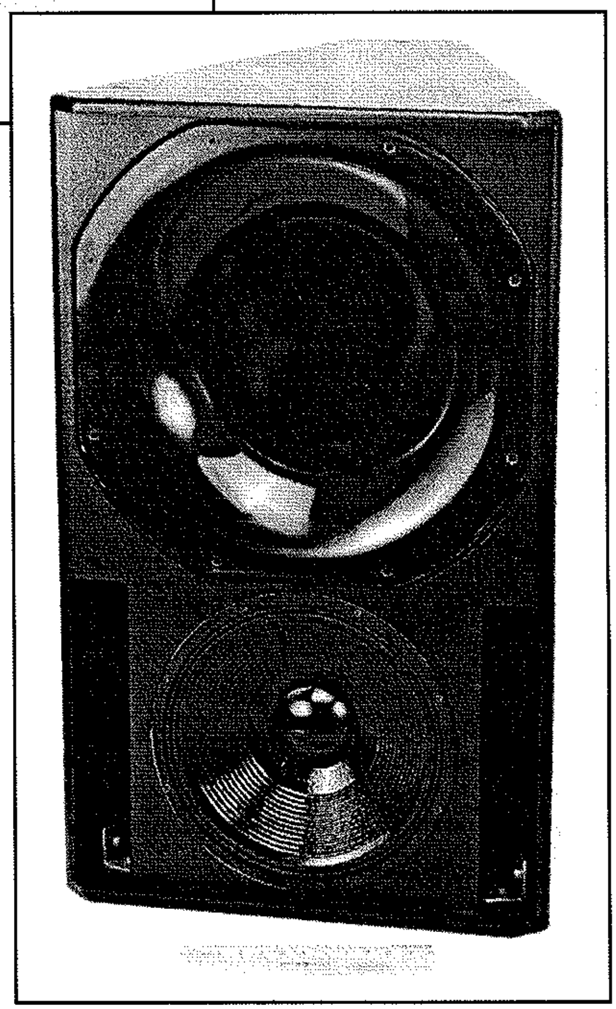


FIGURE 3-59. Large size full-range three-way loudspeaker using one 15 in low-frequency driver, one 8 in mid-range driver, and one 1 in compression driver on Complex Conic™ and CoEntrant™ horn with 60° horizontal by 40° vertical coverage angle (Renkus-Heinz CT5/64). Loudspeaker cabinet has grille removed. Product courtesy of Renkus-Heinz.

Two-way loudspeakers have a crossover frequency in the 1,000 to 2,500 Hz range. In this frequency range, the low-frequency driver has considerable output but its directional pattern is narrow due to high-frequency sound radiating from the woofer. The high-frequency horn provides broad sound coverage in the crossover frequency region. The interaction of on-axis radiation from the woofer and the broad coverage high-frequency horn can result in variations in on- and off-axis sound quality. Typically, these frequency variations are in the 800 to 3,000 Hz range, where the ear is most sensitive. In such cases a three-way loudspeaker may be a better choice for quality reproduction where the crossover frequencies are commonly at 500 and 4,000 Hz.

The performance characteristics of single full-range multi-way loudspeakers include nominal frequency response between 50 and 15,000 Hz, typical mid-range frequency **Q** values between 8 and 15, and power handling of 100 to 500 W. Horizontal and vertical coverage angles are available in a wide range of values based upon the horn type installed within the enclosure. These systems generally perform best in rooms having a $\mathbf{T_{60}}$ less than 1.6 s and may not provide adequate speech intelligibility in more reverberant rooms unless placed close to the listener. Poor directional pattern control at low-frequencies and moderate directional pattern control at high-frequencies are two inherent design compromises, due to the front-mounted woofer and a relatively small high-frequency horn selected to fit within the cabinet. The designer should use separate horn-loaded woofers and large high-frequency horns arranged in a cluster configuration when improved directional pattern control and higher **Q** values are necessary, particularly in large reverberant rooms.

3.9.8 Processor-Controlled Loudspeakers

The latest advances in loudspeaker design use DSP signal processing to optimize the performance of multi-way loudspeakers, typically those in bi-amplified and tri-amplified configurations. Some manufacturers produce companion signal processors tailored to the specifics of their loudspeakers. The signal processor receives input from the signal mixer, or another electronic component, with output to the power amplifiers. Some loudspeaker signal processors can daisy-chain multiple amplifiers to a single signal processor.

The signal processing can provide low-level crossover, high-pass and low-pass filtering, frequency equalization, CD horn compensation, signal delay, level adjustment, loudspeaker overload protection, and other features the manufacturer deems necessary. The audible result is a wide and smooth frequency response with enhanced dynamic range and better signal alignment between the individual drivers.

One particular advantage with processor-controlled loudspeakers is the requirement for external signal processing equipment is reduced, which can lower system installation costs.

3.9.9 Self-Powered Loudspeakers

A variety of loudspeakers have power amplifiers integral within the loudspeaker cabinet. Common are bi- or tri-amplified configurations with a separate power amplifier connected to each low-, mid-, or high-frequency loudspeaker section.

Self-powered (active) loudspeakers have several advantages over passive loudspeakers. The power amplifier is matched to the loudspeaker section and can provide the optimum power and signal drive characteristics. The series load resistance due to a long cable between the power amplifier and loudspeaker is eliminated which results in a greater damping factor for the loudspeaker and less power loss through the cable.

A source of 120 VAC electrical power is necessary to be installed near the loudspeakers to power the amplifiers. The audio signal is routed to the loudspeaker via a line level cable. Figure 3-60 shows a self-powered sub-woofer having a cardioid-shaped directivity pattern which is useful for reducing sound levels on-stage that might exacerbate feedback.

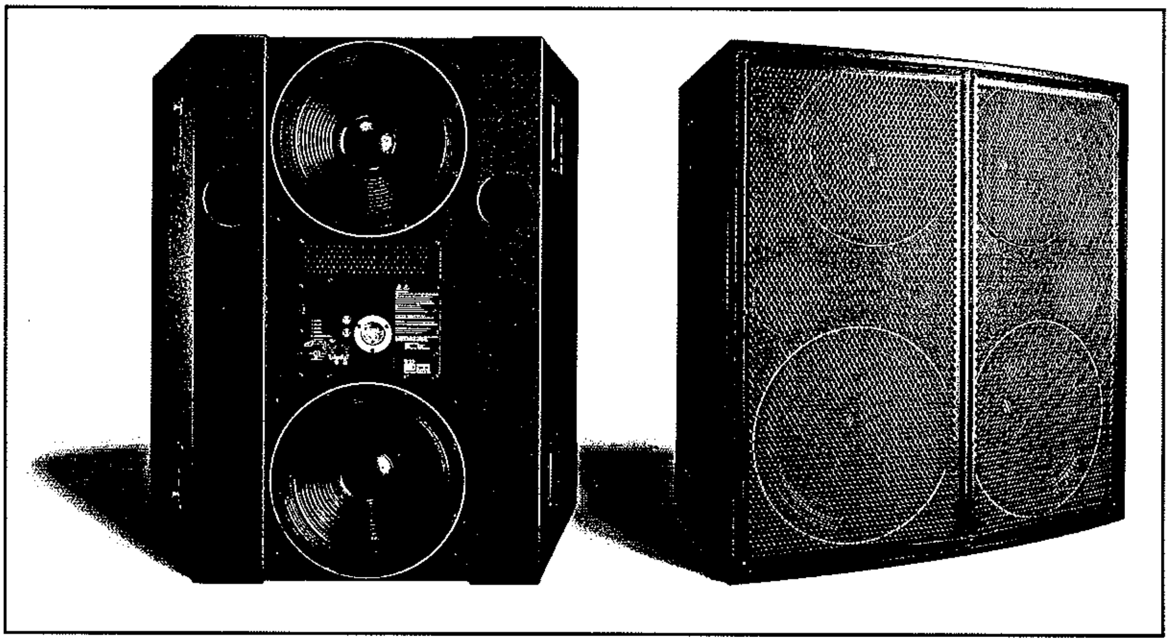


FIGURE 3-60. Self-powered cardioid subwoofer (Meyer Sound Laboratories PSW-6), rear (left) and front (right). Two 15 in drivers are at rear with two 18 in and two 15 in drivers at front. All drivers are in separate ported chambers. The rear two drivers are signal processed to redirect the sound energy towards the front, resulting in a 15 to 20 dB attenuation of the rearward radiating sound energy. Four power amplifiers and active signal processing electronics are contained within the loudspeaker enclosure. Photo courtesy of Meyer Sound Laboratories.

3.10 Assistive Listening Systems

Over 28 million Americans have some form of hearing impairment and less than 20 percent of the hearing impaired population wears a hearing aid. Assistive listening systems (ALS) are wireless electronic devices which distribute an audio signal via a transmitter radiating electro-magnetic energy to listeners wearing an appropriate receiver. Similar to a hearing aid, an ALS makes sound louder, but differs in that it makes desired sounds louder without amplifying the ambient noise near the listener. ALS provisions are required in places of public accommodation and assembly as mandated by the Americans with Disabilities Act (ADA). Figure 3-61 shows the international ALS symbol.

FIGURE 3-61. International symbol indicating that assistive listening equipment is available for use by facility patrons.

The audio signal routed to the ALS can originate from the sound system signal mixer, or from a dedicated microphone, when a sound reinforcement system is not used. The radiated signal from the ALS is picked-up by the listener wearing a small receiver with ear pieces. Individuals with hearing impairment may require ALS to adequately hear the program even when a sound reinforcement system is used. Others with normal hearing acuity may benefit from an ALS where room reverberation or high noise levels reduce speech intelligibility.

Many personal hearing aids have a built-in tele-coil. The purpose of the tele-coil is to pick-up a magnetic field generated by an electrical current flowing through a wire loop. This technique is used in the telephone handset to couple the hearing aid with the telephone loudspeaker and some ALS technologies utilize this same concept. Many hearing aids have a switch that permits changing between the internal microphone (for normal listening to sounds within the room) and the internal tele-coil (for receiving inductively transmitted sound via the tele-coil). Some hearing aids have the provision to listen simultaneously to both signals. The quality of many tele-coils is poor which results in an unpleasant audio quality. Behind-the-ear hearing aids permit a larger tele-coil to be used but the trend with hearing aids is towards smaller less visually obtrusive in-ear devices. The smaller size of these hearing aids can not fit an effective tele-coil. ALS manufacturers have developed a variety of receiver ear pieces to accommodate listeners wearing hearing aids.

3.10.1 Types of ALS

The three broad categories of ALS devices include: (1) radio frequency (FM) broadcast; (2) infrared (IR) light radiation; and (3) magnetic induction loop (IL). Each type of ALS has unique advantages, limitations, and installation requirements. Because of this there is no one system applicable to all facilities. Table 3-4 summarizes characteristics and applications of different ALS technologies.

TABLE 3-4. Summary of ALS Technologies

System Type	Advantages	Disadvantages	Typical Applications
Radio Frequency (FM) **Transmitter** - Transmitter with antenna. **Receiver** - Various types with or without induction coupler and neck loop for tele-coils.	Highly portable. Different channels permit multi-channel programs. Can be integrated into a sound system or used as a portable unit. Moderate cost.	Susceptible to radio frequency interference (RFI) and other radio broadcasts. May require field or factory modification to reduce interference. Lack of security and privacy.	Classrooms Houses of worship Meeting spaces Outdoors Theaters Tour groups
Infrared (IR) **Transmitter** - Emitter with LEDs. **Receiver** - Various types with or without induction coupler and neck loop for tele-coils.	Insures privacy and confidentiality. Different channels permit multi-channel programs. Can be integrated into sound system or used as a portable unit.	Limited portability. Line-of-sight between transmitter and receiver required. Affected by daylight, some lighting ballasts and fluorescent lights. High cost.	Classrooms Courtrooms Houses of worship Meeting spaces Television viewing Theaters
Magnetic Induction Loop (IL) **Transmitter** - Radiating induction wire loop around listening area. **Receiver** - Self-contained induction receiver or personal hearing aid with tele-coil.	Low maintenance. Unobtrusive. Can be integrated into a sound system. Some hearing aids can act as separate receivers. Low cost.	Limited portability. Lack of security and privacy. Signal susceptible to electrical interference. Varying signal strength. Difficult to retrofit in existing space. Older technology.	Classrooms Houses of worship Meeting spaces Retirement centers Television viewing Theaters

3.10.1.1 FM Systems

FM systems use frequency-modulated radio waves to broadcast an audio signal into the room. The system components comprise a transmitter, antenna, power supply, and receivers. The audio signal is input to the transmitter from the signal mixer or a dedicated microphone. The modulated audio signal is routed to the antenna and radiated where it is picked-up by a receiver worn by the listener. The receiver demodulates the signal, amplifies the audio signal, and routes it to ear pieces worn by the listener. Power to the transmitter is from a 120 VAC electrical power receptacle. No license is required to use FM systems, but the manufacturer's equipment must be FCC type approved. Figure 3-62 shows an FM assistive listening system.

FIGURE 3-62. FM assistive listening system comprising transmitter with antenna (Phonic Ear PE 550T left) and receiver unit (Phonic Ear PE 300R right). Products courtesy of Phonic Ear, Inc.

The standard VHF bandwidths assigned by the FCC for FM assistive listening systems are 72 to 76 MHz (low band) and 216.0125 to 216.9875 MHz (high band), both limited to a transmitter power output of 1 W. Up to 57 wideband and narrowband channels are available in each of the low and high transmission bands, with each channel assigned a unique carrier frequency. Wideband channels are available between 72.1 and 75.9 MHz (low band) and 216.025 and 216.975 MHz (high band). Narrowband channels are available between 72.025 and 75.975 MHz (low band) and 216.0125 and 216.9875 MHz (high band).

Different FM systems are available from manufacturers to suit a variety of applications. The major distinctions are whether the system has narrowband or wideband tuning and if the channels operate either in the low or high bands. Assistive listening systems can be operated in low or high bands. Language interpretation and tour guide systems are restricted to the high band. Narrowband ALS devices have advantages of more precise tuning and the potential for less RFI interference. However, they require field adjustment, or with some products, factory modification to change the carrier frequency. Wideband devices use tunable crystals that can be field adjusted to eliminate RFI interference by changing the carrier frequency. Some manufacturers have systems which can be switched between narrowband and wideband tuning. Both narrowband and wideband systems can be used in a multi-channel application, such as large cinema complexes. The number of available channels varies between different manufacturers, with up to 40 channels being common.

Another distinction is the type of antenna connected to the transmitter. The standard antenna is the quarter wave "whip" or "rubber duckie" type which connects directly to the back of the transmitter. These antennae provide adequate field strength for receiver distances up to approximately 300 ft from the transmitter. For distances between 300 and 800 ft it is necessary to use a "large area" antenna. This antenna is remotely connected to the transmitter and is installed in the room the audience is in. For best transmission and reception, the antenna should have a direct line-of-sight to the audience. The remote antenna is commonly provided with a *ground plane* which increases the field strength of the signal to improve reception. In practice, the transmission distances noted above can be reduced by structural steel building elements in the vicinity of the transmitter and antenna. One disadvantage with the remote antenna is it requires proper impedance matched cables. Where distances between the transmitter and antenna are within 100 ft, type RG58/U coaxial cable can be used. Longer distances, up to 300 ft, require type RG8/U coaxial cable.

3.10.1.1.1 Advantages of FM Systems

FM systems have advantages over IR and IL ALS technologies. The audio advantages include greater signal bandwidth and less noise than IL systems, allowing for better fidelity. Functionally, the system has advantages of being able to be used outdoors, or indoors in high ambient light environments, compared to IR systems. Visually, the antenna can be hidden or can be remotely located for distant signal transmission. Each receiver is tuned to a dedicated frequency and it is not possible to pick-up incorrect signal broadcasts.

3.10.1.1.2 Disadvantages of FM Systems

The major disadvantage with FM systems is the lack of security and privacy. Since the signal can radiate into adjacent rooms or outside, it may be possible to eavesdrop on the proceedings if the listener has a receiver tuned to the correct carrier frequency. Another problem is the potential for outside RFI interference from legal and illegal radio sources, particularly in large metropolitan areas. Individuals wearing hearing aids can use FM receivers, but a special tele-coil receiver is required to be compatible with their hearing aid.

3.10.1.2 IR Systems

IR systems use modulated light radiation to broadcast an audio signal into the room. The system components comprise a modulator (sometimes called a transmitter or base station), one or more emitter panels (sometimes called a radiator), power supply, and receivers. Some products combine the modulator and emitter panel into one device. Figure 3-63 shows an IR assistive listening system.

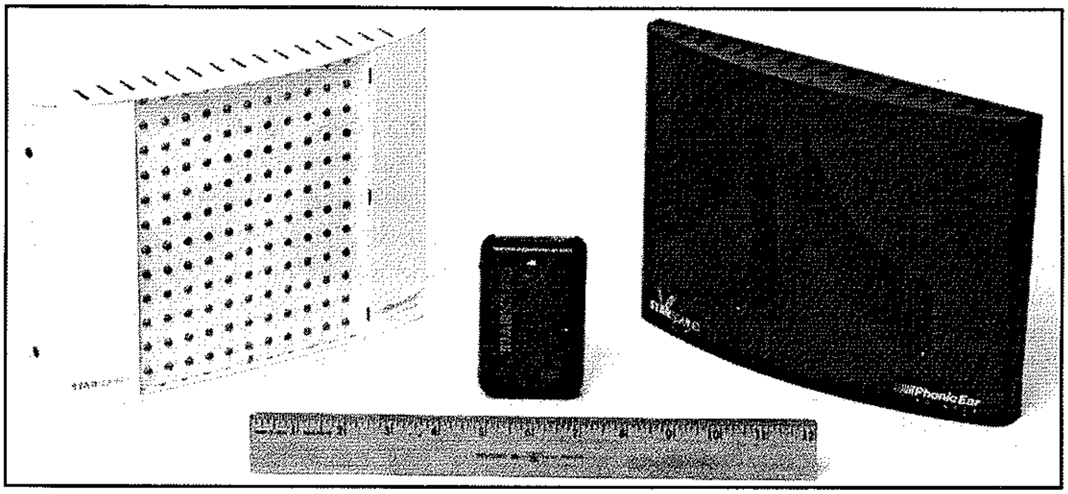


FIGURE 3-63. IR assistive listening system comprising emitter panel (Phonic Ear PE 600E white color left and black color right) and receiver unit (Phonic Ear PE 600R center). Products courtesy of Phonic Ear, Inc.

The audio signal is input to the modulator from the signal mixer or a dedicated microphone. The modulator superimposes a subcarrier signal onto the audio signal to create an amplitude-modulated signal which is routed to the emitter panel(s). The emitter panel has numerous LEDs that convert the amplitude-modulated signal into IR light which radiates into the room. Power to the emitter panel can be from a dedicated 120 VAC electrical power receptacle, or by 24 VDC power generated by the modulator and carried on the audio signal cable connecting the modulator and emitter panel. The latter method may simplify some installation applications. Once radiated into the room, the IR light is picked-up by a receiver worn by the listener. The receiver has an IR light sensitive diode and a filter which attenuates visible light but passes the IR light. The receiver demodulates the IR light into the original audio signal, amplifies the audio signal, and routes it to ear pieces worn by the listener.

Infrared systems are not under FCC jurisdiction because these devices do not emit radio waves. The standard transmission frequencies for IR ALS are 95 to 250 kHz (low band) and 2.3 to 2.8 MHz (high band) which have been sanctioned by the International Electrotechnical Commission (IEC). Both wideband and narrowband transmission modes are available. The wideband carrier frequencies (95 kHz, 250 kHz, 2.3 MHz, and 2.8 MHz) are used for single- and two-channel systems. The latter two carrier frequencies were assigned in response to interference problems from energy-saving lighting ballasts operating at 50 kHz. Wideband two-channel stereo IR systems use 95 kHz or 2.3 MHz for channel one and 250 kHz or 2.8 MHz for channel two. The wideband transmission mode operates with an extended frequency bandwidth, typically between 20 and 18,000 Hz. The narrowband transmission mode is used for up to 32 separate channels, with each channel assigned a unique carrier frequency between 55 and 1,335 kHz. Multi-channel systems are commonly used in language interpretation systems and "Broadway" shows. Currently, no manufacturer offers a transmitter capable of narrowband operation in the higher 2.3 or 2.8 MHz carrier frequency bands. The narrowband transmission mode operates with a reduced frequency bandwidth, typically between 50 and 8,000 Hz.

3.10.1.2.1 Determining Number of Emitter Panels

The number of emitter panels required for a specific installation will depend on the emitter panel coverage area, distance between emitter panel and receiver, audience seating area size and shape, number of channels, light reflectance and roughness of room surfaces, and room ambient light levels.

The IR radiation pattern from most emitter panels is cone-shaped with a coverage angle between 40° and 60° based on the specific manufacturer's product. Newer emitter panels have an 80° coverage angle which provides a wider but shorter throw coverage pattern. The coverage angle and area covered by the emitter panel is based on the type and number of IR LEDs. Emitter panels are available for a variety of different room sizes: (1) "small area" systems are capable of covering up to 400 ft^2 at distances up to 25 ft; (2) "medium area" systems between 400 and 4,000 ft^2 at distances up to 60 ft; and (3) "large area" systems between 8,000 and 12,000 ft^2 at distances up to 150 ft. These distances are for good **S/N** ratio performance, but useable signal levels with higher background noise, extend reception distances.

The primary distinctions between emitter panel performance are the power output, typically 0.5, 2, 5, or 10 W, and the number of LEDs. The emitter panel LEDs are used in multiple arrays to achieve usable radiated power levels. The area covered by each LED varies between 35 and 85 ft^2 depending on the LED design. Higher output levels or a greater number of LEDs will increase the emitter panel coverage area. The area covered by the emitter panel will be halved for each doubling of the number of channels in the IR system. This is because the emitter panel output power is divided evenly between each carrier frequency used for each channel. The manufacturer's data should be reviewed for details on specific emitter panel coverage patterns and areas.

Direct and reflected components of IR light are present in a room, similar to sound. Positioning the emitter panel at the front of the room facing the audience will enhance the direct IR light reception on the receiver LED. The emitter panel should be installed to the left or right of the room centerline at an elevation of 10 to 40 ft above and aimed down towards the audience seating area. Additional emitter panels may be necessary if coverage is not adequate from a single panel. Other emitter panel layout examples are described below.

1. **Emitter Panels Directly Side-by-Side:** In this installation two panels are positioned next to one another facing the audience area. By doubling the radiating power from a common point the transmission distance increases by a factor of 1.41. This installation is effective for receiver positions along the middle of the emitter panel axes.

2. **Emitter Panels Spaced Side-by-Side:** This installation is similar to the above, but the panels are spaced apart to create a square coverage area. The arrangement is effective for movie theaters and auditoria where the audience is normally facing towards the emitter panels. A variant of this is to use an emitter panel in the left and right front corners of the room and aimed in a 45° cross-fired pattern directed at the audience seating area. A narrower coverage area of higher IR light intensity is created by the cross-fired pattern.

3. **Emitter Panels Behind One Another:** This installation places two emitter panels in-line with one another with one panel at the room front and a second panel located about half-way along the room length dimension. The primary advantage of this arrangement is long narrow rooms can be covered, with each emitter panel covering one-half of the room.

4. **Emitter Panels Facing One Another:** This installation places two emitter panels with one panel at the front and a second panel at the rear facing each other. This arrangement is suitable for rooms having varying seating layouts, such as convention centers, hotel ballrooms, and subdivisible spaces. A variant of this is to use four emitter panels, with one in each corner facing towards the room center.

5. **Emitter Panels in Central Cluster:** This installation places several emitter panels in the center of the room aimed at different angles to the audience seating area. This arrangement is suitable for circular-shaped rooms and sports arenas.

6. **Remote Emitter Panels:** Alcoves, balcony, under balcony, transcepts, and other locations not having direct line-of-sight to the primary emitter panels require supplemental emitter panels to provide adequate IR light levels.

Room surfaces which are light colored or smooth will reflect IR energy and increase the total IR field strength in the room. Dark colored or rough textured surfaces will absorb IR light energy and reduce the total IR field strength. Increasing the number of emitter panels by 50 percent is often necessary in some installations to overcome daylight, dark room colors, or rough room surfaces.

Note that in contrast to loudspeaker systems, overlap in the coverage patterns of different emitter panels is not detrimental to reception and the ultimate audible quality heard by the listener. Multiple IR panels which are daisy-chained together need to terminate in a 50 Ω blocking resistor to avoid generation of standing waves in the system.

3.10.1.2.2 Advantages of IR Systems

IR systems have advantages over FM and IL ALS technologies. The audio advantages include a wider signal bandwidth and lower noise levels, which provides better fidelity than the other ALS technologies. Security advantages include IR signal containment within the room. The IR signal behaves as visible light and can not pass through opaque materials or the room enclosure, but can pass through glass and open windows or doors. Additionally, the system is not susceptible to RFI interference from outside sources. The life expectancy of emitter panels is about 100,000 hours before the light output diminishes to approximately 70 percent of its original value.

3.10.1.2.3 Disadvantages of IR Systems

The primary disadvantage with IR systems is their susceptibility to interference from ambient IR light sources such as daylight, incandescent light, and fluorescent light. Daylight contains varying degrees of IR light energy depending on the outside cloud cover. Partially dimmed incandescent lights can have a shift in the red spectrum of the light energy which results in more IR light output from the lighting fixtures. New types of energy efficient fluorescent lighting ballasts may excite light filaments at a frequency between 25 and 50 kHz which generates a signal two times the value (50 to 100 kHz) which can interfere with the 95 kHz carrier frequency of some IR systems.

The ambient IR light guidelines listed below should be followed when specifying IR systems.

1. **Daylight:** Less than 200 foot-candles (fc) for useable operation; potential interference between 200 and 1,000 fc; inoperable above 1,000 fc.

2. **Incandescent Light:** Less than 40 fc for useable operation; potential interference between 40 and 200 fc; inoperable above 200 fc.

3. **Fluorescent Light:** Less than 100 fc for useable operation; potential interference between 100 and 1,000 fc; inoperable above 1,000 fc.

Installing additional emitter panels to strengthen the radiated IR energy intensity will increase the S/N ratio for the IR signal.

The emitter panels and receivers can not be covered or concealed, otherwise the radiation or reception of IR light intensity will be reduced. Installation clearance of 3 in minimum is required at the back, sides, top, and bottom of the emitter panel to dissipate heat build-up from the LEDs. This often results in the panel protruding from the wall surface which may not be aesthetically acceptable. White colored emitter panels are available from some manufacturers which may blend in better with certain room color schemes than the standard black colored units. Figure 3-64 illustrates IR emitter panels in a church.

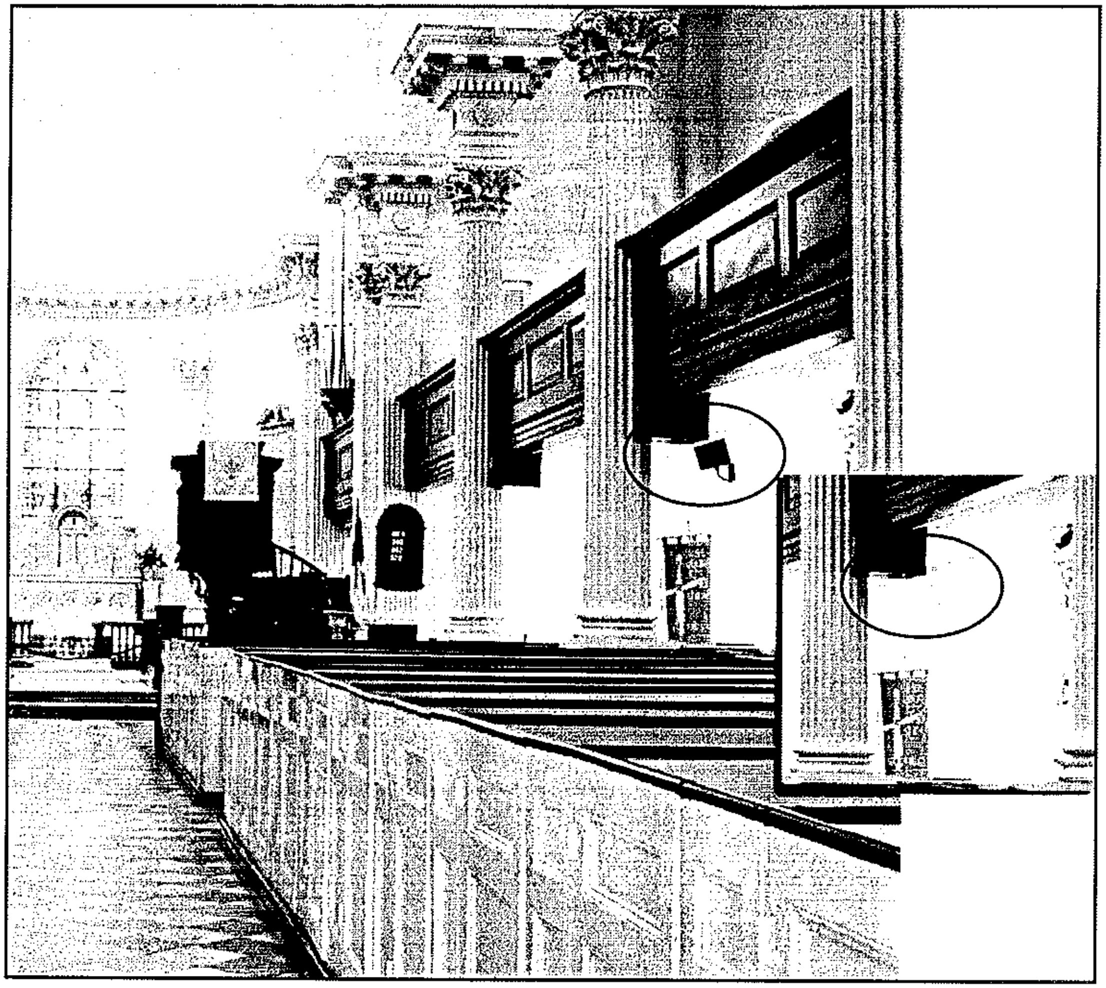


FIGURE 3-64. IR assistive listening emitter panels showing discrete nature of white versus black color in an historic church sanctuary. Note position of panels near window which is not recommended due to potential daylight interference. Photo courtesy of Phonic Ear, Inc.

3.10.1.3 IL Systems

The oldest ALS technology is the IL system which is not as frequently used compared to the newer FM and IR systems. However, IL systems can be applied with good results, and in certain installations, provide superior performance to the other ALS technologies. The system components comprise a power amplifier, autotransformer, wire loop, and receivers.

IL systems use a modulated magnetic field to broadcast an audio signal from a wire routed around the room perimeter. Only the vertical component of the radiated magnetic field is used by IL systems for three reasons: (1) the vertical field strength is greater within the loop area; (2) hearing aid tele-coils are positioned to be more sensitive to the vertical field; and (3) changes in the listener's head position, as when turning, will not alter the pick-up of the vertical field component.

The audio signal is input to the IL system power amplifier from a signal mixer, television (with appropriate connector), microphone, or other electronic component. An alternating electrical current from the power amplifier flows through the perimeter wire loop and generates a modulated magnetic field which is picked up by receivers or the listener's hearing aid, provided it is supplied with a tele-coil. The receiver converts the magnetic field into an alternating current, amplifies the audio signal, and routes it to ear pieces worn by the listener. In the case of the tele-coil-equipped hearing aid, direct coupling is made to the magnetic field and an alternating current results which is amplified by the hearing aid. Special receivers with a magnetic coupling device are available for hearing aids not equipped with tele-coils.

The power amplifier often has a built-in compressor/limiter to reduce the signal dynamic range so as not to clip the amplifier output stage or the tele-coil input. Music will typically use a 4:1 compression ratio and speech up to a 20:1 compression ratio. Both voltage drive and current drive power amplifiers are available. When driving long loops, the outputs of several power amplifiers may need to be summed to overcome the voltage drop across the loop. Manufacturers provide summing networks for their power amplifiers. The autotransformer provides impedance matching of the amplifier output to the wire loop. Standard round single-conductor cable with PVC jacket or flat foil cable for under-carpet application can be used. Protective conduit for the loop wire must be non-metallic, otherwise, magnetic shielding can occur.

For optimum magnetic field distribution around the listener, the loop should be positioned about 10 to 20 percent of the smallest loop dimension above or below the listening plane. Floor installation is the most common, but if ceiling installation is necessary, the ceiling height should be limited to less than 10 ft. The magnetic field strength will diminish from the edge to the center of the loop. The normal limit of acceptability is a 5 dB variation between edge and center locations.

3.10.1.3.1 Advantages of IL Systems

The IL system offers advantages compared to the FM and IR ALS technologies. The perimeter wire loop can be routed around selected areas which may reduce installation complexity. The thin wire is flexible and can be installed under carpets, above acoustical tile ceilings, or in stud walls, making installation invisible. Additionally, direct pick-up of the radiated magnetic signal from properly equipped hearing aids with tele-coils may be useful in some installations, such as retirement centers.

3.10.1.3.2 Disadvantages of IL Systems

The major disadvantages with IL systems are the susceptibility to EMI, radio frequency interference (RFI), and the ability of the magnetic signal to radiate into adjacent spaces. In general, these systems are not designed to be electro-magnetic compatible (*EMC*).

Nearby magnetic fields from lighting ballasts, loudspeakers, and transformers can interfere with the loop radiation. The proximity to building structural steel elements may affect the magnetic signal strength and distribution. The power amplifier current drive may need to be increased to account for magnetic field strength losses. Often, 3 to 12 dB higher current values are necessary over the "no-loss" conditions. The magnetic field strength loss is frequency dependent and is about 3 dB/octave, with up to 15 dB of loss occurring from 150 to 5,000 Hz. To overcome this, the frequency of the modulated magnetic signal can be boosted in a predefined manner using manufacturer's proprietary circuits.

The spillover of the magnetic field can occur at distances up to three or four loop sizes away. Special loop arrays which rapidly reduce the signal strength outside the loop layout pattern can permit loops in adjacent rooms to be used. These loops often use a phase shifting network between the loop arrays or the loops are driven at different levels. Up to 40 dB of isolation between adjacent loop arrays can be achieved.

The final disadvantage with IL systems is only about 12 percent of the hearing impaired population can take advantage of IL technology. Statistics indicated approximately 20 percent of the hearing impaired population wears a hearing aid and only about 60 percent of those hearing aids are equipped with tele-coils.

3.10.2 Receivers and Ear Pieces

ALS receivers and ear pieces are available in a variety of styles to suit specific user needs. Some receivers have integral ear pieces. Others have a separate ear piece which plugs into the receiver via a cable. Figure 3-65 shows different types of receivers and ear pieces.

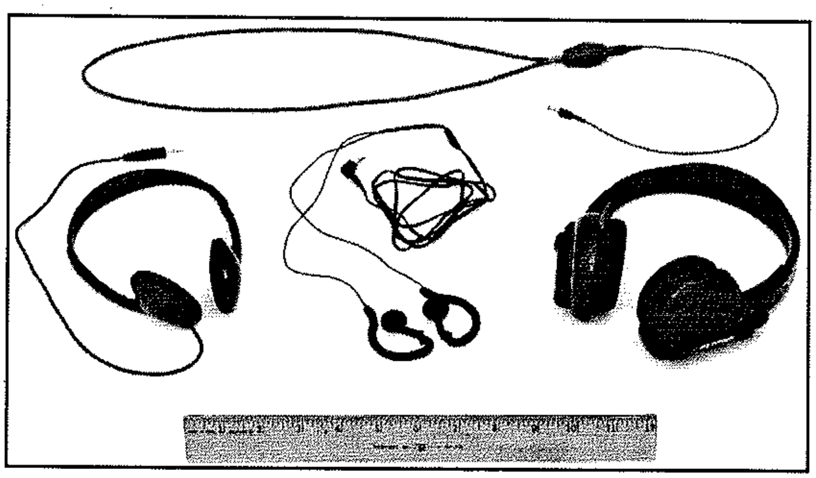


FIGURE 3-65. Ear pieces for use with ALS receivers: induction loop neck coil (Phonic Ear AT163-B top), small double muff headset (Phonic Ear AT541-14 left), miniature ear buds (Sony A340 center), and IR headset (Phonic Ear PE 601R right). Products courtesy of Phonic Ear, Inc. and Sony, Inc.

Receivers are either a small pocket style or an under-the-chin style. Commonly, FM systems use small pocket style receivers and IR systems use under-the-chin style receivers, although some manufacturers offer both styles. For ADA requirements, the total number of receivers must equal four percent of the facility occupancy load capacity.

Caution should be exercised when using IR receivers due to the potential for inadvertent blocking of the IR signal at the receiver detector. Pocket style IR receivers have a pick-up angle of approximately 160°; headset style IR receivers have a pick-up angle of 360°. The headset receiver has the advantage of better pick-up of the IR signal, but this must be weighed against the potential for obstructing sight lines to patrons sitting behind an individual wearing a headset.

Ear pieces are designed for listeners with and without hearing aids. Listeners who wear hearing aids normally use an induction loop coil worn around the neck, an induction coil ear canal earphone, or a miniature ear speaker worn on the pinna. All plug into a small pocket style receiver. Ear pieces for individuals who do not wear hearing aids are commonly miniature single or double "ear buds", lightweight single or double muff headsets, or a stethoscope. The ear pieces plug into under-the-chin or pocket style receivers. Individuals will have a preference to the type of ear piece based on performance, comfort, hygiene, the need to hear room sound not carried by the ALS channel, and whether a hearing aid is worn. Table 3-5 summarizes factors when evaluating the different ear piece styles.

TABLE 3-5. ALS Ear Piece Selection Factors

Type	Ear Position	Performance	Hearing Aid Use	Comfort	Hygiene	Cost
Induction Loop Coil	On shoulders	Excellent	Excellent (requires tele-coil in hearing aid)	High	High	High
Induction Coil Ear Canal Earphone	At entrance to one ear canal	Excellent	Excellent (requires tele-coil in hearing aid)	Moderate	Poor	High
Ear Speaker	Over one ear	Excellent	Good (can be used for tele-coil and non-tele-coil hearing aids)	High	Good	Medium
Single Ear Bud	At entrance to one ear canal	Good	Fair (requires hearing aid to be removed)	Moderate	Poor	Low
Double Ear Bud	At entrance to both ear canals	Good	Fair (requires hearing aid to be removed)	Moderate	Poor	Low
Single Muff Headset	Over one ear	Excellent	Good (can be used for tele-coil and non-tele-coil hearing aids)	High	Good	Medium
Double Muff Headset	Over both ears	Excellent	Good (can be used for tele-coil and non-tele-coil hearing aids)	High	Good	Medium
Stethoscope	At entrance to both ear canals	Excellent	Fair (requires hearing aid to be removed)	Moderate	Poor	Medium

The facility should have a variety of different receivers and ear piece styles to accommodate user preferences. A suggested compliment would comprise: (1) 70 percent pocket style and 30 percent under-the-chin receivers; (2) 25 percent headphones; (2) 20 percent single ear bud earpieces; (3) 20 percent double ear bud earpieces; (4) 10 percent ear speakers; (5) 15 percent stethoscope earpieces; and (6) 10 percent neck loop coils.

3.11 Technical Production Intercom Systems

Technical production intercom systems are used to provide behind-the-scenes voice communication including production equipment cuing, talent cue call, and program monitoring. Application of these systems is primarily limited to arenas, auditoria, exhibition halls, stadia, theaters and other facilities where it is necessary to coordinate the activities of production personnel, such as event directors, stage managers, and lighting, rigging, and sound technicians. Equipment comprises a pick-up microphone located above the stage, a master control station, remote microphone and loudspeaker stations, and associated system interface equipment. Hard-wired and wireless systems are available and both are often used in the same facility.

3.11.1 Production Cuing Equipment

Production cuing equipment permits the event director or stage manager to communicate with technical personnel so they can follow the production from remote locations. Typical applications might include cuing of lights, music and special effects generation, scenery changes, and panning of television/video cameras during drama, music, sporting, and similar events.

The cuing is initiated from a main station located in a side stage area or in a separate production control room. The production technicians wear a single or double muff headset with a microphone connected to a belt pack which plugs into a wall station jack. Locations for wall stations include lighting and sound consoles, followspot positions, lighting booths, catwalks, orchestra pit, rigging positions, gridiron, and scenery construction rooms.

3.11.2 Talent Cue Call Equipment

Talent cue call equipment permits the event director or stage manager to notify talent (actors, emcee, musicians, presenter, et cetera) of upcoming events which require their presence. Typical applications might include notifying talent at the start of a show or new act, end of intermission, directions to assemble in a designated area, or to make a stage appearance.

Cue calls to the talent are initiated from the main station and the announcements are heard through small wall-mounted loudspeaker enclosures. Typical locations for the loudspeakers include all dressing and changing rooms, talent toilet rooms, green room and lounges, stage entrances, and talent assembly/staging areas. A push-to-talk microphone on the loudspeaker enclosure permits communication with the main station. The cue call overrides the program monitoring signal from the stage, even if the loudspeaker monitoring function volume is lowered, ensuring that the talent hears the cue call.

3.11.3 Program Monitoring Equipment

Program monitoring equipment permits the facility administrative staff, talent, technical personnel, and late-arriving audience members to hear the program. A hypercardioid microphone is hung above the stage to pick-up the program event and route the signal to the main station. The signal is amplified and routed to wall- or ceiling-mounted loudspeakers, separately zoned for production (talent and technical personnel) and non-production personnel (facility administration staff and audience). The same loudspeakers used for the talent cuing function provide program monitoring for the talent and technical personnel. Wall- or ceiling-mounted loudspeakers located in the administrative offices, lobby, public lounges, public toilet rooms, coat check room, and ticket booth provide program monitoring for facility non-production personnel and audience members.

3.11.4 Equipment Description

One key design element for technical production intercom systems is to isolate the various subsystems so that only intended personnel communicate with one another or receive their program messages. The subsystems need to be designed so technical production cues are not routed to the talent, intercommunication between technical production personnel are not routed to the monitor system, talent call cues are not routed to technical production personnel, and the like.

The two basic types of technical production intercom systems include point-to-point and conference line systems. Point-to-point systems, sometimes called matrix systems, permit addressing only a designated station or area. Thus, the point-to-point system is a private call system. The audio signal is typically a simplex line (talk, then listen). Conference line systems, sometimes called distributed amplifier systems, permit all parties connected to the system to hear the call and there is no private communication. The audio signal is typically a duplex line (simultaneous talking and listening). Some technical production intercom systems combine both simplex and duplex communication features.

Distributed amplifier systems are the most common technical production intercom system used. These systems have main and remote stations with dedicated microphone preamplifiers and power amplifiers for headsets or loudspeakers. The stations are connected together using balanced microphone cable. The audio program is carried on one signal cable, a second signal cable carries DC power, and the third cable is the system ground connection. Stations bridge the audio line at a high impedance, similar to a 70.7 V amplifier system, and the audio level remains constant as the number of stations across the line changes. The power to the stations is delivered from the main station or a separate power supply.

Technical production intercom systems are available in one-, two-, four-, six-, eight-, and 12-channel configurations. The majority of musical and theatrical productions can be served by a two-channel system. With this configuration, one channel is used for technical personnel production cuing and the other channel is used both for talent cue call and program monitoring. Systems permit simplex, or more commonly, duplex communication. Production intercom equipment can be subdivided into main stations, master stations, remote stations, telephone systems, and wireless systems.

3.11.4.1 Main Stations

Main stations include a power supply, microphone or line level inputs with preamplifiers, program monitor control functions with an integral loudspeaker, and announce paging functions. The power supply provides 12 VDC to the remote loudspeaker and belt pack interconnection stations via standard microphone cable. The program monitoring signal can be mixed with the intercom paging functions and the volume level adjusted at the remote loudspeaker stations. Main stations with program inputs can be configured to permit the talent to monitor the program audio on the intercom channel and permit the stage manager to interrupt the program monitor feed to cue the talent. Figure 3-66 shows a main station.

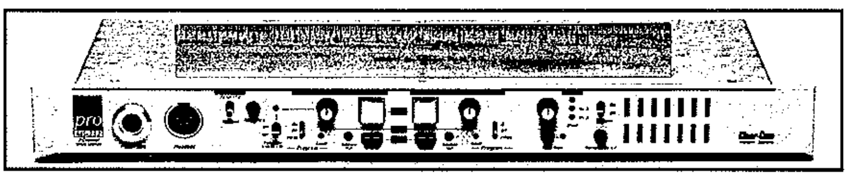


FIGURE 3-66. Two-channel production intercom main station (Clear-Com® MS-232). Product courtesy of Clear-Com®.

3.11.4.2 Master Stations

Master stations are similar to main stations but have multiple channels, typically six-, eight-, and 12-channels acting as a control center. These devices do not provide DC power to remote loudspeaker or belt pack interconnection stations, requiring separate DC power supplies at these locations.

3.11.4.3 Remote Stations

Remote stations are available in belt pack, rack-mount, and wall-mounted configurations. They can be located anywhere in the facility and do not require a dedicated source of 120 VAC electrical power. Power is provided from the main station or a separate DC power supply. Figure 3-67 shows different types of remote stations. The two most common remote stations include combination ear muff and microphone belt pack wall stations and monitor loudspeakers with a push-to-talk microphone. Figure 3-68 shows different types of production intercom communication devices to be used with remote stations.

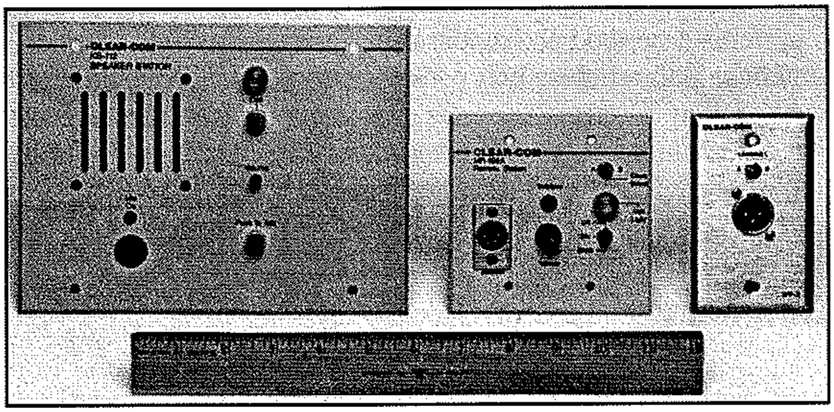


FIGURE 3-67. Production intercom remote stations: two-channel speaker station intended for talent cuing (Clear-Com® KB-111A left), two-channel headset station intended for technical and administrative personnel (Clear-Com® MR-102 center), and single-channel headset station (requires use of a belt pack) intended for technical and administrative personnel (Clear-Com® WP-2 right). Products courtesy of Clear-Com®.

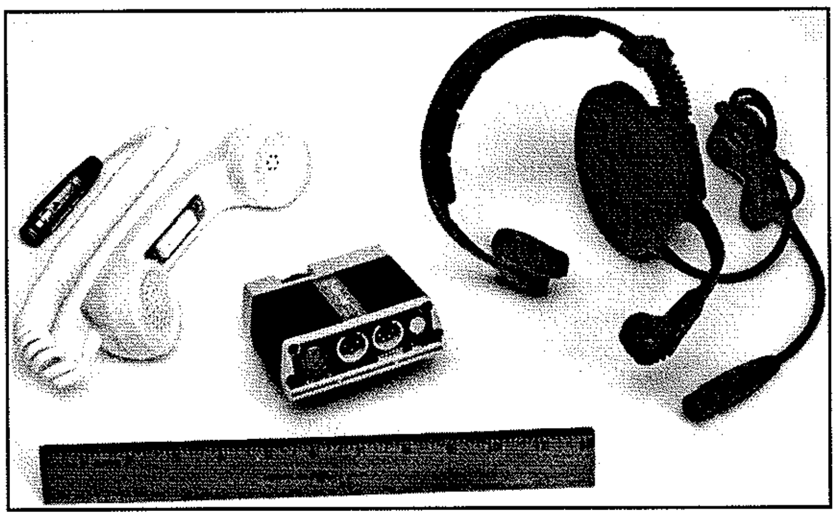


FIGURE 3-68. Production intercom communication devices: telephone-style handset (Clear-Com® HS-6 left), single-channel belt pack (Clear-Com® RS-501 center), and single muff headset with microphone (Clear-Com® CC-85 right). Products courtesy of Clear-Com®.

3.11.4.4 Telephone Systems

A variant of the dedicated technical production intercom system is to provide select communication between administrative offices, production control rooms, house manager, stage, stage manager, and ticket office through a modification of the building telephone system. This may provide a reasonable compromise in some situations, such as an existing facility where it may be difficult to run conduit to distant locations and install wall-mounted stations.

3.11.4.5 Wireless Systems

Wireless production intercom systems are similar in function to the hard-wired intercom systems described above, but usually comprise a base station with send and receive antennae and wireless belt packs which permit connection of a headset and microphone. Often wireless systems are used in conjunction with hard-wired production intercom systems when extra operator stations are required or when installation of conduit and wall-mounted stations is not feasible.

3.12 Chapter Summary

This chapter has introduced the reader to the basic sound system components which are broadly classified as transducers and electronics. Complete sound systems can be assembled from different components to create a system which fulfills the needs of the user.

Transducers, comprising microphones and loudspeakers, convert acoustical signals into electrical signals and vise versa. Electronic components modify the audio signal through mixing and combining, dynamic range adjustment, frequency equalization, signal delay, splitting and routing, and amplification. Newer signal processing technologies implement signal modifications through DSP or computer control. Specialized subsystems, such as assistive listening and production intercom, make use of both transducers and electronics.

Loudspeakers are the most critical element in the sound system. There are many different loudspeaker types which have been optimized for different electro-acoustical performance criteria and room acoustical environments. Selection of loudspeakers should be considered first since they will have the greatest impact on sound system performance, required electronics, and installation.

The next chapter will outline the design process for sound systems to include determining functional requirements, calculation methods to predict performance, correct equipment installation practices, and various system configurations to avoid.

3.13 Technical Notes

3.A Three Dimensional Microphone Polar Patterns

Microphone polar patterns are symmetrical about the polar coordinate system used to represent their pick-up characteristics. The wire-frame models below provide a realistic representation of the microphone sensitivity relative to the frontally oriented diaphragm which is pointed towards the upper right in the photographs.

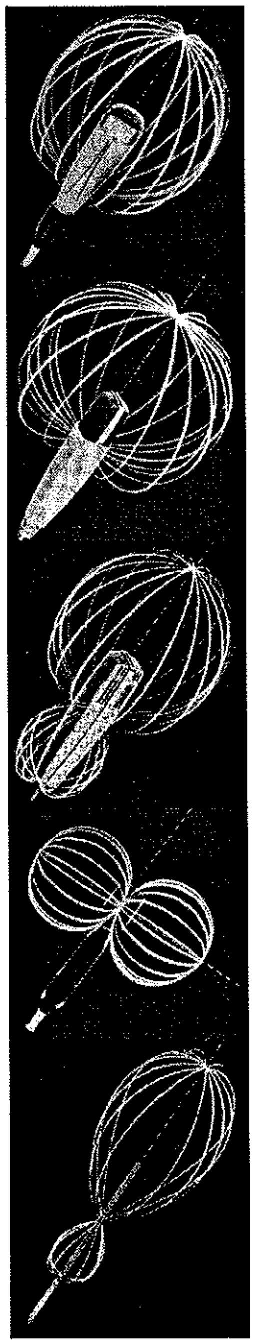


FIGURE 3-A. Three dimensional representation of selected microphone polar patterns (from top to bottom): omnidirectional pattern; cardioid pattern; super cardioid pattern; bi-directional figure-of-eight pattern; and shotgun pattern. Photo courtesty of Sennheiser Electronic Corporation.

3.B Characteristics of Omnidirectional and First-Order Cardioid Microphones

Table 3-B summarizes the major electro-acoustical properties of omnidirectional and first-order cardioid microphones. The relative size of the polar response pattern is shown in the first row for each microphone type. A larger polar response pattern will result in greater angular sound pick-up. The coverage (pick-up) angle at the -3 dB down points is given in the second row. As microphone directionality increases the pick-up angle decreases. The third row lists the incident angle (null angle) relative to on-axis incidence which results in zero voltage output by the microphone. As microphone directionality increases the null angle tends to approach the perpendicular orientation to the diaphragm. The microphone ambient sound sensitivity, sometimes called the random energy efficiency (REE) is listed in the fourth row. It is a measure of the ratio of desired (on-axis)-to-random (off-axis) signals picked-up by the microphone. Lower REE values indicate greater rejection of random off-axis sound and greater emphasis to frontally-arriving on-axis sound. The distance factor in row five indicates the physical distance, relative to an omnidirectional microphone, at which a directional microphone can be placed to maintain the same REE value and have the same ratio of desired-to-undesired sound pick-up. The distance factor increases as microphone directionality increases.

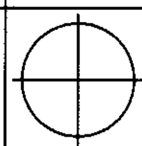
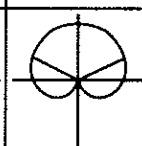
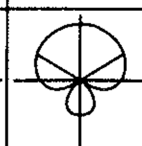
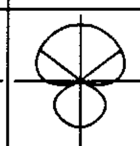
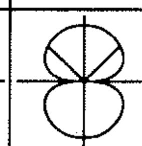
CHARACTERISTIC	OMNI-DIRECTIONAL	CARDIOID	SUPER-CARDIOID	HYPER-CARDIOID	BI-DIRECTIONAL
POLAR RESPONSE PATTERN					
COVERAGE ANGLE	360°	131°	115°	105°	90°
ANGLE OF MAXIMUM REJECTION (NULL ANGLE)	—	180°	126°	110°	90°
AMBIENT SOUND SENSITIVITY (RELATIVE TO OMNI)	100%	33%	27%	25%	33%
DISTANCE FACTOR (RELATIVE TO OMNI)	1	1.7	1.9	2	1.7

FIGURE 3-B. Summary of omnidirectional and first-order cardioid microphone properties. Data after Shure Brothers, Inc.

3.C Balanced Versus Unbalanced Interconnection

Balanced interconnection is necessary to minimize noise pick-up in sound system cables. It is not uncommon for microphone cables to exceed 500 ft. With such a distance, the low signal levels carried in the cables are susceptible to various forms of noise. Cables will act as antennae, with the antenna effect becoming more pronounced for longer cables, increasing the possibility of unwanted noise pick-up.

The balanced signal carried on the two conductors has the same amplitude but opposite polarity. Noise picked-up by the balanced cable is identical in the two conductors. A balanced input on an audio component will amplify the difference between the two signals (the opposite polarity audio signals) and reject signals (noise) which are identical on each conductor. These inputs are referred to as differential inputs.

In contrast, an unbalanced interconnection will reproduce any signal or noise imposed on the conductor. Noise will pass through the unbalanced audio component inputs and will be amplified along with the audio signal.

Figures 3-C and 3-D illustrate balanced and unbalanced cables along with common cable interconnections.

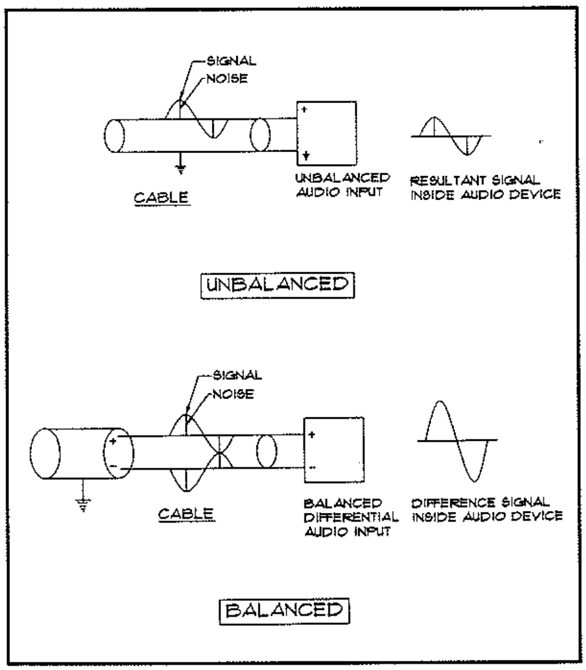


FIGURE 3-C. Operational principles of unbalanced (top) and balanced (bottom) cables.

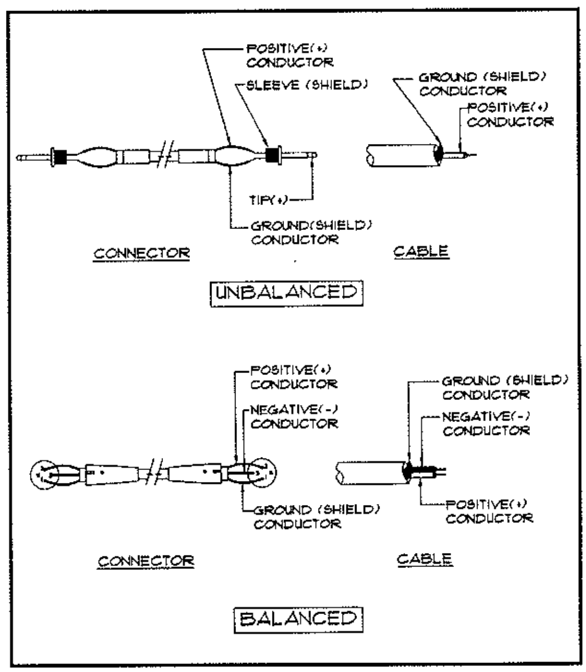


FIGURE 3-D. Convention for unbalanced (top) and balanced (bottom) connectors.

3.D Comb Filtering

Two identical signals with a defined time offset can combine together resulting in frequency-selective addition and cancellation (Law of Superposition) which alters the frequency response of the original signal. A frequency, and whole number multiples of that frequency, which are time delayed by one-half the *period* will cancel partially or completely because the signals have opposite polarity. The cancellation null depth will depend on the relative magnitude of the two signals. If the signals are of equal magnitude an infinitely deep null results.

When viewed on a linear magnitude versus frequency scale, the peaks and nulls appear at regular intervals, which resembles a "comb." When plotted on a logarithmic frequency scale, the peaks and nulls appear closer together at higher frequencies.

Figure 3-E shows how a single reflection can cause a comb filter response on the output of a loudpeaker. Increasing the time offset between the two signals will increase the number of construction and destructive interactions which decreases the spacing between the individual comb filters and results in a more dense comb filter response. Decreasing the time offset will reduce the number of individual comb filters and results in a less dense comb filter response.

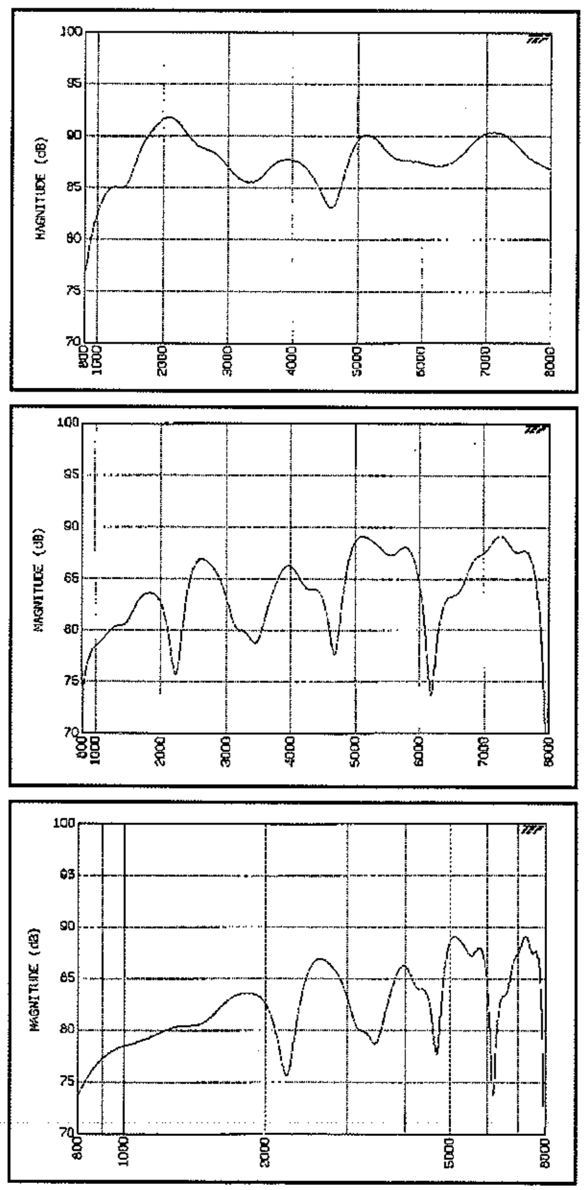


FIGURE 3-E. Comparison of loudspeaker frequency response between 800 and 8,000 Hz with and without comb filtering due to a single reflection from a surface 3 in from the loudspeaker. Frequency response without comb filter with linear scale (top); frequency response showing comb filter with linear scale (middle); and same frequency response showing comb filter with logarithmic scale (bottom).

The audibility of comb filters depends upon the *critical bandwidth* response of the ear. The ear's critical bandwidth is approximately one-third octave wide. The comb filter audibility decreases when the critical bandwidth is greater than the comb filter bandwidth. The ear's critical bandwidth response then does not have sufficient resolution to perceive the comb filter frequency. Increased comb filter audibility occurs when the critical bandwidth is equal to or less than the comb filter bandwidth. In general, a smaller time offset between the two signals will result in more audible comb filters.

The frequency at which comb filters occur can be calculated based upon the time offset between the two signals. The frequency spacing (**f**) between comb filter peaks and nulls can be calculated using the following equation:

$$\mathbf{f} = \frac{\mathbf{1}}{\mathbf{T}} \qquad \textbf{(3.A)}$$

where,

f is the frequency spacing between comb filters, Hz

T is the time offset between the two signals, ms

The frequency at which the first null will occur is at **T**/2. Table 3-A summarizes comb filter peaks and nulls.

TABLE 3-A. Comb Filter Peaks and Nulls

Time Delay, ms	Spacing Between Nulls, Hz	Spacing Between Peaks, Hz
0.1	5,000	10,000
0.5	1,000	2,000
1.0	500	1,000
5.0	100	200
10.0	50	100
50.0	10	20

Comb filtering can be caused by the interaction of electrical or acoustical signals. An example of electrical signal interaction is feeding back a portion of the signal and combining it with the original. Special effects devices called "flangers" are used to create this artistic effect and are commonly used in popular music. Direct and reflected sound arriving at a microphone can result in acoustic comb filtering. Boundary layer microphones have been developed to reduce this problem.

3.E Automatic Microphone Mixing Circuits

Various types of circuits are used in automatic mixers to activate microphones and adjust gain levels among active microphones. Some of these circuit types are described below.

Adaptive Threshold automatic mixers adjust the threshold level based on the ambient noise level in the room. The microphone gain is lowered when a microphone does not sense a signal above the reference threshold. Several schemes have been developed to achieve this including: (1) ambience sensing; (2) high-to-low adaptive threshold scanning; and (3) low-to-high adaptive threshold scanning.

Ambience Sensing automatic mixers use a microphone to sense the level of ambient noise is the space. The threshold level is automatically adjusted to account for the ambient noise level so microphones appropriately trigger on when the voice level exceeds the threshold.

High-to-Low Adaptive Threshold Scanning automatic mixers scan the ambient noise level over an 80 dB range over a 10 ms period. The signal levels of all channels are compared to a reference level and the highest channel level is automatically gated on for approximately 200 ms.

Low-to-High Adaptive Threshold Scanning automatic mixers use a low initial gate threshold which is increased by the input level average. The release time, the time it takes for a gated input to turn off, is gated and adjustable.

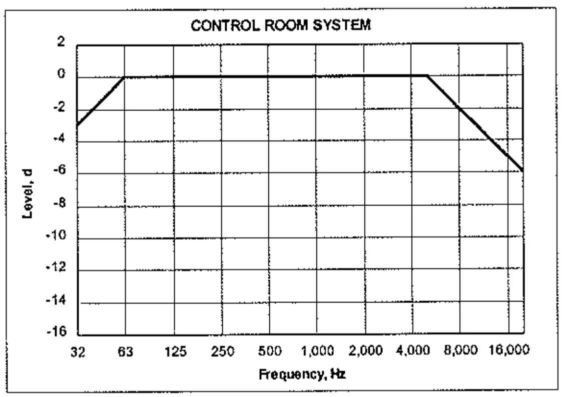
Direction Sensing automatic mixers work in conjunction with special direction sensing microphones. The microphone uses a dual element cardioid capsule which can pick-up sound within a 120° arc relative to the front of the microphone. The sound picked-up within the 120° arc must be at least 9.65 dB higher in level than at the rear of the microphone in order for the microphone to be gated on.

Fixed Threshold automatic mixers are the simplest type and use an adjustable threshold circuit. When the sound level exceeds the set threshold value, the microphone input channel is gated on. Advantages include simplicity of set-up, however for most applications the disadvantages can predominate. Some problems include false triggering of microphones due to audience coughing and other room noises when the threshold is set too low, or cutting out talkers having a weak voice, since their voice level will be below the triggering threshold. The latter condition becomes more prevalent with a greater number of active microphones and the mixer gain is lowered to compensate for the additional open microphones.

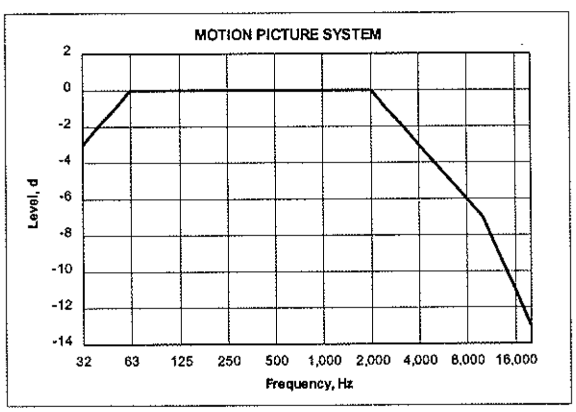
Gain Sharing automatic mixers continuously adjust the mixer gain between the different input channels. No threshold or gating circuits are used to trigger microphone signals. The signal level at each microphone output is used by the mixer to determine the gain distribution between channels. A microphone which has a weaker signal level than the average of the other microphones receives greater gain than the other channels. Internal to this circuit is a reference level established by the sum of all the microphone channel levels.

3.F Frequency Equalization Curves for Different Program Types

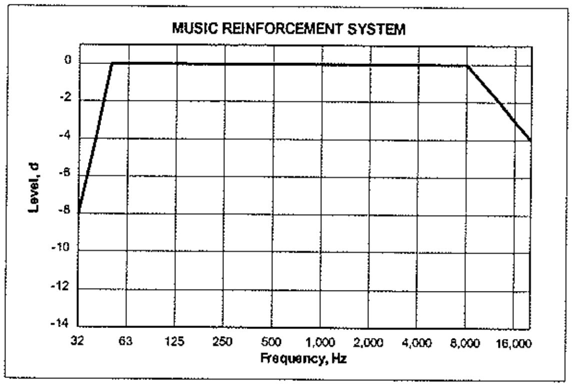
Target frequency response curves ("house curves") have been developed for different sound systems. These curve exhibit different roll-off characteristics for low- and high-frequency sound and wider or narrower flat bandwidth characteristics for mid-range sound. The frequency equalization objective is to roll-off the signal where there is little useful acoustical energy. The particulars of the sound system loudspeakers and space acoustical properties will determine the variation from the target frequency equalization response curves, both for the roll-off points and the mid-range flatness. A variation of ±2 dB is desired within the overall frequency response envelope, but may be difficult to achieve for frequencies below 500 Hz. Figure 3-F shows suggested frequency equalization response curves for different sound systems.



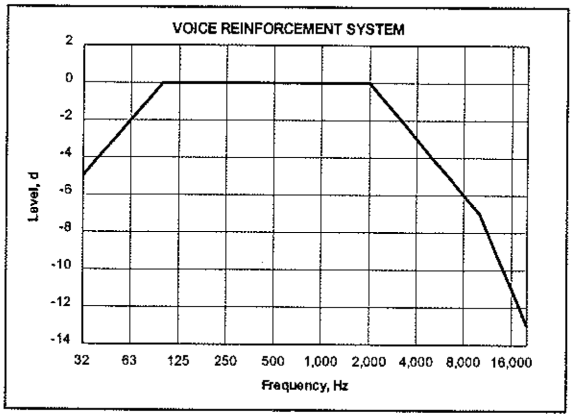
Control Room System
31.5 - 63 Hz: 3 dB/octave
63 - 6,300 Hz: flat
6,300 - 20,000 Hz: 3 dB/octave



Motion Picture System
31.5 - 63 Hz: 3 dB/octave
63 - 2,000 Hz: flat
2,000 - 10,000 Hz: 3 dB/octave
10,000 - 20,000 Hz: 6 dB/octave



Music Reinforcement System
31.5 - 50 Hz: 12 dB/octave
50 - 8,000 Hz: flat
8,000 - 20,000 Hz: 3 dB/octave



Voice Reinforcement System
31.5 - 100 Hz: 3 dB/octave
100 - 2,000 Hz: flat
2,000 - 10,000 Hz: 3 dB/octave
10,000 - 20,000 Hz: 6 dB/octave

FIGURE 3-F. General sound system frequency equalization curves for different program types.

3.H Psycho-Acoustic Basis of Signal Delay

Theory and application of signal delay lines are based on several psycho-acoustic factors relating to the perception of multiple sound sources (loudspeakers or sound reflections) as described below.

1. The hearing mechanism can perceptually combine a primary sound and several secondary sounds arriving within approximately 30 ms to subjectively perceive a single sound of greater loudness.

2. A secondary sound can be perceived as loud as the primary sound, but its level may have to be raised as much as 10 dB relative to the primary sound.

3. Secondary sounds arriving within approximately 30 ms after the primary sound improve speech intelligibility and increase perceived loudness; later sounds impair speech intelligibility.

4. The primary sound can be localized by the Haas/precedence effect. Secondary sounds can distort the perceived localization when they are substantially greater in level or delayed longer than the primary sound.

5. Secondary sounds become more disturbing with increased time delay. The audibility of delayed sound depends on the program type, with speech being more sensitive than music, which can tolerate higher levels and longer delay times. Impulses, such as hand claps, are the most subjectively sensitive, but have little relevance to the quality of speech or music programs.

6. Speech intelligibility can be enhanced or degraded depending upon the level of secondary sounds arriving 30 ms after the primary sound. The hearing mechanism will partially combine the later arriving sounds depending upon their level. However, the later arriving sounds enhance perceived spaciousness and envelopment which is beneficial for music perception.

3.1 Elementary DSP

A DSP chip is a signal processing device which contains hardware, software, and instruction sets optimized for high-speed computing of digital signals in real time. The DSP can replace conventional analog devices such as transistors, resistors, capacitors, inductors, and hard-wired circuit boards.

The basic hardware elements of a DSP system include: (1) analog input; (2) DSP chip; (3) analog output; and (4) optional host interface. The analog input bandwidth limits the incoming analog signal with an anti-aliasing filter with output to an A/D converter. The analog signal is converted to the digital format by the A/D converter. The output of the A/D converter routes the digital signal to the DSP chip where the actual signal processing takes place. The DSP chip must perform all processing on the signal before the next packet of signal information arrives. Immense computer processing power is required to achieve this real time data manipulation. The DSP output is routed to a D/A converter which reconverts the digital signal back to an analog signal. The output from the D/A converter is routed to a reconstruction filter which removes any undesired signal artifacts. An optional computer host interface can be used to control aspects of the signal processing function.

The primary advantage with DSP is the tremendous degree of flexibility in signal processing options. A virtual signal processing device is constructed which contains embedded filter coefficients and algorithms stored in memory. To create new virtual signal processing devices, the designer loads in a new set of filter coefficients or algorithms. Changing audio devices is a matter of software modifications, not hardware modifications.

One downside with DSP processing is the finite precision of the A/D and D/A converters used in the hardware. Because of inherent limitations in sampling and other digital processing functions, rounding errors can result causing some audible degradation and noise. Newer generations of DSP hardware will certainly improve upon these current performance limitations.

3.J Amplifier Classes

Amplifier output stages have different circuit designs which provide varying efficiency and distortion characteristics. Some of these circuit classes are described below.

Class A amplifiers have output transistors which always conduct current even when the signal waveform level is zero. The amplifier is characterized by extremely low distortion, low efficiency and power output, and large amounts of heat generation.

Class B amplifiers were developed to provide greater efficiency than class A amplifiers. The output transistors are operated in pairs and each transistor conducts current only for one-half the signal waveform. One transistor handles the positive going waveform and a complementary transistor handles the negative going waveform. Efficiency is improved, however crossover distortion is generated when low-level signal waveforms cross over from positive to negative and vice versa. The amplifier is characterized by moderately low distortion, improved efficiency, greater power output, and less heat generation than the class A amplifier.

Class AB amplifiers improve upon the class B design by having both transistors conduct current at low signal levels, similar to a class A amplifier. At higher signal levels the transistors alternately conduct current, similar to a class B amplifier. The amplifier is characterized by low distortion, reduced efficiency, lower power output, and greater heat generation than the class B amplifier. The majority of power amplifiers used in sound systems operate in class AB mode.

Class C amplifiers have increased efficiency but generate considerably higher distortion since the transistors conduct current only over a portion of the signal waveform. This amplifier class is normally used only for RF transmission where very high non-audio frequencies are amplified.

Class D amplifiers vary the width of a series of constant amplitude pulses by the signal waveform. The pulses are routed to a low-pass filter which reconstructs the pulsed audio signal to resemble its input, only at a higher amplitude. The amplification process is called pulse width modulation. The amplifier is characterized by low distortion, very high efficiency, high power output, and minimal heat generation than the class AB amplifier.

Class G amplifiers are similar in concept to class B amplifiers. They are more efficient and use two power supply voltages connected to the output transistors which are alternately switched on based on the signal waveform level. At low signal levels the power supply voltage is low and increases with signal level resulting in greater power output. The amplifier is characterized by low distortion, higher efficiency, greater power output, and less heat generation than the class B amplifier.

Class H amplifiers are very similar to class G amplifiers but the output transistor power supply voltage is modulated and operates above the signal waveform level, resulting in better efficiency, greater power output, and less heat generation than the class G amplifier.

3.K Is it a Driver or a Loudspeaker?

The terms "driver" and "loudspeaker" are often used interchangeably, but each has a unique meaning.

A driver is a specific loudspeaker element, such as a full-range unit, woofer, mid-range unit, tweeter, or compression-type unit. Drivers can reproduce broad or restricted frequency ranges. A loudspeaker can comprise individual or multiple drivers which may or may not be mounted in a common enclosure. Loudspeakers normally reproduce a wide frequency range.

A woofer in a separate ported low-frequency enclosure combined with a horn-loaded compression driver can reproduce a wide frequency range but the drivers are separate. Together they are referred to as a loudspeaker system. The same woofer and horn/compression driver can be installed in a common cabinet, with or without an internal crossover, and is normally referred to as a loudspeaker.

3.L Development of the Dynamic Cone Driver

The basis for the design of the dynamic cone driver was by the German engineer E.W. Siemens in 1874. His device was not an actual driver, but rather an electro-mechanical relay. In 1889 the Englishman Sir Oliver Lodge refined Siemens' relay into the first moving coil driver. Lodge referred to his invention as a "bellowing telephone" claiming his device could "fill the room with sound" and would replace the dynamic telephone handset. The dynamic cone driver did not catch on since there was no public demand for a "loudspeaking telephone". Nearly 25 years later this changed with the introduction of radio broadcasting.

In 1925 the General Electric Company engineers Rice and Kellogg published a seminal paper ("Notes on the Development of a New Type of Hornless Loud Speaker") outlining the dynamic cone driver engineering mechanisms. Their paper described a lightweight conical diaphragm driven by a wire coil located in a magnetic field; the mass-controlled nature of the system; the size of the diaphragm being small in comparison to the reproduced wavelength; and the need for an enclosure to prevent cancellation due to the radiation from the rear of the cone at low-frequencies. The first commercial product using their ideas was the "Radiola" Model 104 loudspeaker with an integral 1 W power amplifier introduced in 1926.

While improved and modified from the basic system concepts, the dynamic cone driver remains essentially the same as its original format. This humble little device revolutionized the communications industry and is found in business, educational, entertainment, and military applications.

3.M Port Enclosure Tuning

An analogy can be made between the "tuning" of a bottle and a ported loudspeaker enclosure. Blowing across the lip of a bottle will produce a characteristic frequency, the fundamental resonant frequency, based on the volume of air contained in the bottle. This frequency can be raised by decreasing the air volume by adding liquid to the bottle. The ratio of the air volume in the neck of the bottle (loudspeaker port) and the air volume in the bottle (loudspeaker enclosure) can be manipulated to tune the system resonant frequency. The tuned loudspeaker port works only over a narrow frequency band centered at the system resonant frequency just as in the bottle example.

3.N Thiele-Small Parameters

In early 1970s Richard Small published a series of papers in the *Journal of the Audio Engineering Society* describing a more accurate method of predicting the low-frequency performance of drivers in sealed and ported enclosures. His work expanded on earlier research by A.N. Thiele of Australia in the 1960s. The parameters are derived from measurements of the loudspeaker impedance characteristics, both for free-air and enclosed volume conditions.

The small signal driver parameters Thiele and Small developed permit the frequency response of the combined driver and enclosure to be calculated from the variables listed below.

$\mathbf{f_s}$	free air resonant frequency, Hz
$\mathbf{Q_{ts}}$	total Q (sum of mechanical Q and electrical Q), dimensionless
$\mathbf{V_{AS}}$	equivalent air volume which provides a restoring force equal to the driver compliance, liters
η	half-space acoustic efficiency, dimensionless

The large signal driver parameters Thiele and Small developed permit the maximum power output of the combined driver and enclosure to be calculated from the variables listed below.

$\mathbf{P_{Emax}}$	thermally limited maximum power input, watts
$\mathbf{S_d}$	effective cone area, m^2
$\mathbf{R_E}$	DC resistance of voice coil, ohms

$\mathbf{V_D}$ peak displacement volume of the cone, cm^3

$\mathbf{X_{max}}$ peak linear displacement of the cone, mm

3.0 Common Equations Evaluating Compression Driver Performance

The theoretical efficiency (η_T) of a compression driver is based on the radiation resistance ($\mathbf{R_{ET}}$) and the voice coil resistance ($\mathbf{R_E}$). The value of $\mathbf{R_{ET}}$ can be calculated using the following equation:

$$\mathbf{R_{ET}} = \frac{\mathbf{S_T(B\ell)^2}}{\rho_O \mathbf{c S_D^2}} \tag{3.B}$$

where,

$\mathbf{R_{ET}}$ is as above

$\mathbf{S_T}$ is the phasing plug slit area, m^2

$\mathbf{B\ell}$ is the product of the magnetic flux density, tesla and voice coil length, m

$\rho_O \mathbf{c}$ is the acoustical impedance in air, 415 N•s/m

$\mathbf{S_D}$ is the diaphragm area, m^2

The value of η_T can be calculated using the following equation:

$$\eta_T = \frac{\mathbf{2R_E R_{ET}}}{(\mathbf{R_E} + \mathbf{R_{ET}})^2} \tag{3.C}$$

where,

η_T, $\mathbf{R_E}$, and $\mathbf{R_{ET}}$ are as above

Multiplying the value in equation 3.C by 100 will give η_T in percent.

The mass break-point frequency ($\mathbf{f_{HM}}$) can be calculated using the following equation:

$$\mathbf{f_{HM}} = \frac{(\mathbf{B\ell})^2}{\pi \mathbf{R_E M_{MS}}} \tag{3.D}$$

where,

$\mathbf{f_{HM}}$ is the mass break-point frequency, Hz

$\mathbf{B\ell}$ and $\mathbf{R_E}$ are as above

π is a constant.

M_{MS} is equivalent moving mass of the diaphragm, kg

The second harmonic distortion ($\%HD_2$) can be calculated using the following equation:

$$\%HD_2 = 1.73\left(\frac{f}{f_c}\right)\sqrt{I_T} \times 10^{-2} \quad \text{(3.E)}$$

where,

$\%HD_2$ is the second harmonic distortion, percent

f is the driving frequency, Hz

f_c is the horn cutoff frequency, Hz

I_T is the sound intensity level at the phasing plug/diaphragm, W/m^2

3.P Horn Directional Patterns

The directivity pattern of horns will depend upon the physical size and the type of horn. The three-dimensional polar diagrams illustrate the change in directivity patterns for different horns at the frequencies of 500, 2,000, and 8,000 Hz (top to bottom) in the figures. All horns have a nominal coverage angle of 60° horizontal by 40° vertical (unless otherwise noted) in order to remove the coverage angle as a variable in the directivity pattern comparison. Figures 3-G.1 through 3-G.4 compare the unamplified voice to different size CD horns to illustrate how directional control changes with size.

Figure 3-G.1 Unamplified male voice

Figure 3-G.2 JBL 2385-46 small format CD horn

Figure 3-G.3 JBL 2353-47 medium format CD horn

Figure 3-G.4 JBL 2393-90 large format CD horn

For each frequency, note how the size and shape of the directivity pattern decreases as the horn size increases (viewed left to right, across all pages), indicating greater sound radiation control. Note in all cases, the wider three dimensional polar pattern at 500 Hz, below which directional control for some horns is difficult to maintain.

Figures 3-G.5 through 3-G.8 compare different horn types and sizes to illustrate changes in directional control.

Figure 3-G.5 Renkus-Heinz CDT-350/6 large format CD horn

Figure 3-G.6 Renkus-Heinz CDT-500/64 medium format Complex Conic™ horn

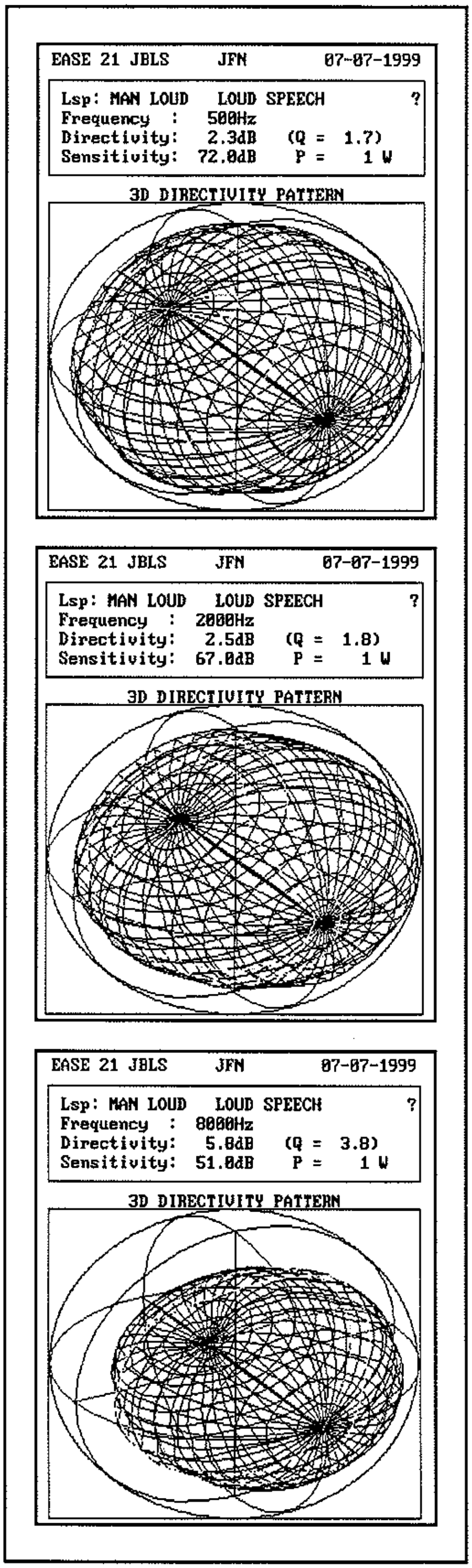


FIGURE 3.G-1

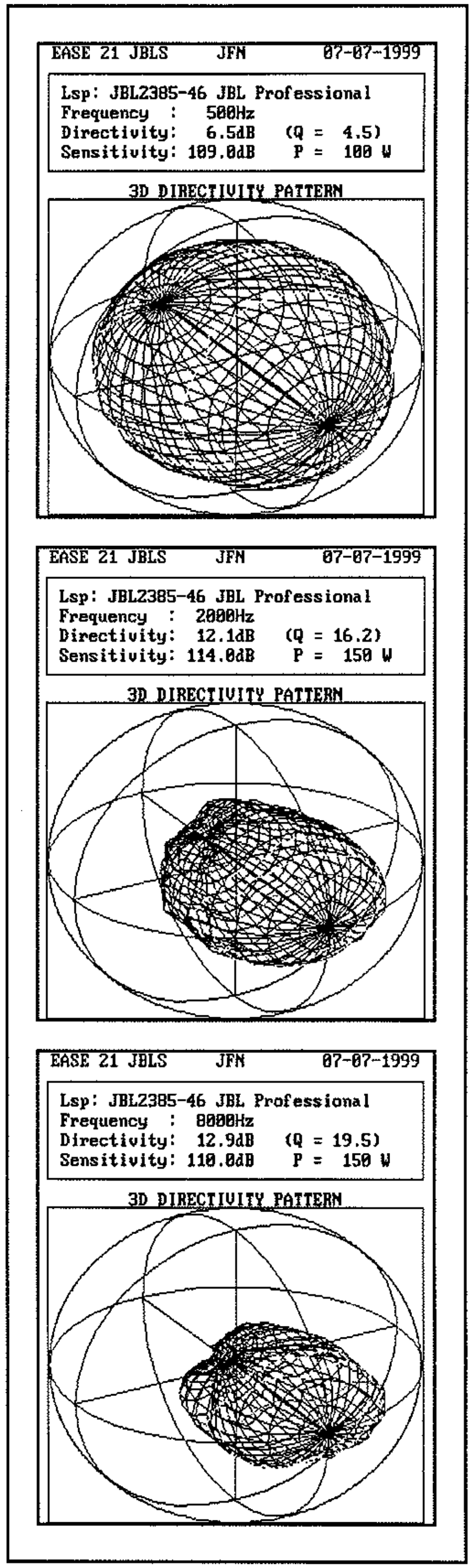


FIGURE 3.G-2

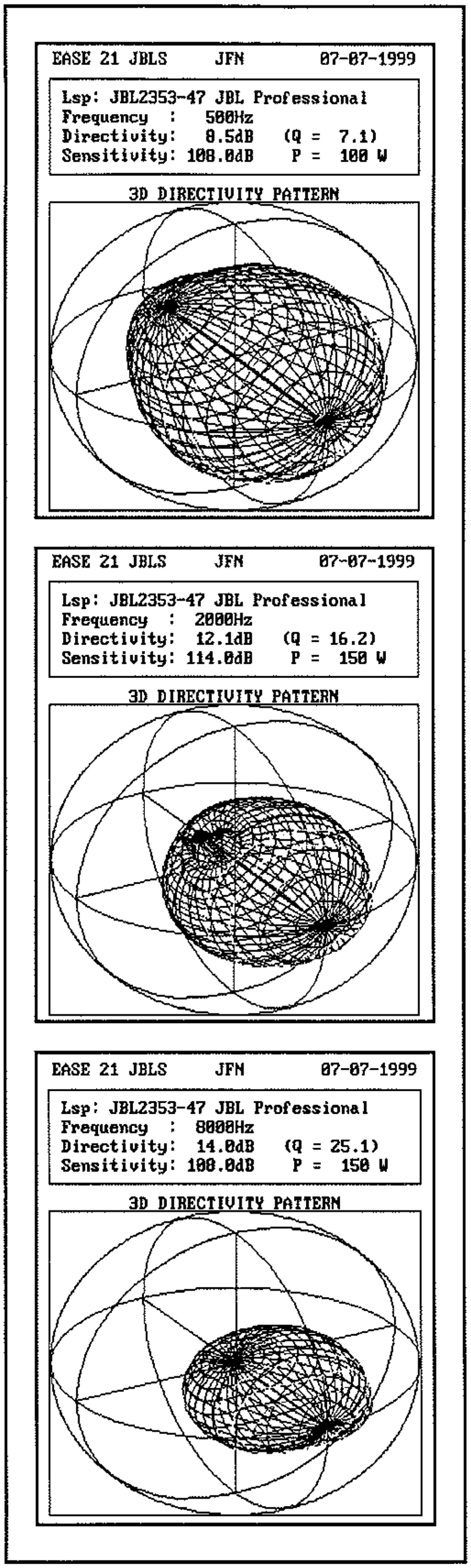


FIGURE 3.G-3

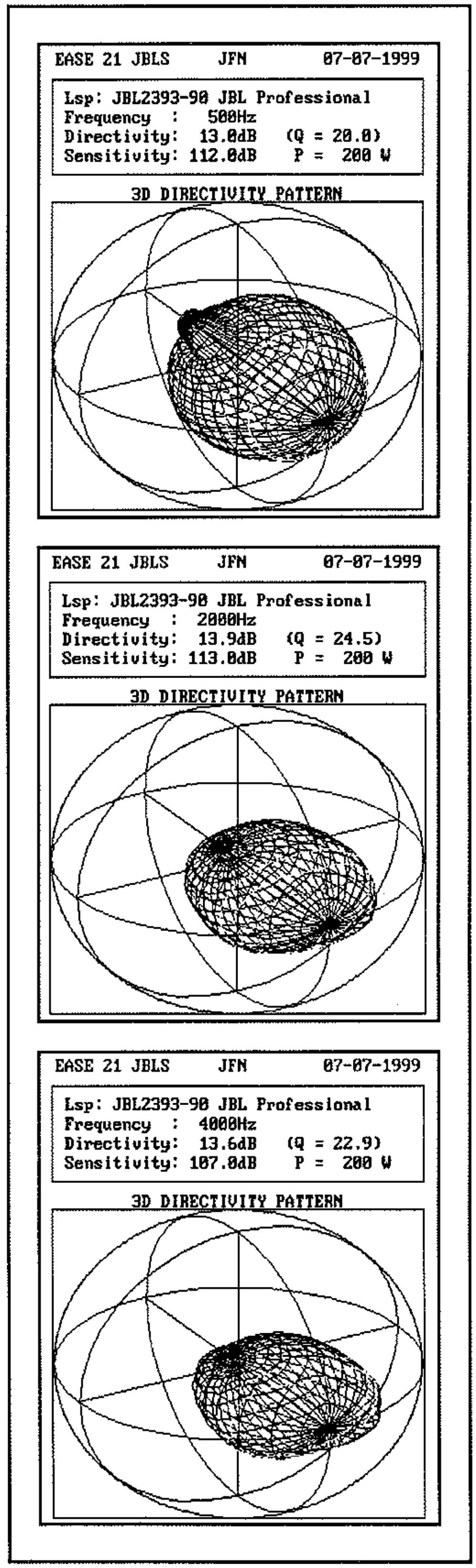


FIGURE 3.G-4

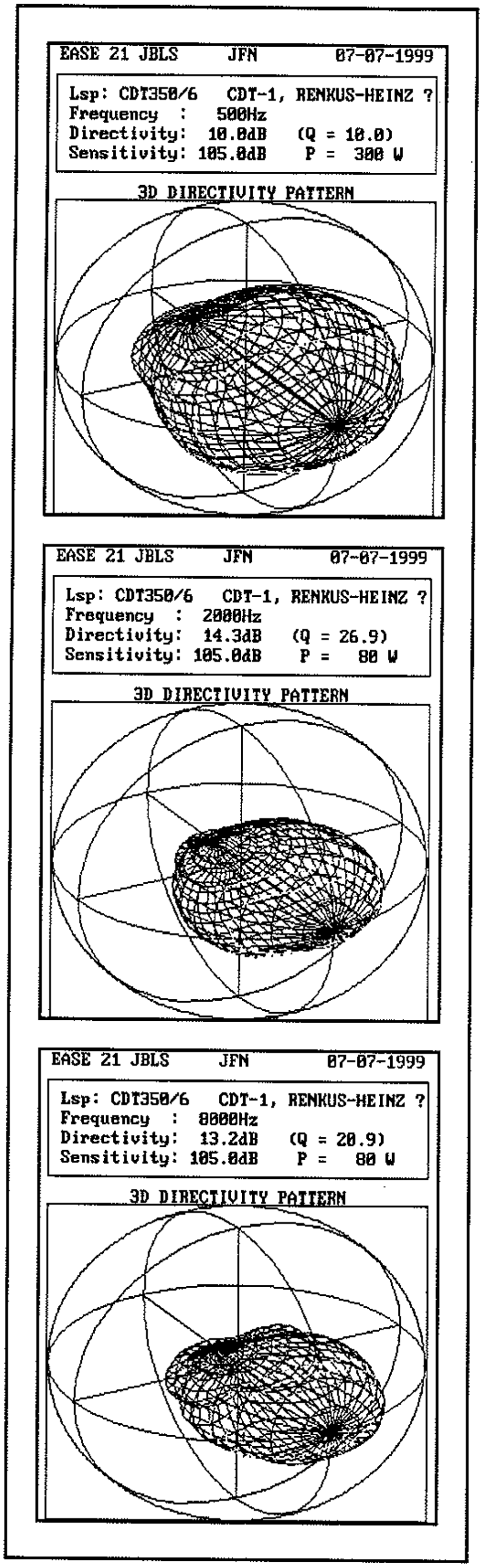


FIGURE 3.G-5

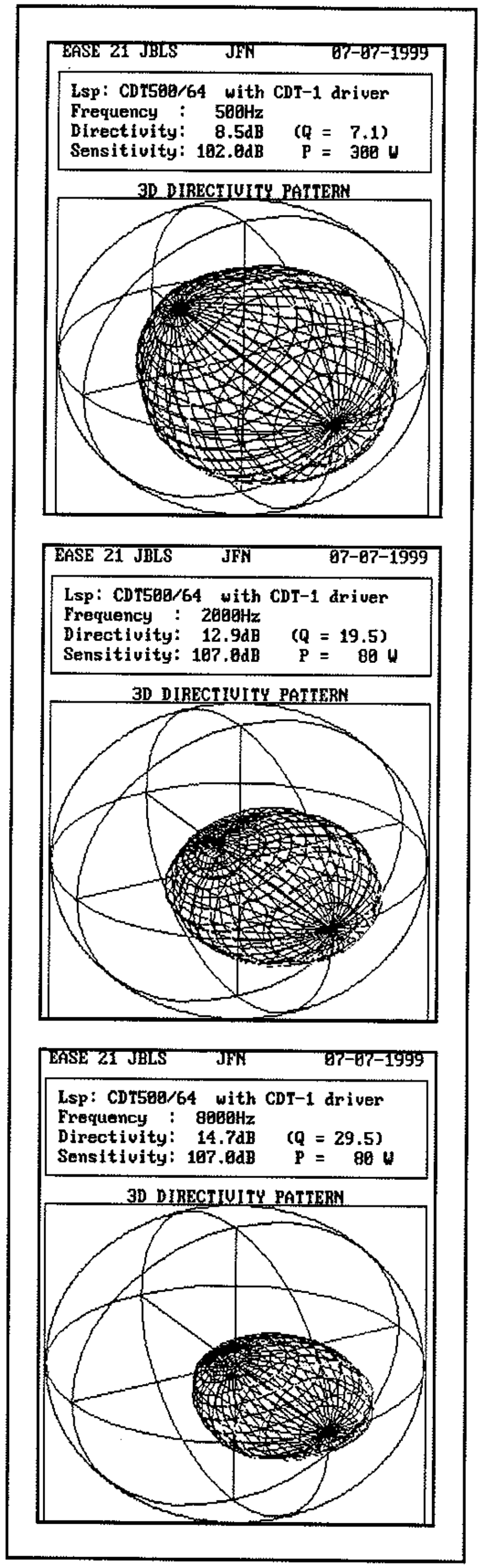


FIGURE 3.G-6

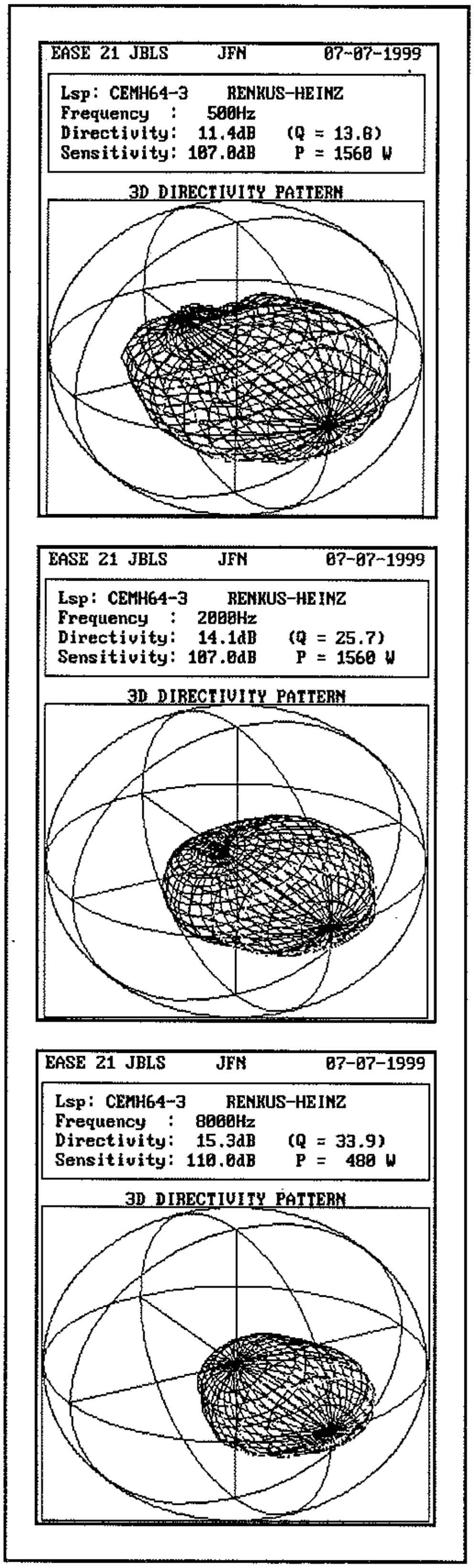


FIGURE 3.G-7

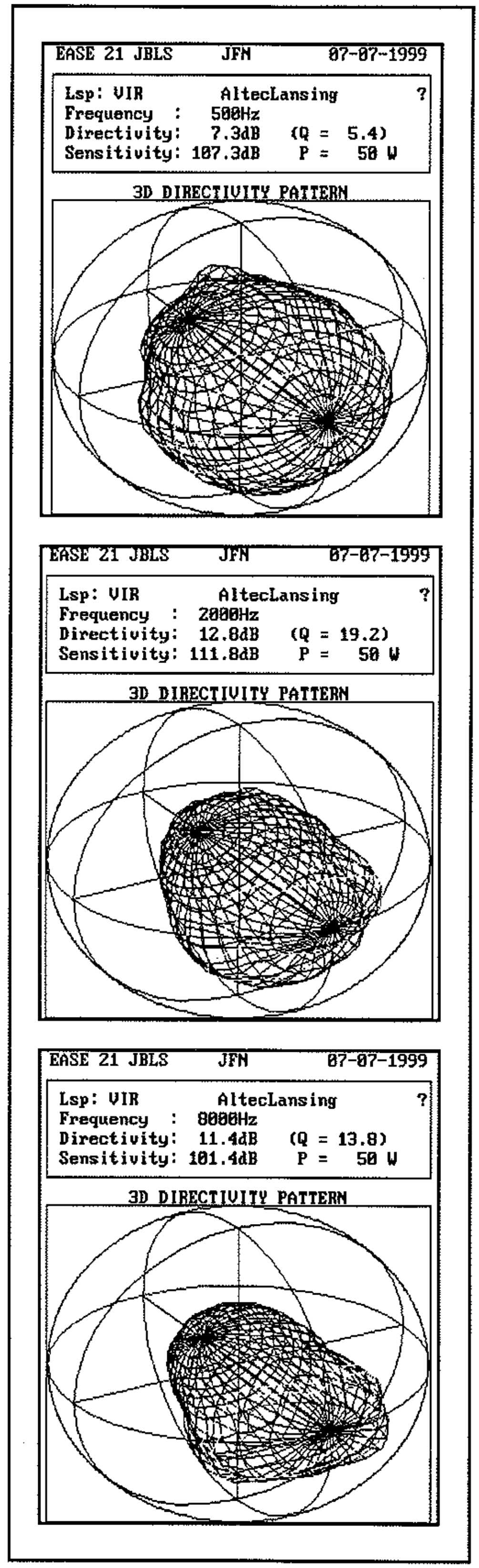


FIGURE 3.G-8

Chapter 4

Design and Installation

"The task of 'putting it all together' calls on experience, training, and individual creativity. New ideas have their birth here. Even failures are full of lessons of what to do differently the next time. Deciphering the end users' real needs and translating them into hardware operable by the actual personnel destined to use it is never a trivial task and is usually a fascinating challenge."

Davis and Davis, Chapter 18, *Sound System Engineering*,
Howard W. Sams & Company, Indianapolis, IN (1987).

4

Sound system design initially involves assessing the functional and user requirements, progressing into calculations to quantify system and equipment performance parameters, selecting appropriate equipment, and finally determining physical installation and interconnection methods between equipment rack-mounted components, microphones, loudspeakers, and other devices.

This chapter will provide the reader with an overview of determining functional and user requirements, basic engineering calculations, system design concepts, and installation practices for different sound systems.

4.1 Determining Functional and User Requirements

One of the first tasks with a sound system design, whether it is a completely new system or modifications to an existing system, is to determine the user's system functional requirements. After this is determined, the designer can begin to develop the sound system design. The list of questions below will assist the designer in determining sound system requirements.

4.1.1 Room Acoustical Properties

1. What is the room reverberation time?
2. Are there any highly reflective surfaces that might give rise to sound reflections resulting in audible echoes?

3. Are there any concave surfaces that might give rise to sound focusing?

4. Will any architectural modifications to the space be necessary to improve room acoustical properties?

5. What is the room ambient noise level?

4.1.2 Environmental Conditions (Exterior-Located Sound Systems Only)

1. What months of the year will the sound system be used?

2. What times of the day or night will the sound system be used?

3. What is the temperature and humidity range?

4. Will loudspeakers and sound system equipment be exposed to adverse weather conditions?

5. Will the loudspeakers be removed and stored at the end of the performance season or will they be installed year-round?

6. Will the loudspeakers be located in areas exposed to wind?

7. Is there pollution or salt spray in the atmosphere?

8. What is the distance to the nearest property line that loudspeakers will be located?

9. Are there any buildings or site topography conditions in the loudspeaker directional pattern that might result in sound reflections?

10. What is the exterior ambient noise level?

4.1.3 Electrical Infrastructure

1. What is the available electrical power to support the sound system?

2. Is there an adequate number of circuits on the electrical panel board or will new circuits be required?

3. How many and at what locations will electrical power receptacles be required?

4. Is there adequate provision to provide both electrical safety and technical systems grounding?

5. Are there existing conduits in the building that can be used or will new conduits be required?

6. How difficult is it to route new conduits between the sound system equipment racks, loudspeakers, microphone, line level, and control plates?

7. Can "plenum rated" cable be used in lieu of cable in conduits?

8. Are long cable runs required that might result in excess voltage drop or signal loss?

9. Will the sound system equipment or conduits be located near potential sources of EMI, RFI, or other electrical interference producing equipment?

10. Are there any radio, television, police, or taxicab antennae in the vicinity of the sound system?

4.1.4 Architectural Conditions for Equipment Installation

1. What architectural restrictions are there to install the sound system, particularly loudspeakers?

2. Will separate audio control rooms, amplifier rooms, and equipment closets be required?

3. Is a dedicated sound mixing position in the audience area to be provided?

4. Should all equipment racks be in the same location or should remote equipment racks be used?

5. Is there adequate space for installation of equipment racks and mixing consoles at scheduled locations?

6. Will there be adequate space around equipment racks to permit servicing?

7. Is there adequate heating, cooling, and dehumidification in locations where equipment racks are to be located?

8. Are any sources of standing water or water pipes located near equipment racks?

9. Will the sound system operator have direct line-of-sight to the loudspeakers?

10. Is the capacity of the building structural members adequate to carry the weight of loudspeaker and rigging equipment?

11. Is any motorized rigging required for the loudspeakers to permit servicing?

12. Are there concerns for security to prevent theft, vandalism, or unauthorized tampering with the sound system equipment?

4.1.5 Codes

1. What local and national Codes are applicable to the sound system design?

2. Are fire-rated equipment rooms or a fire suppression system required?

3. Does equipment need to have a CSA, FM, or UL rating?

4. Are fire-rated loudspeaker enclosures or floor boxes required?

5. Does cable need to be routed in conduit?

6. What classification of cable is needed to satisfy local Code officials?

4.1.6 Sound System Operators

1. Will there be a trained sound system operator or will non-technical staff have to operate the system?

2. Will the sound system operator be responsible for maintenance and adjustment of the sound system?

3. Will any touring groups with their own sound system operator use the system?

4.1.7 Sound System Functions

1. Is the sound system for an existing or new facility?

2. If the sound system is to replace an existing system, what operational functions and features were liked and disliked?

3. If the sound system is to replace an existing system, what aspects of its electro-acoustic performance were liked and disliked?

4. What is the anticipated life expectancy of the sound system?

5. Are future sound system modifications anticipated to accommodate additional equipment and new technologies, such as video/projection systems or teleconferencing?

6. What building locations will require sound systems?

7. What functions are the sound system to provide (sound reinforcement, sound distribution, or sound reproduction)?

8. What are the objective performance parameters for the sound system?

9. Is there to be any interface between the sound system and other building technical systems such as fire alarm, lighting, teleconferencing, or video/projection?

10. Is remote control required?

11. Is computer control necessary?

12. Should automatically adjusted controls or manual controls be provided for the sound system operator?

13. Should the sound system operator have access to adjust components such as crossovers, frequency equalizers, and signal delays?

4.1.8 Input and Signal Processing Equipment

1. How many microphone, line level inputs, and remote control stations are required?

2. Where are microphone and line level inputs to be located?

3. What auxiliary audio sources, such as CD and tape reproducers, are required?

4. Will automatic or manual signal mixers be used?

5. Is user-adjustable signal processing equalization needed?

6. Will signal delay be required for remote loudspeakers?

4.1.9 Loudspeaker Requirements

1. What type of loudspeaker systems are desired?

2. Where are loudspeakers to be located?

3. Are certain loudspeaker systems impractical for the facility due to room acoustical properties, architectural conditions, or user expectations?

4. Is the ceiling sufficiently high for a central cluster system?

5. Is the ceiling too high for a ceiling distributed system?

6. Are any special loudspeaker requirements such as stereo, multi-channel, or surround sound required?

7. Are loudspeakers to be zoned to permit signal routing or muting of unused loudspeakers?

8. How loud must the loudspeakers operate?

9. Is accessibility to the loudspeakers of concern?

4.1.10 Contract Documents

1. Is the sound system to be a design/bid/install or a design/build procurement?

2. Are existing architectural and sound system drawings and specifications available?

3. Will standard American Institute of Architects or owner-developed contracts between the owner and the sound system contractor be used?

4. Will performance-type or proprietary-type specifications be used?

5. What level of detail should the drawings be prepared?

6. How will contractor-proposed equipment substitutions be handled?

7. How will Change Orders and unforseen conditions be handled?

8. What will be the procedure for reviewing contractor submittals on the sound system and related electrical and architectural infrastructure items?

9. What insurance and bonding requirements are necessary?

10. Are there any contractual requirements for hiring of minority, disadvantaged, or woman-owned businesses?

11. Are liquidated damages applicable if the sound system installation is not installed by the contract completion date?

4.1.11 Installation and Contractor Coordination

1. Will the owner contract directly with a sound system contractor or will the sound system contractor be a subcontractor to the electrical contractor or general contractor?

2. Are there any particular project phasing requirements that might affect installing the sound systems?

3. What coordination with other building trades will be required to install the sound system?

4. Will the sound system contractor have access to building electrical power, elevators, and scaffolding provided by other trades?

4.1.12 Budget and Time Frame

1. What is the financial budget for sound system equipment and installation labor?

2. What is the project duration?

3. What are the anticipated start, substantial completion, and final completion dates for installing the sound systems?

4.1.13 System Acceptance

1. What procedures are required for system acceptance?

2. Are electro-acoustic performance tests required as part of system acceptance?

3. Who will perform the system acceptance testing?

4. Are electro-acoustic performance tests results to be provided as written documentation?

5. Is the owner to attend the system acceptance testing?

4.1.14 Maintenance and Operation

1. Are detailed operation and maintenance manuals required?

2. What spare parts and extra equipment should be provided?

3. What training and systems orientation will be provided to the users?

4. Is a service contract to be provided by the sound system contractor?

5. What is the warranty period for individual sound system equipment components and the system as a whole?

4.2 Design Calculations

Sound system design calculations are necessary to determine system performance and to select proper equipment. Many calculations require making initial assumptions to approximate a design solution. The initial calculation results can be refined with additional data to obtain a more assured and comprehensive result. Computer technology has automated the calculation process and provides "mapping" of calculation results for the room. Some programs offer *auralization* capabilities, enabling the designer to simulate the "sound" of the completed system prior to installation.

The three major calculations include: (1) gain-before-feedback, to determine the sound level increase before the onset of feedback; (2) sound level and coverage, to determine the spatial uniformity of frequency dependent sound in the audience seating area; and (3) speech intelligibility, to determine the ability to understand speech as reproduced by the sound system.

These calculations can be completed using simple algebraic equations, estimated or known room acoustic properties, and a knowledge of the sound system layout in relation to the room geometry.

4.2.1 Concepts of Acoustical Gain

The acoustical gain of a sound reinforcement system is the increase in the source sound level provided by the system above the unamplified source sound level at the same listener location. In a simplistic sense acoustical gain is a measure of the "amplification" provided by the sound reinforcement system. Many sound system design and performance factors are affected by acoustical gain including selection of microphones and loudspeakers, requirements for narrowband frequency equalization, maximum achievable sound levels, and the probability of acoustical feedback. The concepts of acoustical gain and feedback are only applicable for sound reinforcement systems where open microphones are used.

Acoustical gain is determined by the relative sound attenuations, expressed in dB, between the unamplified source-to-farthest listener distance ($\mathbf{D_O}$), the microphone-to-closest distance ($\mathbf{D_1}$), the loudspeaker-to-farthest listener distance ($\mathbf{D_2}$), and the source-to-microphone distance ($\mathbf{D_S}$). The geometry for gain calculations is shown in Figure 4-1.

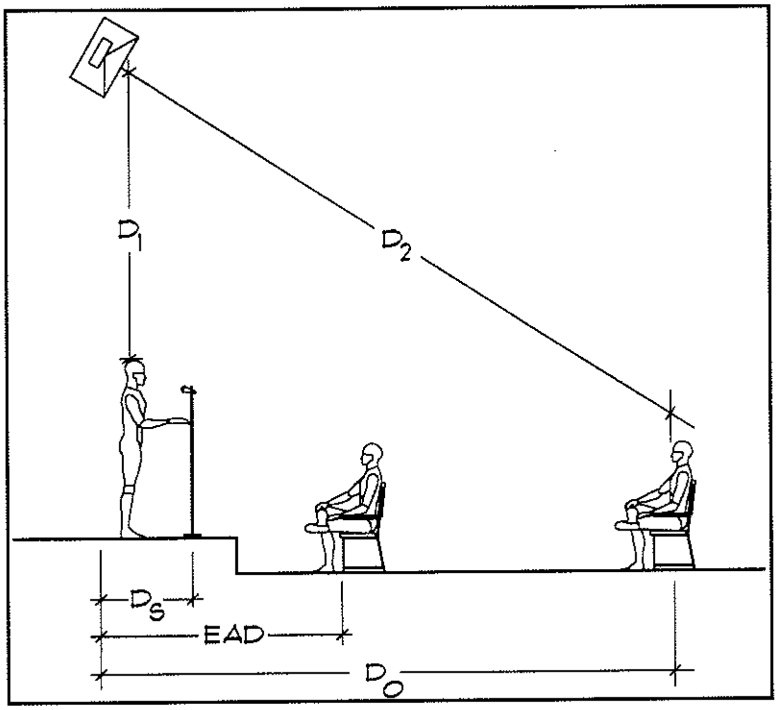


FIGURE 4-1. Geometry for acoustical gain calculations with nomenclature as described in text.

Adequate acoustical gain results when a feedback free sound level occurs at the farthest listener position and the sound level is similar to what the listener would experience near the unamplified source. The maximum acoustical gain is controlled by the *Nyquist criteria* for oscillation. A sound system will become unstable and oscillate, and go into feedback, when the sound system is at unity gain and the phase angle at the oscillation frequency is zero. Unity gain occurs when the loudspeaker sound level at the microphone is equal to the unamplified source sound level at the microphone. Ideally, less than unity gain is desired at the microphone and greater than unity gain is desired in the audience seating area. In order for the designer to evaluate acoustical gain, the concepts of *Equivalent Acoustical Distance* (***EAD***), *Needed Acoustical Gain* (***NAG***), and *Potential Acoustical Gain* (***PAG***) need to be introduced.

4.2.1.1 Concept of EAD

In a given acoustical environment there will be a distance between the sound source and the listener beyond which unamplified communication is not possible. The increased sound level provided by the sound system, compared to the unamplified source sound level, perceptually "moves" the listener closer to the source and improves listening conditions. The **EAD** is the distance from the sound source at which the farthest listener can perceive the sound source without reinforcement. In typical sound system design the **EAD** requirements will vary between 5 and 20 ft depending on the source sound level, ambient noise level, room reverberation time, and the listener hearing acuity. The **EAD** is a useful first step in determining how much acoustical gain is required from the sound system. The sound system **PAG** must equal or exceed the **NAG** to minimize feedback.

*(See Technical Notes, Section 4.A, at the end of this chapter, for additional information on **EAD**.)*

4.2.1.2 Calculation of NAG

The **NAG** is the increase in sound level required to compensate for the distance attenuation between the **EAD** and the listener's location. As $\mathbf{D_O}$ increases, the unamplified source sound level decreases. The factors which affect **NAG** include $\mathbf{D_O}$ and **EAD**. The **NAG** can be calculated using the following equation:

$$\mathbf{NAG} = 20\log_{10}\left[\frac{\mathbf{D_O}}{\mathbf{EAD}}\right] \tag{4.1}$$

where,

NAG is the needed acoustical gain, dB

$\mathbf{D_O}$ is the unamplified source-to-farthest listener distance, ft

EAD is the equivalent acoustical distance, ft

Equation 4.1 shows that **NAG** is an inverse-square law relationship between distances $\mathbf{D_O}$ and **EAD**. Increasing $\mathbf{D_O}$ or decreasing **EAD** will require greater gain from the sound system. Greater gain is needed at the rear audience seating area than the front, due to larger $\mathbf{D_O}$. Likewise, greater gain is required for high ambient noise levels where a smaller **EAD** value is required to maintain good listening conditions. Equation 4.1 is applicable to both indoor and outdoor sound reinforcement systems where the effect of "room gain" from reverberant sound is not considered. This equation conservatively estimates **NAG** for indoor sound reinforcement systems.

4.2.1.3 Calculation of PAG

The **PAG** is the maximum acoustical gain possible from the sound system before it will go into feedback. The factors which affect **PAG** include $\mathbf{D_O}$, $\mathbf{D_1}$, $\mathbf{D_2}$, $\mathbf{D_S}$, the *Number of Open Microphones* (***NOM***), and the *Feedback Stability Margin* (**FSM**). **PAG** can be calculated using the following equation:

$$\mathbf{PAG} = 20\log_{10}\left[\frac{\mathbf{D_1 D_O}}{\mathbf{D_2 D_S}}\right] - 10\log_{10}(\mathbf{NOM}) - \mathbf{FSM} \tag{4.2}$$

where,

PAG is the potential acoustical gain, dB

$\mathbf{D_O}$ is as above

$\mathbf{D_1}$ is the microphone-to-closest loudspeaker distance, ft

$\mathbf{D_2}$ is loudspeaker-to-farthest listener distance, ft

D_S is the source-to-microphone distance, assumed to be 6 ft for footstage microphones, 2 ft for lectern-mounted microphones, 0.5 ft for lavalier and handheld microphones, and 0.25 ft for headset-mounted microphones

OM is the number of open microphones, assumed to be one for automatic microphone mixers, or for manual mixing, the actual number of open microphones

FSM is the feedback stability margin, normally 6 dB

Equation 4.2 shows that **PAG** is an inverse-square law relationship between the product of distances D_1 and D_O and the product of distances D_2 and D_S less the **NOM** and **FSM** factors. Increasing D_1 and/or D_O, or decreasing D_2 and/or D_S, will increase the acoustical gain from the sound system. Acoustical gain is independent of the source sound level but is inversely proportional to D_S and is directly proportional to D_1. Three inexpensive ways of increasing **PAG** are to decrease D_S and D_2, or increase D_1. Halving D_S will result in a 6 dB increase in **PAG**. The **NOM** factor shows that for each doubling of the number of open microphones, **PAG** decreases by 3 dB. This would suggest that in many sound systems an automatic microphone mixer can be an effective means to increase acoustical gain. The **FSM** factor is important since the sound system should not be operated at the point just below feedback but should have a safety margin to provide a natural quality to the amplified sound. Equation 4.2 is applicable to both outdoor and indoor sound reinforcement systems where the effect of "room gain" from reverberant sound is not considered. This equation conservatively estimates **PAG** for indoor sound reinforcement systems.

Equation 4.2 can be adjusted for an indoor sound reinforcement system where both the microphone and farthest listener are in the reverberant sound field of the loudspeaker and the microphone is in the direct sound field of the source. The adjusted **PAG** can be calculated using the following equation:

$$\mathbf{PAG} = 20\log_{10}\left[\frac{D_{CT}}{D_S}\right] - 10\log_{10}(\mathbf{NOM}) - \mathbf{FSM} \tag{4.3}$$

where,

PAG is the potential acoustical gain, dB

D_{CT} is the critical distance of the sound source in ft, with an assumed directivity factor **Q** of 2.0 for a human voice

D_S, **NOM**, and**FSM** are as above

Equation 4.3 shows that **PAG** is an inverse square law relationship between the distances D_{CT} and D_S. The only effective way of increasing **PAG** is to decrease D_S, since D_{CT} is controlled by the room acoustical properties. Obviously, adding acoustical absorption to the room will increase D_{CT}, but this can be an expensive proposition since a doubling of D_{CT} will require quadrupling the area of acoustically absorptive room materials.

4.2.1.3.1 Effects of Directional Microphones and Loudspeakers

The above discussion of **PAG** has not accounted for the directional properties of microphones and loudspeakers. Equations 4.2 and 4.3 will result in conservative values of **PAG** since it assumes an omnidirectional microphone pick-up pattern and an omnidirectional loudspeaker radiation pattern. Cardioid-type microphones with the microphone back oriented towards the loudspeaker can in theory result in 5 to 6 dB of additional acoustical gain. The -6 dB loudspeaker coverage angle when oriented towards the microphone can also be used to raise the acoustical gain by another 6 dB.

These incremental gain values might not be fully realized with an actual sound reinforcement system due to the following: (1) at frequencies below approximately 250 Hz the loudspeaker will begin to have an omnidirectional radiation pattern; (2) cardioid-type microphones do not maintain their directional patterns below approximately 250 Hz and start to exhibit omnidirectional pick-up behavior; (3) misaligned cluster loudspeakers can project a sound lobe down towards the microphone or a microphone might have a pattern lobe in the loudspeaker direction; and (4) sound reflections from distant or nearby surfaces can arrive at the microphone. Experience suggests that feedback will be more probable in the low- to mid-range frequency region, below approximately 250 Hz. Thus, allowances for the directional effects of microphones and loudspeakers are best not to be used in calculations.

4.2.1.4 Refined Gain Calculations for Indoor Sound Systems

The above discussion of **PAG** and **NAG** has centered on calculating these parameters for sound reinforcement systems where the reverberant sound has been ignored (simplified indoor systems) or is not applicable (outdoor sound systems). The exception to this is Equation 4.3 where a simplified indoor sound system has been examined for both the microphone and listener in the reverberant field.

For indoor installations the direct and reverberant sound will be present and both will be equal at $\mathbf{D_C}$. Beyond $\mathbf{D_C}$ the reverberant sound controls and the total sound level will be constant throughout the space. At distances much less than $\mathbf{D_C}$ the direct sound will control and sound levels will decrease with increasing distance from the sound source. Thus, the **PAG** and **NAG** gain equations described above need to be modified to account for the reverberant sound and the attendant "room gain."

Equation 4.1 can be modified to account for the sound absorption in the room, in terms of **R**, and the loudspeaker **Q** value. The **NAG** can be calculated using the following equation:

$$NAG_{IN} = 10\log_{10}\left[\frac{\dfrac{Q}{4\pi EAD^2}+\dfrac{4}{R}}{\dfrac{Q}{4(\pi D_O^2)}+\dfrac{4}{R}}\right] \tag{4.4}$$

where,

NAG_{IN} is the needed acoustical gain indoors, dB

Q is the frequency dependent directivity factor of the loudspeaker, dimensionless

R is the frequency dependent room constant, ft^2

D_O and EAD are as above

Equation 4.4 shows that NAG_{IN} increases when **EAD** decreases and when D_O increases. Less intuitive is that NAG_{IN} increases as **R** increases, due to lower reverberant sound levels from the greater acoustical absorption. Increasing **Q** is an inexpensive means to decrease NAG_{IN}.

Equation 4.2 can be modified to account for **R** and **Q**. The **PAG** can be calculated using the following equation:

$$PAG_{IN} = 10\log_{10}\left[\frac{Q}{4\pi D_2^2}+\frac{4}{R}\right] + 10\log_{10}\left[\frac{Q}{4\pi D_S^2}+\frac{4}{R}\right] - 10\log_{10}\left[\frac{Q}{4\pi D_1^2}+\frac{4}{R}\right] - 10\log_{10}\left[\frac{Q}{4\pi D_O^2}+\frac{4}{R}\right] - 10\log(NOM) - FSM \tag{4.5}$$

where,

PAG_{IN} is the potential acoustical gain indoors, dB

Q, D_1, D_O, D_2, D_S, R, NOM, and FSM are as above

Equation 4.5 shows that as for reasons described above PAG_{IN} will increase as D_2, D_S, D_O, **R**, and **NOM** decrease and **Q** and D_1 increase.

A refinement to Equations 4.4 and 4.5 would be to substitute D_C, based on the loudspeaker **Q**, in place of D_O assuming D_C is less than D_O. This will be typical in reverberant rooms where the total sound level is controlled by the reverberant sound. The net effect of this substitution will yield a slightly greater value of **PAG** and a slightly smaller value of **NAG**.

4.2.2 Sound Levels and Loudspeaker Coverage

The sound level in the audience seating area due to the loudspeaker sound power can be calculated for the direct and reverberant sound levels at a given frequency. Repeating the calculations at different locations and frequencies will determine the loudspeaker sound level coverage in the audience seating area. Based on these results the designer can make adjustments in loudspeaker selection, location, or aiming to improve the sound level coverage. The goal is to provide the audience seating area with as even a sound level coverage as possible for both the amplitude and frequency domains. Ideally, the sound levels will be a minimum of 25 dB above the ambient noise level in the audience seating area.

4.2.2.1 Loudspeaker Sound Power Calculation

The loudspeaker sound power (**W**) is a function of the electrical power applied to the loudspeaker and the nominal efficiency (η) of the loudspeaker. For a needed sound power, either the electrical power to the loudspeaker can be increased, which will be limited by the maximum loudspeaker power handling capacity, or a more efficient loudspeaker can be selected. For a loudspeaker, **W** can be calculated using the following equation:

$$\mathbf{W} = (\mathbf{P})(\eta) \quad \textbf{(4.6)}$$

where,

W is the acoustical power of the loudspeaker, acoustical watts re 10^{-12} watts

P is the electrical power delivered by the power amplifier to the loudspeaker, electrical watts

η is the nominal efficiency of the loudspeaker, (%)

The more common loudspeaker acoustical power level ($\mathbf{L_W}$) in dB can be calculated from the following equation:

$$\mathbf{L_W = 10log_{10}(W) + 120\ dB} \quad \textbf{(4.7)}$$

where,

$\mathbf{L_W}$ is the loudspeaker acoustical power level, dB re 10^{-12} acoustical watts

W is as above

120 is a constant

Equations 4.6 and 4.7 shows that greater acoustical power from the loudspeaker can only be achieved by greater amplifier **P** or higher loudspeaker η . Typical full-range cone drivers have nominal efficiencies approximately one-fifth less than horn loaded cone or compression drivers. This explains why horn loaded drivers are preferred where high sound levels are needed.

(See Technical Notes, Section 4.B, at the end of this chapter, for additional information on loudspeaker efficiency.)

4.2.2.2 Direct Sound Level Calculation

The direct sound level ($\mathbf{L_D}$) can be calculated for both indoor and outdoor sound systems. As the distance between the loudspeaker and the listener increases the direct sound level will decrease. For outdoor sound systems the ambient sound level may exceed the direct sound level at distant locations from the loudspeaker. The $\mathbf{L_D}$ at the listener from a loudspeaker with a known **Q** and a distance between the loudspeaker and listener (**r**) can be calculated using the following equation:

$$\mathbf{L_D = L_W + 10log_{10}\left[\frac{Q}{4\pi r^2}\right] + 10.5\ dB} \quad (4.8)$$

where,

$\mathbf{L_D}$ is the direct sound level at the listener, dB

r is the distance between the loudspeaker and the listener, ft

$\mathbf{L_W}$ and **Q** are as above

10.5 is a constant

Equation 4.8 shows that $\mathbf{L_D}$ is inversely proportional to $\mathbf{r}^2$ and directly proportional to the loudspeaker $\mathbf{L_W}$ and **Q**. Doubling **Q** or halving **r** will result, respectively in a 3 and 6 dB increase in $\mathbf{L_D}$.

4.2.2.3 Reverberant Sound Level Calculation

The reverberant sound level ($\mathbf{L_R}$) can be calculated for indoor sound systems, but not for outdoor sound systems since there is no reverberant field outdoors. Beyond $\mathbf{D_C}$ the reverberant sound will dominate and the sound level remains constant in the room. The $\mathbf{L_R}$ at the listener from a number of loudspeakers in a room with a given **R** can be calculated using the following equation:

$$\mathbf{L_R = L_W + 10log_{10}\left[\frac{4N}{R}\right] + 10.5\ dB} \quad (4.9)$$

where,

$\mathbf{L_R}$ is the reverberant sound level at the listener, dB

N is the ratio of the total number of loudspeakers radiating sound in the room to the number of loudspeakers radiating direct sound to the listener

$\mathbf{L_W}$, **R**, and **10.5** are as above

Equation 4.9 shows that $\mathbf{L_R}$ is inversely proportional to **R** and directly proportional to the loudspeaker $\mathbf{L_W}$ and **N**. Doubling **R** and halving **N** will each result in a 3 dB decrease in $\mathbf{L_R}$.

4.2.2.4 Total Sound Level Calculation

The total sound level ($\mathbf{L_T}$) at the listener's location can be determined by adding the direct and reverberant sound levels as described in Section 1.2.

4.2.2.5 Loudspeaker Coverage of Audience Area

The calculation of loudspeaker sound levels L_D, L_R, or L_T determines how even the sound level coverage is in the audience area. One factor not explicitly noted in Equations 4.8 and 4.9 is the assumption the loudspeaker is aimed at the audience. By aiming the loudspeaker at the audience, L_D is maximized while the acoustically absorptive audience reduces L_R.

Determining the approximate loudspeaker coverage and aiming angles can be through simple geometry taken in both the horizontal and vertical planes of the room. A series of triangular templates with different angles is useful for making an initial estimate of the needed loudspeaker coverage and aiming angles. An example of these templates is shown in Figure 4-2. Once determined, computerized sound system design programs can refine the original selection.

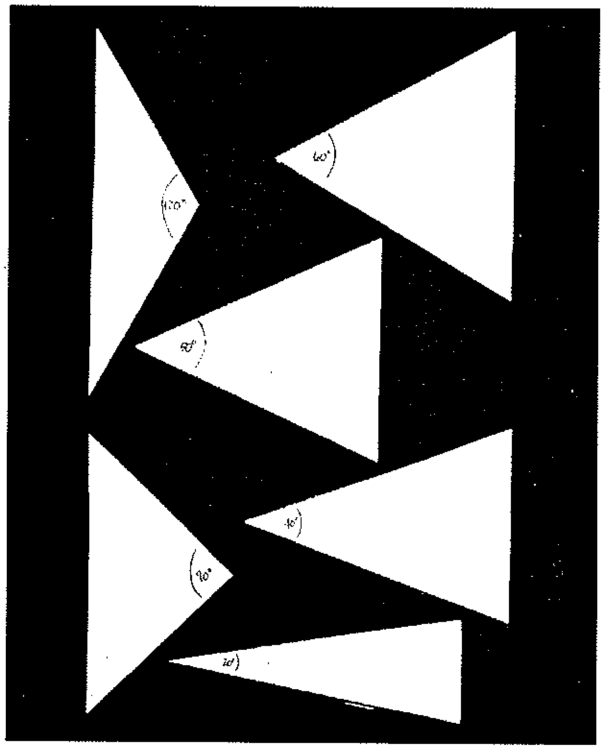


FIGURE 4-2. Triangular templates with different angles used for initial selection of loudspeaker coverage and aiming angles.

4.2.2.6 Computerized Calculations of Loudspeaker Coverage and Sound Levels

Manual calculation of L_D, L_R, and L_T to determine loudspeaker coverage and sound levels is a tedious process. The calculations need to be performed at different octave band frequencies and listener locations to fully evaluate loudspeaker performance. Computerized sound system routines speed up the calculation and design process. Figures 4-3, 4-4, and 4-5 show the results from such calculations.

(See Technical Notes, Section 4.C, at the end of this chapter, for additional information on modeling loudspeaker coverage using a light source.)

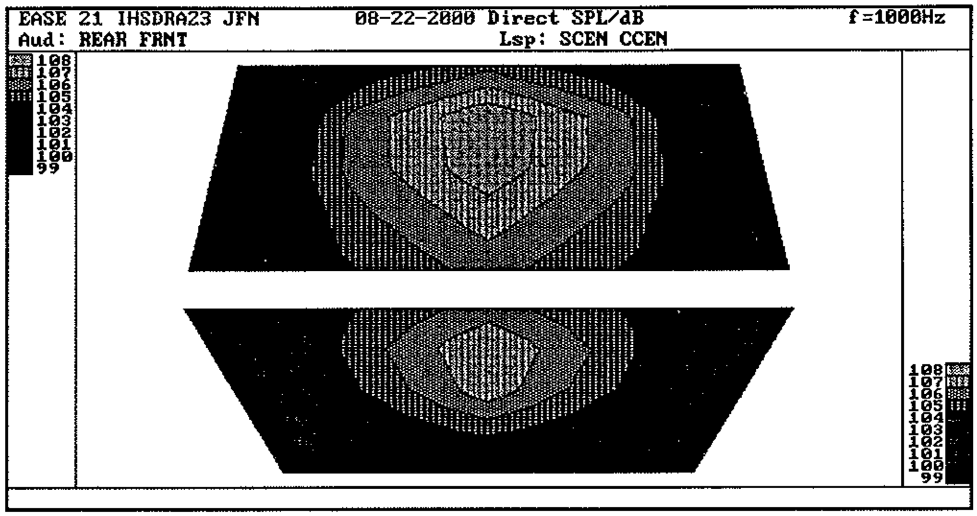


FIGURE 4-3. Computerized modeling of L_D, in an auditorium seating area. Note the bulls-eye pattern showing the two loudspeakers covering the front and rear halves of the room. Sound levels decrease towards the room sides.

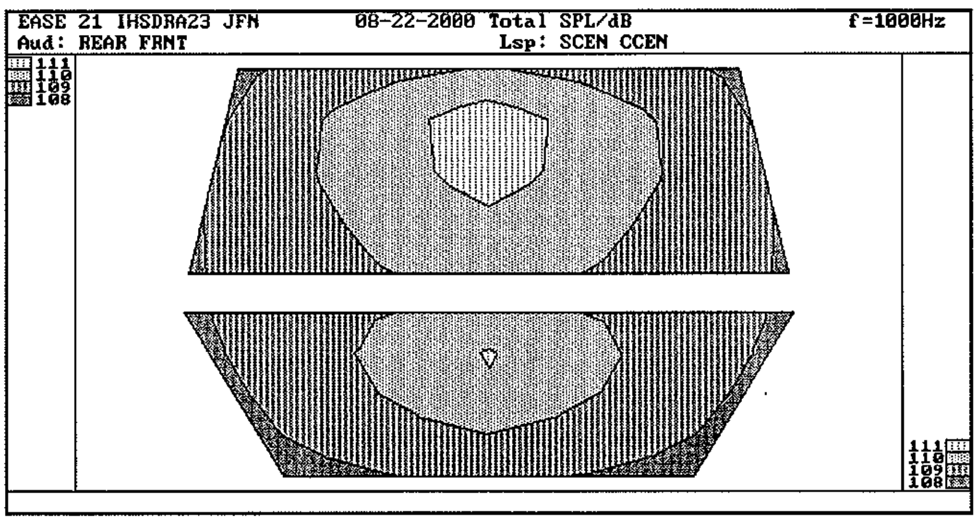


FIGURE 4-4. Computerized modeling of L_T in the same auditorium. Note the increase in sound level showing contribution of L_R.

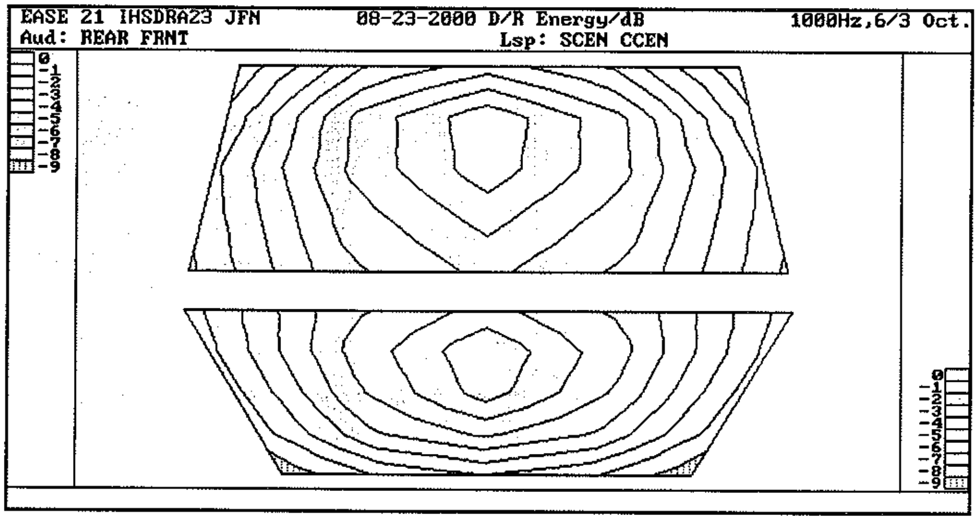


FIGURE 4-5. Computerized modeling of **D/R** level in the same auditorium. Note the decrease in **D/R** level towards the room sides.

4.2.2.7 D/R Ratio Calculation

The **D/R** ratio can be calculated by taking the arithmetic difference between $\mathbf{L_D}$ and $\mathbf{L_R}$. The value of $\mathbf{L_R}$ is frequently greater than $\mathbf{L_D}$, resulting in negative values of **D/R** ratio, which is common except in small non-reverberant rooms, or in cases where $\mathbf{D_2}$ is very small and **Q** is high.

(See Technical Notes, Section 4.D, at the end of this chapter, for additional information on sources of sound system computer modeling software.)

4.2.3 Speech Intelligibility

The speech intelligibility of a sound system relates to the listener's ability to correctly understand spoken words as reproduced by the sound system. The perception of speech intelligibility is largely dependent on understanding consonant sounds. The acceptance of a sound system often centers on its speech intelligibility performance. Pending modifications to life safety Codes will likely mandate minimum speech intelligibility performance standards for sound systems intended to transmit emergency or warning messages. .

4.2.3.1 Factors Affecting Speech Intelligibility

Many variables, broadly categorized as sound system, room acoustical, and subjective factors, affect the perception of speech intelligibility. These include: (1) adequate sound level; (2) frequency response of system; (3) signal distortion; (4) **D/R** ratio; (5) room reverberation time; (6) presence of discrete reflections; (7) ambient noise levels; (8) speech articulation by the talker; and (9) listener hearing acuity.

1. **Sound Level:** The sound system should provide a sufficient sound level above the ambient noise level so that low-level high-frequency speech consonants are not masked by noise. A good design goal is to have the total sound level a minimum of 25 dB above the ambient noise level. In most places of public assembly this can be easily achieved. Exceptions include sports and similar facilities where crowd noise levels can range from 85 to 100 dBA. Speech intelligibility depends on the absolute sound level and begins to decrease at levels above 95 dBA due to non-linearity response conditions from the ear's physiology.

2. **Frequency Response:** The sound system should provide a minimum frequency response between 300 and 4,000 Hz to adequately convey speech signals, with a frequency response between 150 and 8,000 Hz for naturalness. The majority of voice sound energy is contained between 250 and 600 Hz but the majority of speech intelligibility is conveyed between 1,000 and 4,000 Hz, which has considerably less acoustical energy.

3. **Signal Distortion:** The sound system should provide relatively low levels of distortion since this affects the system linearity. Harmonic distortion degrades speech intelligibility less than intermodulation distortion and gross non-linearity distortions, which can occur as the sound system is driven into overload.

4. **D/R Ratio:** The strength of direct sound relative to reverberant sound will affect speech intelligibility since the latter will mask the speech direct sound which strongly determines speech intelligibility.

5. **Reverberation:** Spoken speech will be degraded by reverberation since the signals will be masked by the decaying syllables spoken previously. Speech intelligibility becomes more problematic in rooms which have a reverberation time that exceeds 1.6 s.

6. **Reflections:** Strong specular reflections from room surfaces can reduce speech intelligibility through problems ranging from comb filtering, resulting in a "hollow" sound quality, to audible echoes. Reflections which occur within approximately 30 ms after the direct sound tend to improve speech intelligibility by subjectively making the direct sound louder. Reflections beyond 50 ms will degrade speech intelligibility.

7. **Ambient Noise:** High ambient noise levels above NC-40 indoors or 55 dB(A) outdoors will reduce speech intelligibility due to sound masking effects and reducing the **S/N** ratio as discussed above.

8. **Talker Articulation:** The ability of the talker to speak clearly has an affect on the perceived speech intelligibility. A trained talker will

adjust the vocal effort and spoken word delivery to suit the room acoustical environment.

9. **Listener Hearing Acuity:** Clearly an attentive listener with normal hearing acuity will perceive greater speech intelligibility than an inattentive listener or one with some degree of hearing impairment.

4.2.3.2 Speech Intelligibility Metrics

The two major speech intelligibility metrics commonly used for sound system evaluation include the **%AL$_{CONS}$** and **STI**. Sometimes the shorter form of **STI (RASTI)** is used. Both metrics can be calculated and measured, but the **%AL$_{CONS}$** lends itself to simpler calculations and will be discussed below. One disadvantage of the **%AL$_{CONS}$** calculation is the assumption of a 25 dB or greater **S/N** ratio, which may not be applicable in some sound system designs such as sports facilities where crowd noise can be quite high.

4.2.3.3 Speech Intelligibility Calculation

The speech intelligibility based on the **%AL$_{CONS}$** metric can be evaluated using the room volume, acoustical properties, loudspeaker electro-acoustical properties and quantity, and distance to the listener. As a minimum the calculation should be performed for the 2,000 Hz octave frequency band. Additional calculations at 500, 1,000, and 4,000 Hz are useful for a more thorough sound system performance evaluation. When $\mathbf{D_2}$ is less than or equal to 3.16$\mathbf{D_C}$, the **%AL$_{CONS}$** can be calculated using the following equation:

$$\%\mathbf{AL}_{\mathbf{CONS}} = \frac{\mathbf{656D_2^2T_{60}^2N}}{\mathbf{VQ}} \tag{4.10}$$

where,

$\%\mathbf{AL}_{\mathbf{CONS}}$ is the frequency dependent speech intelligibility, percent

$\mathbf{D_2}$, $\mathbf{N}$, and $\mathbf{Q}$ are as above

$\mathbf{T_{60}}$ is the frequency dependent room reverberation time, s

$\mathbf{V}$ is the room volume, ft^3

656 is a constant

Examining Equation 4.10 shows that $\mathbf{D_2}$, $\mathbf{T_{60}}$, and **N** are directly proportional and that **V** and **Q** are inversely proportional to speech intelligibility. To improve speech intelligibility, the two parameters that are the least expensive to implement include decreasing $\mathbf{D_2}$ and increasing **Q**. Halving $\mathbf{D_2}$ will improve the **%AL$_{CONS}$** by a factor of four, whereas doubling **Q** will improve **%AL$_{CONS}$** by a factor of two. Note also that decreasing $\mathbf{T_{60}}$ will improve the **%AL$_{CONS}$** by a factor of four, but this will require adding a substantial amount of sound absorptive materials to the room which may be prohibitively expensive.

In large reverberant rooms or where low **Q** sound sources are used and where $\mathbf{D_2}$ is greater than $3.16\mathbf{D_C}$, the $\%\mathbf{AL_{CONS}}$ can be calculated using the following equation:

$$\%\mathbf{AL_{CONS}} = \mathbf{9T_{60}} \tag{4.11}$$

where,

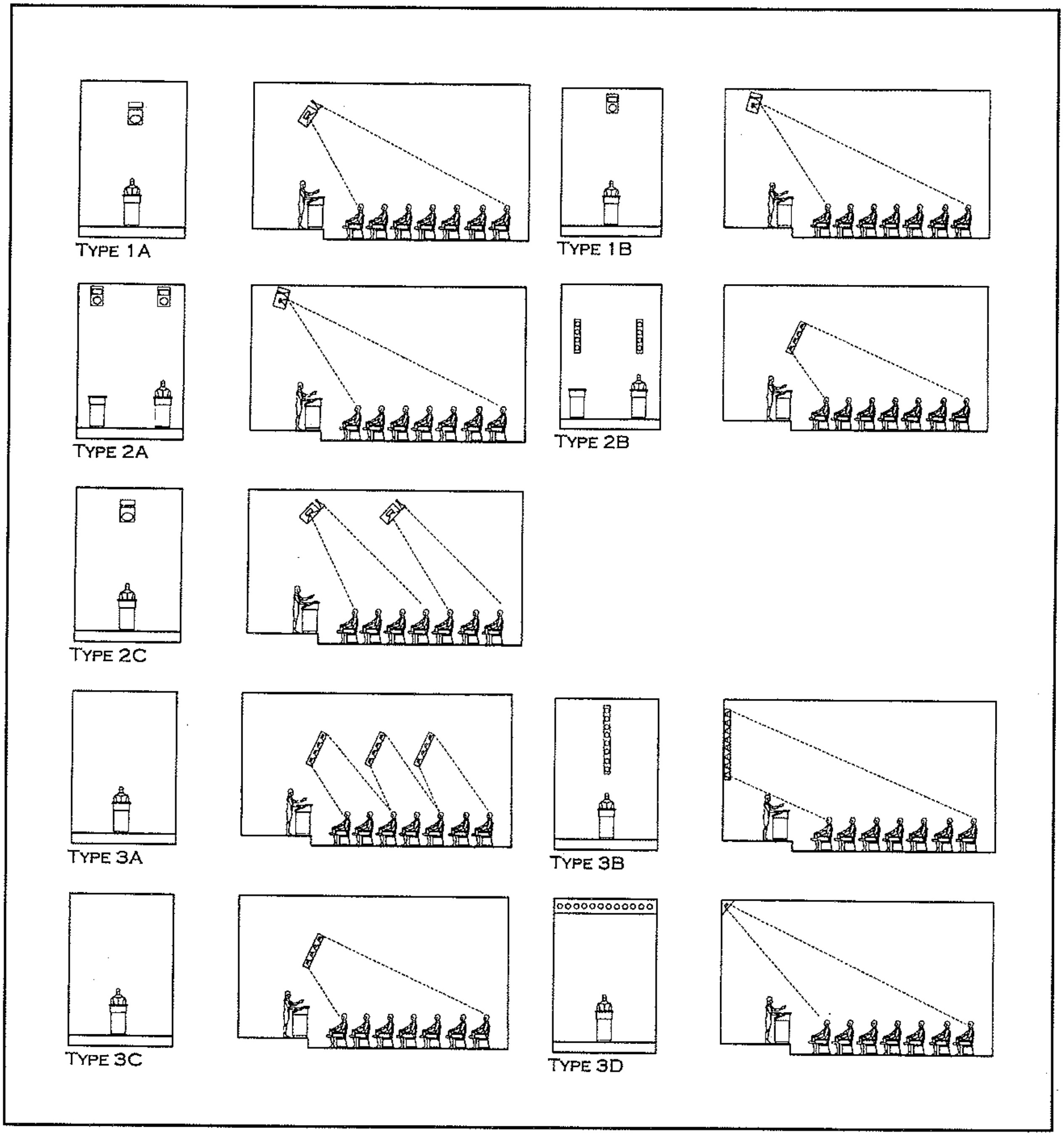
9 is a constant

$\%\mathbf{AL_{CONS}}$ and $\mathbf{T_{60}}$ are as above

4.3 Loudspeaker System Configurations

Loudspeaker system configurations vary by type, electro-acoustic performance, application, and aesthetics. There is no one perfect loudspeaker configuration for a sound system or a room. The designer needs to evaluate often conflicting criteria such as user requirements, visual aesthetics, quality of reproduction, sound level output, coverage area, ease of installation, and economic costs to determine the most appropriate loudspeaker system to specify. This process invariably results in compromise with some of the above considerations. Loudspeaker system selection is probably the most important consideration in sound system design. A related consideration is to ensure the loudspeaker system is installed at the correct location relative to the audience.

The most common loudspeaker configurations include: (1) central cluster system; (2) split system; (3) line source system; (4) multi-channel system; (5) ceiling distributed system; and (6) seat-back system. Within these broad categories are several subcategories. Central cluster systems can comprise an array of loudspeakers at the same location each covering a different frequency range or audience seating area (Type 1A system) or a single full-range loudspeaker covering the entire audience seating area (Type 1B system). Split systems often comprise full-range multi-way loudspeakers on either the left or right sides of the room front wall (Type 2A system), small column loudspeakers on both sides of the room front wall (Type 2B system), and multiple small clusters deployed around the seating area (Type 2C system). Line source systems include standard column loudspeakers along the room length (Type 3A system), digital directivity controlled column loudspeakers at the room front (Type 3B system), Bessel array loudspeakers at the room front (Type 3C system), and a horizontal line source array along the room front (Type 3D system). Multi-channel systems typically comprise stereo two-channel (Type 4A system), three-channel (Type 4B system), and five-channel (Type 4C system) with loudspeakers located above or to the sides of the proscenium. Ceiling distributed systems include small full-range transformer-coupled cone drivers (Type 5A system), and less commonly, larger horn-loaded full-range drivers (Type 5B system). Seat-back systems (Type 6A system) use small full-range drivers positioned in the seat directly in front of a small group of listeners. These different loudspeaker system configurations are illustrated in Figure 4-6 along with a summary of their design and application factors in Table 4-1.


TYPE 1A
TYPE 1B
TYPE 2A
TYPE 2B
TYPE 2C
TYPE 3A
TYPE 3B
TYPE 3C
TYPE 3D

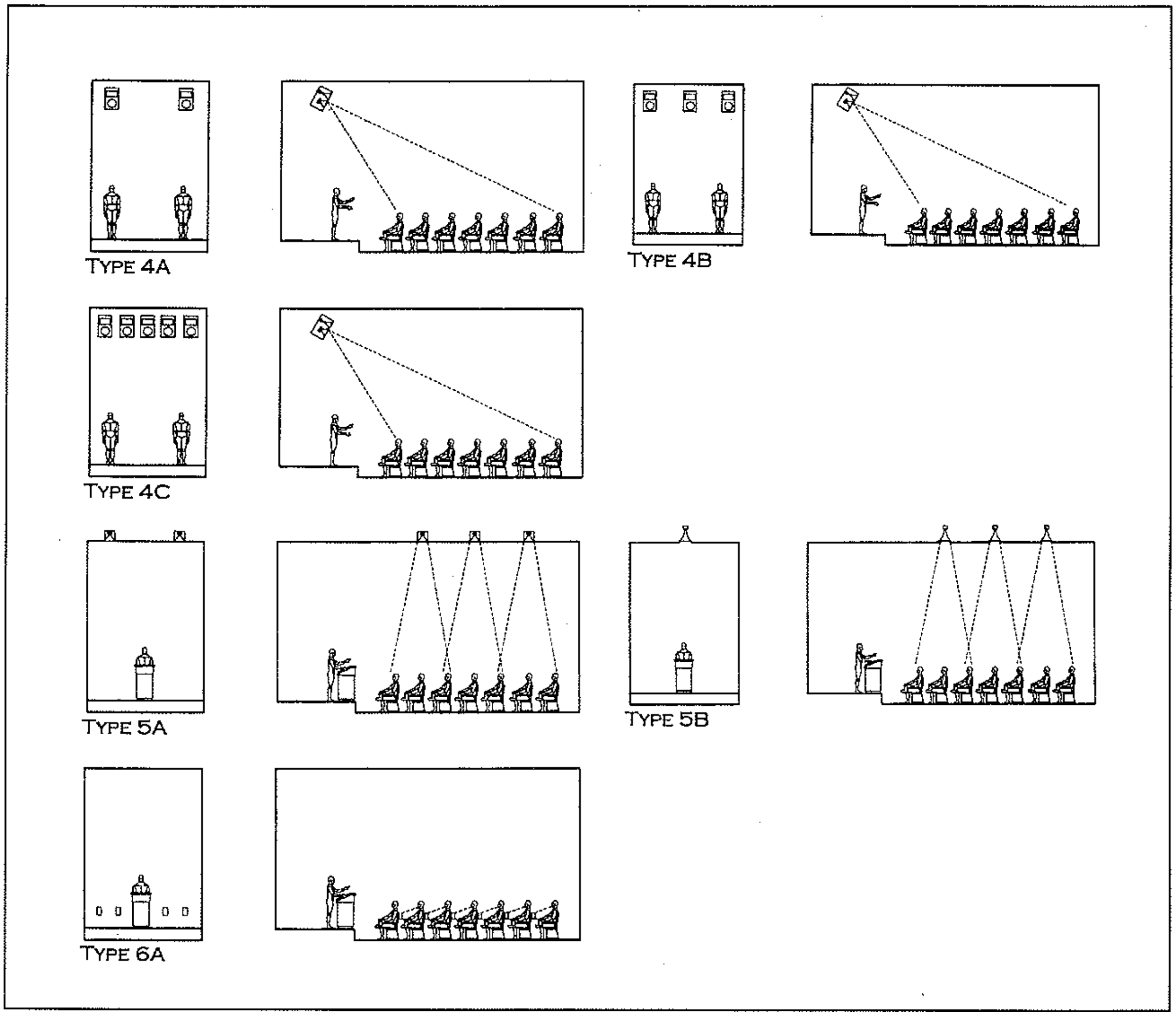


FIGURE 4-6. Loudspeaker system configurations used in sound systems design. See text for system configuration description and Table 4-1 for application, design, and installation factors.

TABLE 4-1. Characteristics of Different Loudspeaker Configurations

System Type	Application	Design Factors	Directional Realism	External Signal Delay	Loudspeaker Visibility	Relative Cost
Type 1A Central Cluster System Using Separate High- and Low-Frequency Loudspeakers	Preferred system where architecture permits use.	Line-of-sight to listeners; large horns required for directional control with long room reverberation time; loudspeaker coverage angle critical; aim loudspeakers at audience; loudspeakers should be no higher than 45 ft; may require ceiling distributed system for under balcony and other remote locations.	Excellent	Yes (driver alignment)	High (if exposed) Low (if recessed)	Low
Type 1B Central Cluster System Using Single Full-Range Multi-Way Loudspeakers				No (internal crossover may have delay)		Low
Type 2A Split System Using Left and Right Full-Range Multi-Way Loudspeakers	Used for separate left and right speaking positions.	As Type 1A; do not use both loudspeakers simultaneously; use left loudspeaker when talker is at left side and use right loudspeaker when talker is at right side; potential for greater realism than central cluster; loudspeakers should be no higher than 45 ft.	Excellent	No (internal crossover may have delay)		Moderate
Type 2B Split System Using Left and Right Column Loudspeakers		As Type 2A; column loudspeaker needs to be aimed at audience.	Good	No		Low
Type 2C Split System Using Multiple Small Loudspeaker Clusters	Used to provide close-in sound to specific areas.	Line-of-sight to listeners; closer proximity to listeners permits smaller loudspeakers to be used; helps to minimize room reverberant sound; aim loudspeakers at audience.	Good (if delay used) Poor (if no delay used)	Yes (driver alignment and to synchronize distributed loudspeakers)	High	High
Type 3A Line Source System Using Column Loudspeakers	Used down length of long narrow rooms.	Distance between left and right loudspeakers no greater than 45 ft; aim loudspeakers at audience; custom loudspeakers tailored to room acoustics and geometry work best; larger column size provides better directional control; realism improves with signal delay.	Good (if delay used) Poor (if no delay used)	Yes (to synchronize distributed loudspeakers)	High (if exposed) Moderate (if painted to match interior surfaces)	Moderate-to-high

TABLE 4-1. Characteristics of Different Loudspeaker Configurations

Type 3B Line Source System Using Digital Directivity Controlled Column Loudspeakers	Useful in very reverberant rooms; long throw applications; minimum installation depth from wall.	Internal DSP-based signal processing electronics need to be adjusted on-site; loudspeaker coverage and aiming by DSP-based electronics permits flush mounting loudspeaker to wall; different audience areas can be covered by same loudspeaker with software controlled directivity patterns.	Excellent	No (drivers delayed internally)	High (if exposed) Moderate (if painted to match interior surfaces)	Moderate-to-high
Type 3C Line Source System Using Bessel Array Loudspeakers	*far-field* or wide coverage applications.	Increased upper frequency limit and greater power handling than standard column loudspeaker; aim loudspeakers at audience; 5-element array works best.	Good	Yes (to synchronize distributed loudspeakers)		Low-to-moderate
Type 3D Line Source System Using Horizontal Line Source Array Loudspeakers	Used where desire is to not have loudspeakers exposed in view of architectural elements.	High density driver layout required; provides 3 dB per doubling of distance attenuation; localization is at driver closest to listener; useful above proscenium or front of stage locations.	Moderate	No	Low	High
Type 4A Mutli-Channel System Using Two-Channel Stereo Loudspeakers	Performance type spaces requiring the ultimate in sound quality.	As Type 1A or 1B; select each loudspeaker to cover entire audience; all loudspeakers to have same tonal characteristics.	Excellent	Yes (driver alignment)	High (if exposed) Low (if recessed)	High
Type 4B Mutli-Channel System Using Three-Channel (Left-Center-Right) Loudspeakers		As Type 4A; left and right loudspeakers used for music and center loudspeaker used for voice.				
Type 4C Mutli-Channel System Using Five-Channel Loudspeakers		As Type 4A; system provides low gain when floor-mounted microphones used.				
Type 5A Ceiling Distributed System Using Cone Loudspeakers	Intended for low ceiling rooms and under balcony areas.	Primarily for speech systems; best results in non-reverberant rooms; loudspeakers should be no higher than 20 ft; realism improves with delay; can be constant voltage system.	Good (if delay used) Poor (if no delay used)	Yes (to synchronize distributed loudspeakers)	Low	Low-to-moderate
Type 5B Ceiling Distributed System Using Horn Loudspeakers	High ceiling rooms; desire to confine sound to specific areas.	Works best for reverberant rooms; horn loudspeakers cover specific area; greater sound level than cone drivers; loudspeakers should be no higher than 45 ft; realism improves with signal delay; can be constant voltage system.			High (if exposed) Low (if recessed)	Moderate-to-high
Type 6A Seat-Back System Using Cone Loudspeakers	Where desire is not to have loudspeakers exposed in view of architectural elements; useful in very reverberant rooms.	Large number of drivers spaced every 3-to-4 listeners apart; small cone drivers in custom mounted enclosures; drivers must face listener; realism improves with signal delay; constant voltage system.	Good (if delay used) Poor (if no delay used)	Yes (to synchronize distributed loudspeakers)	Low (visible only at seat locations)	High

4.3.1 Central Cluster Systems (Type 1A and 1B Systems)

A central cluster system comprises a single or multiple loudspeakers at the horizontal room centerline elevated between 20 and 45 ft above the floor. These systems are generally located slightly forward of the sound source and aimed at the audience below. The number and type of loudspeakers depends on the audience size, desired sound level, and the frequency range to be reproduced. The Type 1A system comprises separate high-frequency horn(s) and low-frequency enclosure(s). The Type 1B system comprises single full-range multi-way loudspeaker systems integral with high-frequency horn(s), low-frequency loudspeaker(s), and internal crossover.

The most common central cluster systems are two-way arrays for voice reinforcement comprising separate high-frequency and low-frequency drivers which are crossed over at 800 Hz. Both Type 1A and 1B systems can be used for this application. Figures 4-7, 4-8, and 4-9 show typical installations of the Type 1B systems.

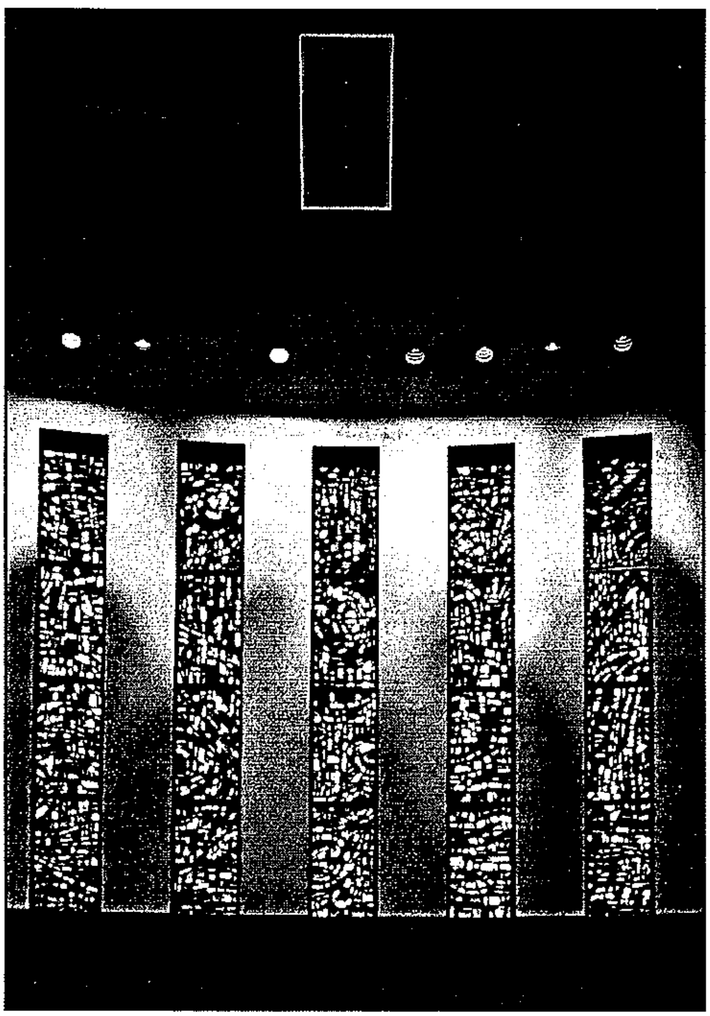


FIGURE 4-7. Type 1B central cluster system comprising one Renkus-Heinz CE125T loudspeaker at the American University Kay Chapel Spiritual Life Center in Washington, DC. System design by the author.

FIGURE 4-8. Type 1B central cluster system comprising four Renkus-Heinz TC-3 RPA reference point array loudspeakers at the Irvine Presbyterian Church in Irvine, CA. System design by Ron Sauro of Northwest Audio and Acoustics. Photo courtesy of Renkus-Heinz.

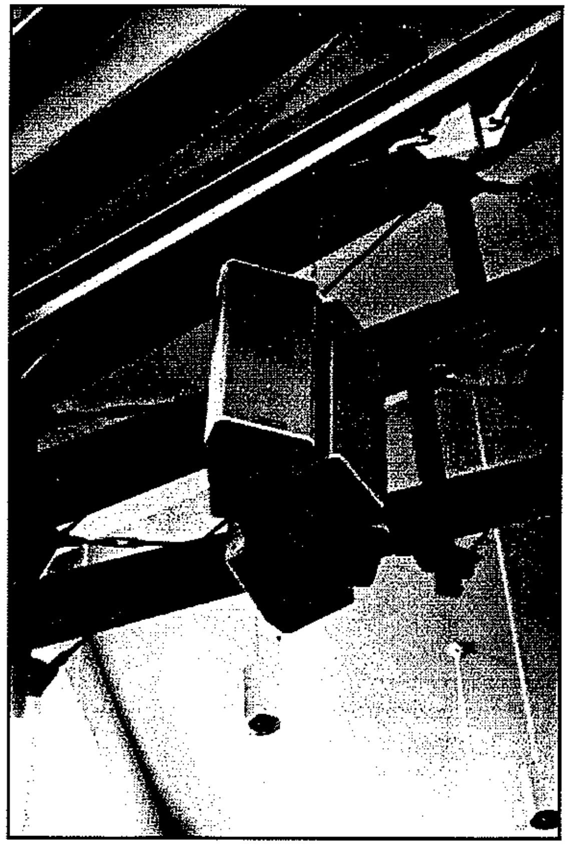


FIGURE 4-9. Close-up of the loudspeaker installation of Figure 4-8. Note custom color of loudspeaker cabinets which blends in with other architectural elements. Photo courtesy of Renkus-Heinz.

Three-way (low-, mid-, and high-frequency) and four-way (very low-, low-, mid-, and high-frequency) clusters are used where it is necessary to extend the reproduced frequency range for musical or special effects applications. Typical crossover frequencies for three-way systems are 800 and 2,500 Hz and four-way systems are 80, 800, and 2,500 Hz. The nominal **Q** value for horns used in central cluster systems varies between 10 and 35.

The selection of horn type is based on the distance between the loudspeaker and the audience and the room acoustical properties. For close-in locations near-throw horns are used. Distant locations, such as balconies, use far-throw horns. *Mid-throw* horns cover the audience between these two limits. Non-reverberant spaces can be served by small format near-throw horns where pattern control below 1,000 Hz not critical.

More reverberant spaces can be served by medium format near-throw and large format far-throw horns. Spaces with long reverberation times are best served by large format near-throw and far-throw horns. Some manufacturers produce proprietary horns which combine near-throw and far-throw patterns within a single horn similar in concept to that shown in Figures 4-10 and 3-49. An advantage with this horn type is the cluster size can be reduced.

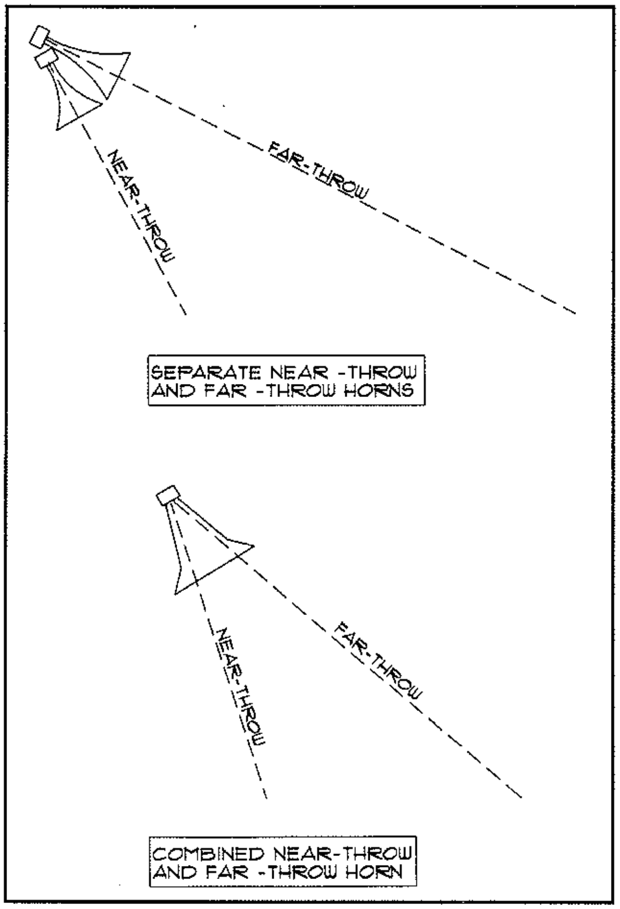


FIGURE 4-10. Conceptual relationship of combined near-throw and far-throw patterns within a single horn.

The best result with central cluster systems occurs where the room length does not exceed approximately three times the cluster height above the floor. For preliminary planning the farthest distance between loudspeaker-to-listener ($\mathbf{D_2}$) in rooms with a reverberation time less than 2.0 s can be calculated using the following equation:

$$\mathbf{D_2 \cong 0.1\sqrt{\frac{QV}{T_{60}}}} \qquad \textbf{(4.12)}$$

where,

$\mathbf{D_2}$, $\mathbf{Q}$, $\mathbf{V}$, and $\mathbf{T_{60}}$ are as above

More detailed calculations to determine the speech intelligibility should be performed once an initial distance has been calculated.

The interaction of individual drivers will have an impact on the performance of large central cluster arrays. Numerous variables characteristic to the drivers themselves, and when used in specific driver combinations or array geometries, give rise to system-specific interactions. It is presently not feasible to accurately calculate the effect these interactions will have on the performance of central cluster arrays, even with the currently-available computer modeling programs. Therefore, it is best to evaluate the performance of individual drivers over their passband for the audience area to be covered and to repeat the procedure for the other drivers in the array. Calculations that can be performed include direct and reverberant sound levels and speech intelligibility as described above in Equations 4.8 through 4.11.

4.3.1.1 Adjustment to Loudspeaker Coverage Angles

The loudspeaker horizontal coverage angle must be adjusted to account for the loudspeaker aiming angle when calculating the direct sound level coverage for an element in a central cluster array. A loudspeaker with a nominal horizontal coverage angle of 90° when positioned parallel to the floor (no aiming angle) will have a projected horizontal coverage angle of 90° on the audience. Aiming the loudspeaker down towards the audience will increase the projected horizontal coverage angle and widen the horizontal coverage area. The loudspeaker design horizontal coverage angle (θ) remains unchanged, but the projected horizontal angle (θ') at the audience increases due to the loudspeaker aiming angle (ϕ). Figure 4-11 illustrates the foreshortened horizontal angle geometry when aiming a loudspeaker. The projected horizontal coverage angle (θ') can be calculated using the following equation:

$$\theta' = 2\arctan\left[\frac{\tan(\theta/2)}{\cos(\phi)}\right] \tag{4.13}$$

where,

θ' is the projected horizontal coverage angle, degrees

θ is the nominal design horizontal coverage angle from the loudspeaker specifications, degrees

ϕ is the aiming angle of the loudspeaker, degrees

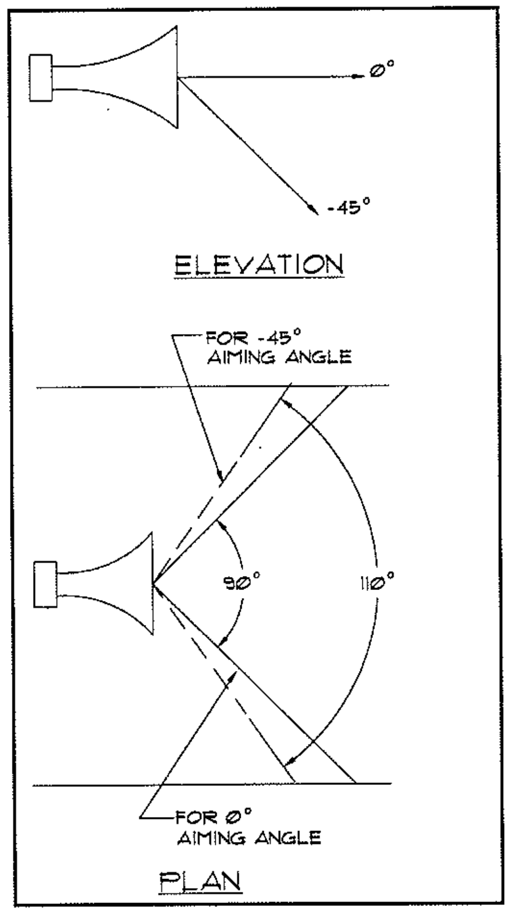


FIGURE 4-11. Effect of aiming angle on the projected horizontal coverage angle for a horn with a 90° horizontal coverage angle. For 0° aiming angle the projected horizontal coverage angle is 90°. For a -45° aiming angle the projected horizontal coverage angle increases to 110°.

4.3.1.2 Advantages of Central Cluster Systems

The two primary advantages of central cluster systems are good directional realism and the potential for high sound levels with even coverage. Directional realism is achieved by taking advantage of human hearing characteristics which can localize sound sources in the horizontal plane with great accuracy but can not localize sound sources in the vertical plane with great precision. A well designed central cluster system will have the perception of increasing the sound level in the source direction. With the loudspeakers elevated above the audience, good coverage is assured assuming loudspeakers having correct coverage angles are selected. The sensitivity and power handling capacity of horn-type drivers can result in high sound levels in the audience seating area. Caution should be exercised in the system design to ensure the sound level at the front seats is not overly loud, due to the seats being closer to the loudspeakers, compared to the more distant seats. Secondary advantages of central cluster systems include simplicity of design, installation, and low cost since fewer loudspeakers and power amplifiers are required compared to other loudspeaker configurations.

4.3.1.3 Design Objectives of Central Cluster Systems

The primary design objectives of central cluster systems are to select devices which provide the necessary coverage angle and **Q** in the audience seating area. Coverage angles should be selected which minimize the overlap of loudspeaker directional patterns in the audience seating area and do not radiate excess acoustical energy on wall and ceiling surfaces. Overlapping of loudspeaker directional patterns will give rise to detrimental audible comb filtering. Too wide a coverage angle, which results in sound radiated on wall and ceiling surfaces, will increase the reverberant sound level, decrease the **D/R** ratio, and lower speech intelligibility. When selecting horns the designer should examine potential reflections off the rear wall caused by the upper half of the vertical coverage pattern striking this surface. If reaiming the horn is not possible, it will be necessary to provide acoustically absorptive material on the rear wall surface to control sound reflections. For preliminary planning, placement of loudspeakers and determining the number of devices to cover the audience seating area can be calculated using the following equation:

$$\left[\frac{\mathbf{D_F}}{\mathbf{D_N}}\right] \le \mathbf{2} \qquad \textbf{(4.14)}$$

where,

$\mathbf{D_F}$ is the far-throw distance to the farthest listener, ft

$\mathbf{D_N}$ is the near-throw distance to the closest listener, ft

More detailed loudspeaker aiming and coverage calculations should be performed once initial locations and loudspeaker quantities have been determined.

Multiple loudspeakers should not be used to cover the same audience seating area even if higher sound levels are required. It is better to select a loudspeaker with a higher sensitivity rating, greater power handling capacity, and a coverage pattern which more closely approximates the audience seating geometry. The last recommendation can be solved by using proprietary defined coverage-type horns. The advantage with these devices is fewer horns are required for good audience coverage which increases the **D/R** ratio and improves speech intelligibility. If it is necessary to use multiple horns, they should be selected and aimed so the -6 dB down points of their coverage patterns overlap, ideally in aisles or other areas where the audience is not normally present.

The listeners need to have line-of-sight conditions to the mouth of the central cluster systems (where the sound originates) in order to have good coverage at high-frequencies which control speech intelligibility. Listeners at off-axis positions from the central cluster, while being able to see the loudspeakers, may have attenuated high-frequency coverage due to their location which does not afford an on-axis orientation to the horn mouths. The sound system operator must have direct line-of-sight conditions to the central cluster systems to experience the same sound quality as the audience in order to make adjustments to the sound to suit technical and artistic requirements.

In rooms with balconies it is common for many seats below the balcony to be occluded from the central cluster system. In such cases, it is necessary to use separate loudspeakers on a signal delay line to provide coverage to seats below the balcony. Figures 4-12, 4-13, and 4-14 show typical under balcony loudspeaker installations.

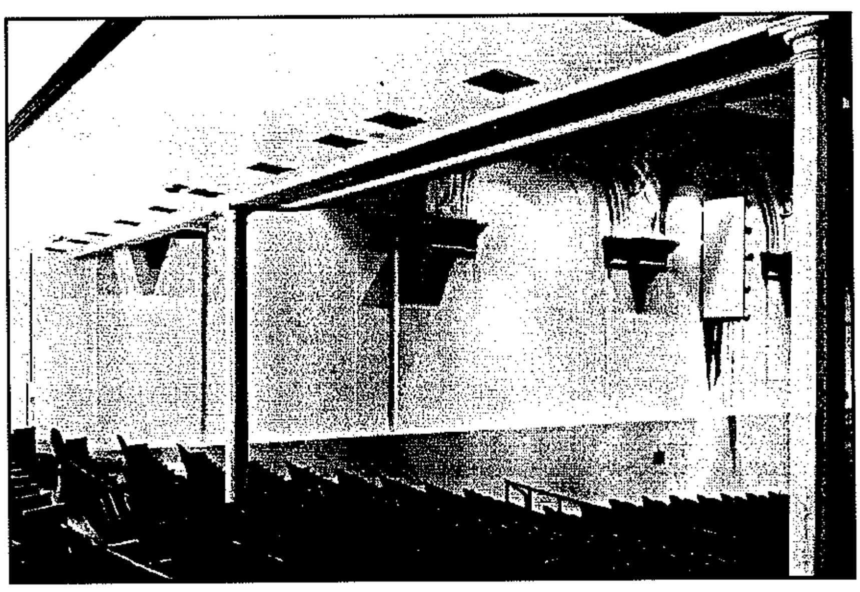


FIGURE 4-12. Under balcony loudspeakers comprising 8 in coaxial loudspeakers on 70.7 V distribution system in the Friedburg Concert Hall at the Peabody Conservatory of Music in Baltimore, MD. System design by David Lloyd Klepper of Klepper Marshall King Associates, Ltd.

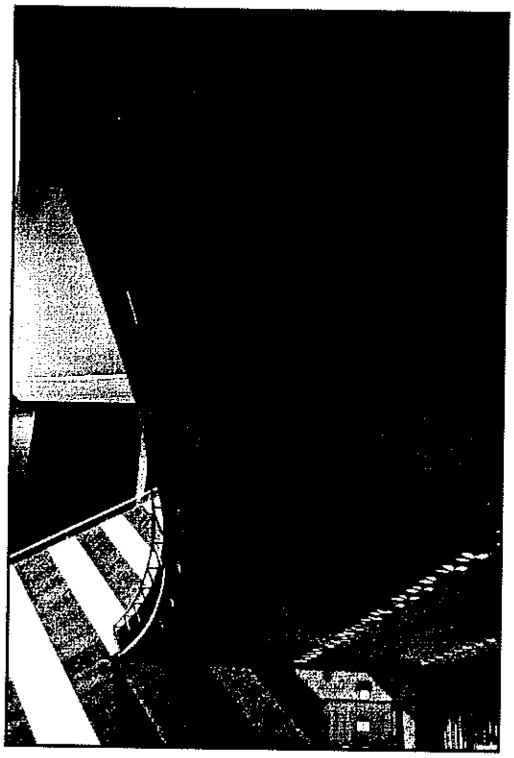


FIGURE 4-13. Under balcony loudspeakers comprising EAW JF60 small full-range multi-way loudspeakers in the Auditorium de Dijon in Lyon, France. System design by Tom Young of ARTEC Consultants, Inc. Photo courtesy of ARTEC Consultants, Inc.

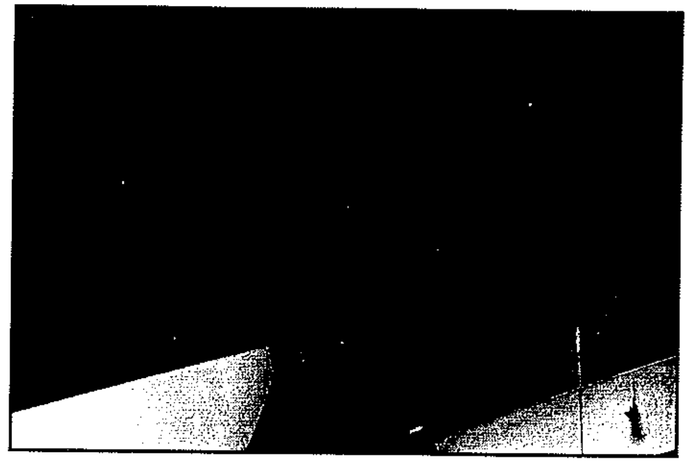


FIGURE 4-14. Close-up of the loudspeaker installation of Figure 4-13. Note the recessed loudspeaker installation to minimize visible appearance and the downward aiming angle. Photo courtesy of ARTEC Consultants, Inc.

4.3.1.4 Modifying Coverage Patterns

The coverage angles of horns can be modified, intentionally or not, depending how they are installed. Both horizontal and vertical coverage angles can be widened or narrowed depending on the horn arrangement. Often the horn array **Q** increases by stacking and splaying multiple horns. The designer should be aware of the potential for acoustic interference between multiple devices, resulting in lobing, from combining multiple horn patterns.

4.3.1.4.1 Widening Coverage

Widening the horizontal coverage angle can be achieved by stacking and splaying loudspeakers along their -6 dB horizontal coverage angle. Thus a pair of loudspeakers having a 60° horizontal coverage angle can be extended to 120° by stacking and splaying the loudspeakers at 60°. The vertical coverage angle remains essentially unchanged when widening the horizontal coverage angle. Some proprietary loudspeaker systems have been designed to be used in a "tight-pack" array, such as shown in Figures 4-8 and 4-9 where each loudspeaker covers a defined horizontal angle. The total horizontal coverage angle of the array can be approximated by the sum of the individual loudspeaker coverage angles. Loudspeakers in this category typically have a horizontal coverage angle between 20° and 40°.

The vertical coverage angle can be widened by arranging a pair of horns so they are "cross-fired" along their -6 dB vertical coverage angle. This arrangement results in the smoothest coverage with minimal lobing when the horns are splayed at the mouths rather than at the compression drivers. The horizontal coverage angle remains essentially unchanged when widening the vertical coverage angle. Figure 4-15 shows techniques for widening loudspeaker coverage and Figure 4-16 shows an installation example.

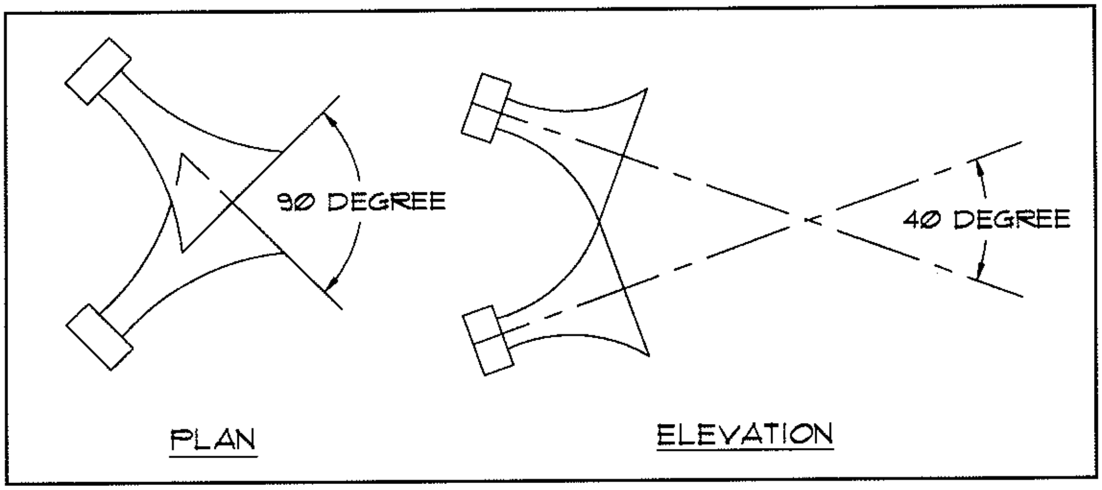


FIGURE 4-15. Techniques for widening loudspeaker coverage angle by stacking and splaying in the horizontal plane (left) and cross-firing in the vertical plane (right).

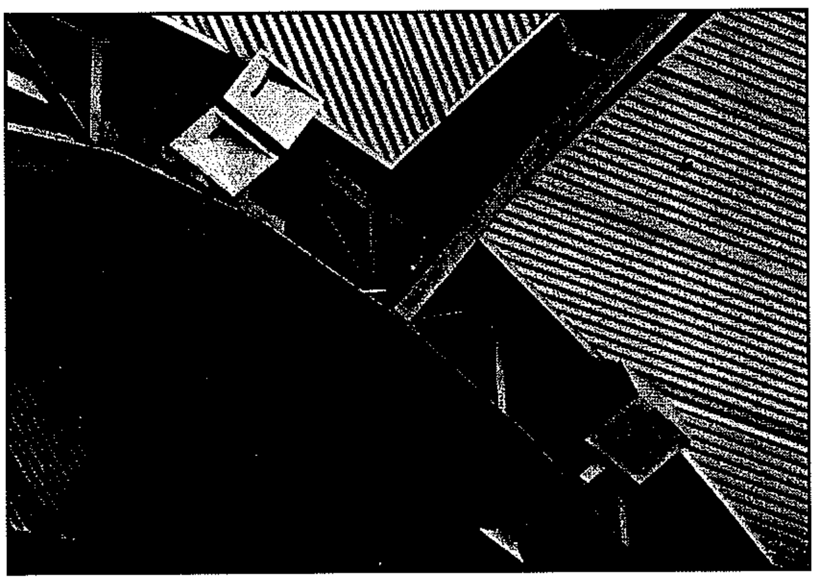


FIGURE 4-16. Widening of loudspeaker coverage angle by cross-firing in the vertical plane small format constant directivity horns (Electro-Voice 640 with DHIA compression driver) and 12 in low-frequency driver (in custom enclosure) at Reagan National Airport in Washington, DC. Note the small low-frequency enclosure at the opposite side of the horns. Low-frequency cabinets are arranged with the driver facing the opposite horns. Omni-directional nature of low-frequency sound permits this installation concept. System design by Bob Ledo of Coffeen Fricke Associates.

4.3.1.4.2 Narrowing Coverage

Placing horns horizontally side-by-side will narrow the horizontal coverage angle but causes appreciable lobing. The vertical coverage angle remains essentially unchanged when the horizontal coverage angle is narrowed. Better results, with reduced potential for lobing, are achieved by selecting a single horn with a narrower horizontal coverage angle to replace the multiple horizontal loudspeakers.

Stacking the drivers in a line source array will narrow the vertical coverage angle. Again, lobing will be present due to the multiple devices. The horizontal coverage angle remains essentially the same when the vertical coverage angle is narrowed. Figure 4-17 shows techniques for narrowing loudspeaker coverage.

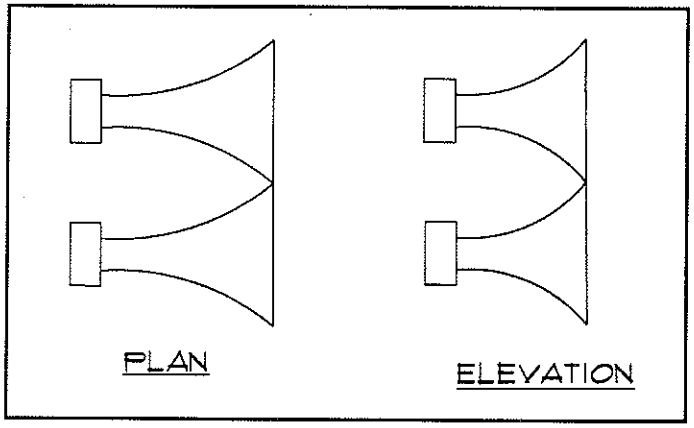


FIGURE 4-17. Techniques for narrowing loudspeaker coverage angle by stacking and splaying in the horizontal (left) and vertical (right) planes.

4.3.2 Split Systems (Type 2A, 2B, and 2C Systems)

Split systems comprise a variety of configurations including left and right loudspeakers either side of the room front wall (Type 2A and 2B systems) and small distributed loudspeaker clusters (Type 2C system). One of these systems is often used when a large central cluster system is visually too imposing for the room architectural design. They are also used when it is necessary to provide better directional realism by locating the loudspeaker close to the talker, locating loudspeakers closer to the listener to maximize sound levels, improve the **D/R** ratio, or to increase speech intelligibility. The design of split systems using single full-range multi-way loudspeaker system (Type 2A system), or individual high-frequency horns and low-frequency enclosures (Type 2C system), is similar to central cluster systems. Column loudspeakers (Type 2B system) have unique design requirements and are discussed below.

4.3.2.1 Left and Right Loudspeakers (Type 2A and 2B Systems)

Left and right loudspeakers widely spaced at the front of a room is an acceptable practice, provided only one loudspeaker operates at a time. This configuration is commonly used on either side of a auditorium stage or worship house platform. Source localization is preserved when the loudspeaker closest to the talker is active and the opposite loudspeaker is muted. If operated simultaneously acoustic interference between the two loudspeakers will result in the audience seating area as discussed in detail below. Left and right loudspeakers should be selected and located so each loudspeaker will independently cover the entire audience area. Single full-range multi-way loudspeakers (Type 2A system) or separate high-frequency horns and low-frequency enclosures in a loudspeaker cluster (Type 2C system) are appropriate for larger rooms. Figure 4-18 shows an installation example. Column loudspeakers (Type 2B system) comprise multiple small full-range drivers vertically aligned and mounted in a single enclosure. They are commonly used in small narrow rooms where mounting to side walls or building columns is feasible.

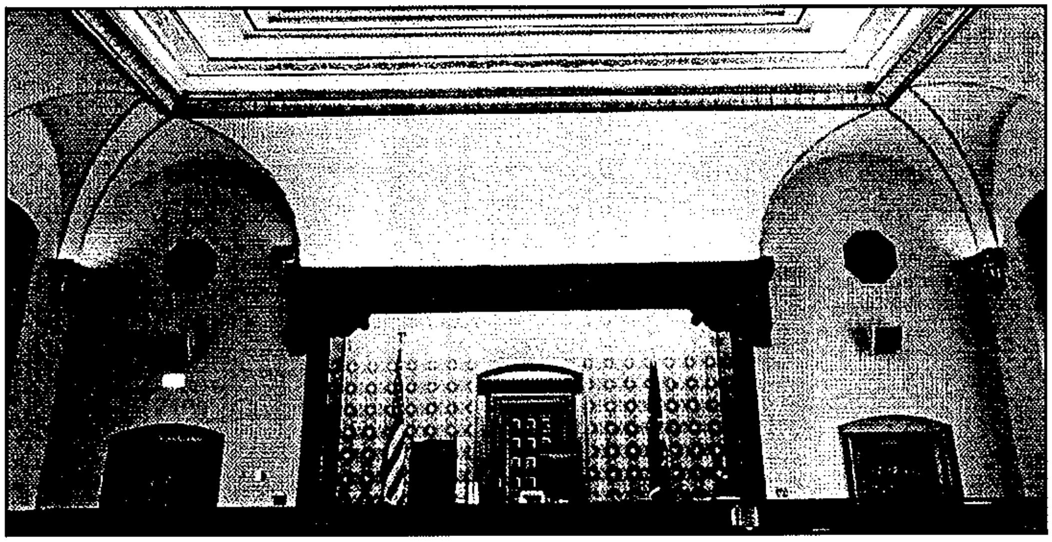


FIGURE 4-18. Type 2A left and right system comprising Frazier CAT-56 medium size coaxial loudspeakers located either side of the Judge's Bench on the front wall of a courtroom. System designer not known. Photo courtesy of Lenscape.

4.3.2.2 Distributed Clusters (Type 2C System)

Distributed clusters (Type 2C system) are similar to smaller versions of central cluster systems that are deployed in multiple units around the audience seating area. Their application is common in sports and convention facilities where the audience is seated in a bowl configuration facing towards a center activity zone. Figures 4-19, 4-20, and 4-21 show installation examples.

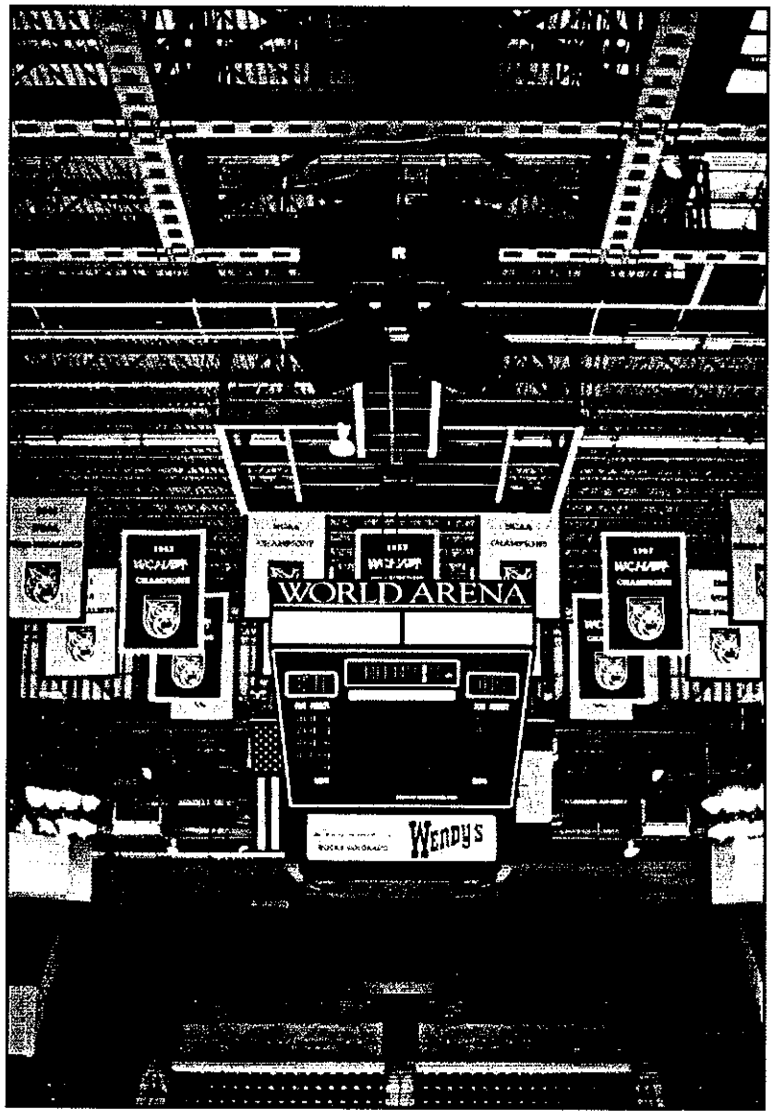


FIGURE 4-19. Type 2C distributed cluster system comprising four Renkus-Heinz CEMH medium-high frequency medium format CoEntrant™ horns and four Renkus-Heinz CELF ported low-frequency loudspeakers at the World Arena in Colorado Springs, CO. System design by Ron Baker of Wrightson, Johnson, Haddon & Williams, Inc. Photo courtesy of Renkus-Heinz.

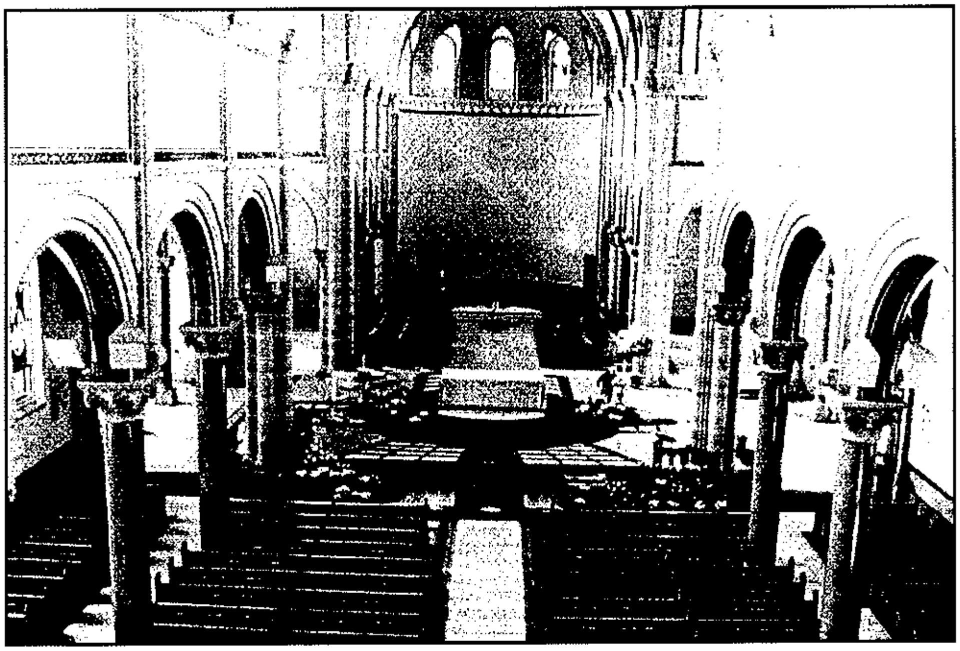


FIGURE 4-20. Type 2C distributed loudspeaker cluster system comprising two Frazier CAT-56 medium size coaxial loudspeakers on every other column at the St. Vincent's College Basilica in Latrobe, PA. System design by Gary Czajkowski of CZ Sound. Photo courtesy of Frazier Loudspeakers.

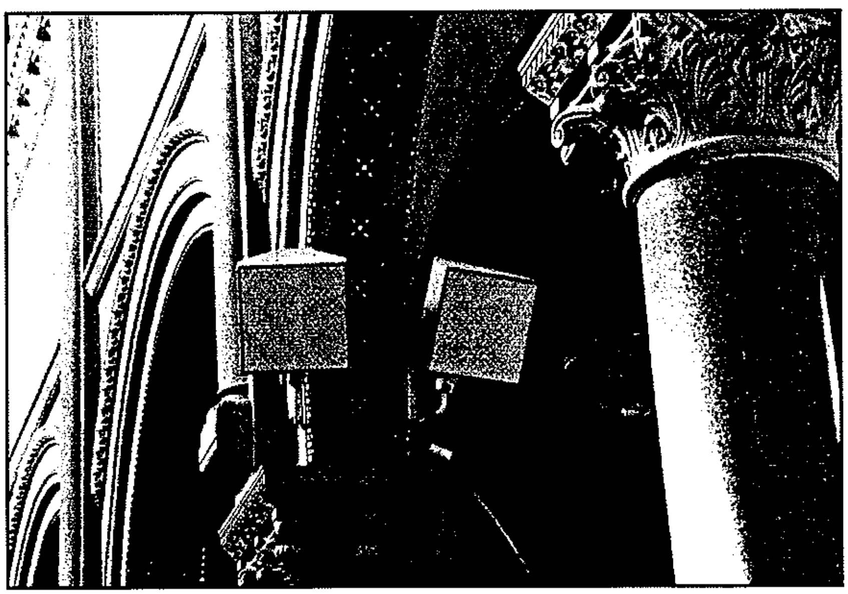


FIGURE 4-21. Close-up of the loudspeaker installation of Figure 4-20. Note white color of loudspeaker cabinets and small mounting brackets which blends in with other architectural elements. Photo courtesy of Frazier Loudspeakers.

Often the designer is faced with the option of selecting a large central cluster positioned above the activity zone or smaller distributed clusters located above the audience. Costs are generally less for the central cluster system compared to the distributed clusters, but greater control of sound levels is achieved with the latter system. This can be a major factor when it is necessary to comply with sound emission levels at the property line of an outdoor facility or on the playing floor/field as mandated by some professional sports associations

4.3.2.3 Advantages of Split Systems

Split systems share most advantages of central cluster systems. One advantage with the left and right loudspeakers is improved naturalness and localization for stage left and right podium positions since the loudspeaker is closer to the talker, compared to the central cluster which places the loudspeaker at the room centerline. One advantage distributed clusters have over a central cluster system is it places the loudspeakers closer to the audience, thus decreasing $\mathbf{D_2}$. This results in improved speech intelligibility, less reverberant sound energy radiated into the room, smaller amplifier power requirements, and smaller loudspeaker arrays.

4.3.2.4 Design Objectives of Split Systems

Most design objectives for split systems are similar to central cluster systems. Separate left and right loudspeaker systems should be selected and positioned to cover the entire audience seating area. The electronics system for the loudspeakers should be designed so that when a talker is located near the left loudspeaker, sound originates only from that loudspeaker, and vice versa. Distributed clusters should be aimed to provide coverage only to the audience seating areas. When coverage patterns have to overlap, they should do so in aisles and other non-audience areas. Another concern with distributed clusters is not to provide overly loud sound at the front seating, due to the loudspeaker being closer to the audience, relative to the rear seating. Finally, it is possible to degrade speech intelligibility and reduce the **D/R** ratio with multiple loudspeaker clusters due to the higher **N** factor value. The design trade-off involves using fewer clusters with a larger coverage angle and lower **Q** compared to a greater number of clusters with a smaller coverage angle and higher **Q**.

4.3.3 Line Source Systems (Type 3A, 3B, 3C, and 3D Systems)

A line source system comprises many small full-range drivers closely spaced together within a common loudspeaker enclosure. Line source systems are unique in that they can be used independently or as part of more complex central cluster, split system, or ceiling distributed systems. Some line source systems include: (1) standard column loudspeakers (Type 3A system); (2) digital directivity controlled column loudspeakers (Type 3B system); (3) Bessel array loudspeakers (Type 3C system); and (4) horizontal line source arrays (Type 3D system). With the exception of the horizontal line source array, all other line source systems have a vertical orientation, i.e., their height is much larger than their width or depth dimensions.

Line sources are characterized by a narrow coverage angle in the plane parallel to the driver array orientation and a broad coverage angle in the plane perpendicular to the driver array orientation. The linear arrangement of multiple small drivers radiating in-phase with each another creates destructive interference which concentrates the acoustical energy into a sound beam with a narrow coverage angle. As the reproduced frequency increases the individual drivers tend to become more directional which increases the **Q**. The lack of destructive interference in the opposite plane, and the narrow width dimension of the loudspeaker cabinet, results in a sound beam with a broad coverage angle similar to that of an individual driver.

Applications of line source systems are more common in Europe than the United States, probably owing to a smaller base of European horn loudspeaker manufacturers. Both commercial and custom line source systems are available. However, most commercial products in the United States market are of poor quality and may not be suitable for some rooms or quality installations. For these applications, custom line source systems tailored to the room acoustical properties and audience seating geometry are required. When properly designed, line source systems can provide sound reproduction that rivals horn loudspeakers without the characteristic "horn sound."

4.3.3.1 Standard Column Loudspeakers (Type 3A System)

A standard column loudspeaker (Type 3A system) is a sound source which is large in the vertical plane in comparison to the wavelength it reproduces. The system comprises multiple 4 to 6 in diameter drivers closely spaced in a vertical columnar arrangement. The driver array is housed in a narrow enclosure between 6 to 10 in wide and deep. Column loudspeakers vary between 3 and 9 ft long, with the longer devices having better vertical directivity control since a greater number of drivers are used. Average coverage angles in the speech frequency range are between 100° and 140° in the horizontal plane and between 15° and 30° in the vertical plane, but are frequency dependent. A 5 ft column loudspeaker has a vertical coverage angle of approximately 60° at 220 Hz. At an octave higher (440 Hz) the vertical coverage angle narrows to approximately 30°. Further reductions in coverage angle will be realized at higher frequencies. The nominal **Q** value for a standard column loudspeaker varies between 8 and 12. Figures 4-22 and 4-23 show standard column loudspeakers and Figure 4-24 shows an installation application of the Type 3A system.

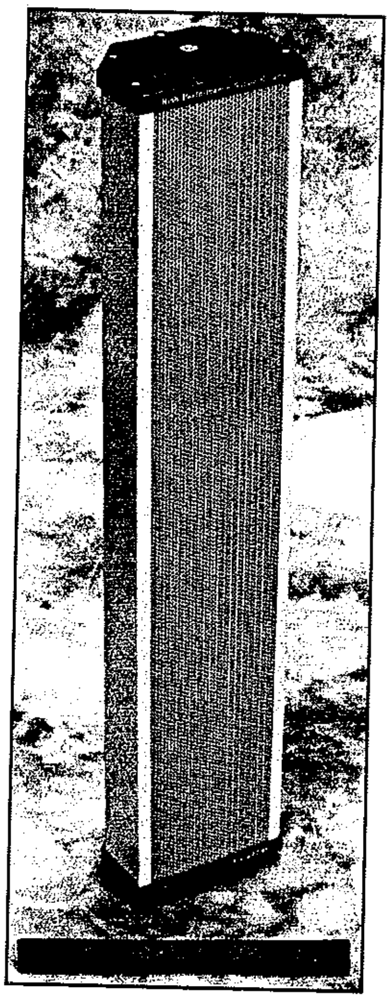


FIGURE 4-22. Type 3A standard column loudspeaker (PASO Sound Products C304T). Product courtesy of PASO Sound Products, Inc.

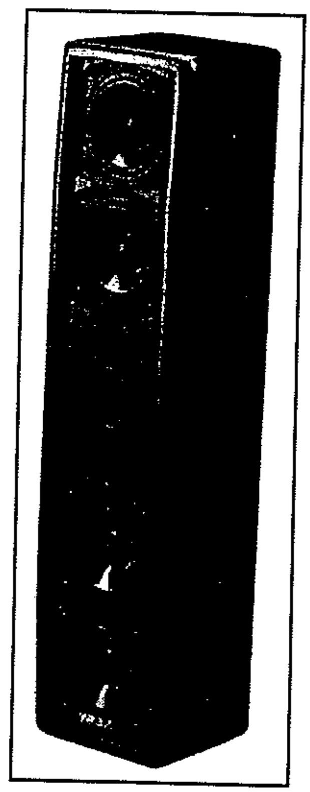


FIGURE 4-23. Type 3A standard column loudspeaker with separate low-frequency and high-frequency drivers for improved performance (EAW LS432). Height is approximately 4 ft. Photo courtesy of Eastern Acoustic Works.

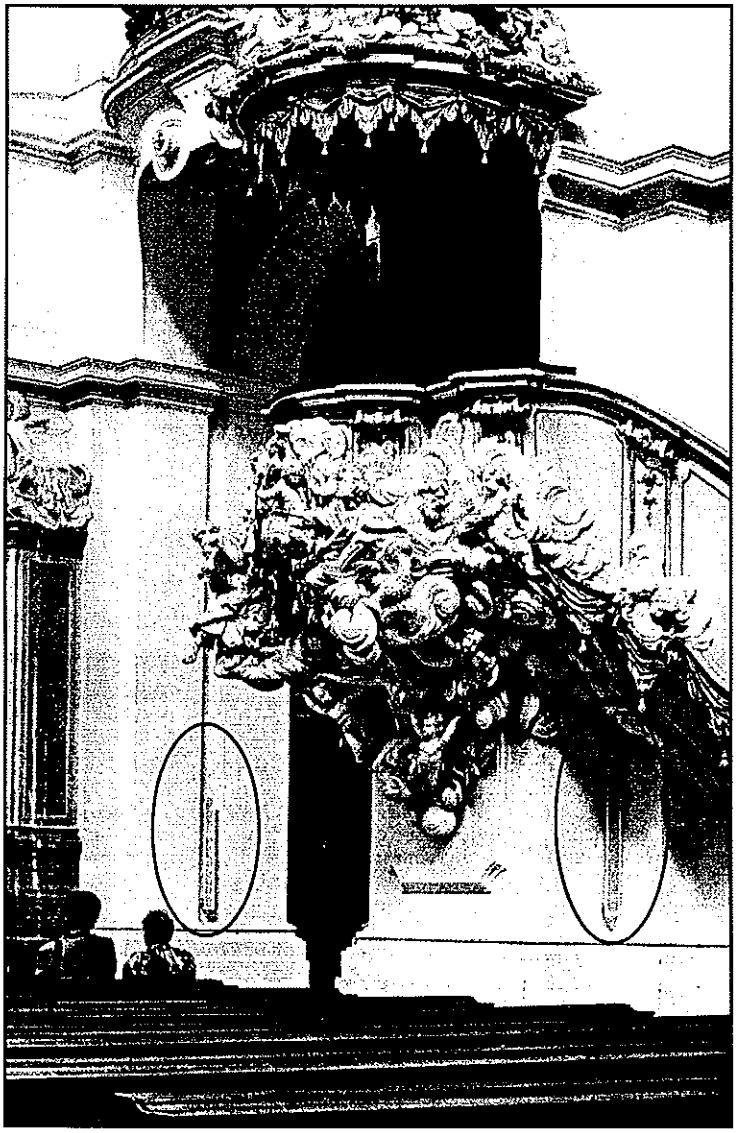


FIGURE 4-24. Type 3A standard column loudspeakers at the Katholische Hofkirche in Dresden Germany. System designer not known. Note low position of the column loudspeakers to the seating area necessary to maintain high **D/R** ratio. Photograph courtesy of Ihlow.

For a column loudspeaker of length **L**, the direct sound field will be attenuated approximately 3 dB per doubling of distance from the source up to a distance of L/π. Beyond this distance the direct sound field will be attenuated similar to a point source with 6 dB per doubling of distance attenuation. It can be seen that the benefits of decreased direct sound field attenuation will be realized only for large line arrays and small line arrays will approximate point source attenuation.

Standard column loudspeakers are low power devices that require careful matching to a separate power amplifier. They are best suited for voice reinforcement systems where reproduction of frequencies below 150 Hz is not critical. Mutual coupling between the drivers in the array extends the low-frequency response. The high-frequency response is controlled by the relatively small diameter of the individual drivers and does not provide much acoustical output beyond 8,000 Hz. In practice this is adequate for quality voice reproduction.

(See Technical Notes, Section 4.E, at the end of this chapter, for additional information on standard column loudspeakers.)

4.3.3.1.1 Basic Line Source Array Theory

The design of a column loudspeaker is a compromise between the individual driver spacing and the line array length. The spacing between the drivers is taken as the physical (acoustical) center between the devices. When the line array length is small compared to the wavelength there is little directional control. As frequency increases and the wavelength decreases, **Q** will increase, since the ratio of the array length to wavelength increases. For a line source to produce a constant **Q** value, its effective length must vary as the inverse of frequency. Since the signal input will vary with frequency, this would suggest the physical length of the line array needs to vary to optimize directional control. Techniques for achieving this are described below.

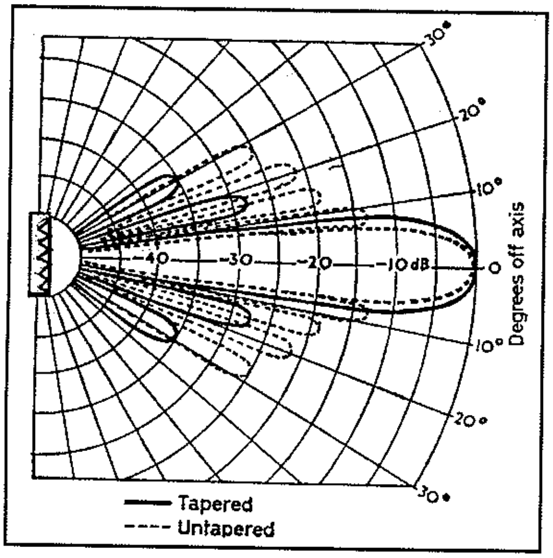


FIGURE 4-25. Directivity of a 10 ft column loudspeaker showing lobing pattern at 1,000 Hz. Dashed lines show reduced side lobes due to electrical tapering of individual drivers. Data after Peter H. Parkin.

At low-frequencies the column loudspeaker length must be at least one-fourth the wavelength of the lowest frequency at which directional control is desired. A longer line source with a greater number of drivers will have better vertical directional control than a shorter line source having fewer drivers. The upper frequency limit is controlled by the spacing between the acoustic centers of the individual drivers. When the driver spacing is greater than one-fourth the wavelength of the highest frequency at which directional control is desired, the drivers decouple and interference results. At driver spacing distances greater than one wavelength severe lobing occurs. Line arrays with a large number of drivers produce high-frequency side lobes which are generally less than 10 dB below main directional pattern. Short length line arrays with fewer drivers will exhibit greater lobing, with the magnitude of the side lobes approaching the main directional pattern. Lobing from a large column loudspeaker is shown in Figure 4-25.

The response ($\mathbf{R}(\theta)$) of a line source array in the far-field can be calculated using the following equation:

$$\mathbf{R}(\theta) = \frac{\sin[(N\pi d/\lambda)\sin\theta]}{N\sin[(\pi d/\lambda)\sin\theta]} \qquad \textbf{(4.15)}$$

where,

- $\mathbf{R}(\theta)$ is the ratio of the sound level at angle θ relative to the normal of the line array
- $\mathbf{N}$ is the number of point source elements in the line array
- $\mathbf{d}$ is the spacing between the point source elements in the line array, ft
- λ is the wavelength of interest, ft
- θ is the angular location relative to the normal of the line array

The quantities in the brackets are evaluated in radians

An ideal line source array will result when **N** approaches infinity and **d** approaches zero. When this occurs the product of **N** and **d** will equal the length of the array (**L**). Equation 4.15 can then be simplified to the more general case of:

$$R(\theta) = \frac{\sin[(\pi L/\lambda)\sin\theta]}{(\pi L/\lambda)\sin\theta} \tag{4.16}$$

where,

$R(\theta)$, λ, and θ are as above

L is the length of the line array, ft

The quantities in the brackets are evaluated in radians

In practice Equation 4.16 can be used with sufficient accuracy providing the distance between the driver acoustic centers in the line array is small compared to the reproduced wavelength.

4.3.3.1.2 Modifications to Line Source Directivity

The simplest way of connecting the individual drivers in a standard column loudspeaker is for each driver to receive the same in-phase input signal. Several problems result when a column loudspeaker is used for full-range reproduction: (1) as frequency increases, and the wavelength decreases to become comparable to the spacing between the individual drivers, the radiated sound breaks up into several lobes of varying strength; (2) above still a higher frequency the directional pattern of the individual drivers narrows which causes increased on-axis high-frequency sound levels (beaming) and reduced off-axis high-frequency sound levels; and (3) the loudspeaker will become increasingly directional in the horizontal plane as frequency increases. Of these problems, the first is the most severe. The lobing problem, with its attendant comb filtering, may result in reduced gain-before-feedback, contribute to a low **D/R** ratio, and reduce the off-axis frequency response. The other two high-frequency directivity-related problems typically occur at frequencies approaching or beyond which most column loudspeakers are used.

Several techniques have been developed to improve the performance aspects of column loudspeakers by reducing the lobing, extending the useable frequency response, and providing better directional control: (1) electrical tapering through level adjustment; (2) mechanical tapering using physical methods; (3) frequency response shaping and subdivision; and (4) signal delay of individual drivers. These techniques attempt to make the line array behave with frequency-dependent variable length properties.

Electrical tapering of individual drivers can be done with transformers or electrical resistance networks to progressively reduce the voltage across the drivers towards the outermost drivers. For a nine-element line array (one center driver and four drivers above and below), the center driver voltage is not reduced and each successive driver is reduced in voltage by 20 percent, until the outermost driver is tapped at 20 percent of the center driver voltage.

Mechanical tapering uses fiberglass wedges, which creates a varying acoustical impedance (loading) across the individual drivers, or some form of physical angling of individual drivers to modify the directional pattern. The fiberglass wedge mechanical tapering technique was pioneered by David Lloyd Klepper. This technique is illustrated in Figure 4-26 and uses medium density fiberglass of approximately 2 in thickness covering the outermost array drivers and narrowing to a fraction of an inch at the driver just above and just below the array center driver.

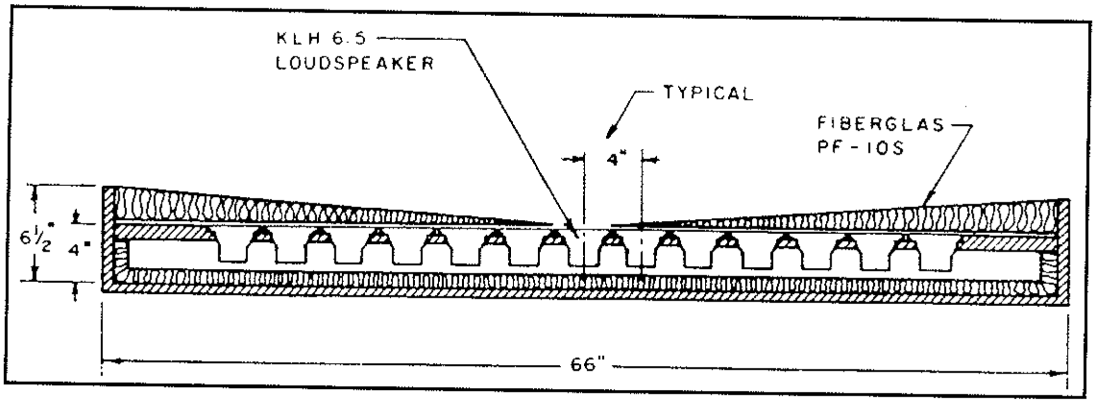


FIGURE 4-26. Mechanical tapering of column loudspeakers by use of fiberglass wedges. Drawing courtesy of Audio Engineering Society.

The physical angling of the drivers can take two forms. The first uses the increasing **Q** of individual drivers at higher frequencies to reduce the line array length by angling the individual drivers in the horizontal array plane. The second technique, shown in Figure 4-27, developed by the late Paul S. Veneklasen, angles the drivers in the horizontal array plane, but uses unequal vertical spacing between the drivers. This latter design results in better directional control from the column loudspeaker. The column length controls the low-frequencies, the curvature controls the mid-range frequencies, and the axes of the splayed units controls the high-frequencies.

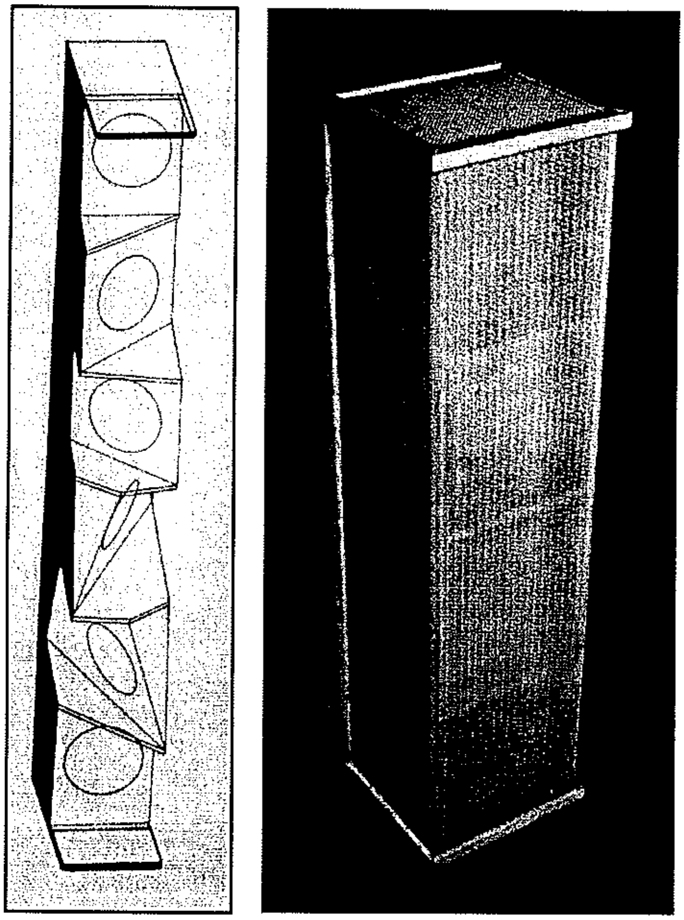


FIGURE 4-27. Curved line source array loudspeaker schematic (left) and photograph (right) designed by the late Paul S. Veneklasen. Note grille cloth on multiple sides of loudspeaker. Drawing courtesy of Paul S. Veneklasen Foundation and photograph courtesy of Michael Kreiser Photography.

Frequency tapering can be achieved by using full-range drivers which are bandpass filtered to reduce the high-frequencies to the outer drivers, with the center drivers receiving the full high-frequency input. Alternately, inductance coils can be wired in series to selected outer drivers to restrict the high-frequencies to the center drivers. The third technique is to design the column loudspeaker as a two-way device with a high-frequency center section and low-frequency outer section, as shown in Figure 4-27. These techniques attempt to make the column loudspeaker acoustically "short" at high-frequencies and "long" at low-frequencies.

Applying progressive signal delay to individual drivers can be used to improve directivity characteristics and to "steer" the loudspeaker acoustic energy beam towards a desired location.

4.3.3.2 Digital Directivity Controlled Column Loudspeakers (Type 3B System)

The latest development with column loudspeakers utilizes DSP signal processing to augment and improve upon the natural acoustical properties of these devices. The Type 3B system can incorporate the DSP signal processing and power amplifiers within the loudspeaker enclosure, or alternately, separate DSP signal processing and power amplifier components can be installed in an equipment rack. Integrating the electronics within the loudspeaker enclosure can simplify some installations and reduce costs compared to using separate components.

The DSP signal processing provides a means of adjusting the loudspeaker horizontal and vertical coverage angles through sophisticated "beam steering" using a combination of electrical tapering, frequency response shaping and subdivision, and signal delay to the separate drivers similar to that described above. The signal processing functions occur in real-time and dynamically change with the signal to maintain the desired directional coverage.

One such beam steering loudspeaker is the AXYS® Intellivox system developed by Duran Audio of the Netherlands. One model uses an asymmetrical non-uniform layout of sixteen to thirty-two 4 in diameter drivers. The exact number of drivers depends on the desired coverage angle and system low-frequency cutoff. The loudspeaker electronics vary the acoustical length of the array. Electrical filtering of the individual drivers continuously changes as a function of the frequency. At lower frequencies more drivers are active, due to the longer wavelengths. At higher frequencies, fewer drivers are active, due to the shorter wavelengths. Each driver has a low-pass constant group delay filter and an adjustable delay. This permits the individual drivers to converge at a given location or to achieve a specified coverage angle. The individual drivers each have separate power amplifiers.

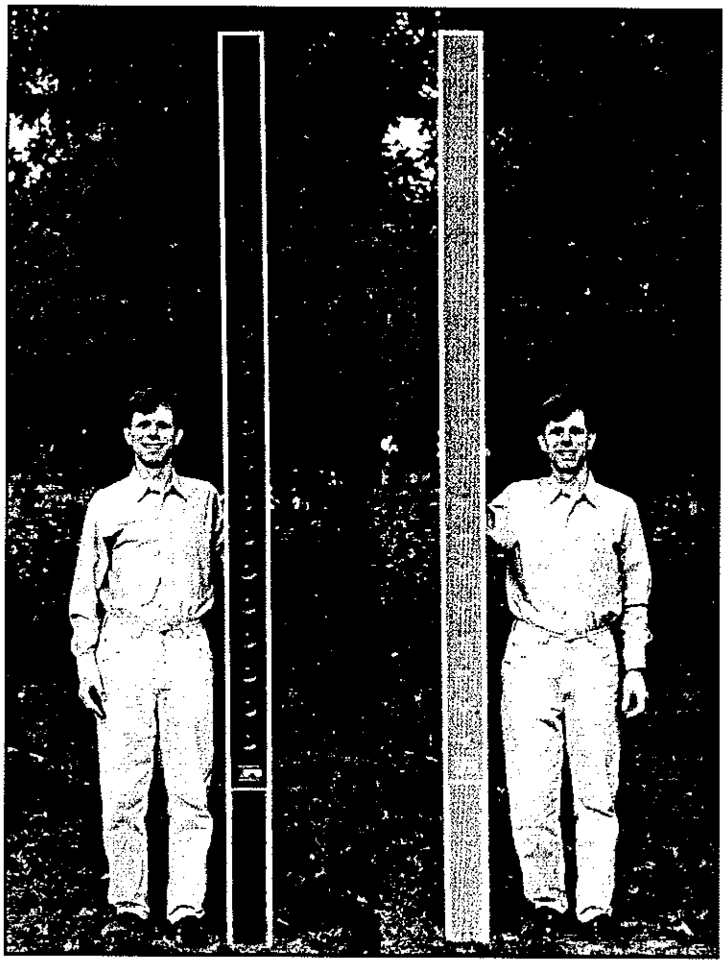


FIGURE 4-28. Type 3B digital directivity controlled column loudspeaker with 16 four in full-range drivers in asymmetrical spacing pattern (AXYS® Intellivox Model 2C) with and without grille cover. Product courtesy of Duran Audio and Performance Devices.

The most commonly installed AXYS® Intellivox loudspeaker is the Model 2C system which uses sixteen 4 in drivers arranged in an asymmetrical vertical array with integral DSP-controlled signal processing and power amplifiers built into the loudspeaker enclosure bottom. The system uses the RS-485 communication protocol for remote monitoring and control of frequency equalization, delay settings, volume and gain configurations. A temperature humidity sensor is also available. The nominal frequency response is 130 to 10,000 Hz with a continuous sound level output of 90 dB at 100 ft. The system has a nominal coverage angle of 140° horizontal and an adjustable vertical coverage angle between. Figure 4-28 shows a digital directivity controlled loudspeaker and Figure 4-29 shows an installation example of the Type 3B system.

Advantages with digital directivity controlled column loudspeakers include: (1) the loudspeaker can be installed flush to wall and no physical aiming is required; (2) the directivity and coverage angle can be changed *in-situ*; (3) the backward-radiated acoustical energy can be steered in the same direction as the front beam which increases the sound level at the listener's location; and (4) these systems can be electrically adjusted to have more than one main lobe, which is useful for covering the main floor and balcony seating areas.

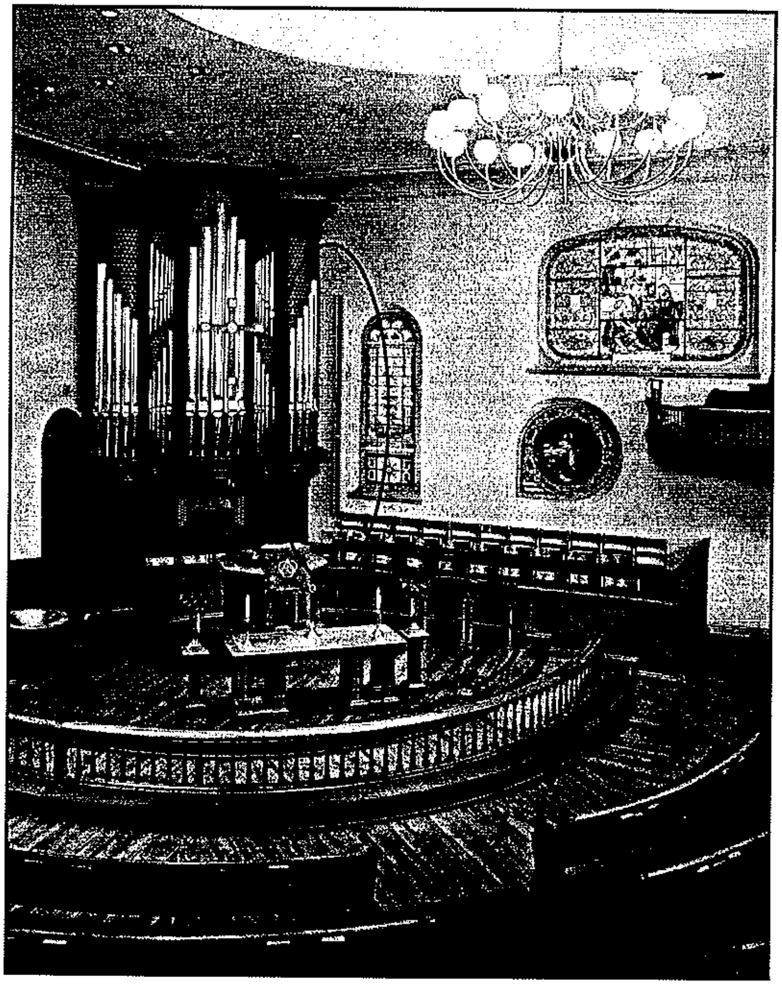


FIGURE 4-29. Type 3B digital directivity controlled column loudspeaker AXYS® Intellivox Type 7-SYM installed at the West Market Street United Methodist Church in Greensboro, NC. Note flush mounting of the loudspeaker adjacent to the Dobson Pipe Organ Builders, Inc. organ case. This loudspeaker covers the entire sanctuary using one lobe projected on the main floor and a second lobe projected to the upper balcony. The symmetrical driver arrangement of the Model 7-SYM produces less off-axis lobing than an asymmetrical driver array. Sound system design by Dan Clayton and room acoustics design Jerry Marshall of Marshall/KMK. Photo courtesy of Dobson Pipe Organ Builders, Inc.

One concern with digital directivity controlled column loudspeakers is their narrow vertical coverage angle can miss the target audience seating area. This can happen if the room temperature and humidity drastically change from the temperature and humidity conditions which were present when the system signal delay lines were initially calibrated. The temperature and humidity variation results in a change in the characteristic speed of sound in the room which changes the relative delay offset between individual drivers. This can be overcome by installing a room temperature and humidity sensor which automatically readjusts the delay settings to individual drivers to maintain the calibrated beam steering coverage angle.

4.3.3.3 Other Line Sources (Type 3C and 3D Systems)

Two other line sources available to the designer include Bessel arrays (Type 3C system) and the horizontal line source array (Type 3D system). These arrays are used less frequently due to their unique operating characteristics and can provide good electro-acoustical performance when correctly applied.

4.3.3.3.1 Bessel Arrays (Type 3C System)

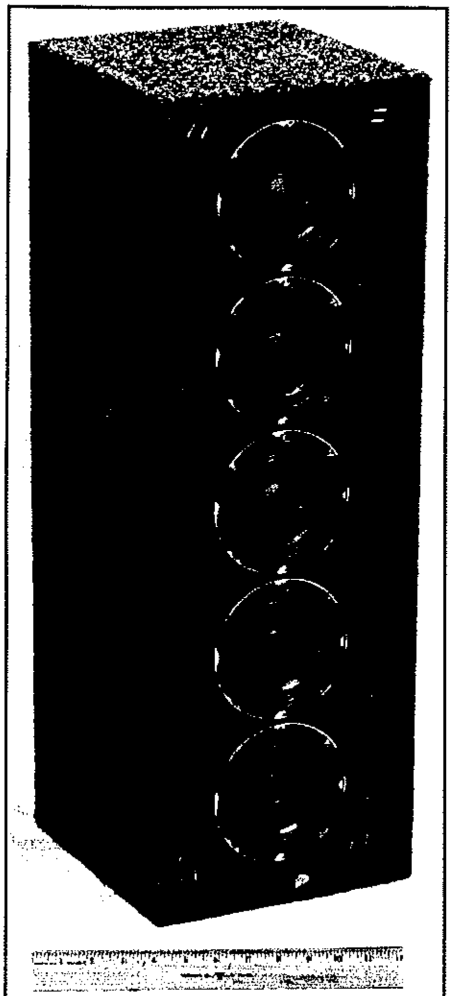


FIGURE 4-30. Type 3C Bessel array column loudspeaker with five 4 in full-range drivers in symmetrical spacing pattern (J.W. Davis column Bessel array) without grille cover. Product courtesy J.W. Davis Company.

Bessel arrays (Type 3C system) comprise five, seven, or nine identical small full-range cone drivers vertically aligned and equally spaced in a narrow enclosure. From the exterior a Bessel array loudspeaker is identical in appearance to a standard column loudspeaker. In contrast to a standard column loudspeaker they produce polar patterns similar to that of a single driver in the array, typically characterized by a radial sound pattern having broad horizontal and vertical coverage angles. Bessel array loudspeakers are easy to construct but use special poling and level control so the drive signal input to each driver has a unique "weighting factor." These systems were developed by the Philips Company in the Netherlands in 1983 and are covered by international patent rights. Figure 4-30 shows a Bessel array loudspeaker.

The drive levels (Bessel coefficients) applied to each individual driver randomizes the polarity of the drivers. Some drivers are connected in reverse polarity which reduces the entire loudspeaker efficiency compared to a standard column loudspeaker. The Bessel coefficients extend the directional operating bandwidth into the region where the array behaves as if it were many wavelengths long, compared to its physical size. This acts to improve the loudspeaker pattern control.

The performance and high-frequency limit of Bessel arrays improves at greater distances from the array. Thus they perform best as far-field devices. Bessel arrays do not have a sharply defined *near-field*/far-field behavior as does a standard line source array. In the near-field, the performance characteristics of a standard line source will appreciably change with distance from the array. In contrast, the far-field performance characteristics of a standard line source will remain fairly constant with distance from the line array. The advantages of a Bessel array are realized when the listener is positioned away from the loudspeaker a minimum of 20 times the array length. One problem with the Bessel array is the phase versus direction and the phase versus frequency characteristics at off-axis locations are poor and exhibit non-linear behavior. This makes these systems difficult to match with other loudspeaker types.

The five-element Bessel array offers better performance than seven- or nine-element Bessel arrays. The five element Bessel array provides superior performance to standard five element column loudspeakers. Modeling of Bessel arrays by Don Keele notes performance parameter differences between a five-element Bessel array and a standard five-element column array as described below.

1. **Efficiency:** Column loudspeaker times that of Bessel array.

2. **Input Power Handling:** Column loudspeaker 1.4 times that of Bessel array.

3. **Maximum Operating Frequency:** Bessel array 28 times that of column loudspeaker.

4. **Efficiency Bandwidth Product:** Bessel array 6 times that of column loudspeaker.

5. **Power Bandwidth Product:** Bessel array 4.4 times that of column loudspeaker.

Thus, the Bessel array sacrifices efficiency and power input to provide a higher upper frequency operating limit with more efficiency and greater power handling capabilities across a wider frequency bandwidth.

4.3.3.3.2 Horizontal Line Source (Type 3D System)

A horizontal line source (Type 3D system) is an array of 4 or 8 in diameter full-range cone drivers closely spaced in a horizontal linear arrangement. One application of a horizontal line source is the "infinite" line source array in which the drivers span across the entire front of the room. Reflected "phantom" sound images are created from the side walls which increases the apparent acoustical array length in a manner similar to two parallel mirrors successively reflecting their images. When this occurs the horizontal line source acts as an infinitely long source. Direct sound attenuation from the infinite line source is 3 dB per doubling of distance compared to a point source having 6 dB direct sound attenuation per doubling of distance. Infinite line source systems provide wide horizontal and vertical coverage. Two disadvantages with these systems include poor source localization and restricted frequency range. Source localization occurs at the drivers closest to the listener, which may not correspond to the actual source location. These systems are restricted to voice amplification application due to the small diameter drivers which are used. A fine example of an infinite line source array has been installed in the Friedburg Concert Hall at the Peabody Conservatory of Music in Baltimore, MD. It was designed in order to satisfy historic preservation requirements because the originally proposed central cluster system was rejected as being too intrusive. Figure 4-31 shows an installation example of the Type 3D system.

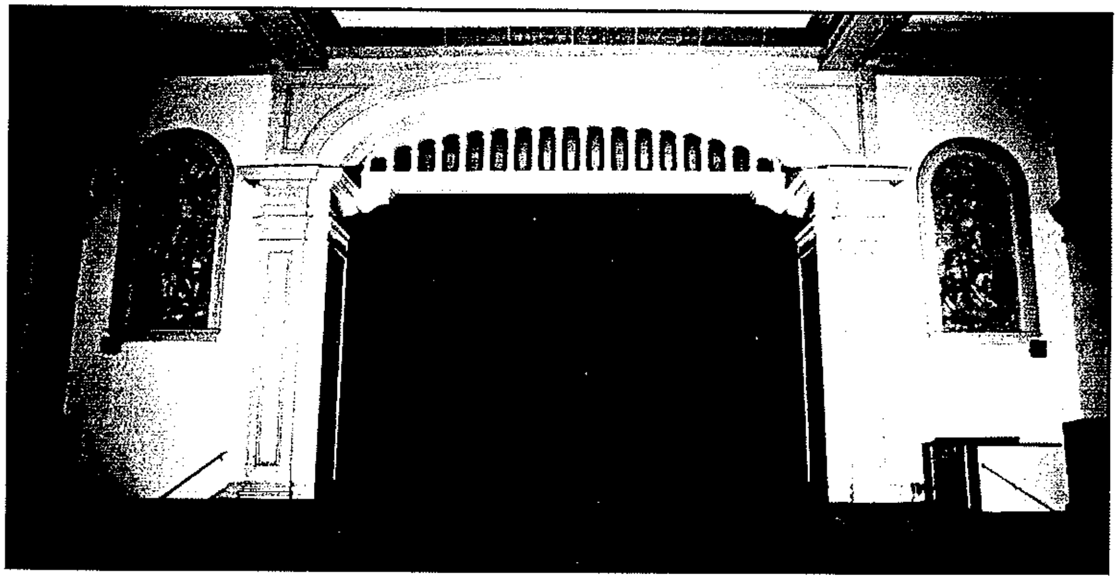


FIGURE 4-31. Type 3D horizontal line source array installed in the Friedburg Concert Hall at the Peabody Conservatory of Music Concert Hall in Baltimore, MD. This system uses 40 eight in diameter JBL LE8T-H drivers within an enclosure above the stage proscenium and extending between the two side walls. Note the location of the loudspeaker enclosure between the cornice molding and the ceiling. Sennheiser® infrared emitter panels for ALS are below stage-left and stage-right bas-relief sculptures. Sound system design by David Lloyd Klepper of Klepper Marshall King Associates, Ltd.

Another horizontal line source application is the stage front to provide sound coverage for the first seat rows. Often these seats are not adequately covered by a central cluster array. Depending on the stage geometry, the drivers may not extend between the two side walls to create an infinite line source array. When the stage is less wide than the room a finite length array results. This can result in narrowing of the coverage pattern and lobing due the phase cancellation between the drivers at the end and the array center. Listeners at the center of the array will experience more infinite line behavior than listeners towards the array end because of the diminished level and cancellation from the end drivers at the array center.

4.3.3.4 Advantages of Line Source Systems

Line sources, when they are correctly applied, have several advantages over other loudspeaker systems. The wide variety of line source types gives the designer flexibility when selecting these devices. Improvements in speech intelligibility and increasing the **D/R** ratio result when the line source is positioned close to the listener as is commonly done in large house of worship installations. Excellent control of vertical coverage can be achieved with one of the tapering line source systems or by using the digital directivity controlled systems. When long horizontal line sources are used, the direct sound attenuation can be less than that of a point source. The narrow dimensional profile of line source systems can be an advantage in some installations and permit the column loudspeaker to be recessed into the room architecture. From a subjective standpoint, a well designed column loudspeaker may sound more natural and lifelike to some listeners than certain horn-type loudspeakers.

4.3.3.5 Design Objectives of Line Source Systems

Design objectives of line sources should first consider the frequency range to be reproduced as this will determine the overall array length, the spacing between the individual drivers, and the individual driver size. Secondly, the requirements for vertical and horizontal coverage angles needs to be determined since there are a variety of line sources, each with characteristic coverage patterns. The number of line sources and the location needs to be coordinated with the audience seating and overall room geometry. Finally, these systems are often used in conjunction with specific signal processing equipment such as bandpass filters and signal delay lines.

4.3.4 Multi-Channel Systems (Type 4A, 4B, and 4C Systems)

Most central cluster sound systems are operated in a single-channel monophonic mode. These systems are adequate for speech and certain music programs which do not require reproduction of frequencies below 80 Hz. Often, complex music and drama programs suffer from poor spatial realism, inadequate frequency response, low dynamic range, and compromised speech intelligibility when played over monophonic sound systems. Multi-channel systems provide improvements in these and other performance parameters which result in superior sound fidelity. The major types of multi-channel systems include stereo two-channel (Type 4A system), three-channel (Type 4B system), and five-channel (Type 4C system).

The major disadvantages with multi-channel systems are increased costs, due to a larger number of loudspeakers and power amplifiers, and difficulty with integrating the loudspeakers into the room architecture. Like monophonic central cluster systems, multi-channel systems must be designed to provide the listener with line-of-sight conditions to all loudspeakers. Each loudspeaker must be selected to cover the entire audience seating area. The nominal **Q** value for horns used in multi-channel systems varies between 10 and 35.

(See Technical Notes, Section 4.F, at the end of this chapter, for a brief history of multi-channel sound.)

4.3.4.1 Stereophonic Systems (Type 4A System)

Stereophonic systems (Type 4A system) operate using two channels. Loudspeakers are positioned at the far left and right locations, often placed directly on the stage floor or elevated mid-way between the stage floor and ceiling. Visual obstruction to the stage is often not a problem with these locations and the loudspeaker systems can be large to provide extended low-frequency reproduction.

The reduced elevation has the advantage of improving localization since the loudspeakers are closer to the source height. In order for the stereophonic effect to work, the loudspeakers must provide a maximum overlap of both left and right loudspeaker coverage patterns. This can cause comb filtering for many seating locations, assuming the same signal is routed to each loudspeaker. Routing of dissimilar signals to each loudspeaker will minimize comb filtering.

4.3.4.2 Three-Channel Systems (Type 4B System)

Three-channel systems (Type 4B system) are the most common multi-channel format. Three separate loudspeaker systems are used for the left, center, and right channels. The left and right channels are used for music and special effects and the center channel is used for speech or a musical soloist. Three-channel systems were pioneered by Bell Telephone Laboratories and the motion picture industry in the 1930s.

With the advent of true three-channel sound mixing consoles, this format is becoming more common in high-end sound reinforcement systems. The primary advantages with a three-channel system are the creation of a strong center image, improved localization, and greater speech intelligibility. Secondary advantages include reduced loudspeaker sizes, enhanced spaciousness, and improved sound quality. The general design objectives are to select loudspeakers which can each cover the entire audience seating area, have the same frequency response and tonality characteristics, and provide the same sound level output.

Routing the speech signal to the center loudspeaker promotes speech intelligibility because fewer loudspeakers are used to carry the speech signal. In essence the center channel operates in a monophonic mode. This increases the **D/R** ratio at the listener and eliminates speech signal comb filtering since only one loudspeaker location is used. Because there is little low-frequency energy in speech below 100 Hz, the center channel loudspeaker can be rolled-off starting at this frequency, or a smaller loudspeaker with reduced low-frequency response can be selected. Frequency equalization can be applied to the center channel loudspeaker which will result in improved speech intelligibility and clarity without affecting the tonal balance of the left and right music loudspeakers.

Improved localization results from loudspeakers at the stage right and left sides which are closer to the source location. This can be particularly useful when a large musical cast or ensemble is on stage. The sound system operator also has the ability to *pan* the signal, such as from a moving vocalist, to the left or right loudspeaker. Finally, some of the center channel voice signal can be routed to the left and right loudspeakers, which has the effect of pulling the perceived image downward from the overhead loudspeaker while improving coverage to the front audience seats.

The left and right loudspeakers are operated full-range and usually have subwoofers to extend the low-frequency response below 80 Hz. The constraints of loudspeaker aiming can be relaxed somewhat and the effects of room reverberation can be used to improve the perception of spaciousness. Positioning left and right loudspeakers is typically about 20 ft off the room centerline, with the objective of limiting the time of arrival of the two loudspeaker signals to the listener position to less than 20 ms. Figures 4-32, 4-33, and 4-34 show installation examples of the Type 4B system.

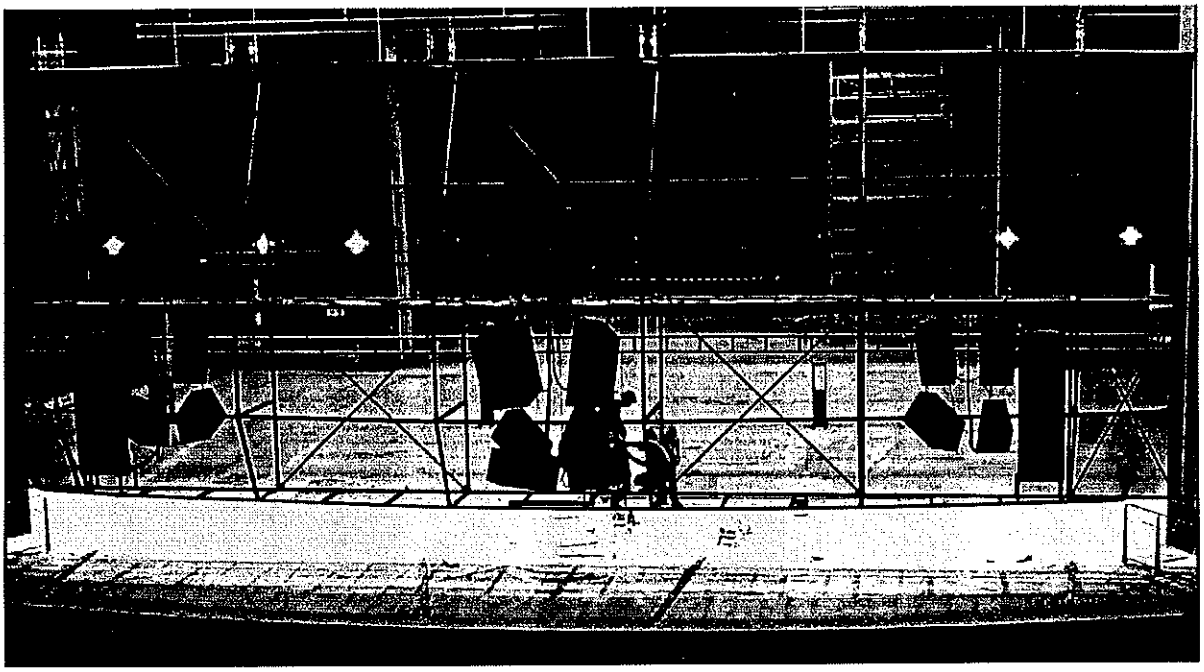


FIGURE 4-32. Type 4B proscenium three-channel system (left, center, and right) covering the front seating area comprising at center two EAW KF853 and four KF650 large format full-range loudspeakers and at left and right one EAW SB840 subwoofer, two EAW DS153, and two EAW SM155 large format full range loudspeakers in the Auditorium de Dijon in Lyon, France. Note the acoustically transparent bridge which houses the loudspeakers and the size of the loudspeakers compared to the stagehands. The bridge is lowered to the stage level for servicing the loudspeakers. System design by Tom Young of ARTEC Consultants, Inc. Photo courtesy of ARTEC Consultants, Inc.

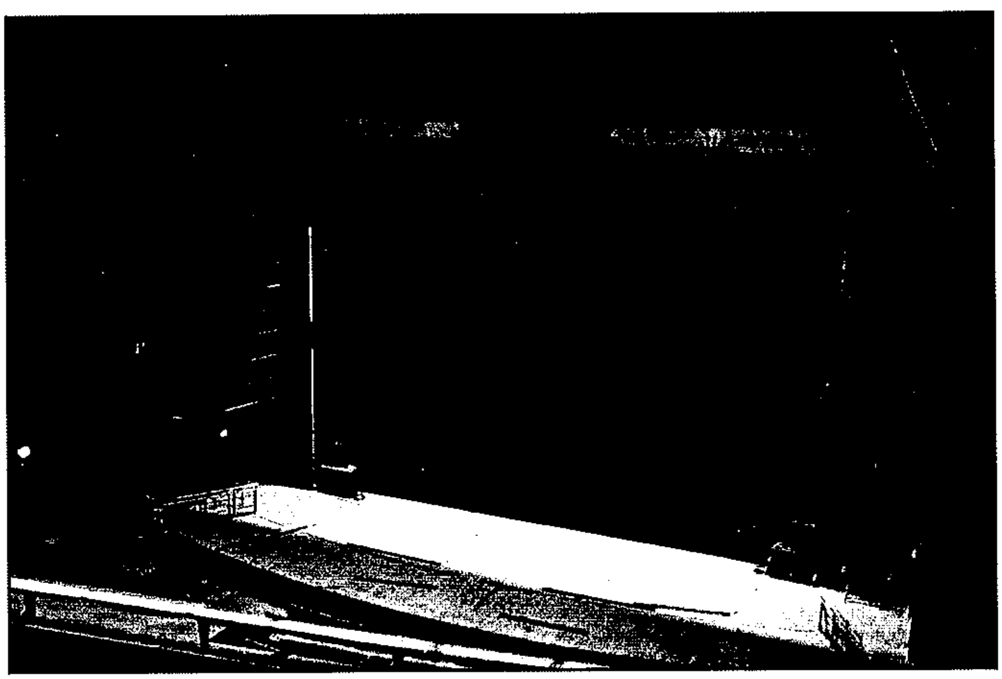


FIGURE 4-33. Same loudspeaker installation as in Figure 4-32 but raised to the normal operating position. Note the portable music loudspeakers at stage left and stage right locations adjacent to the proscenium. System design by Tom Young of ARTEC Consultants, Inc. Photo courtesy of ARTEC Consultants, Inc.

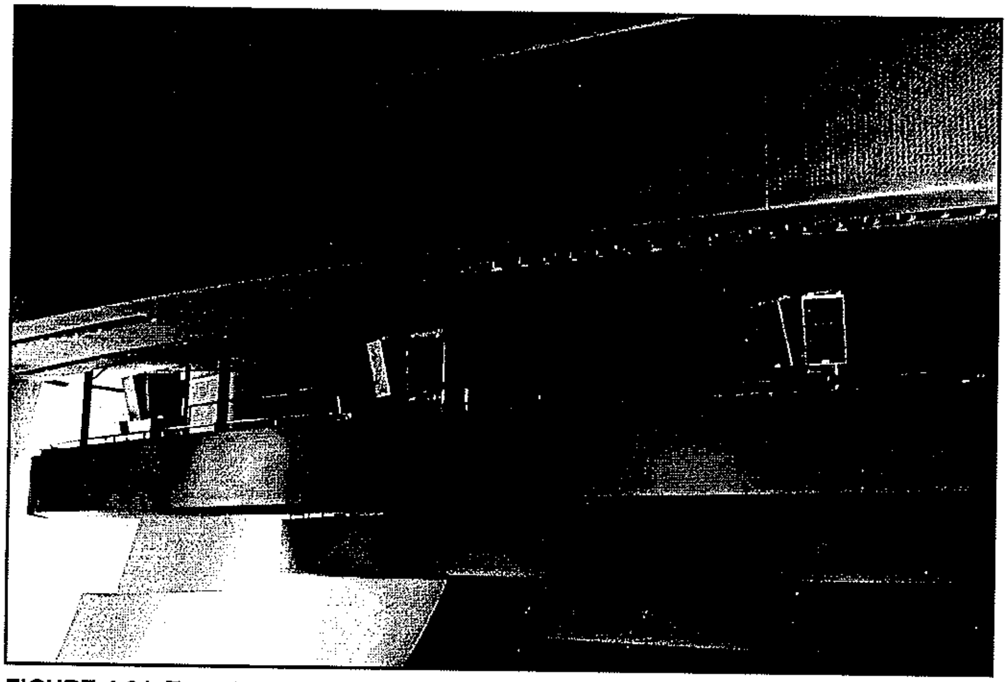


FIGURE 4-34. Type 4B delay three-channel system (left, center, and right) covering the rear seating comprising EAW KF650 large format loudspeakers in the Auditorium de Dijon in Lyon, France. System design by Tom Young of ARTEC Consultants, Inc. Photo courtesy of ARTEC Consultants, Inc.

4.3.4.3 Five-Channel Systems (Type 4C System)

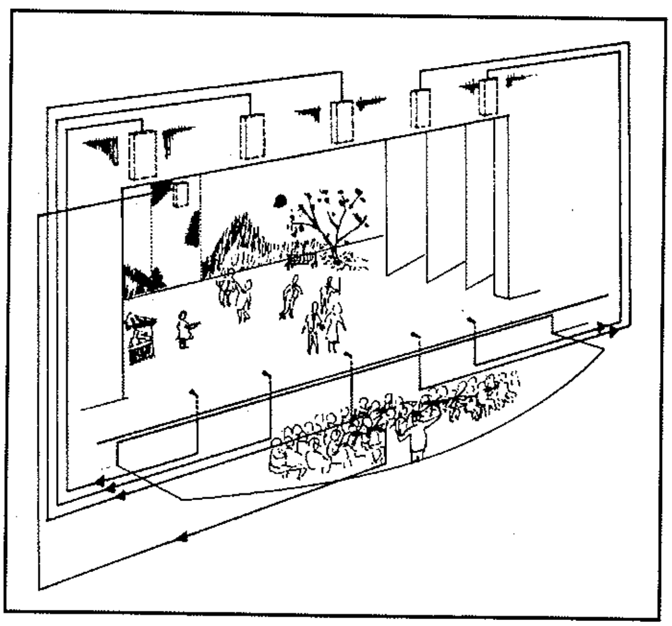


FIGURE 4-35. Operational concept of the Type 4C five-channel system showing the dedicated microphones and loudspeakers for each channel, plus orchestra pit-to-stage sound transfer for soloist performer synchronization with the orchestra by loudspeaker on upstage wall. Drawing courtesy of Paul S. Veneklasen Foundation.

The five-channel system (Type 4C system) was promoted by the late Paul S. Veneklasen and is an extension of the three-channel system. One additional loudspeaker is located on both the left and right sides of the center loudspeaker. The left, left-center, center, right-center, and right loudspeakers are all equally spaced no more than 25 ft apart. The loudspeakers are used in conjunction with floor-recessed microphones on moveable plaques positioned at downstage locations. Each microphone is input to a dedicated mixer channel, amplified, and routed to the proscenium-mounted loudspeaker directly above it. Figure 4-35 shows the operational concept of the Type 4C system.

The intended function of the five-channel system is to provide improved localization with a minimum of operator intervention.

As the performer moves towards a microphone the sound level at the loudspeaker above increases. Since D_S is large compared to more conventional sound reinforcement systems, the gain-before-feedback is limited. These systems are intended for "voice lift" applications where a modest amount of gain (less than 8 dB) is required and the desire is to minimize the listener's perception that the source is being amplified. This impression is further enhanced with the five loudspeakers being recessed into the proscenium so as not to be visible. Figure 4-36 shows an installation example of the Type 4C system.

FIGURE 4-36. Type 4C proscenium five-channel system (left, left-center, center, right-center, and right) comprising multi-cell horns similar to Figure 3-45 and multiple 15 in low-frequency enclosures in the Hancher Auditorium at the University of Iowa in Iowa City, IA. Note the acoustically transparent grille cloth covering the loudspeakers. System design by the late Paul S. Veneklasen. Photo courtesy of Michael Kreiser Photography.

4.3.4.4 Advantages of Multi-Channel Systems

The primary advantages of multi-channel systems are improved source localization and the ability to separate the talker or solo performer from the other sound sources, by panning the voice to the center channel. Highlighting this individual can improve the auditory experience for the listener. Secondary advantages include wider frequency extension, particularly at low-frequencies, greater sound levels with less distortion, and potentially smaller loudspeakers than conventional monophonic systems.

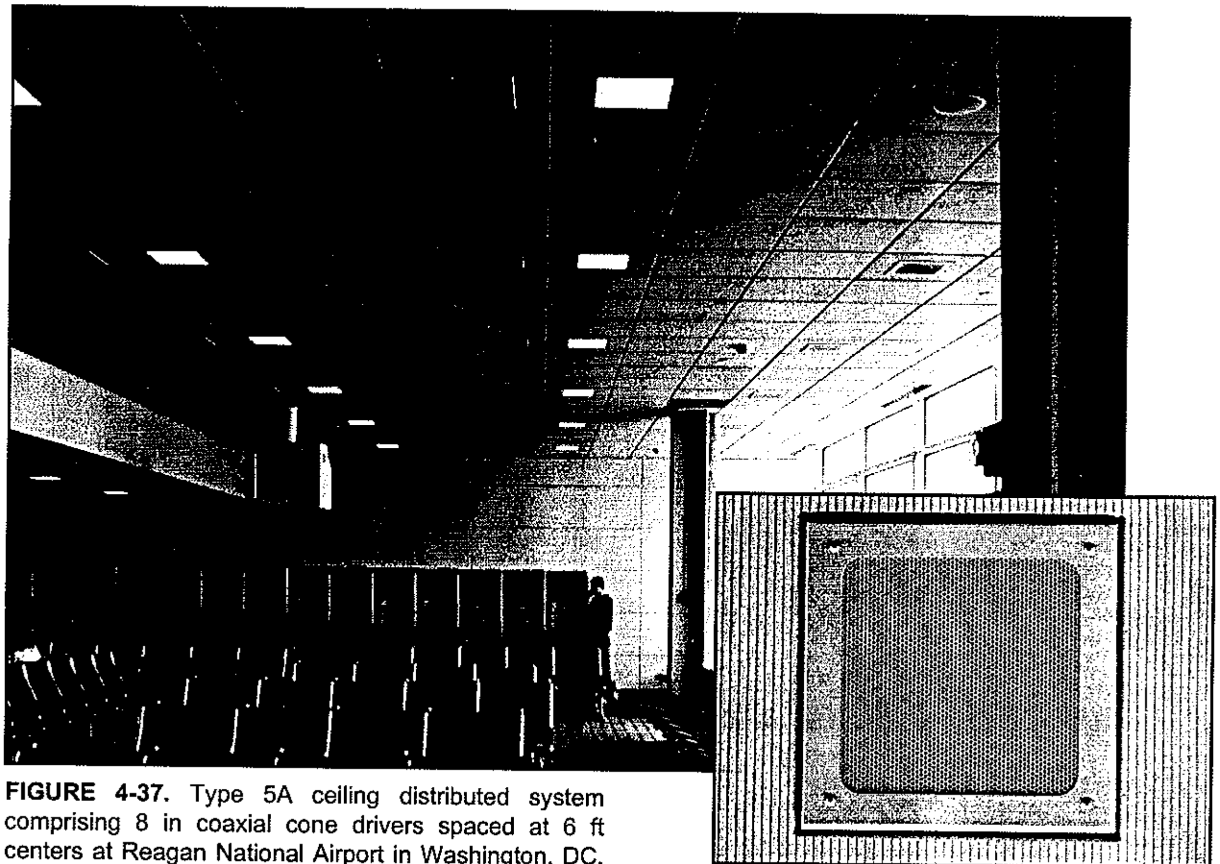


FIGURE 4-37. Type 5A ceiling distributed system comprising 8 in coaxial cone drivers spaced at 6 ft centers at Reagan National Airport in Washington, DC. System design by Bob Ledo of Coffeen Fricke Associates.

FIGURE 4-38. Close-up of the loudspeaker installation in Figure 4-37.

4.3.4.5 Design Objectives of Multi-Channel Systems

The design objectives of multi-channel systems are essentially similar to central cluster systems. Indeed, multi-channel systems can be considered to be multiple monophonic sources with respect to the design objectives of even sound level coverage in the audience seating area, uniform sound levels, and extended frequency response.

4.3.5 Ceiling Distributed Systems (Type 5A and 5B Systems)

A ceiling distributed system comprises a group of loudspeakers installed at the ceiling plane which are operated at a low volume level and aimed at the audience below. The simplest ceiling distributed system uses small full-range cone drivers (Type 5A system). Less common is a system which uses small directional horn drivers (Type 5B system). Most ceiling distributed systems use transformer-coupled loudspeakers connected to a constant 70.7 V power amplifier. The transformer taps permit adjusting the power delivered to the driver to set the sound level output. Ceiling distributed systems are intended primarily for voice reinforcement and paging applications, but with high quality transformers and larger diameter drivers, can be suitable for music reinforcement or reproduction. The loudspeaker closest to the listener will convey the necessary speech intelligibility, but the remaining loudspeakers decrease the **D/R** ratio at the listener which diminishes speech intelligibility, particularly in reverberant rooms.

The number and type of loudspeakers needed depends on the audience size, ceiling height, loudspeaker coverage angle, desired overlap in loudspeaker coverage, and the frequency range to be reproduced. Type 5A loudspeakers vary between 4, 6½, 8, 12, and 15 in diameter sizes, with 8 in diameter coaxial devices being the most common. The smaller diameter drivers (4, 6½, and 8 in) are appropriate for speech applications or use in small rooms. The larger diameter drivers (12 and 15 in) are used in rooms with high ceilings, for music reproduction, or where greater sound levels are required. Small-to-medium format directional horns with coverage angles between 40° by 40° and 90° by 60° are used for the Type 5B system. The nominal **Q** value for drivers used in ceiling distributed systems varies between 4 and 16 for cone drivers and between 8 and 25 for horn drivers.

The Type 5A system performs best when the ceiling and floor surfaces are acoustically absorptive, the ceiling height is no more than 20 ft, and the room reverberation time is less than 1.4 s. Rooms with higher ceilings and longer reverberation times are suitable for the Type 5B system, however the loudspeakers should not be installed more than 45 ft above the floor. Figures 4-37 and 4-38 show an installation example of the Type 5A system.

4.3.5.1 High Impedance/Constant Voltage Systems

Most ceiling distributed systems are operated in a high impedance/constant voltage power distribution configuration. This provides higher voltage and lower current to the loudspeaker line than conventional low impedance loudspeaker distribution.

The constant voltage concept was developed in the early 20th century by the electrical power industry as a means to minimize energy loss in cables extending long distances. The constant voltage technique makes use of "step-up" and "step-down" transformers. At the power station, a step-up transformer raises the voltage and decreases the current which reduces the energy loss in the cable. At the receiving end a step-down transformer reduces the voltage and increases the current to values which can be used with standard appliances. This concept is the basis of constant voltage loudspeaker systems. The step-up transformer is located at the power amplifier, which increases the voltage to a nominal value, usually 70.7V, but other values such as 25 and 100 V are possible. A step-down transformer is located at each driver which matches the line voltage to the driver impedance. The loudspeaker step-down transformer primary is paralleled across the secondary of the power amplifier step-up transformer. Figure 4-39 illustrates the constant voltage concept.

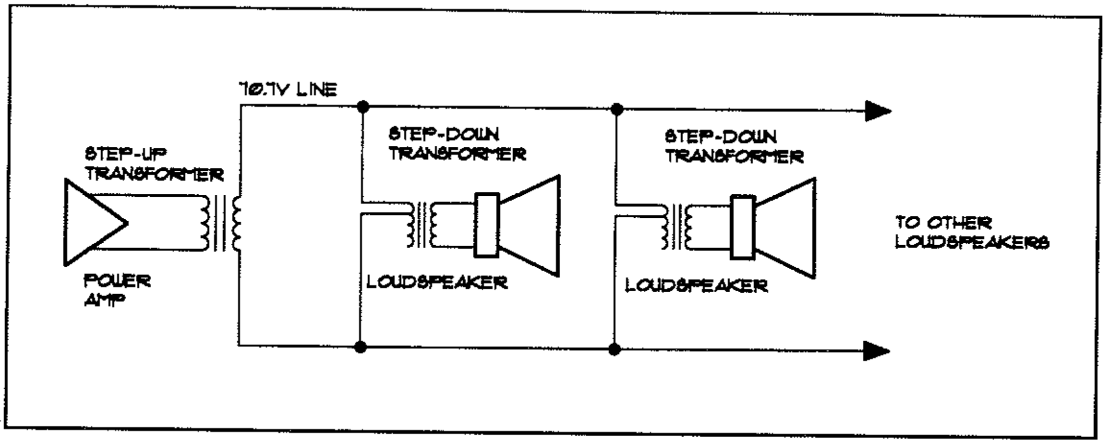


FIGURE 4-39. Typical constant voltage system showing step-up transformer at power amplifier, 70.7 V distribution line, parallel connection of loudspeakers across loudspeaker line, and step-down transformer at each loudspeaker.

High impedance loudspeaker power distribution systems offer advantages over conventional low impedance power distribution systems. Some advantages include the following:

1. Many local Codes do not require the loudspeaker cable to be installed in conduit which can result in significant installation cost savings. However, special fire-resistant loudspeaker cable may be required by some Codes.

2. Failure of one or more drivers or transformers will not result in failure of the entire loudspeaker distribution system. Loss of sound occurs locally where drivers or transformers have failed. All other loudspeakers will continue operating.

3. Loudspeakers can be adjusted in volume level by setting their transformer taps. This can provide different volume levels in separate areas while all loudspeakers are connected to the same power amplifier.

4. Simplified installation results because loudspeakers are wired in parallel across the amplifier. Adding loudspeakers does not pose a problem as long as the sum of the loudspeaker transformer taps power rating does not exceed the power amplifier output rating, with 80 percent of the power amplifier output rating providing a margin of safety.

5. Higher operating voltage and lower current in the loudspeaker cable reduces cable losses and permits smaller gauge less costly cable to the used.

6. Cable resistance is less which allows for longer cable runs with minimal power loss.

4.3.5.2 Loudspeaker Transformers

The input to each loudspeaker in a constant voltage ceiling distributed system is fitted with a line-matching transformer connected in parallel across the loudspeaker cable, similar to electrical power outlets connected across a 120 VAC power line.

Two line-matching transformer configurations are normally used. The first type permits different line level voltage sources to be connected across the primary with the loudspeaker power adjusted at the power taps of the secondary. The second type is intended for a constant line level voltage source with the loudspeaker power adjusted at the power taps of the primary and the loudspeaker impedance matched at the secondary. Larger tap settings for both transformer types will result in higher sound levels. Figure 4-40 illustrates both loudspeaker transformer types and Figure 4-41 shows different transformers.

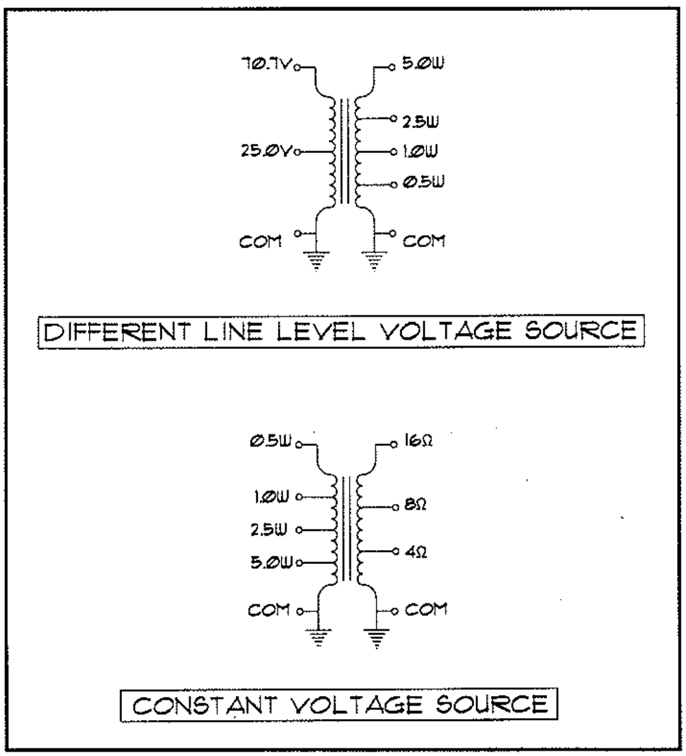


FIGURE 4-40. Typical loudspeaker line matching transformers. The different line level voltage source type (top) has primary tap settings of 25.0 and 70.7 V plus common and secondary tap settings of 0.5, 1.0, 2.5, and 5.0 W plus common. The constant voltage type (bottom) has primary tap settings of 0.5, 1.0, 2.5, and 5.0 W plus common and secondary tap settings of 4, 8, and 16Ω plus common.

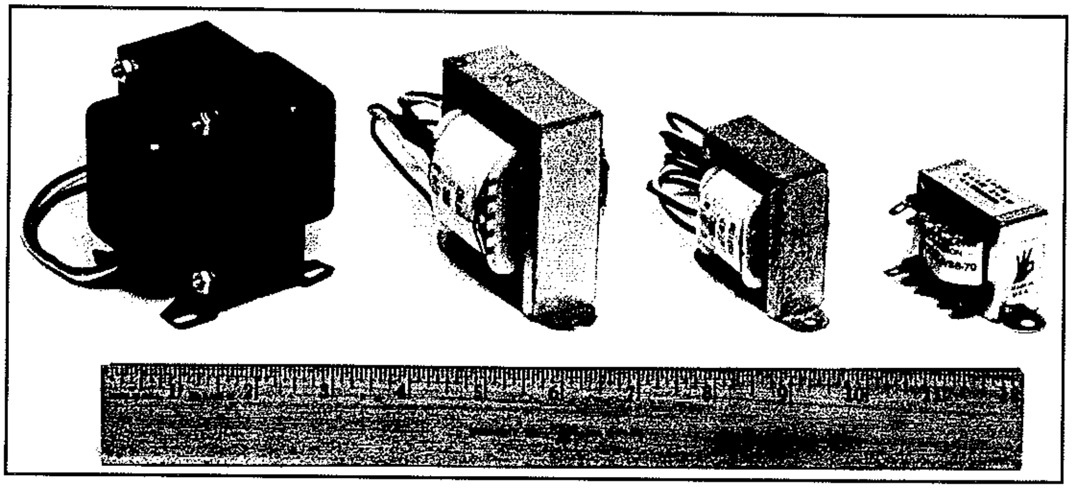


FIGURE 4-41. 70.7 constant voltage loudspeaker line matching transformers 8 to 32 W primary taps (Soundolier HT327 left), 4 to 16 W primary taps (Soundolier HT167 left-center), 1 to 8 W primary taps (Soundolier HT87 right-center) and 2 to 8 W primary taps (EDCOR EM4065 right). Products courtesy of Atlas/Soundolier, Inc. and EDCOR Electronics.

The different line level voltage source transformer is intended for a fixed loudspeaker impedance. The loudspeaker transformer primary has multiple voltage taps to maintain a constant voltage across the loudspeaker. Typical primary tap values are 25.0, 70.7, and/or 100.0 V. The loudspeaker transformer secondary has taps at various power levels to drive the loudspeaker. Typical values 0.25, 0.5, 1.0, and 2.0 W for small transformers, 1.0, 2.0, 4.0, and 8.0 W for medium transformers, and 4.0, 8.0, 16.0, and 32.0 W for large transformers

The constant voltage source transformer for 70.7 V operation has typical input impedances between 5 and 5,000 Ω which corresponds to 1,000 to 1 W for a power amplifier rated at 70.7 V. The loudspeaker transformer primary has multiple power taps to adjust the electrical power delivered to the loudspeaker. Typical primary tap values are 0.25, 0.5, 1.0, and 2.0 W for small transformers, 1.0, 2.0, 4.0, and 8.0 W for medium transformers, and 4.0, 8.0, 16.0, and 32.0 W for large transformers. The loudspeaker transformer secondary has taps at various impedances to match the loudspeaker. Typical values are 4, 8, and 16 Ω.

Loudspeaker transformers have several importance performance parameters which must be satisfied across a given design bandwidth: (1) equal power is drawn at all frequencies; (2) a reasonably flat impedance characteristic is present and the impedance does not drop significantly in specific frequency regions; (3) low overall harmonic distortion; and (4) low *insertion loss*. The first three performance factors are directly interrelated by the physical size of the transformer core. The insertion loss is due primarily to resistance heating which is related to the number of windings on the transformer primary and secondary. Eddy currents around the transformer core are another source of insertion loss.

The performance of inexpensive transformers worsens at low-frequencies. Frequencies below approximately 100 Hz are difficult for many transformers to drive. Transformers will saturate their cores at low-frequencies which results in a decrease in the transformer load impedance causing more current to be drawn from the power amplifier. This may cause the power amplifier to activate its protection circuitry, go into current limiting, or to fail. When current limiting occurs, negative voltage spikes, called flyback, are developed in the transformer and flow back to the power amplifier. The negative voltage spikes result in severe harmonic distortion which degrades the sound quality. To avoid these problems a high-pass filter can be used at the signal mixer output to attenuate frequencies below 100 Hz. If the sound system requires good low-frequency performance the use of wide bandwidth transformers will be necessary, but this will significantly increase the system costs.

The transformer insertion loss will limit the power that can be delivered to the driver. An insertion loss of 3 dB will result in one-half of the amplifier power being delivered to the driver. The other half is essentially lost as heat within the transformer. Typical insertion loss values are 1 dB for medium quality transformers and less than 0.5 dB for high quality transformers.

The power rating of transformers can be ambiguous since two methods are used to establish this parameter. The designer should know which rating method is used by the transformer manufacturer. Otherwise under powering the driver or overdriving the power amplifier can result. The former condition is generally preferred to the latter.

The first rating scheme establishes the power drawn by the transformer primary is the power drawn from the line, not the power delivered to the driver. This method does not compensate for the transformer insertion loss and the transformer will deliver less power to the driver, by an amount equal to the insertion loss. The sound level from the driver will less than the value calculated. The labeled value on the power tap is not the same as that delivered to the driver.

The second rating scheme establishes the power drawn by the transformer primary is the power delivered to the driver. This method compensates for the transformer insertion loss by adding extra windings to the primary and the transformer will draw a little more power than the values indicated on the power taps. This may overload the power amplifier if many transformers are used and the sum of transformer power taps equals the power amplifier rating. The power delivered to the driver is the labeled value on the power tap.

To determine if the transformer is insertion loss compensated the power should be calculated when the driver is connected to a constant voltage source. If the calculated power of the transformer (**P**) is less than the rated power it can be concluded the transformer is not compensated for insertion loss. The value of **P** can be calculated using the following equation:

$$\mathbf{P} = \frac{\mathbf{V}^2}{\mathbf{Z}} \tag{4.17}$$

where,

P is the calculated power drawn by the transformer, watts
V is the constant voltage source, usually 70.7 volts
Z is the total load impedance, ohms

4.3.5.3 Cable Losses

Another loss associated with all sound systems is the resistive loss of the cable ($\mathbf{R_D}$) connecting the power amplifier to the loudspeakers. Larger diameter cable will have less resistance and therefore a smaller signal loss through the cable. The tradeoff is cost since larger cable is more expensive. A rule-of-thumb is to limit loudspeaker cable losses to no more than 0.5 dB. The total cable resistance should be limited to no more than 30 percent of the transformer load and driver resistances. Values of cable resistance per 1000 ft are listed in manufacturer's catalogs. Table 4-2 provides typical values as a function of wire gauge.

The cable resistance for the installation can be estimated by determining the cable run length. This value should be doubled to account for positive and negative sides of the loudspeaker circuit which the cable is connected to. The value of $\mathbf{R_D}$ can be calculated using the following equation:

$$\mathbf{R_D} = \left[\frac{\mathbf{R_C}}{\mathbf{1000}}\right](\mathbf{L}) \tag{4.18}$$

where,

$\mathbf{R_D}$ is the resistance of the cable for a given length, ohms
$\mathbf{R_C}$ is the characteristic resistance of the cable per 1000 ft, ohms
L is the length of the cable from amplifier to loudspeaker which has been doubled for full loudspeaker circuit, ft
1000 is a constant for 1000 ft cable

TABLE 4-2. Loudspeaker Cable Resistance/1000 Ft (Ohms) vs Gauge

Resistance/1000 Ft, Ohms	Gauge, AWG
0.6	8
1.0	10
1.6	12
2.5	14
4.0	16
6.4	18
10.2	20
16.0	22

(See Technical Notes, Section 4.G, at the end of this chapter, for additional information on loudspeaker cable signal level loss.)

4.3.5.4 Loudspeaker Back Boxes

A back box is necessary to provide acoustical loading for the driver to extend the low-frequency response and to improve the efficiency. If no back box is provided, the rearward motion of the driver cone will radiate sound into the ceiling plenum resulting in less sound radiated to the audience below. The reverberant sound from the plenum can enter the room through ceiling openings such as return air grilles and light fixtures or through the acoustical ceiling tile and driver cone. This reverberant sound will degrade the sound quality and speech intelligibility.

Loudspeaker back boxes are available in a variety of sizes based on driver dimensions and the needed low-frequency extension. Most back boxes are of metal construction. Quality back boxes have coated damping material applied to the interior walls to lower resonances and sound radiation through the cabinet. Internal fiberglass lining is used to reduce reflections within the back box from the rearward cone motion. Building Codes prohibit wood back boxes from being installed in ceiling plena due to the material combustibility.

Small loudspeaker back boxes are cylindrical and large loudspeaker back boxes are rectilinear in shape. The smaller back boxes for 4, 6½, and 8 in diameter drivers can usually be mounted directly behind GWB or acoustical ceiling tile surfaces. Back boxes for 12 and 15 in diameter drivers can be quite large and heavy and require attachment to the building structure. Figures 4-42 through 4-46 show back boxes of varying sizes and for different installation applications. Table 4-3 provides recommendations on minimum back box sizes. Some manufacturers provide packaged loudspeaker assemblies complete with the back box integral with the driver, line matching transformer, and grille. Figures 4-47 and 3-54 show examples of these systems.

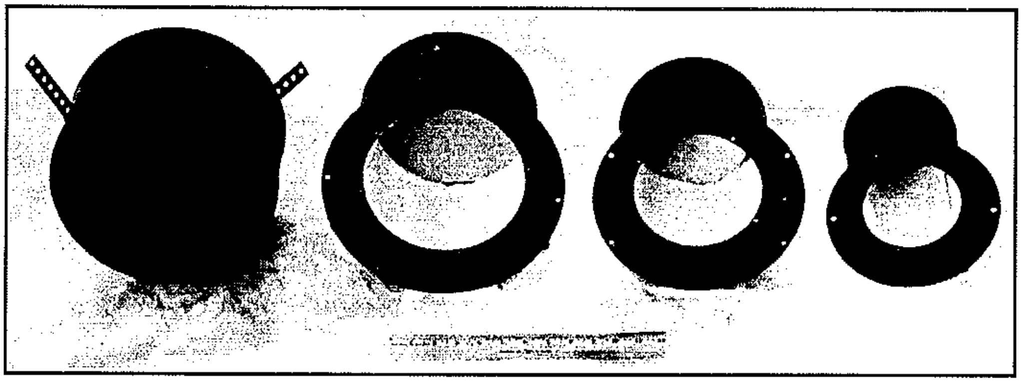


FIGURE 4-42. Loudspeaker back boxes for 4 in diameter drivers with 0.3 ft^3 internal volume (Lowell DX104 left) and 0.1 ft^3 internal volume (Soundolier FA97-4 right), 8 in diameter drivers with 0.25 ft^3 internal volume (Soundolier FA97-8 left-center), and 6½ in diameter drivers with 0.17 ft^3 internal volume (Soundolier FA97-6 right-center). Products courtesy of Atlas/Soundolier, Inc. and Lowell Manufacturing Company.

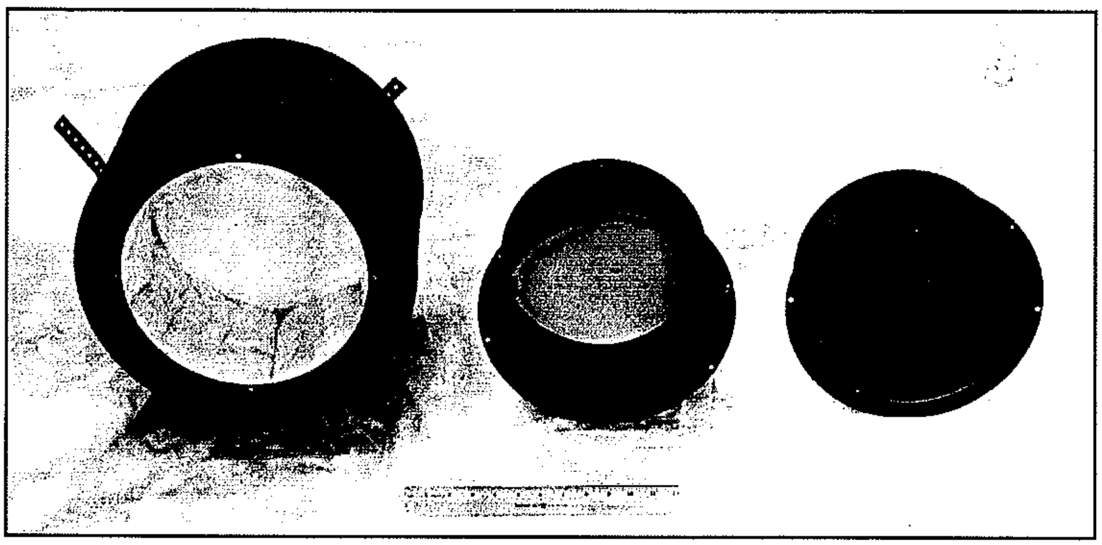


FIGURE 4-43. Loudspeaker back boxes for 8 in diameter drivers with 1.0 ft^3 internal volume (Lowell DX108 left), 0.4 ft^3 internal volume (Lowell XPC187 center), and 0.25 ft^3 internal volume (Lowell XPC85 right). Products courtesy of Lowell Manufacturing Company.

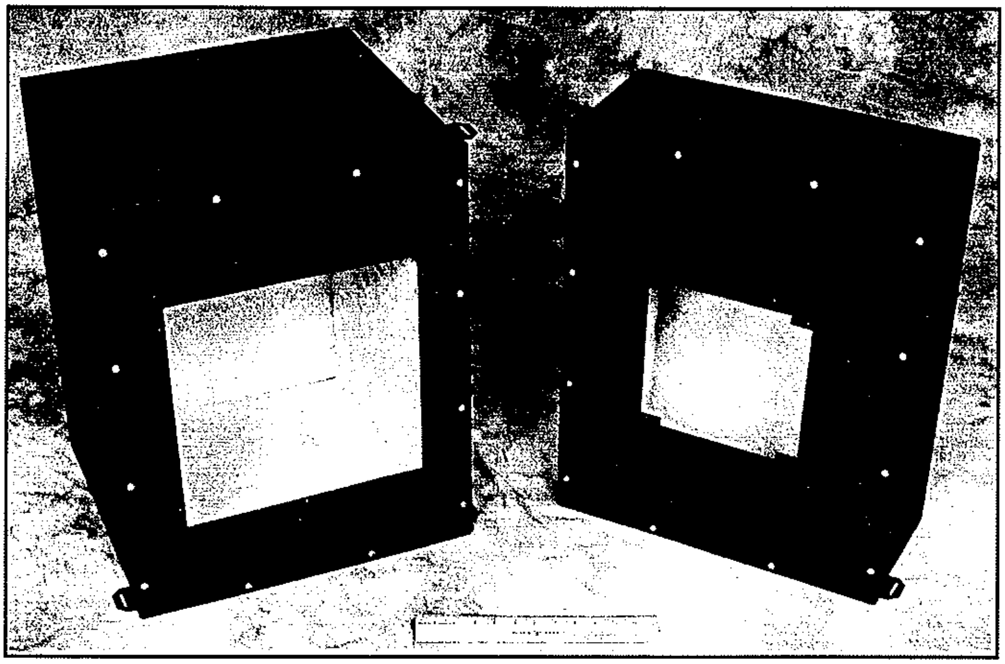


FIGURE 4-44. Loudspeaker back boxes with 4.0 ft^3 internal volume for 12 in diameter drivers (Lowell DX1612 left) and 1.5 ft^3 internal volume for 8 in diameter drivers (Lowell DX118 right). Products courtesy of Lowell Manufacturing Company.

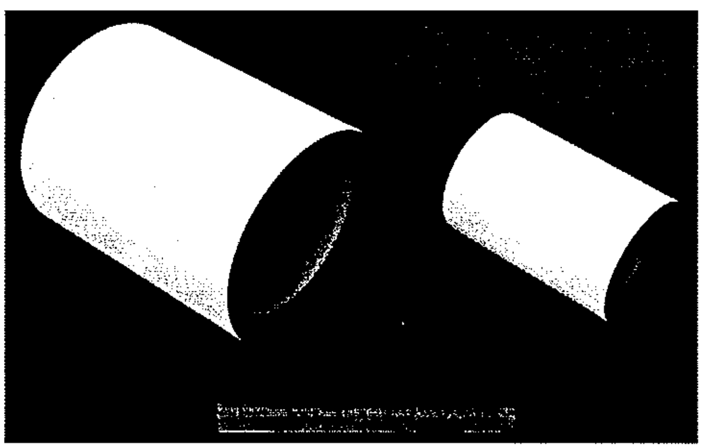


FIGURE 4-45. Pendant hanging loudspeaker back boxes with 0.6 ft^3 internal volume for 8 in diameter drivers (Soundolier X8414W left) and 0.2 ft^3 internal volume for 4 in diameter drivers (Soundolier X44W right). Products courtesy of Atlas/Soundolier, Inc.

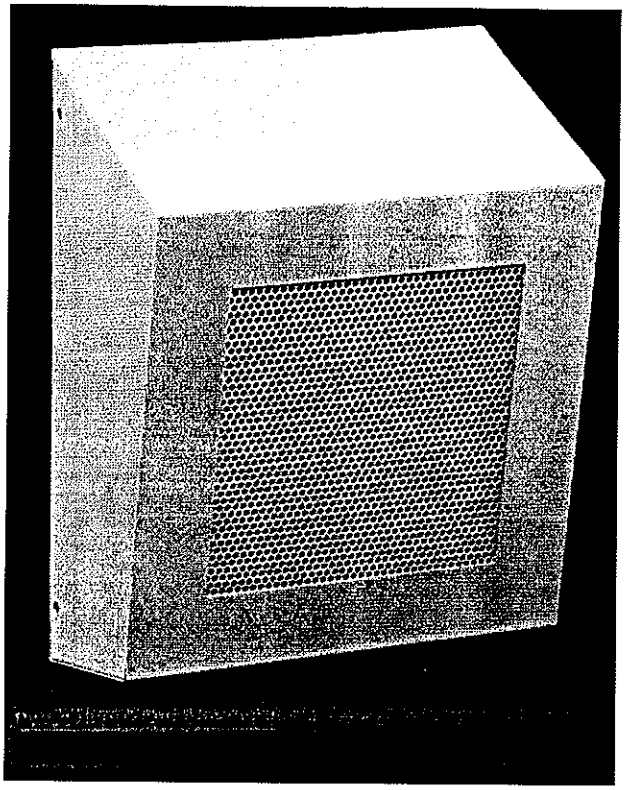


FIGURE 4-46. Wall mounted loudspeaker back box with 0.4 ft^3 internal volume for 8 in diameter driver (Lowell CEK-8M). Product courtesy of Lowell Manufacturing Company.

TABLE 4-3. Driver Size vs Back Box Size for Voice and Music Applications

Driver Size, in	Back Box Size, Ft^3
Low Quality Voice Reproduction or Paging	
4	0.2
8	0.5
Low Quality Music	
4	0.5
8	0.5 to 1.0
High Quality Voice Reproduction	
8	1.0 to 2.0
12	2.0 to 3.0
High Quality Music	
8	2.0 to 3.0
12	3.0 to 4.0
15	4.0 to 6.0

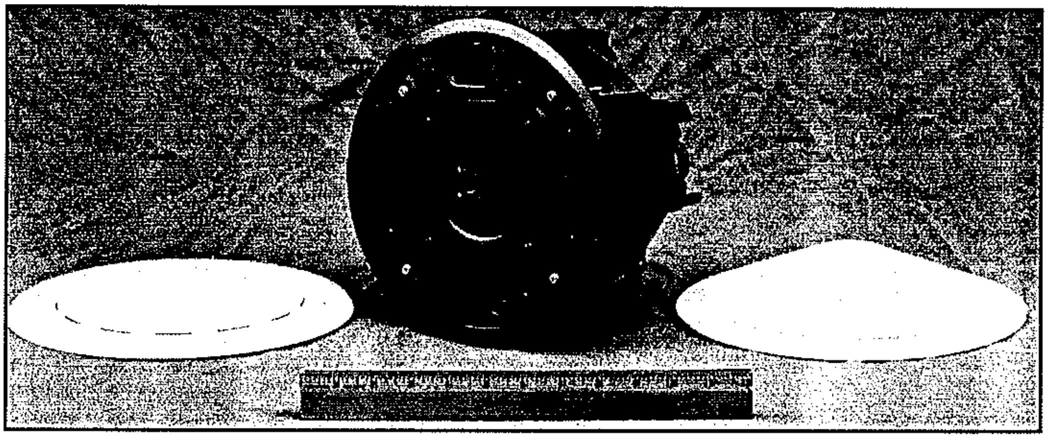


FIGURE 4-47. Packaged loudspeaker assembly (TOA F-121C/F-121CM) complete with tuned ported back box integral with 4.7 in driver, line matching transformer, and grille. Note conical shaped sound diffusing grille on right which enhances high-frequency sound coverage. Standard grille is on left. Products courtesy of TOA Electronics, Inc.

4.3.5.5 Loudspeaker Layout

A standard cone driver aimed directly downward will have a coverage pattern resembling a circular cone with the driver at the apex. Some listeners will be directly below (on-axis) to the driver, while others will be at off-axis locations. As frequency increases, the driver coverage pattern will narrow, resulting in attenuated high-frequency coverage for off-axis locations which progressively worsens as the off-axis angle increases. Listeners at off-axis locations will experience a subjectively unbalanced sound spectrum, reduced naturalness, and poorer speech intelligibility. To counter this tendency some proprietary loudspeaker designs use sound diffusing cones or other techniques to increase the coverage angle and limit high-frequency attenuation at off-axis locations.

To provide for good listening conditions, individuals at off-axis locations should be within the driver's -6 dB coverage angle at the listening plane for the 4,000 Hz octave frequency band. The manufacturer's product data should be reviewed to determine the specific coverage angle for the driver under consideration.

Loudspeaker coverage guidelines are summarized below.

1. Smaller drivers tend to have a wider coverage angle than larger drivers since the cone is small compared to the wavelength over a wider frequency bandwidth.

2. Coaxial drivers will provide a wider coverage angle at higher frequencies than single cone drivers.

3. An assumed -6 dB coverage angle of 60° should be used when designing high quality voice and music systems.

4. An assumed -6 dB coverage angle of 90° should be used when designing low quality voice paging and background music systems.

5. Conservative design practice would suggest limiting the assumed -6 dB coverage angle to no greater than 120° regardless of what the manufacturer claims.

The loudspeaker density is critical in optimizing the performance of ceiling distributed systems. A high density is necessary to provide even sound level coverage and to fill in the comb filtering between individual loudspeakers. One major problem with many ceiling distributed systems is an inadequate number of loudspeakers. Contrary to intuition, a low ceiling room will require a higher loudspeaker density than a high ceiling room. This is analogous to shining a flashlight down on a surface. The lower the light is to the surface, the greater the light intensity, but a smaller area is illuminated, thus requiring a larger number of light sources for even illumination. Increasing the flashlight height decreases the light intensity and increases the area illuminated, reducing the number of light sources required for even illumination. Downward pointing loudspeakers behave in a similar manner.

Small format horn loudspeakers will have a coverage pattern determined by the horn geometry. As frequency increases, the area covered by the horn tends to decrease which will result in listeners at off-axis locations receiving attenuated high-frequency sound, similar to a cone driver. Several small and medium format horns have been designed specifically for ceiling installation and feature equal coverage angles in both the horizontal and vertical planes. Typical coverage angles include 40° by 40°, 60° by 60°, and 90° by 90°.

4.3.5.5.1 Coverage of a Single Cone Driver

The coverage geometry for a downward radiating cone driver is shown in Figure 4-48. The coverage area (**A**) of a single cone driver oriented normal to the listening plane can be calculated using the following equations:

$$\mathbf{r} = (\mathbf{h} - \ell)\tan\left(\frac{\alpha}{2}\right) \tag{4.19}$$

where,

r	is the coverage area radius, ft
h	is the floor-to-ceiling height, ft
ℓ	is the height of the listening plane taken as 4 ft for sitting and 6 ft for standing
α	is the driver coverage angle at the listening plane, degrees

$$A = \pi r^2 \tag{4.20}$$

where,

A is the coverage area of a single driver, ft^2

r is as above

Equation 4.19 determines the cone driver coverage area radius while the coverage area is determined from Equation 4.20, assuming a circular coverage pattern.

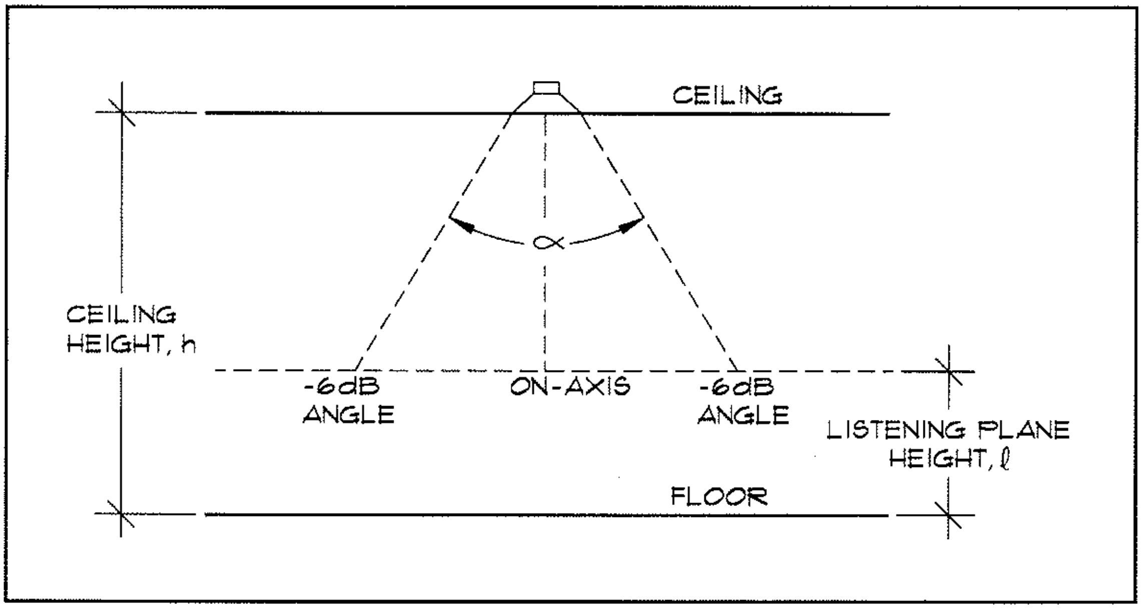


FIGURE 4-48. Coverage geometry for a downward radiating cone driver. The listening plane height (ℓ) is taken as 4 ft for sitting and 6 ft for standing. The -6 dB coverage angle (α) is defined at the listening plane, not the standard polar coverage angle taken at 1 m from the driver.

The concept of coverage angle needs to be clarified when dealing with cone loudspeakers aimed directly downward. Conventional practice uses the polar coverage angle which defines the coverage angle at the -6 dB level relative to the on-axis level on a one m sphere from the measurement microphone. For downward radiating loudspeakers the coverage angle is defined at the -6 dB level relative to the on-axis level at the listening plane. For most ceiling loudspeaker installations, the distance between the listening plane and the loudspeaker will exceed the 1 m polar coverage angle distance.

Aiming a loudspeaker downward will result in greater attenuation at the off-axis locations than the -6 dB polar coverage angle would suggest. The actual coverage angle on the listening plane is less than the coverage angle given by the -6 dB coverage angle definition when it is projected onto the listening plane. The path length to off-axis locations using the -6 dB coverage angle will be greater than the path length to off-axis locations using the listening plane coverage angle. Thus, attenuation of direct sound will be substantially higher using the -6 dB coverage angle definition. For example, a loudspeaker rated with a 140° polar coverage angle will have approximately a 100° listening plane coverage angle. Thus, the standard polar coverage angle data published in manufacturer's literature will result in prediction errors if used to calculate loudspeaker coverage and sound levels at the listening plane.

Some manufacturers provide the loudspeaker coverage angle based on the listening plane. Others use the standard -6 dB polar coverage angle. The data in Table 4-4 can be used to adjust the -6 dB polar coverage angle data to listening plane angle data when calculating sound levels.

TABLE 4-4. Correction to Polar Coverage Angle to Obtain Listening Plane Coverage Angle

-6 dB Polar Off-Axis Angle, degrees	Listening Plane Correction Factor, dB
20	-0.5
25	-0.9
30	-1.3
35	-1.7
40	-2.3
45	-3.0
50	-3.8
55	-4.8
60	-6.0
65	-7.5
70	-9.3
75	-11.7
80	-15.2

For example assume a loudspeaker has a polar coverage angle of 100° at the -6 dB down points. The off-axis angle will be 50°. From Table 4-4, a correction factor of -3.8 dB needs to be added to the -6 dB value to arrive at an off-axis level of -9.8 dB. In this example, the sound level will be almost one-half as loud at the off-axis location compared to the on-axis location.

Conversely, the -6 dB coverage angle at the listening plane can be approximated using the polar coverage angle and the data in Table 4-4. Using the loudspeaker -6 dB polar plot data, add the 5° correction values in the table to each 5° increment on the polar diagram to arrive at the total value which equals -6 dB (sum of polar and correction dB values) to estimate the coverage angle at the listening plane.

4.3.5.5.2 Loudspeaker Layout Patterns

Several loudspeaker layout patterns have been developed which provide varying degrees of coverage overlap at the listening plane. Standard distances of 4 ft for sitting and 6 ft for standing are assumed for the listening plane. The layout patterns in order of decreasing coverage overlap include: (1) center-to-center; (2) minimum overlap; and (3) edge-to-edge. These correspond, respectively, to approximately 50 percent, 20 percent, and 0 percent coverage overlap at the -6 dB loudspeaker coverage angles. Additionally, the three loudspeaker patterns can be laid out in square or hexagonal geometries. Thus, there are six common ceiling loudspeaker layout patterns.

The center-to-center layout has the -6 dB coverage of one loudspeaker coinciding with the center of an adjacent loudspeaker. This layout usually results in too great a loudspeaker quantity for practical installations. Excessive cost and conflicts with coordinating loudspeaker locations with other ceiling fixtures (lights, HVAC diffusers, sprinkler heads, and smoke detectors) are common. The minimum overlap layout has all areas within the -6 dB coverage of at least one loudspeaker. The edge-to-edge layout results in the -6 dB coverage of each loudspeaker just touching. This layout usually results in too few loudspeakers and compromises sound level coverage and uniformity. Thus, for most practical situations, the minimum overlap layout provides the best solution between conflicting issues of cost, installation, and electro-acoustic performance.

The square spacing lays out the loudspeakers evenly in a row and column geometry, which may simplify layout in a suspended ceiling grid. The hexagonal spacing staggers the loudspeaker rows to create a criss-cross (equilateral triangle spacing) between the loudspeakers. Figure 4-49 shows the square loudspeaker spacing for the three loudspeaker layout patterns. Figure 4-50 is similar but for the hexagonal loudspeaker spacing.

Loudspeaker layout patterns and sound level output for different applications are summarized below.

1. **Background Music or Low Quality Paging:** Edge-to-edge providing 5 dB increase in sound level above ambient noise.

2. **Background Music with High Quality Paging:** Minimum overlap providing 15 dB increase in sound level above ambient noise.

3. **Speech Reinforcement or High Quality Music:** Center-to-center (where possible) providing 25 dB increase in sound level above ambient noise.

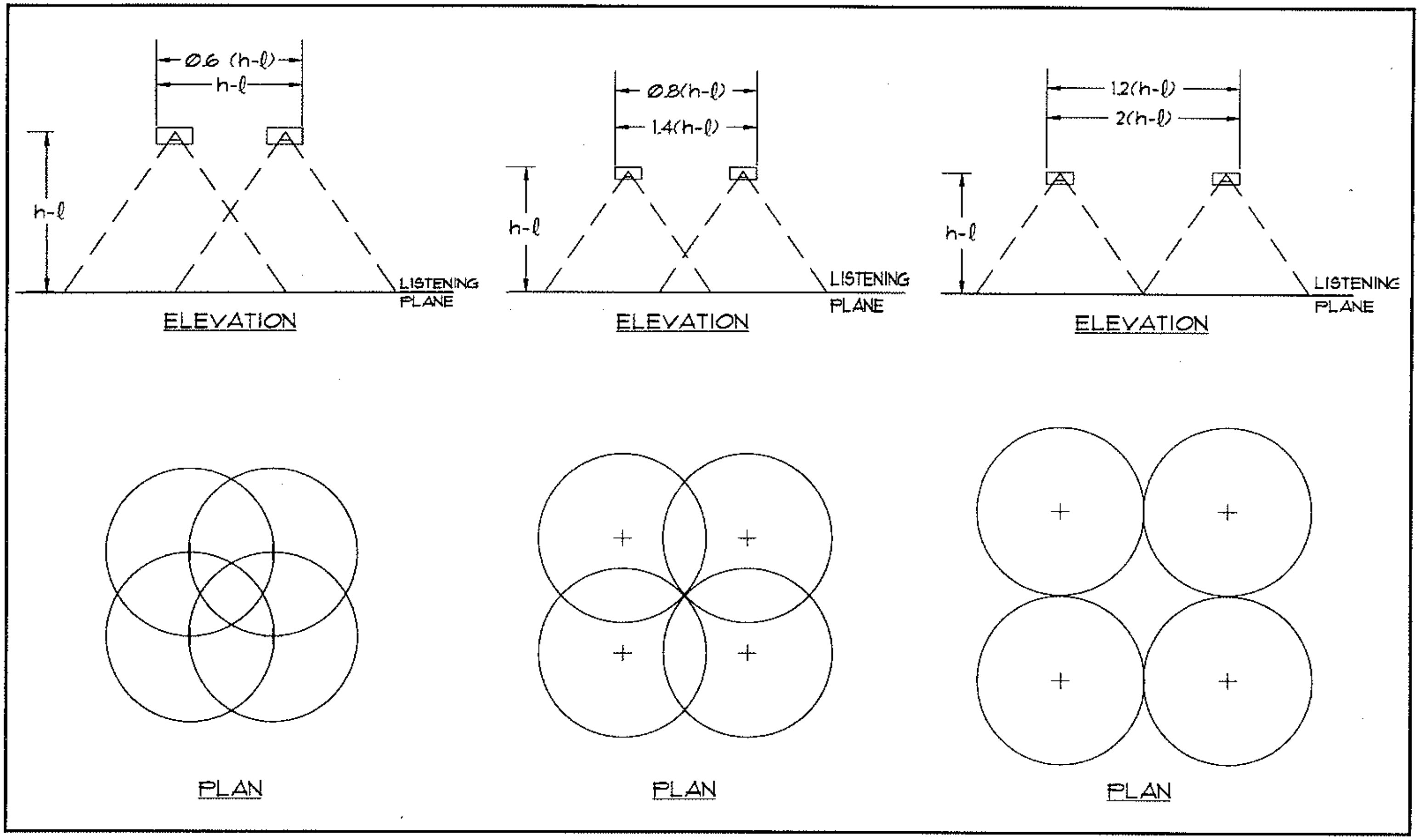


FIGURE 4-49. Square loudspeaker spacing for center-to-center (left), minimum (center), and edge-to-edge (right) overlap layouts. The top equation in the figure is for 60° coverage angle drivers. The bottom equation in the figure is for 90° coverage angle drivers.

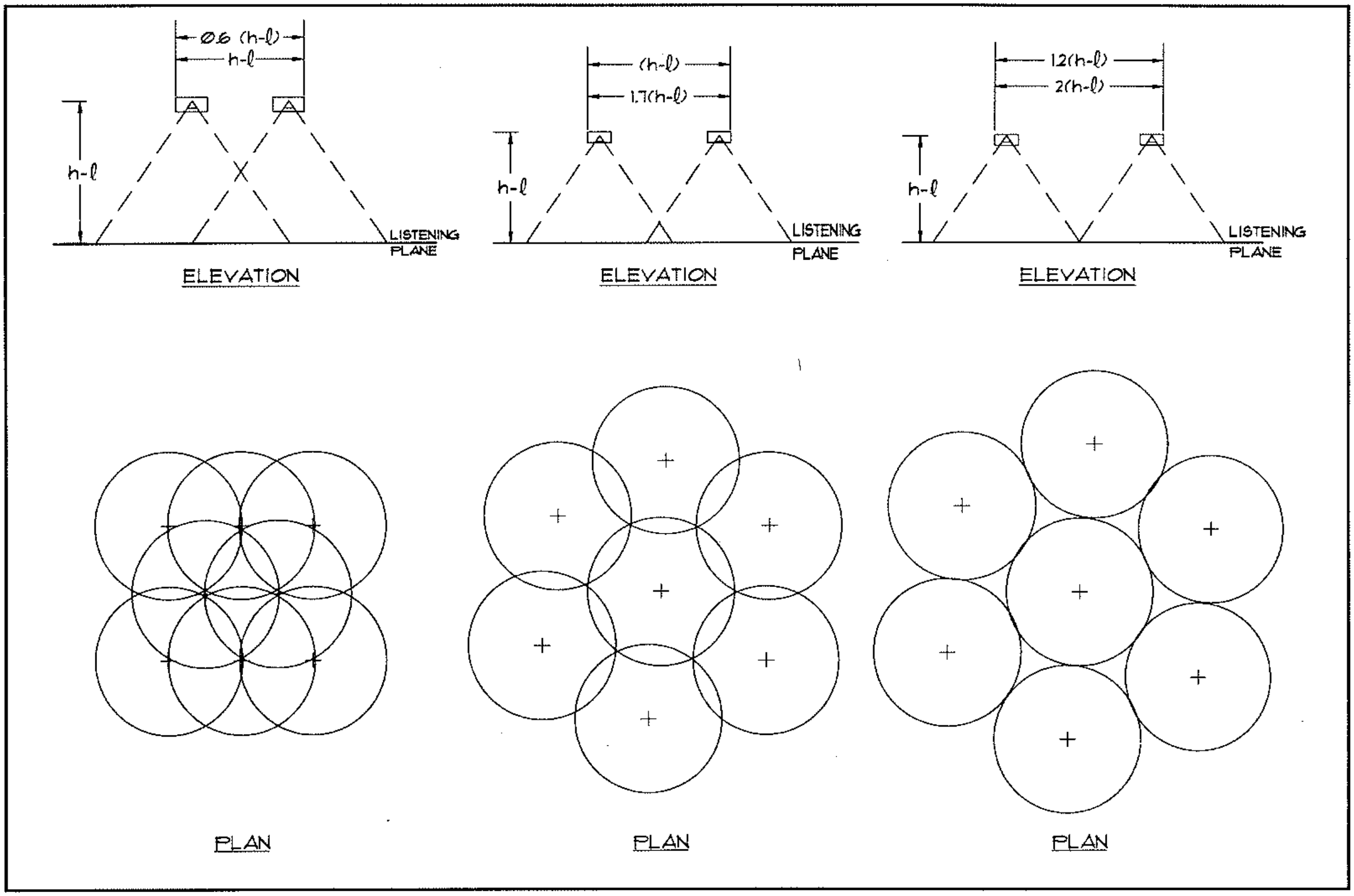


FIGURE 4-50. Hexagonal loudspeaker spacing for center-to-center (left), minimum (center), and edge-to-edge (right) overlap layouts. The top equation in the figure is for 60° coverage angle drivers. The bottom equation in the figure is for 90° coverage angle drivers.

4.3.5.5.3 Coverage by Multiple Loudspeakers

The number of loudspeakers required to cover an audience area can be estimated from the data in Tables 4-5 and 4-6 for the six ceiling loudspeaker pattern layouts. The loudspeaker quantity data are based on the -6 dB polar coverage angle. The loudspeaker quantity will increase when calculating based on the coverage angle at the listening plane, since the actual coverage area at the -6 dB listening plane will be less.

TABLE 4-5. Loudspeaker Density for Different Layout Patterns for 60° -6 dB Coverage Angle Loudspeakers

h-ℓ	Area (Ft^2) Covered by One 60° Loudspeaker	Number of 60° Loudspeakers per 1000 Ft^2					
		Center-to-Center		Minimum Overlap		Edge-to-Edge	
		Square Spacing	Hexagonal Spacing	Square Spacing	Hexagonal Spacing	Square Spacing	Hexagonal Spacing
4	17	188	217	94	72	47	54
5	26	120	139	60	46	30	35
6	38	83	96	42	32	21	24
7	51	62	71	31	24	15	18
8	67	47	54	23	18	12	14
9	85	37	43	19	14	9	11
10	105	30	35	15	12	8	9
11	127	25	29	12	10	6	7
12	151	21	24	10	8	5	6
13	177	18	21	9	7	4	5
14	205	15	18	8	6	4	4
15	236	13	15	7	5	3	4
16	268	12	14	6	5	3	3

TABLE 4-6. Loudspeaker Density for Different Layout Patterns for 90° -6 dB Coverage Angle Loudspeakers

h-ℓ	Area (Ft^2) Covered by One 60° Loudspeaker	Number of 90° Loudspeakers per 1000 Ft^2					
		Center-to-Center		Minimum Overlap		Edge-to-Edge	
		Square Spacing	Hexagonal Spacing	Square Spacing	Hexagonal Spacing	Square Spacing	Hexagonal Spacing
4	50	63	72	31	24	16	18
5	79	40	46	20	15	10	12
6	113	28	32	14	11	7	8
7	154	20	24	10	8	5	6
8	201	16	18	8	6	4	5
9	255	12	14	6	5	3	4
10	314	10	12	5	4	3	3
11	380	8	10	4	3	2	2
12	452	7	8	4	3	2	2
13	531	6	7	3	2	2	2
14	616	5	6	3	2	1	2
15	707	4	5	2	2	1	1
16	804	4	5	2	2	1	1

A more precise determination of the required number of loudspeakers can be made by calculating the loudspeaker-to-loudspeaker spacing (**S**) as a function of the ceiling (**h**) and listening plane (ℓ) heights and the loudspeaker coverage angle. The value of **S** can be calculated using the following equations:

Center-to-center overlap pattern with square or hexagonal geometries

$$\mathbf{S} = \mathbf{A}(\mathbf{h} - \ell) \qquad \textbf{(4.21)}$$

(**A** is 0.6 and 1.0 respectively for 60° and 90° -6 dB coverage angles)

Minimum overlap pattern with square geometry

$$\mathbf{S} = \mathbf{A}(\mathbf{h} - \ell) \qquad \textbf{(4.22)}$$

(**A** is 0.8 and 1.4 respectively for 60° and 90° -6 dB coverage angles)

Minimum overlap pattern with hexagonal geometry

$$\mathbf{S} = \mathbf{A}(\mathbf{h} - \ell) \tag{4.23}$$

(**A** is 1.0 and 1.7 respectively for 60° and 90° -6 dB coverage angles)

Edge-to-edge overlap pattern with square or hexagonal geometries

$$\mathbf{S} = \mathbf{A}(\mathbf{h} - \ell) \tag{4.24}$$

(**A** is 1.2 and 2.0 respectively for 60° and 90° -6 dB coverage angles)

where,
S is the loudspeaker-to-loudspeaker spacing, ft
h and ℓ are as above

The value of **S** is used to create the loudspeaker layout pattern over the desired coverage area. When designing with horn drivers the manufacturer's coverage data should be used to create the loudspeaker layout pattern.

The listener will receive sound from multiple loudspeakers within the coverage area with diminishing level contributions from the more distant loudspeakers. Table 4-7 provides correction factors to account for summation of sound levels from nearby multiple loudspeakers. Within the coverage area the sound level will vary due to the coherent and incoherent summation from multiple sources. Table 4-8 provides correction factors to account for the level variation within the coverage area. The values in these two tables can be used for more precise calculations of sound levels within the loudspeaker coverage area.

TABLE 4-7. Correction Factors for Summation of Multiple Loudspeakers

Loudspeaker Pattern	Correction Factor, dB
Square Loudspeaker Layout	
Center-to-Center Overlap	+5.2
Minimum Overlap	+2.0
Edge-to-Edge Overlap	+0.7
Hexagonal Loudspeaker Layout	
Center-to-Center Overlap	+5.4
Minimum Overlap	+1.4
Edge-to-Edge Overlap	+1.0

TABLE 4-8. Correction Factors for Level Variation from Multiple Loudspeakers

Loudspeaker Pattern	Correction Factor, dB
Square Loudspeaker Layout	
Center-to-Center Overlap	-1.4
Minimum Overlap	-2.0
Edge-to-Edge Overlap	-4.4
Hexagonal Loudspeaker Layout	
Center-to-Center Overlap	-1.2
Minimum Overlap	-2.6
Edge-to-Edge Overlap	-5.4

4.3.5.6 Advantages of Ceiling Distributed Systems

The main advantages of ceiling distributed systems are the potential for very even sound level coverage and installation simplicity. A secondary advantage is the flexibility to zone the loudspeakers to suit different space configurations, as occurs with hotel ballrooms having operable partitions. These systems are often used in conjunction with central cluster or multi-channel systems to provide coverage to under balcony locations which would be occluded from the main loudspeakers.

Directional realism is often compromised with ceiling distributed systems since the sound radiates from above the listener. The overhead sound is usually at a higher level and arrives earlier in time than the direct sound from the talker or performer. Thus, the listener perceives the sound as originating overhead rather than frontally. Directional realism can be improved by using a signal delay line so the direct sound from the talker or performer arrives first, followed by the delayed sound from the overhead loudspeakers. In large rooms it may be necessary to subdivide the loudspeakers into "zones" with each zone on a different signal delay setting. In cases where the direct sound from the talker or performer is anticipated to be weak, a small directional loudspeaker positioned above the source can provide localization cues, assuming this sound arrives prior to the sound from the overhead loudspeakers.

One disadvantage with ceiling distributed systems used in rooms where quality reproduction is required is a higher cost than central cluster or similar centralized loudspeaker systems due to the larger quantity of amplifiers, signal processing, cable, and loudspeakers required.

4.3.5.7 Design Objectives of Ceiling Distributed Systems

The primary design objectives of ceiling distributed systems are to select loudspeakers and transformers which reproduce the necessary frequency range and to arrange the loudspeakers in a sufficiently dense pattern which ensures even sound coverage in the audience area.

Ceiling distributed systems using cone drivers may not perform to their full potential due to four factors: (1) the quality of the driver and transformer; (2) the loudspeaker layout is based on too large an assumed driver coverage angle; (3) an insufficient number of drivers are used; and (4) the driver lacks a properly sized back box or does not have one at all.

The loudspeaker and transformer quality, adequate number of loudspeakers, and the need to use a conservative loudspeaker coverage angle have been discussed above. Calculations relating to direct and reverberant sound level and speech intelligibility are equally applicable to ceiling distributed systems.

4.3.6 Seat-Back Systems (Type 6A System)

Seat-back systems (Type 6A system) comprise 3 or 4 in diameter full-range cone drivers in enclosures installed on the back of the seat directly in front of the listener. The drivers are connected to a constant voltage power amplifier and are operated at a low sound level. Seat-back systems are intended for voice reproduction due to the small size drivers and the limited low-frequency extension. They are commonly called "pew-back" loudspeaker systems. David Lloyd Klepper is credited with applying this system type to reverberant houses of worship.

Drivers are spaced every three or four listeners apart which results in a maximum listener-to-driver distance of less than 10 ft. The close position of the drivers to the acoustically absorptive listeners helps minimize exciting the room reverberant sound field. Thus, these systems have the potential for high speech intelligibility in the most reverberant of spaces where the $\mathbf{D_C}$ may be on the order of less than 5 ft.

The drivers are housed in small enclosures, usually custom designed for the specific installation. Within the enclosure is a small transformer normally tapped at 1 W or less and filled with fiberglass to reduce enclosure resonances. The loudspeaker enclosures can be either separate units or continuous along the length of the pew as shown in Figure 4-51.

FIGURE 4-51. Type 6A seat back system using 3 in cone drivers spaced at 5 ft centers at National City Christian Church in Washington, DC. System design by R. Lawrence Philbrick, Jr. of Acentech, Inc.

It is recommended that the drivers should not be installed under the seats to conceal their presence. Doing so will result in the listeners receiving only reflected sound, and not direct sound, which is important for speech intelligibility. The reflected sound will be attenuated in the mid- to high-frequency range by absorption from the floor finishes, padded knee rests, and the clothing on the listener's legs.

4.3.6.1 Advantages of Seat-Back Systems

Seat-back systems have electro-acoustical performance, aesthetic, and adaptability characteristics not realized with other system types. Speech intelligibility is very high with these systems and excellent gain-before-feedback is achieved since the drivers are positioned remotely from the microphones, resulting in a large value of $\mathbf{D}_1$. With the drivers positioned behind the seats there is no visual obstruction to decorative architectural elements as might occur with central cluster or distributed column systems. This can be advantageous in worship houses having stained glass windows or other religious symbols which must remain visible. Seat-back systems can provide sound coverage to listeners located behind the talker or performer where it might otherwise be difficult to locate loudspeakers. A common application is in choir seating located behind the lectern or pulpit.

One disadvantage with seat-back systems is their high cost. This is due to the large number of drivers required and the common practice of using custom loudspeaker enclosures to minimize the visual appearance of the system. Fitting loudspeaker enclosures and cables to the back of seats requires careful installation practices to achieve a pleasing appearance. An additional factor contributing to the high cost is the need for signal delay lines, multiple power amplifiers, and extensive cable and conduit runs to drive the numerous zoned loudspeakers.

4.3.6.2 Design Objectives of Seat-Back Systems

The primary objective of seat-back systems is to provide uniformity of coverage with close driver spacing. A signal delay line is normally used with these systems in order to preserve source localization. Each loudspeaker zone is progressively delayed between 10 and 20 ms.

Often delayed reverberant sound "wash" from the rear wall will return to the front half of the room when a distributed delay system is adjusted to "pull" the sound source image towards the front. This can decrease the speech intelligibility at the front half of the room due to the delayed sound not synchronizing with the frontally-arriving sound. The speech intelligibility can be improved by maintaining a constant **D/R** ratio in the audience seating area. The loudspeaker output level can be reduced from the front to the rear zones which minimizes the amount of delayed reverberant sound returning to the room front. By reducing the zoned loudspeaker levels it is possible to decrease the delay time necessary to maintain source localization. Typically, a level reduction of 0.5 dB per zone is sufficient, which typically results in the farthest zone having levels between 3 to 4 dB less than the frontmost zone. A 3 dB level reduction in sound levels at the rear wall is equivalent to sound reflecting off a surface with a sound absorption coefficient of 0.50.

Subjective improvement for all delay zones can be achieved through frequency equalization having a flat response between 160 and 8,000 Hz, in contrast to the normal practice of rolling off high-frequencies above 2,000 Hz at 2 to 3 dB per octave. Frequency limiting below 160 Hz and above 8,000 Hz is recommended. Signal delay lines should be adjusted to synchronize the sound from the talker or localizer loudspeaker but not include the 15 to 20 ms additional signal delay (Haas/precedence delay) common with other systems.

A localizer loudspeaker near the talker can often improve directional realism when the sound from this loudspeaker reaches the listener before the sound from the seat-back loudspeaker. The sense of directionality can be enhanced by equalizing the localizer loudspeaker for an extended frequency bandwidth between 160 and 8,000 Hz and bandwidth limiting the seat-back loudspeakers between 300 and 5,000 Hz.

4.4 System Configurations and Installation Practices to Avoid

The designer may encounter existing sound system configurations and installation practices that result in poor electro-acoustical performance. Some of the more common mistakes are described below.

4.4.1 Simultaneously Operating Multiple Monophonic Loudspeakers

Two widely spaced loudspeakers operated simultaneously with a monophonic input signal will cause severe comb filtering in the audience seating area except along the room centerline. The comb filtering is due to differences in the arrival time of sound from the two loudspeakers at the listener position. The path length difference results in frequency selective sound cancellation which degrades speech intelligibility and naturalness. Another byproduct is poor sound source localization. Listeners close or on-axis to a loudspeaker will perceive the sound to come from that loudspeaker and not the sound source. These listeners may perceive an echo from delayed sound from the other loudspeaker if the signal is sufficiently loud and is delayed greater than 50 ms. Often speech intelligibility and naturalness can be improved by turning off one of the loudspeakers. Figure 4-52 illustrates the problems with a split system with monophonic loudspeakers.

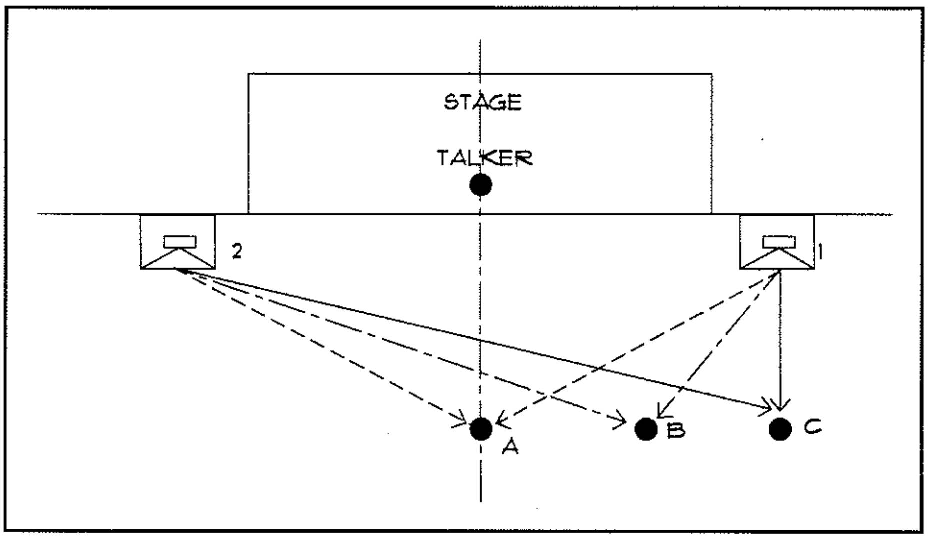


FIGURE 4-52. Problems with a split system with monophonic loudspeakers left and right of the stage. Listener A experiences no comb filtering (equal path length between loudspeakers 1 and 2) and good localization only when the talker is at the room centerline. Listener B experiences comb filtering (unequal path length between loudspeakers 1 and 2) and poor localization. Listener C experiences comb filtering (unequal path length between loudspeakers 1 and 2) and poor localization, with sound from loudspeaker 1 dominating and potential for audible echoes from loudspeaker 2.

4.4.2 Cross-Fired Loudspeakers

Cross-fired loudspeakers facing each another result in poor electro-acoustical performance for reasons similar to split systems. This configuration is often used in an attempt to improve sound coverage to the back of a room when the front-facing loudspeakers are inadequate to cover the entire room. Localization will be especially poor since the listener will receive sound from both the front and the back while the source originates from the front. Due to the closer proximity of the rear loudspeakers, the sound level for listeners at the back will be higher, which further compounds poor localization. Audible and visual cues need to be synchronized so the directional origination of the sound corresponds to the visual location of the sound source.

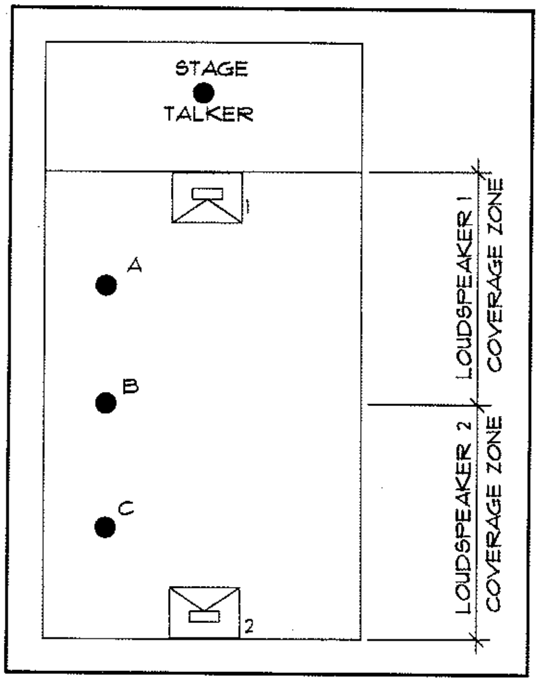


FIGURE 4-53. Problems with cross-fired loudspeakers at the front and back of the room. Listener A experiences good coverage by loudspeaker 1 and good localization of the talker with moderate comb filtering from loudspeaker 2. Listener B experiences poor coverage by loudspeakers 1 and 2, poor localization of the talker, and severe comb filtering from loudspeakers 1 and 2. Listener C experiences good coverage by loudspeaker 2 and bad localization of the talker with moderate comb filtering from loudspeaker 1.

The use of bi-directional loudspeakers in airport and train station circulation spaces is frequently encountered. Often these spaces are large and reverberant and the need for good loudspeaker pattern control is critical. Here the concern is for high speech intelligibility and naturalness, not directional realism. Bi-directional loudspeakers will result in comb filtering which will reduce speech intelligibility. Figure 4-53 illustrates the problems with cross-fired loudspeakers.

4.4.3 Inappropriate Loudspeaker Types

A common mistake is installing inappropriate loudspeakers for the room acoustical environment. This is particularly true for rooms with a long reverberation time where physically large loudspeakers having high **Q** values are necessary for good electro-acoustic performance. The sound system designer must not be tempted to cave in to architectural mandates that dictate only small loudspeakers be used when these are not suitable for the room. There are generally several loudspeaker types which can provide the needed electro-acoustical performance for a given acoustical environment.

Loudspeakers must have their frequency response tailored for the program type. Speech reinforcement systems do not require loudspeakers having low-frequency response below 300 Hz to convey speech intelligibility, but voice naturalness will be enhanced with loudspeakers having response down to 125 Hz. Frequently, music is played back at high levels over small loudspeakers which can not handle the low-frequency content, resulting in high levels of harmonic distortion and damaged loudspeakers.

When challenged by the building owners or architect on the need for physically large or elaborate loudspeaker systems, the designer can demonstrate the subjective affects of different loudspeakers, both appropriate and inappropriate for the room acoustical characteristics, using sound system auralization software.

4.4.4 Loudspeaker Location

Inappropriate loudspeaker location can compromise sound system performance. Loudspeakers which are located greater than 45 ft above the floor can result in an echo at some audience seating locations due to the path length difference between the loudspeaker-to-listener and stage-to-listener distances. The path length difference between the direct sound from the on-stage source and the loudspeaker should be less than 35 ft to provide the best sound quality.

Another situation involves installing low **Q** loudspeakers at distances remote from the listeners with an attendant loss of speech intelligibility due to the increase in reverberant sound level. Often improvements will result by locating the loudspeakers closer to the listeners.

Excessive sound levels can result with central cluster systems in low-ceiling rooms. The front seating is covered by the near-throw (bottom half) of the loudspeaker vertical coverage pattern and the rear seating is covered by the far-throw (top half) of the vertical coverage pattern. Sound levels will be greater at the front because of proximity to the loudspeaker and less at the rear due to the greater distance from the loudspeaker. Another problem occurs in long low-ceiling rooms when the loudspeaker is turned up to be heard at the room rear, resulting in overly loud sound levels at the room front. Figure 4-54 illustrates some problems with loudspeaker location.

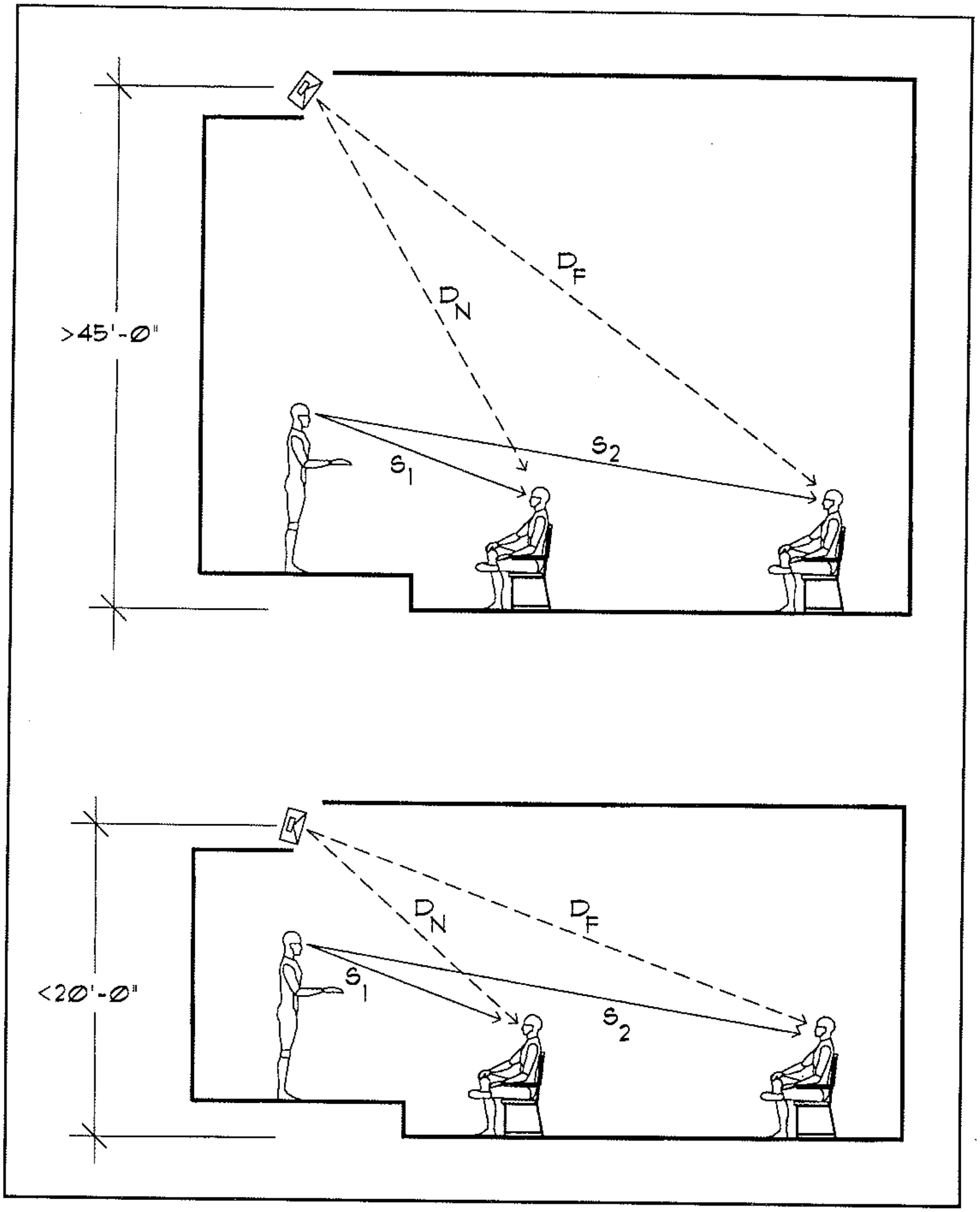


FIGURE 4-54. Problems with loudspeaker locations which are either too high or too low. In the top figure the loudspeaker is greater than 45 ft above the floor. The path length difference at the front between the source direct sound (**S_1**) and the direct sound from the loudspeaker near-throw vertical coverage pattern (**D_N**) may exceed 35 ft. A similar condition exists at the rear of the room between direct sound (**S_2**) and the loudspeaker far-throw vertical coverage pattern (**D_F**). In the bottom figure the loudspeaker is less than 20 ft above the floor. The sound level is too high at the front of the room from (**D_N**) but acceptable at the rear from (**D_F**).

4.4.5 Number of Loudspeakers

The correct number of loudspeakers is necessary for good electro-acoustical performance. Mistakes are made by both over and under specifying the loudspeaker quantity. Too many loudspeakers with narrow coverage patterns are frequently used in central cluster or split systems where fewer loudspeakers with wider coverage patterns would be more appropriate. The penalty paid for this type of over design is an increase in reverberant sound levels, a decrease in **D/R** ratio, and reduced speech intelligibility due to comb filtering from the overlap of adjacent loudspeaker patterns. Commonly, too few loudspeakers are used in ceiling distributed systems. This type of under design results in inadequate sound level coverage at off-axis locations for frequencies above 2,000 Hz which degrades speech intelligibility.

4.4.6 Obstructed Loudspeaker Path

Loudspeakers require line-of-sight to the listener to provide good direct sound coverage. Compromises in the high-frequency response at the listener's location will result if the high-frequency driver is not visible to the listener. The exception to this are those installations where the loudspeakers are concealed behind an acoustically *transondent* material.

Central cluster systems recessed into the ceiling require an opening larger than the physical size of the loudspeakers. This is to provide adequate clearance to permit the full horizontal and vertical loudspeaker radiation patterns to pass unobstructed through the opening. If the opening is not sufficiently large, portions of the radiation patterns will be blocked by the ceiling opening, resulting in poor direct sound coverage in some audience seating locations. Figure 4-55 illustrates good design practice for recessing loudspeakers into ceiling or wall surfaces.

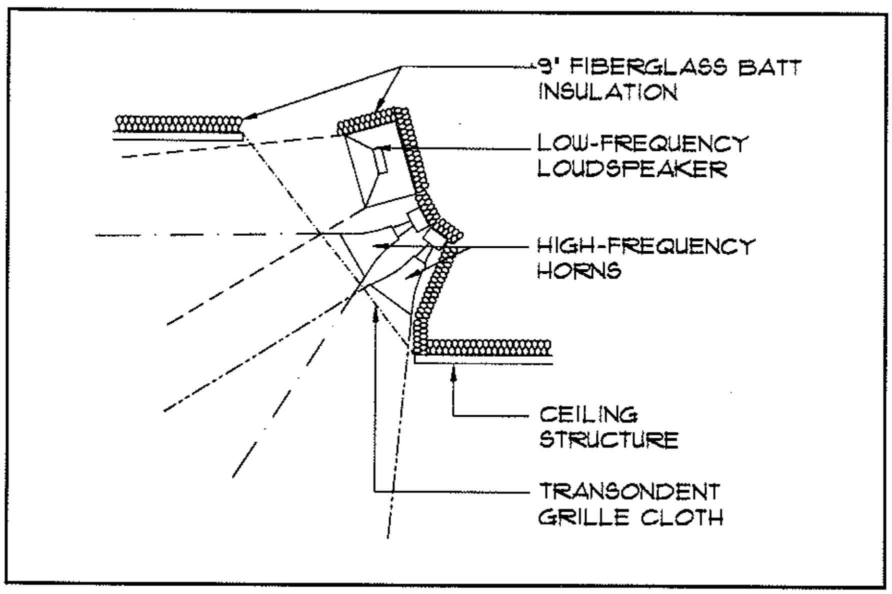


FIGURE 4-55. Good design practice for loudspeakers recessed into the ceiling or wall to permit sound to pass through unobstructed. The back of the cavity is provided with thick fiberglass insulation to absorb incidental sound reflections from nearby surfaces or backward radiating sound from the loudspeakers.

Small physical obstructions directly in front of high-frequency drivers such as wood or metal framing used as part of grille cover assemblies, can block the transmission of high-frequency sound. This can degrade speech intelligibility or reduce musical clarity. A rule-of-thumb is not to have obstructions larger than ½ in wide or deep in front of high-frequency drivers. These obstructions should be installed normal to the driver centerline. Figure 4-56 illustrates common materials used to fabricate transondent openings for loudspeakers.

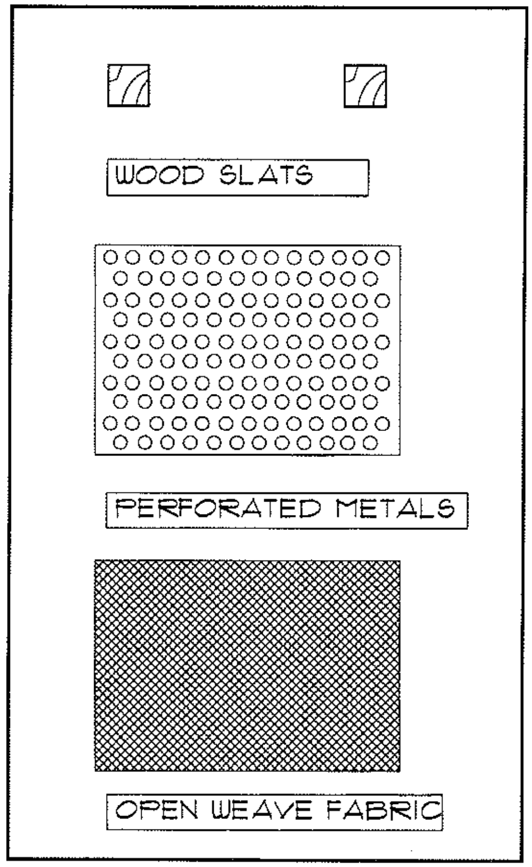


FIGURE 4-56. Different materials for transondent loudspeaker openings. Top is maximum ½ in wide and deep wood slats on minimum 2 in spacings perpendicular to loudspeakers. Middle is perforated metal minimum 50 to 60 percent open area with minimum $^{5}/_{32}$ in diameter staggered holes. Bottom is open weave burlap, "monk's cloth," polyester, fiberglass, or fabric

4.4.7 Loudspeaker/Microphone Interaction

When a loudspeaker radiates sound into a microphone, the sound system gain-before-feedback is limited, or worse, the sound system goes into uncontrolled feedback. Common causes include having the microphone in front of the loudspeaker where it is in the loudspeaker vertical coverage pattern, or having the microphone in the same horizontal plane as the loudspeaker, where it is in the loudspeaker horizontal coverage pattern. Note that these problems can be exacerbated at frequencies below 250 Hz where loudspeaker pattern control is significantly less than at high frequencies. Users of mobile wireless microphones should be advised not to inadvertently walk into the loudspeaker coverage pattern. Figure 4-57 illustrates common loudspeaker and microphone positioning errors and their corrections.

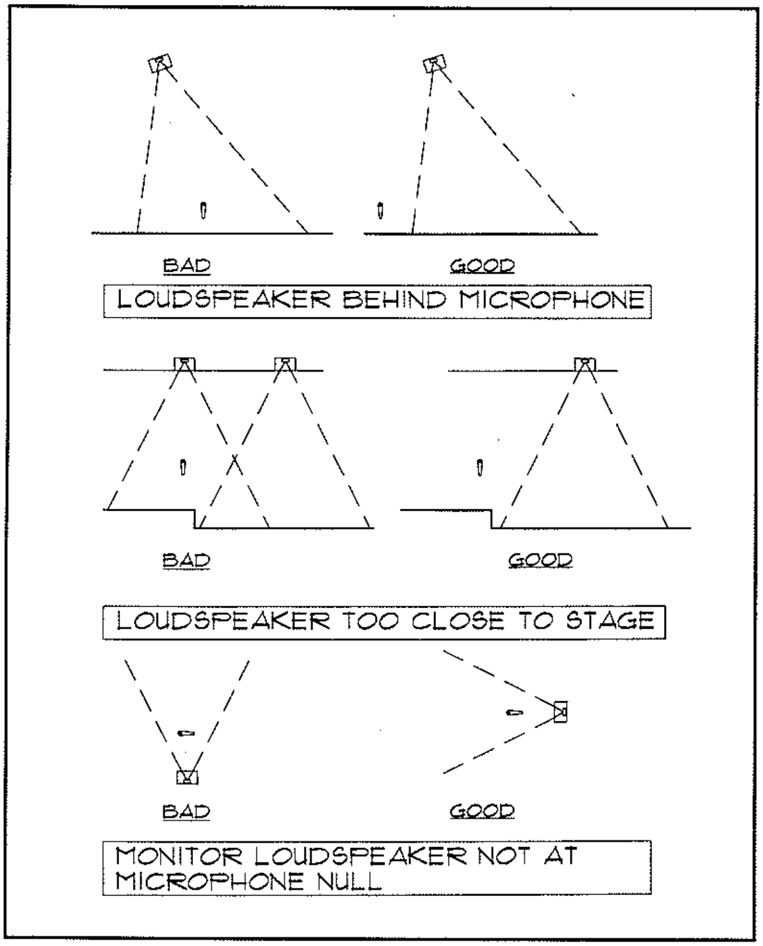


FIGURE 4-57. Common loudspeaker and microphone positioning errors. Top has microphone behind loudspeaker coverage pattern, common with some central cluster systems. Middle has loudspeaker too close to stage, common with distributed ceiling systems. Bottom has loudspeaker not in the microphone directivity pattern null, common with on-stage monitor loudspeaker systems.

4.4.8 Room Acoustical Considerations

Selection and aiming loudspeakers needs to consider room surfaces that sound might reflect from. Ideally, loudspeakers will be selected and aimed to direct acoustical energy only onto the audience. However, this is not always feasible and the need for corrective room acoustical treatment may be necessary where problematic reflections occur. Large flat or concave rear wall or balcony surfaces beyond 60 ft from the stage can cause audibly detrimental sound reflections back to the stage or front audience seating locations. Acoustical treatment can take the form of sound absorbing wall panels or sound diffusing surfaces, when it is desired not to decrease the room reverberation time by adding acoustical absorption.

4.5 Microphone Usage

Application and use of microphones will affect the overall sound quality of reinforced or recorded programs. Placement of the microphone(s) is critical since any signal loss due to incorrect microphone use will not be gained later in the audio signal chain. Guidelines for microphone usage are summarized below for speech and music including reinforcement and recording applications. The same microphones can serve both reinforcement and recording functions in some installations.

(See Technical Notes, Section 4.H, at the end of this chapter, for additional information on common microphone application errors and corrections.)

4.5.1 Use of Multiple Microphones

Reinforcement or recording a sound source requires evaluating the number of microphones needed to pick-up the source. In most applications, only one microphone is necessary when it is properly placed. Multiple microphones picking-up the same source typically degrades the reinforced or recorded signal, primarily due to comb filtering effects. If microphone signals need to be routed to different locations, a microphone "splitter" should be used.

One common rule-of-thumb when using multiple microphones is the "three-to-one" rule. This concept separates nearby microphones a minimum of three times the distance between the microphone and the source as shown in Figure 4-58. The idea with the three-to-one rule is the direct sound will be attenuated approximately 10 dB at the more distant microphone due to inverse square law losses compared to the microphone closer to the source.

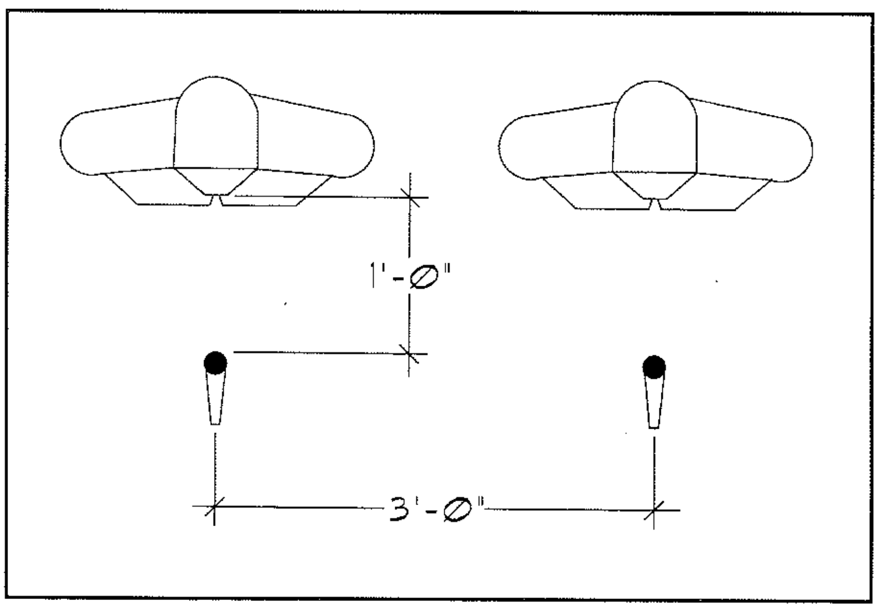


FIGURE 4-58. Three-to-one rule illustrated. The direct sound from the source to the farther microphone will be attenuated approximately 10 dB.

4.5.2 Speech Applications

Speech applications for microphones include reinforcement and recording in different rooms and programs. The primary concern is accurate pick-up of the speech signal to maximize gain-before-feedback and to maintain speech intelligibility.

4.5.2.1 Speech Reinforcement

Speech reinforcement is necessary in virtually all rooms where actors, lecturers, and conference attendees are located.

On-stage microphones used for actors include omnidirectional or cardioid lavaliere-types concealed on clothing or in hair and the cardioid boundary layer-type placed at the stage edge. Figure 4-59 shows application of boundary layer microphones for on-stage pick-up.

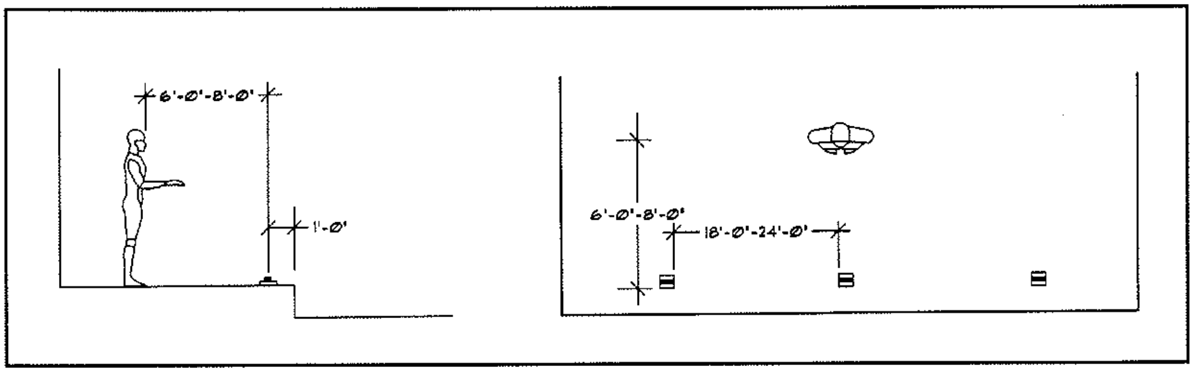


FIGURE 4-59. Use of boundary layer microphones for on-stage pick-up. Microphones are spaced between 6 and 8 ft from the actor and spaced 18 to 24 ft apart. Modest gain up to 8 dB is achievable with this arrangement.

Speech reinforcement from a fixed position can use a thin profile lectern, boundary layer, or stand-mounted handheld microphone. The thin profile lectern microphone is commonly used due to its narrow silhouette and ability to be positioned close to the source. The boundary layer microphone is the least obtrusive type, but provides the lowest gain due to a greater distance to the talker. The stand-mounted handheld microphone can result in the best audio quality, due to the larger diaphragm, with certain manufacturers producing low proximity effect designs. Cardioid-type microphones are commonly used for fixed position sources. Figure 4-60 shows applications of speech reinforcement microphones at a lectern.

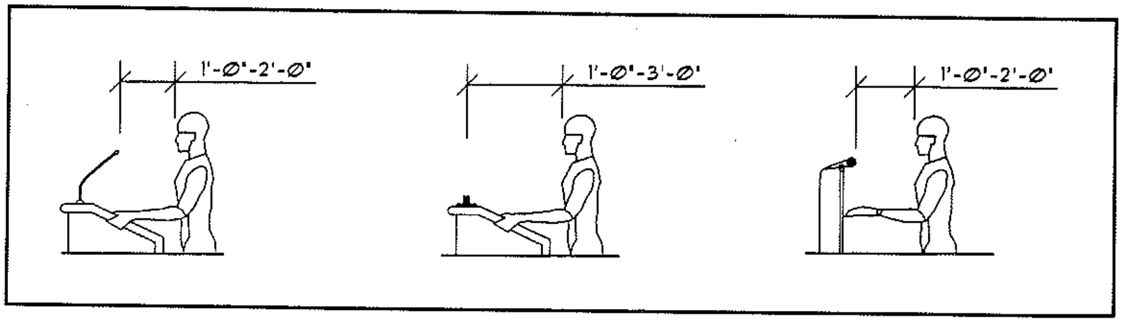


FIGURE 4-60. Use of thin profile lectern (left), boundary layer (center), and stand-mounted handheld microphones (right). Microphones are spaced 1 to 2 ft from the talker for the thin profile lectern and stand-mounted handheld types. The boundary layer microphone is positioned 1 to 3 ft from the talker.

Conference room microphone applications for speech reinforcement or teleconferencing are similar to the fixed position application described above. Improved voice pick-up is achieved with thin profile lectern microphones rather than boundary layer microphones. Use of omnidirectional microphones can reduce the number of microphones necessary for voice pick-up compared to cardioid microphones. Regardless of microphone type used, the three-to-one rule should be observed. Figure 4-61 shows application of speech reinforcement microphones at a conference table.

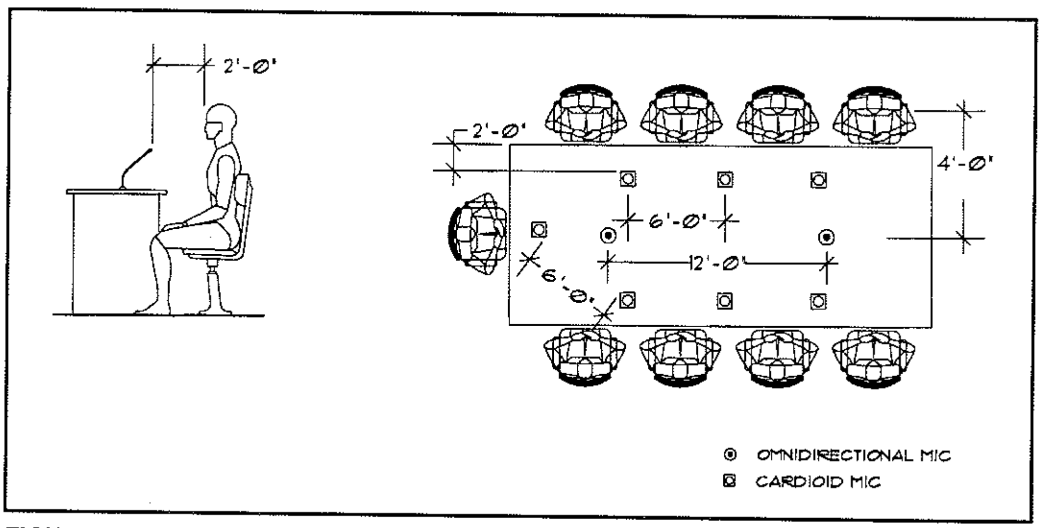


FIGURE 4-61. Use of omnidirectional and cardioid microphones for conference table voice pick-up. Microphones are spaced 2 ft from the talker for the cardioid microphones and 3 to 4 ft from the talker for omnidirectional microphones. Positioning adjacent microphones follows the three-to-one rule.

4.5.2.2 Speech Recording

Recording speech programs requires similar microphone usage as in reinforcement applications. The guidelines above and shown in Figures 4-59 through 4-61 should be used.

4.5.3 Music Applications

Music applications for microphones are similar to speech applications described above. The primary concern is accurate pick-up of voice or instrumental signals while minimizing pick-up of extraneous signals such as audience noise, other voices, and instruments near the primary recording microphone(s).

4.5.3.1 Music Reinforcement

Music reinforcement may be necessary for certain instrumental or choral groups which do not have adequate sound level, as is common with amateur groups. Low gain levels are the goal so the audience is not aware of the subtle reinforcement. Popular music reinforcement with high sound levels requires a different design philosophy and will not be covered here.

Microphones used for music reinforcement are usually small element ceiling hung types, or in some cases, a *piezoelectric* type clipped on to the instrument might be used. Cardioid-type microphones are preferred for ceiling hung microphones. Figure 4-62 shows application of ceiling hung microphones for pick-up of a choral ensemble.

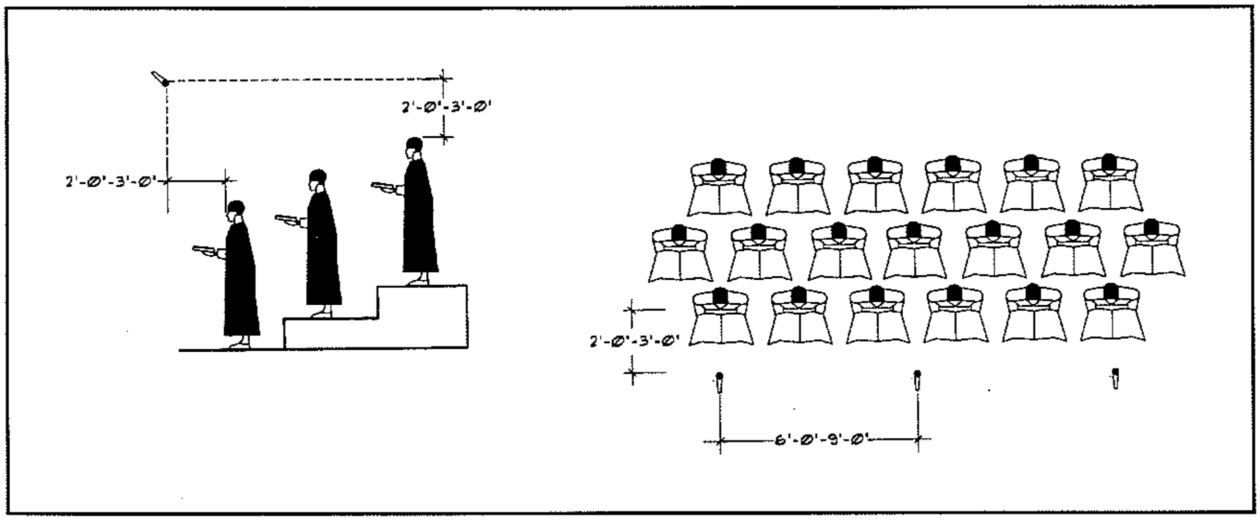


FIGURE 4-62. Reinforcement of a choral ensemble using ceiling hung microphones. The microphones are elevated 2 to 3 ft above the back chorister row and an equal distance from the front chorister row. Positioning adjacent microphones follows the three-to-one rule.

4.5.3.2 Music Recording

Recording musical ensembles uses a variety of different microphone arrangements, from closely spaced cardioids to widely spaced omnidirectional microphones. Music auditoria often have permanent recording microphones placed forward and above the stage. There is generally no firm rule on microphone placement as recording music involves a balance between picking-up the direct sound from the performers and the room reverberant sound. Closer microphone placement will enhance the direct sound pick-up yielding more clarity. More distant placement will pick-up more reverberant sound and enhance spaciousness. Contemporary music sounds best with greater clarity. Classical-type music sound best with some reverberant sound.

High-quality recording microphones are larger than the small overhead ceiling microphones used for sound reinforcement. Additionally, the microphone cables are larger in diameter and the microphones are stabilized with clear nylon or thin stranded "aircraft" cable to prevent the microphones from swaying. These factors result in the microphones being more visible than the small overhead ceiling microphones. Figure 4-63 shows application of music recording microphones and Figure 4-64 shows an installation example.

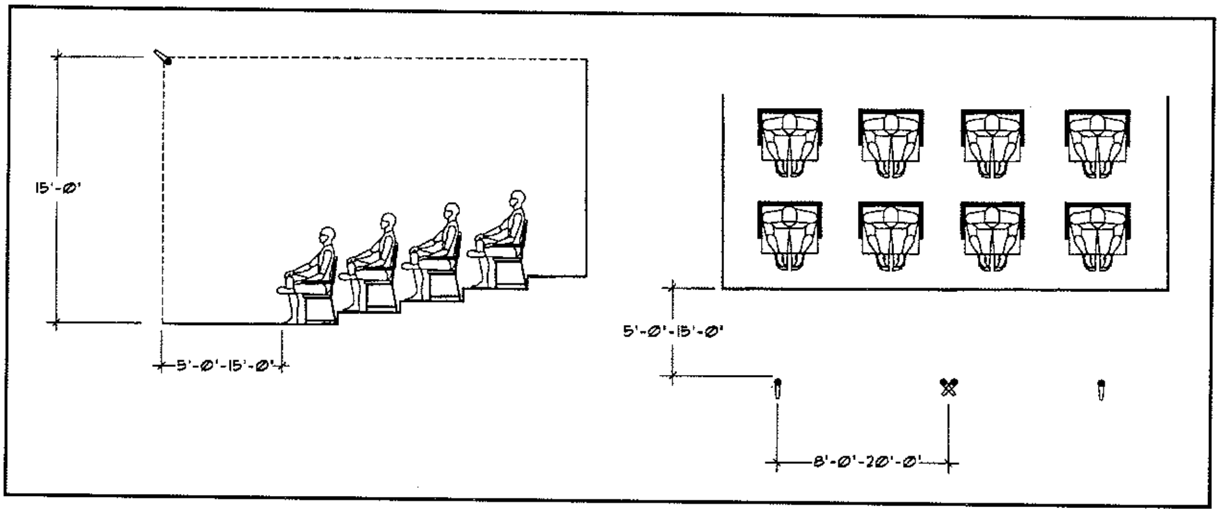


FIGURE 4-63. Recording a musical ensemble with overhead microphones. The microphones can be placed 15 ft above the floor level and between 5 to 15 ft from the performers. The plan view shows a pair of crossed cardioid microphones at the center with spaced omnidirectional microphones at the ends.

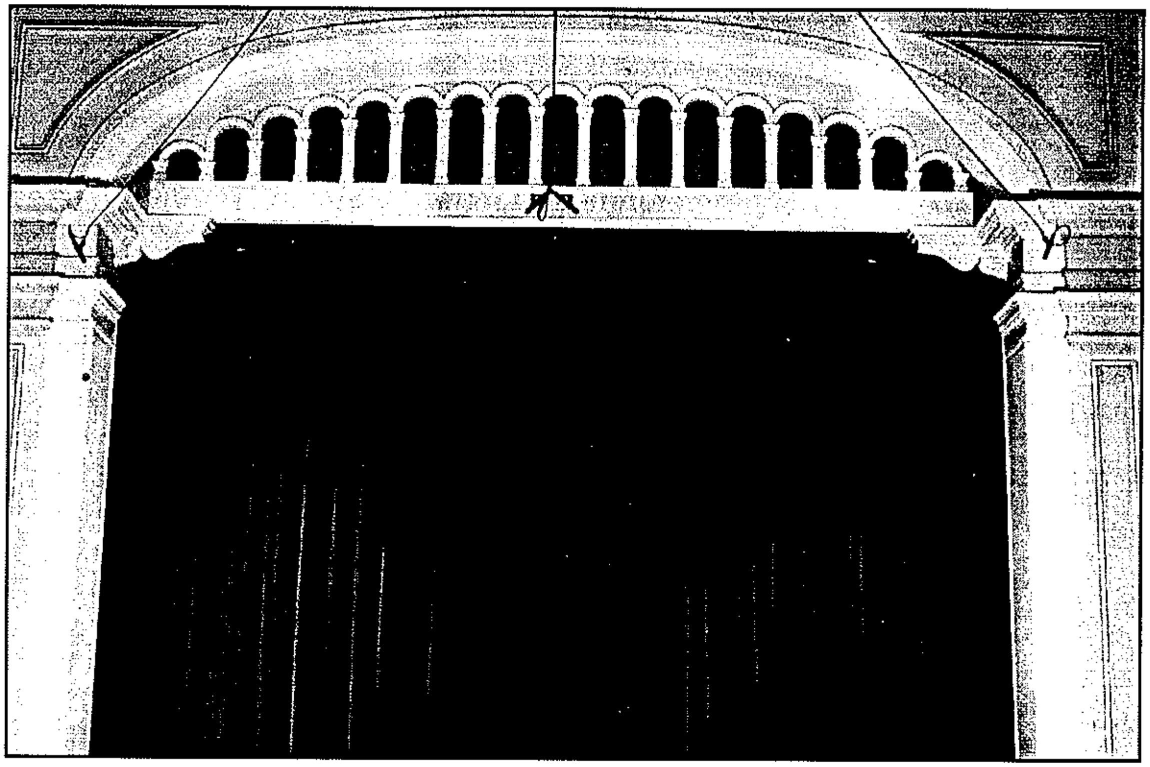


FIGURE 4-64. Recording microphones suspended from the ceiling of the Friedburg Concert Hall at the Peabody Conservatory of Music in Baltimore, MD. The center is a closely spaced cardioid microphone pair (Sennheiser MKH 40) with widely spaced left and right omnidirectional microphones (Sennheiser MKH 20). Microphone cables are suspended from the ceiling with microphone hangers and clear nylon cable (visible in the photograph) used to stabilize the microphones at the side walls to prevent swaying.

4.6 Signal Delay Lines

The objective of a signal delay line is to preserve the source localization by delaying a secondary sound source loudspeaker relative to the primary direct sound source, either talker or loudspeaker. If the signal delay line connected to the secondary source is not correctly adjusted, localization is reduced and naturalness of reproduction suffers. During the design process it is common to perform calculations to determine the required signal delay time for the secondary source(s).

4.6.1 Determining if Delayed Secondary Source is Required

A delayed secondary source loudspeaker is necessary when one or more of the conditions listed below is present.

1. Under or over balcony loudspeakers are used in conjunction with a central cluster system.

2. Several loudspeakers are used to cover the audience, each positioned at least 30 ft farther from each other relative to the primary source.

3. Distributed ceiling or sidewall loudspeakers are used in a narrow room having an aspect ratio exceeding 3-to-1 (length-to-width) with the length greater than 60 ft.

4. Large audience areas are located outdoors, such as in amphitheaters, music sheds, and plazas.

5. Increased sound levels are necessary at selected audience areas remote from the primary loudspeakers.

4.6.2 Calculating Signal Delay Time

The generalized case of calculating the required signal delay time for the secondary source loudspeaker is described below and shown in Figure 4-65.

1. Determine the distance($\mathbf{D_2}$) between the main loudspeaker and the farthest listener.

2. Determine the distance($\mathbf{D_D}$) between the secondary loudspeaker and the farthest listener.

3. Determine the difference($\Delta\mathbf{D}$) in distances between($\mathbf{D_2}$) and ($\mathbf{D_D}$), i.e.,($\mathbf{D_2}$) - ($\mathbf{D_D}$).

4. Calculate the relative arrival time $\Delta\mathbf{T}$) in milliseconds between($\mathbf{D_2}$) and($\mathbf{D_D}$) using the following formula:

$$\Delta \mathbf{T} = \left[\frac{\Delta \mathbf{D}}{\mathbf{1130}}\right]\mathbf{1000} \qquad \textbf{(4.25)}$$

where,

$\Delta\mathbf{T}$ is the relative arrival time difference between $\mathbf{D_2}$ and $\mathbf{D_D}$, ms

$\Delta\mathbf{D}$ is the difference between $\mathbf{D_2}$ and $\mathbf{D_D}$, ft

1130 is the speed of sound, ft/s

1000 is a constant that converts s to ms

5. If the value of $\Delta\mathbf{T}$ is less than 30 ms, no signal delay is necessary.

6. If the value of $\Delta\mathbf{T}$ exceeds 30 ms, signal delay is recommended.

7. Determine the final value of signal delay by adding 15 ms to the value(s) of $\Delta\mathbf{T}$ exceeding 30 ms.

8. Using *in-situ* listening, fine tune the calculated signal delay settings on the signal delay line by ear to obtain the most natural sound. This will require a compromise for the closest and farthest seats in the delay zone.

The calculation procedure described above can be used for under and over balcony loudspeakers, ceiling distributed loudspeakers, and column loudspeakers.

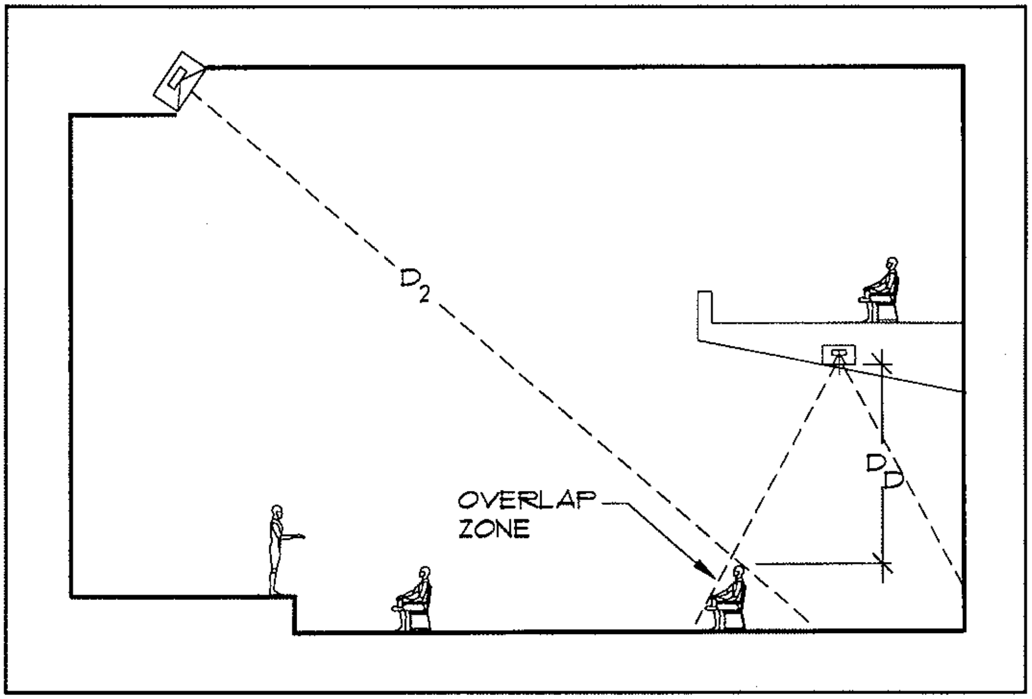


FIGURE 4-65. Geometry for signal delay calculations with nomenclature as described in text.

4.7 Power Amplifiers

The electrical power requirements of power amplifiers should be determined during the design stage. Too little power can result in low sound levels, or if the amplifier is driven into clipping, distortion and potential loudspeaker damage. Too much power can result in overly loud sound levels and possible loudspeaker damage. The required electrical power is a function of the loudspeaker sensitivity, distance between loudspeaker and listener, desired sound level, uniformity of sound level coverage, and amplifier headroom based on the program type.

4.7.1 Calculating Required Power Amplifier Output

Calculating the power amplifier output is a multi-step process and is described below using the concept of amplifier gain, similar to that of acoustic gain discussed earlier.

1. Determine the loudspeaker sensitivity (L_{SENSI}) from manufacturer's product data. Normally, this is expressed in terms of a sound pressure level (in dB) for 1 W input power measured at a 1 meter (3.281 ft) distance.

2. Determine the distance between the listener and the farthest loudspeaker (D_2) and calculate the inverse square law loss from the following equation:

$$\Delta D_X = 20\log_{10}\left[\frac{D_2}{D_{SENSI}}\right] \tag{4.26}$$

where,

ΔD_X is the change in sound level due to distance, dB
D_2 is the distance between the listener and the farthest loudspeaker, ft
D_{SENSI} is the distance used to rate the loudspeaker sensitivity, usually 3.281 ft

3. Determine the minimum program sound level (L_{PROG}) at the farthest listener location based on the ambient noise level at the listening position using the following equation:

$$L_{PROG} = L_N + 25 \tag{4.27}$$

where,

L_{PROG} is the desired minimum program sound level, dB
L_N is the ambient noise level, dB(A)
25 is a constant accounting for acceptable program **S/N** ratio

4. Account for uniformity of sound level at off-axis locations (L_{OA}). For example, if the variation in level is specified as ±2 dB, L_{OA} would be 4 dB.

5. Account for the headroom factor (L_H) based on the program type as summarized in Table 4-9.

TABLE 4-9. Headroom Factors for Different Program Types

Program Type	Headroom Factor, dB
Speech, with dynamic processing	3 to 6
Speech, no dynamic processing	10 to 12
Background or rock music, with dynamic processing	3 to 6
Classical, "Broadway," and pops, no dynamic processing	10 to 18

6. Calculate the amplifier gain using the following equation:

$$\mathbf{L_{AMP}} = (\Delta \mathbf{D_X} + \mathbf{L_{PROG}} + \mathbf{L_{OA}} + \mathbf{L_H}) - \mathbf{L_{SENSI}} \tag{4.28}$$

where,

$\mathbf{L_{AMP}}$ is the required amplifier gain, dB

$\mathbf{L_{PROG}}$, $\Delta\mathbf{D_X}$, $\mathbf{L_{OA}}$, $\mathbf{L_H}$, and $\mathbf{L_{SENSI}}$ are as above

7. Convert the required amplifier gain into the amplifier electrical power required (**EPR**) using the following equation:

$$\mathbf{EPR} = (\mathbf{P_{SENSI}})(\mathbf{10}^{\mathbf{L_{AMP}/10}}) \tag{4.29}$$

where,

EPR is the amplifier electrical power required, watts

$\mathbf{P_{SENSI}}$ is the amplifier power used in the loudspeaker sensitivity specification, usually 1 W

$\mathbf{L_{AMP}}$ is as above

The calculated value of **P** should be compared with the loudspeaker maximum power handling rating along with common sense to assess if the power rating will be within the safe operating limits of the loudspeaker. If it appears the power amplifier rating is excessive, moving the loudspeaker closer to the listener or using a compressor/limiter to reduce program headroom can help reduce the apparent required amplifier power.

4.8 System Interconnection

Audio systems are interconnected in either a balanced or unbalanced configuration, with the former preferred since it reduces pick-up of spurious electrical noise. Different signal levels are encountered in the sound system between the microphone and the loudspeaker. Some components, such as microphones, are extremely sensitive to pick-up of signals radiated by other sound system equipment or outside EMI and RFI sources.

Wherever possible, the sound system should be designed using equipment with balanced inputs and outputs. For some specific equipment items, such as consumer audio components, this may not be possible. In these cases routing the signal into a signal converter/line balancer circuit within an audio distribution, buffer, or summing amplifier is recommended.

Other techniques for reducing pick-up of noise and interference include: (1) using twisted pair cable to reduce coupling of magnetic energy; (2) not coiling excess cables to reduce the loop area susceptible to noise pick-up; (3) using cables with 100 percent braided shield wire over the conductor wire(s); (4) following proper signal grounding procedures; (5) installing cables in either thin wall electrical metal tubing (EMT) or rigid steel conduit (RSC), the former having approximately 13 dB shielding and the latter approximately 19 dB shielding; and (6) maximizing the distance between sensitive signal cables and potential noise sources.

Table 4-10 provides guidelines on recommended separation distances between cables in conduit having different sensitivity to noise and interference.

(See Technical Notes, Section 4.I, at the end of this chapter, for information on characteristics of audio, video, control, and power signals.)

TABLE 4-10. Minimum Separation Distance Between Cables in Conduit, in

Cable Characteristic	Highly Sensitive	Very Sensitive	Moderately Sensitive	Not Sensitive
Highly Sensitive	0	6	12	18
Very Sensitive	6	0	6	12
Moderately Sensitive	12	6	0	6
Not Sensitive	18	12	6	0

4.9 Equipment Racks

Audio components should be permanently installed in metal equipment racks. The equipment racks provide a means to mount equipment at ergonomically convenient operating heights, provide a means to secure equipment against theft, and interconnect signal, power, and grounding cables between equipment.

4.9.1 Location

Equipment racks are located within building spaces and normally provided with conditioned and filtered air similar for human comfort requirements. The location can be selected based on user operational needs or proximity to microphones and loudspeakers to reduce costs for conduit and cable. Some common locations include closets, storage rooms, dedicated equipment rack rooms, audio control rooms, and within the casework of occupied rooms. Equipment racks should not be installed in spaces having high humidity or sources of running water due to electrical shock hazzard and potential for equipment damage. Electrical equipment rooms such as switchgear rooms, UPS rooms, transformer rooms, power company "vault" rooms, and similar spaces should not be selected due to potential EMI interference with the audio components.

Adequate space should be provided around the perimeter of equipment racks for maintenance access and equipment ventilation with dimensions as described below.

1. **Front:** 12 in (if in a closet), 30 in (if near a wall), or 62 in (for ADA compliance).

2. **Rear:** 12 in (if in a closet and on casters), 30 in (if near a wall and floor anchored), or 62 in (for ADA compliance).

3. **Sides:** 24 in (minimum), 30 in (preferred), or 62 in (for ADA compliance). Multiple equipment racks can be ganged together at their sides to conserve space. Normally the panels on the common side where the equipment racks are joined are removed to facilitate equipment interconnection and cooling.

4.9.2 Equipment Rack Standard

Equipment racks are based on Electronic Industries Alliance (EIA) Recommended Standard 310C, "Racks, Panels, and Associated Equipment". This standard addresses common dimensions which affect the installation of equipment and compatibility of equipment rack products from different manufacturers. The standard does not address depth and height dimensions, although there are industry default sizes for equipment racks.

Two factors in the standard are the *rack unit,* used to determine the height of the equipment rack, and the 19 in spacing of the two vertical equipment mounting rails at the sides of the rack. These equipment racks are commonly referred to as 19 in racks. Many audio electronic component manufacturers produce products with 19 in wide face plates predrilled at designated rack unit spacings for direct mounting to equipment racks. Others provide accessory rack mounting "ears" which are attached to the audio components and are mounted in the equipment rack.

4.9.3 Types of Equipment Racks

Different equipment racks are available to suit a variety of equipment installation conditions. Equipment racks can be broadly subdivided into fixed permanently installed racks and portable racks. Fixed equipment racks include: (1) floor-mounted (enclosed and open styles); (2) wall-mounted; and (3) desktop-mounted. Portable equipment racks are smaller versions of enclosed equipment racks with casters or movable raised pedestals.

4.9.3.1 Fixed Equipment Racks

Fixed floor-mounted equipment racks include enclosed and open style racks. Both rack styles are of all steel construction. Enclosed equipment racks are available as fully welded or "knock-down" assemblies. The fully welded enclosed equipment rack provides a more robust assembly than the "knock-down" rack. They are initially more costly than the "knock-down" rack, but are more cost-effective when field labor to assemble the pieces is included in with the "knock-down" rack price. Open equipment racks are available only as welded assemblies.

Enclosed equipment racks are complete with ventilated side panels and a locking rear door assembly, which can be removed, or inverted 180° for left or right door swings. The side panels are integrally welded or screw-attached to the equipment rack frame. Optional locking front door assemblies with reverseable swings are also available. Conduit knock-outs, typically 1 and 2 in sizes, are provided at the top and bottom of the equipment racks. Enclosed equipment racks should be specified where there is the potential for equipment damage, unauthorized equipment adjustment, or theft. Heat build-up within enclosed equipment racks should be determined and supplemental ventilation provided where necessary.

Open equipment racks consist of open steel vertical and horizontal U-channels or square sections which are welded together to form the assembly. Side, rear, or front panels are normally not provided, although some manufacturers offer these as accessory items. Because side and rear panels are normally not used, open equipment racks are less stable and may require bolting to floor or wall surfaces. Equipment within open equipment racks is more likely to be damaged or tampered with. These racks should be installed in dedicated audio equipment rooms accessible only to technical personnel.

Exterior dimensions of fixed floor-mounted enclosed and open equipment racks range between 22 and 25 in wide by 22 and 30 in deep by 42 (21 rack units) and 84 in (44 rack units) high, depending on manufacturer. Weights, less audio components, varies between 95 and 169 lbs for enclosed equipment racks and between 40 and 65 lbs for open equipment racks. Figure 4-66 shows an enclosed equipment rack and figure 4-67 shows an open rack.

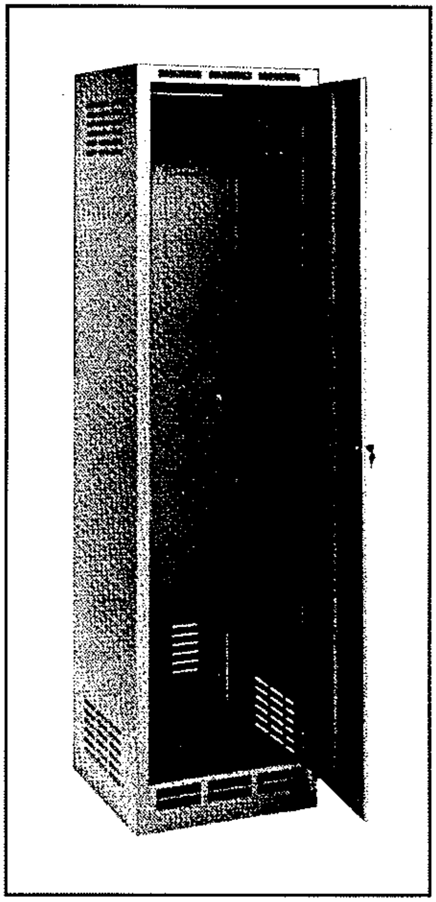


FIGURE 4-66. Enclosed equipment rack with front and rear doors (Lowell Series L65). Unit is 84 in tall and has 44 rack space units. Photo courtesy of Lowell Manufacturing Company.

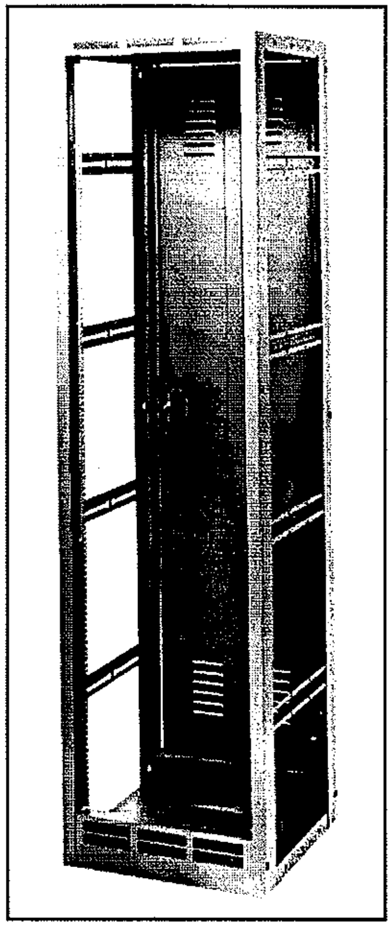


FIGURE 4-67. Open equipment rack with rear locking door suitable for ganging together with other equipment racks (Lowell Series L75). Unit is 84 in tall and has 44 rack space units. Photo courtesy of Lowell Manufacturing Company.

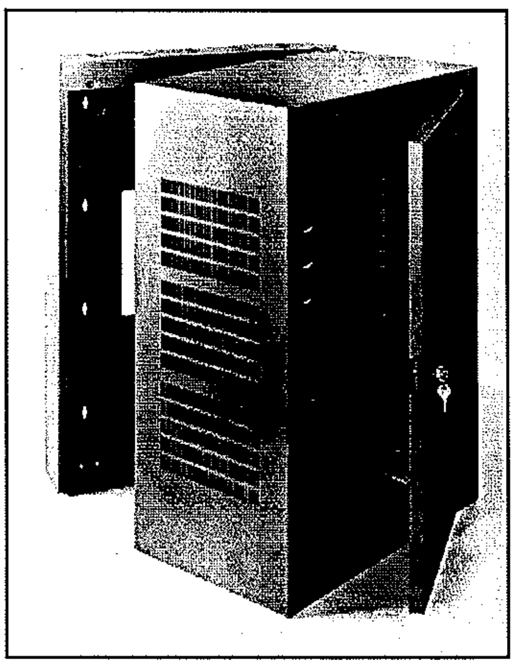


FIGURE 4-68. Wall-mounted equipment rack with front locking door (Lowell Series L50). Surface mounted back box is at rear. Unit is 45 in high and has 24 rack space units. Photo courtesy of Lowell Manufacturing Company.

Wall-mounted equipment racks use a fixed surface-mounted back box which attaches to the wall. A hinged center section cabinet with top, side, and bottom panels attaches to the back box and contains the audio components. Optional locking front doors are available from some manufacturers. Opening the equipment rack provides access to the back of the audio components and cabling. Clearance, equal to the width and depth dimensions of the center section cabinet, is required for access opening. The back box can be inverted 180° for left or right equipment rack swings. Conduit knock-outs, typically 1 and 2 in sizes, are provided at the back box top, bottom, and sides. The primary advantage with this rack type is the smaller size which permits installation in rooms where service clearance is restricted. One disadvantage with this rack type is the large combined weight of the equipment rack and the audio components may require wall reinforcement. Exterior dimensions of wall-mounted equipment racks with back boxes range between 22 and 25 in wide by 25 and 28 in deep by 15 (7 rack units) and 65 in (35 rack units) high, depending on manufacturer. Weights, less audio components, varies between 60 and 220 lbs. Figure 4-68 shows a wall-mounted equipment rack.

Desktop equipment racks are similar to miniature welded floor-mounted equipment racks. They are complete with ventilated side panels and rubber feet. Optional front and rear doors are available. Racks are available in straight front or sloped (10° to 15°) front cabinets for improved equipment visibility. Exterior dimensions of desktop equipment racks range between 22 and 25 in wide by 12 and 19 in deep by 7 (3 rack units) and 33 in (18 rack units) high, depending on manufacturer. Weights, less audio components, varies between 15 and 50 lbs. Figure 4-69 shows desktop equipment racks.

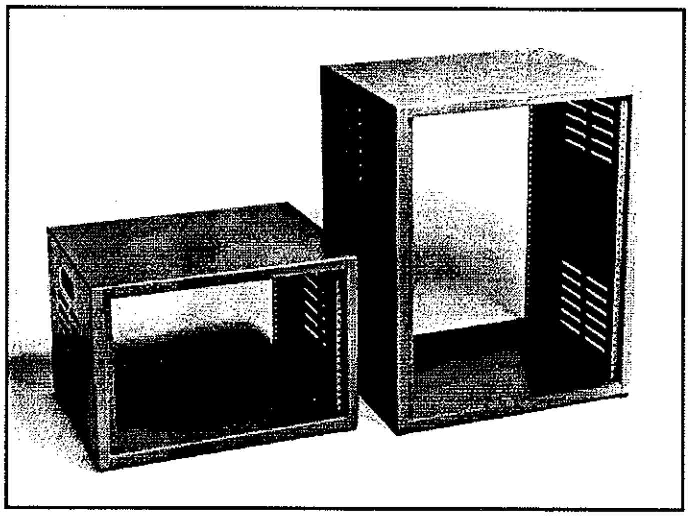


FIGURE 4-69. Desktop equipment racks (Lowell Series L70). Unit on left is 26 in tall and has 14 rack space units. Unit on right is 16 in tall and has 8 rack space units. Photo courtesy of Lowell Manufacturing Company.

4.9.3.2 Portable Equipment Racks

Portable equipment racks are commonly used to house audio components which are shared between different control rooms or operators. These equipment racks are similar to desktop equipment racks but are provided with casters (for taller racks) or moveable raised pedestals (for smaller racks). Enclosed or semi-open styles are available. Both rack styles consist of welded all steel construction. Enclosed equipment racks are complete with ventilated side panels and a locking rear door assembly. An optional locking front door is available. Semi-open equipment racks consist of open steel vertical and horizontal U-channels or square sections which are welded together with sheet metal side panels. No rear or front panels are provided.

Exterior dimensions of portable enclosed equipment racks range between 22 and 25 in wide by 22 and 25 in deep by 35 (14 rack units) and 50 in (21 rack units) high, depending on manufacturer. Weights, less audio components, varies between 85 and 120 lbs for enclosed equipment racks. Exterior dimensions and weights of portable semi-enclosed equipment racks are similar to desktop equipment racks described above. Figure 4-70 shows a portable equipment rack.

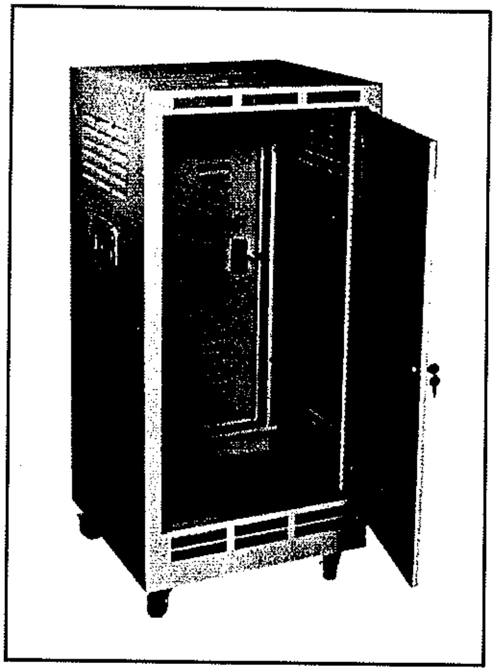


FIGURE 4-70. Portable equipment rack with front and rear locking doors (Lowell Series L58). Unit is 47 in tall and has 21 rack space units. Photo courtesy of Lowell Manufacturing Company.

4.9.4 Equipment Rack Accessories

Equipment rack manufacturers provide a variety of accessories to facilitate audio component installation. Some of these items include: (1) mounting rails; (2) casters/wheels; (3) shelves and drawers; (4) ventilation and blank panels; (5) cooling fans; (6) work lights; (7) electrical power outlet strips; (8) ground terminals/bars; and (9) conduit entrances.

Mounting rails are standard on the front of equipment racks. Rear mounting rails and accessory mounting rails are offered for some equipment racks. Rear mounting rails are identical to front mounting rails. Accessory mounting rails are installed 90° horizontal, and behind the rear mounting rails, towards the equipment rack interior. Both mounting rails are useful for attaching electrical power outlet strips, small audio components which do not require user adjustment, sliding drawers, and shelves. Another benefit of these rails is the ability to support the back of heavy audio components such as power amplifiers, which may deform the front mounting rails when only supported at the front.

Enclosed and open floor-mounted equipment racks can be supplied with casters to permit movement. Casters are available in a variety of types depending on the extent the equipment racks are to be moved. For medium-to-large equipment racks which are moved moderate distances, as between different rooms, should use front-locking swivel casters and rear fixed casters, both with 3 in diameter neoprene wheels. Small portable equipment racks can be provided with 3 in diameter neoprene swivel casters at the front and rear, with two locking-type front casters. Heavy equipment racks which are normally stationary but may require occasional movement for servicing can be provided with 3 in diameter polyolefin non-locking casters at the front and rear.

Shelves and drawers can be useful items for storing audio CDs and cassettes, equipment operation manuals, spare parts, and loose items such as microphones, cables, and headphones. A literature pocket is useful for storing equipment wiring diagrams and other frequently-used sound system documentation.

Perforated metal ventilation panels and solid sheet metal blank panels are available to install between audio components. Ventilation panels work by convection or in conjunction with forced air from a fan. Good practice is to close off unused spaces in the equipment rack and spaces above and below heat-producing items, such as power amplifiers, with ventilation panels. Solid sheet metal blank panels are installed at other rack locations and may be used for mounting switches, attenuators, and other control items. Both panel types are available in a variety of sizes, with one, two, and three rack unit sizes finding the most common application.

Cooling fans are installed on panels for direct mounting to equipment racks. They are often necessary when convection cooling is not adequate, as in the case of amplifier racks. The fans can exhaust hot air from the equipment rack or draw cooler room air into the equipment rack. Exhaust air fans are typically mounted at the top of the equipment rack since heat rises. Intake fans are typically mounted at the bottom of the equipment rack to draw cooler air in towards the warm audio components. With the latter application, it is necessary to install replaceable filters on the fans. Dusty environments may preclude the use of fans at the bottom of the equipment rack altogether. Air volume per fan is around 100 cfm and some cooling fan assemblies use two or three fans for greater airflow. Accessories such as finger guards and neoprene-in-shear vibration isolators are available.

Work lights are a convenience feature useful when the equipment rack is located in a dark room, either because of poor lighting conditions or to minimize visual distraction to the audience as in a theater control room. Work lights are available as front rack-mounted devices to facilitate viewing equipment controls. Interior equipment rack work lights are available from manufacturers.

Electrical power outlet strips are commonly installed in equipment racks to provide a means of powering audio components. Power strips comprise multiple electrical power outlets, typically between four and six duplex receptacles, within a metal enclosure. Commercial or hospital grade electrical power receptacles should be used in the outlet strips since they will maintain contact spring tension longer than residential grade devices. The electrical power outlet safety ground must be maintained to eliminate shock hazzard. This ground connection may also be used for the technical audio ground, and if defective, may make the sound system more susceptible to RFI and electrical noise interference. The electrical power outlet strips are connected to an appropriately sized panel breaker either directly through a direct conduit connection or by plugging into a wall-mounted electrical power outlet connected back to the panel breaker.

Grounding for the equipment rack can take the form of a ground bar mounted to the equipment rack bottom or a ground bus mounted to the equipment rack vertical side. Both the audio signal ground and the AC power ground from each audio electronic component connects to the ground bar or ground bus. Ground bars are cut to a suitable length for the number of audio components requiring grounding. Typical dimensions of ground bars are 1½ in wide by ¼ in thick. Ground busses run the full height of the equipment rack and are typically ¾ in wide by ¼ in thick.

Many equipment racks have conduit knockouts and cable entrances to facilitate connection of conduit and pulling cable. When openings are not the correct size or location on the equipment rack, additional openings can be installed using a metal punch. Rubber or plastic grommets should be installed in bare metal openings to prevent cutting cable insulation. When the equipment racks are not to be electrically connected to the building ground, the conduits entering the equipment racks must be isolated, since the conduits are part of the grounding system and are connected to the building steel. Accessory isolating sleeves and panels are available to isolate equipment racks from the conduit.

4.9.5 Equipment Rack Installation

Installation of audio components and rack accessories should be made based on approved shop drawings or good engineering practice. Installation guidelines are described below.

1. Determine the number and size of equipment racks needed by counting the rack space units required by all of the audio components, blank panels, and ventilation panels.

2. Verify adequate space is available to install equipment racks in scheduled locations.

3. Separate and group audio components and cables having different levels and functions (microphone, line level, loudspeaker level, control, computer, video, and electrical power).

4. Group like audio components within the same equipment rack or in the same proximity if only one equipment rack is used.

5. Locate heavy audio components, such as power amplifiers, at the equipment rack bottom.

6. Separate power amplifiers a minimum of two rack spaces away from other audio components due to potential for inducing hum from radiated magnetic fields.

7. Locate audio components which have frequently used operator controls, such as mixer units, cassette and CD players, and frequency equalizers, at eye level.

8. Locate audio components having antennae, such as wireless microphone receivers and FM ALS transmitters at the top of the equipment racks, and remotely locate the antennae from the equipment rack.

9. Install adequate ventilation panels around heat producing audio components. Use single rack unit vent panels above and below each power amplifier, unless specifically required otherwise by the power amplifier manufacturer.

10. Provide forced air cooling for equipment racks which contain power amplifiers using convection cooling or side-mounted fans.

11. Use ventilation panels or blank panels to close off unused areas at the equipment rack front.

12. Anchor all audio equipment components, ventilation panels, and blank panels to equipment rack mounting rails with screws and nylon washers between the rail and the equipment face panel.

4.9.6 Connection to Equipment Racks

Conduit and cable trays are commonly terminated at equipment racks. Less common techniques include the use of floor troughs and raised access floors. Regardless of the connection method to the equipment racks, proper interfacing is necessary to route and protect cables which are connected to the audio components.

4.9.6.1 Conduit

Conduit connects to the top, sides, and bottom of equipment racks. Ferrous metal is necessary for conduit to provide adequate EMI and RFI protection for audio cables. The most common conduit is EMT. RSC is less commonly used but may be necessary in high-EMC environments. Non-metallic tubing conduit (NMC) is not acceptable for permanent cable installation, since it provides no EMI and RFI protection. It can be used for temporary cable pulls for one-night shows and other short duration events requiring physical protection of cables.

Conduit is connected at the back of floor boxes, wall boxes, audio receptacle plates, and similar devices and routed to a large junction box. A larger conduit carries the cables from the junction box to the equipment rack. It is important that only cables of like type and signal level are routed in the same conduit and junction box. This is necessary to preserve signal integrity and minimize *crosstalk* and electrical signal pick-up between cables. Installation will usually have several conduits housing different cables routed to each equipment rack. Figure 4-71 shows conduit and conduit fittings, and Figure 4-72 shows an installation example.

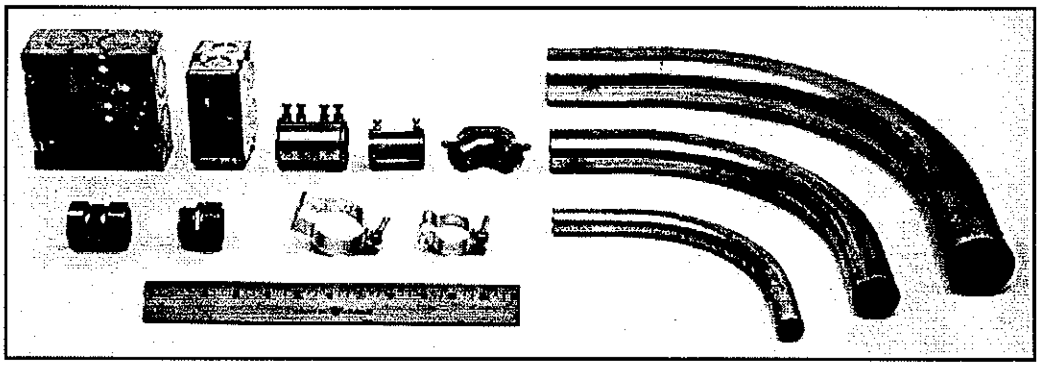


FIGURE 4-71. Metal conduit, fittings, and accessories of various sizes. Double gang and single gang back boxes (far left rear), ½ in and ¾ in set screw couplings and connectors (center rear), ¾ in, 1½ in, and 2 in diameter EMT conduit (right front and rear), ¾ in compression couplings (left front), and ¾ in and 1½ in threaded hangers (right front).

Advantages of conduit include: (1) meets most Code requirements; (2) physically protects cables from building trades and rodents; (3) provides electrical separation of cables having different signal levels; (4) relatively inexpensive cost; and (5) provides a neat and finished appearance.

Disadvantages of conduit include: (1) equipment racks must be in place to connect conduit; (2) once conduit is connected, equipment racks are difficult to move; (3) precautions must be taken to isolate equipment racks from the building electrical ground; and (4) pulling additional cables can be difficult.

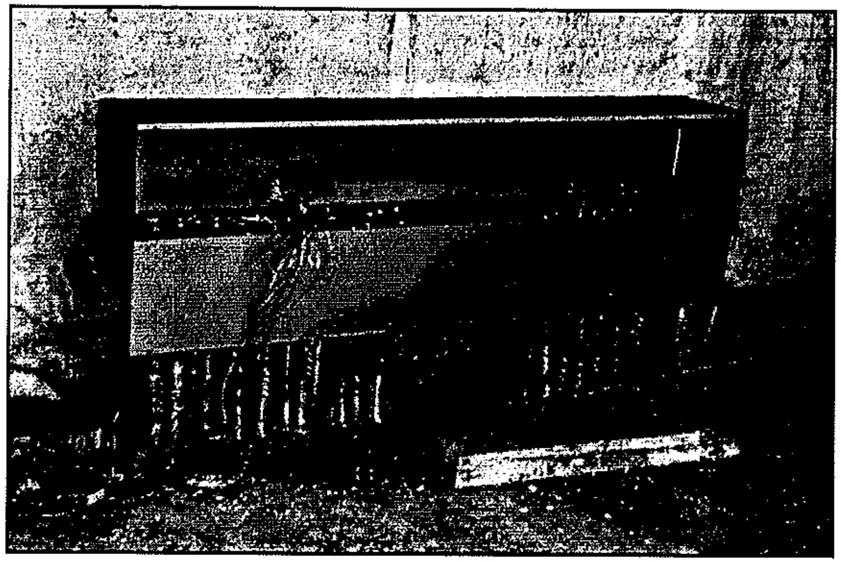


FIGURE 4-72. Conduit routed through the floor to a large wall-mounted electrical box prior to cable pulling, termination, and panel installation. System design by Tom Young of ARTEC Consultants, Inc. Photo courtesy of ARTEC Consultants, Inc.

4.9.6.2 Cable Trays

A common alternative to conduit is the use of cable trays. As the name implies, they are open trays into which cables are laid and include overhead and floor types. Overhead cable trays are supported from the structural slab above and are routed near the equipment rack top from which the cables drop down. Floor cable trays are routed on the floor surface or along the walls near the floor from which the cables rise up. Some floor cable trays are provided with metal covers to protect the cables.

Cable trays do not afford any electronic or physical protection to cables, although some products have shallow subdividers to separate the cables. Because of this, high quality shielded cable must be used and the cable tray routing needs to be carefully planned to avoid potential damage to the cables. Cable trays are often used in dedicated audio equipment rooms where large numbers of cables make conduit awkward, particularly if cables are frequently changed. Some Codes may prohibit electrical power and other high voltage cables from being routed in cable trays, so dedicated EMT conduits must be used for these cables. Figure 4-73 shows a cable tray and Figure 4-74 shows an installation example.

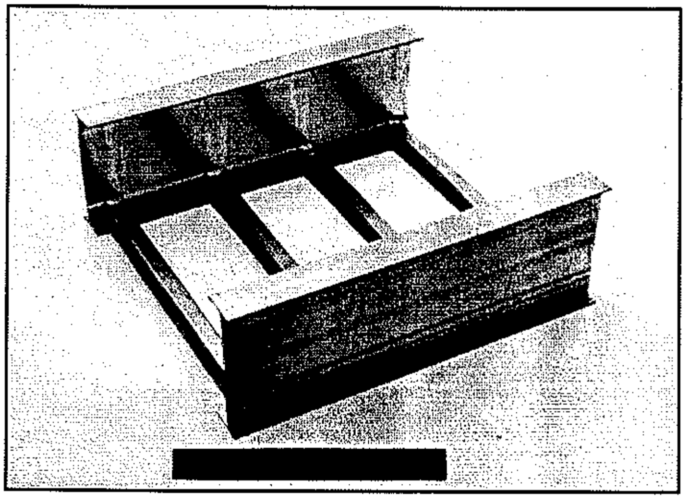


FIGURE 4-73. Section of a metal cable tray showing cross members at 8 inches on center between "I" channel profile side pieces.

Advantages of cable trays include: (1) ease of installing new and removing old cables; (2) equipment racks do not have to be in place to install the cable trays; (3) equipment racks are isolated from the building electrical ground; and (4) relatively inexpensive cost, particularly when numerous cables are to be installed.

Disadvantages of cable trays include: (1) possible signal contamination between different cables; (2) future cable installation requires care during cable placement for best signal isolation; (3) inferior electrical and physical protection compared to conduit; (4) adequate height is required for overhead cable trays and floor cable trays can be a tripping hazard; and (5) provides messy finish appearance.

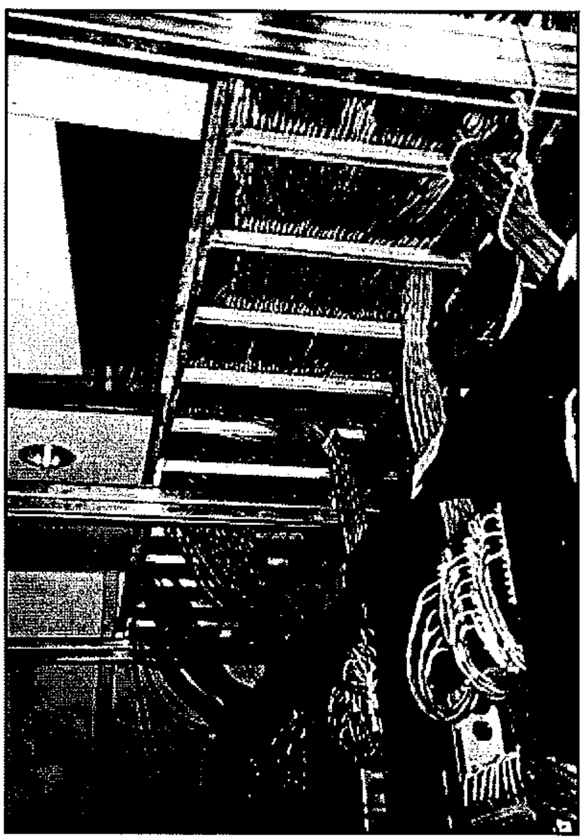


FIGURE 4-74. Overhead cable trays suspended from metal channels below the finished ceiling line with cable drops to equipment racks below within an equipment rack room. Location and system designer not known.

4.9.6.3 Cable Troughs and Raised Access Floors

Cable troughs are passageways cut into the floor which can contain conduit, cable trays, or bare cables. A removable metal cover protects the cable and provides access. Cable troughs are generally feasible only in new construction since they are customized for the installation. They require careful coordination and their use should be brought the attention of the project structural engineer early in the design process. Some floor systems, such as post-tensioned concrete slabs and acoustical "floating" floor assemblies, are incompatible with cable troughs.

Advantages of cable troughs include: (1) ease of installing and removing cables, and (2) good finish appearance.

Disadvantages of cable troughs include: (1) custom installation which can be expensive, and (2) potential for EMI and RFI with bare cables.

4.9.6.4 Raised Access Floors

Raised access floors (computer floors) comprise 24-in by 24-in heavy tiles installed on adjustable metal legs in a grid system. The floor tiles are removable from the grid to gain access to the space below. Cables may be run in conduit, cable tray, or laid directly on the floor.

Advantages of raised access floors include: (1) ease of installing and removing cables; (2) cable separation is greater than with cable trays; (3) shorter cable connections can be made between equipment racks; and (4) when the raised floor space is used as a return air plenum, portions of floor tiles near equipment racks can be removed for cooling.

Disadvantages of raised access floors include: (1) special planning is needed to assure continuous surfaces which meet other floor types and different elevations; (2) raised floor must be in place before the equipment racks are installed; (3) the equipment rack must be isolated from the raised floor to eliminate potential connection to the building ground; (4) possible sound transmission through the open raised floor plenum; and (5) raised floor is expensive.

4.10 Chapter Summary

This chapter has introduced the reader to the sound system design and installation process which includes determining functional and user requirements, calculations to predict electro-acoustical performance, evaluating one of the six major loudspeaker categories, and installation practices for both sound system items and building construction elements.

The most critical design factor is the selection of loudspeaker system and determining its placement within the room so sound is evenly distributed to the audience. Sometimes, the ideal choice from an electro-acoustical standpoint may not be desirable because of aesthetics, architectural or electrical installation difficulties, costs, and other reasons. The variety of different loudspeaker types and installation options within the six loudspeaker categories should provide the architect and sound system designer with acceptable alternate designs with a modest compromise in electro-acoustical performance. Once the loudspeaker system concept is selected, other sound system design factors such as microphones, signal processing, and power amplification need to be addressed. Finally, some sound system concepts violate established design practices but continue to be used with unfortunate consequences for the listeners. Correct installation practices can reduce susceptibility to noise and interference while resulting in a professional and workmanlike appearance.

The next chapter will cover testing and adjustment of sound systems to optimize the performance potential.

4.11 Technical Notes

4.A Determining EAD

Figure 4-A can be used to determine the **EAD** knowing the ambient noise level and the expected vocal effort of the talker.

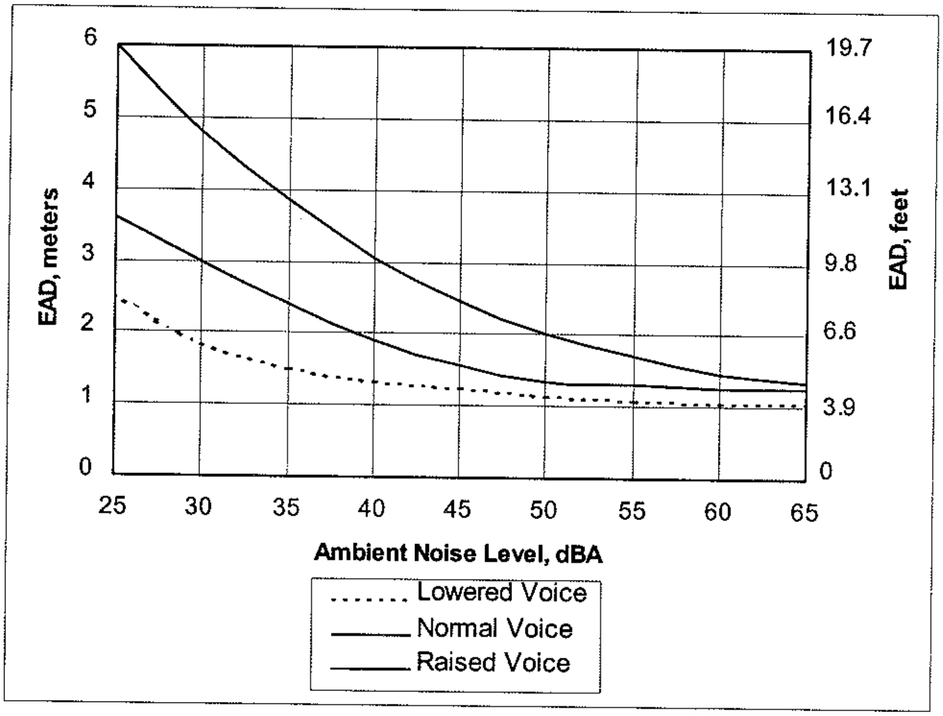


FIGURE 4-A. EAD as a function of ambient noise level and different vocal efforts at 1 meter (lowered voice, 60 dB(A); normal voice 65 dB(A); and raised voice 70 dB(A)).

4.B Efficiency (η) Values of Loudspeakers

Table 4-A provides typical η values for a variety of loudspeaker types. The values are averages of cataloged data from a variety of loudspeaker manufacturers. The tabulated η values can be used in calculations, however manufacturer's data sheets should be reviewed to determine if actual η values are available for the specified loudspeaker.

TABLE 4-A. Nominal Efficiency (η) Values for Loudspeakers

Loudspeaker Type and Size	η percent
Single Cone Drivers	
4 in	0.55
8 in	1.3
12 in	4.5
Coaxial Drivers	
4 in	0.60
8 in	1.8
12 in	2.0
12 in with Small Format Horn	4.2
15 in with Small Format Horn	4.8
Full-Range Loudspeakers	
8 in Woofer with Small Format Horn	2.1
12 in Woofer with Medium Format Horn	5.2
Two 12 in Woofers with Medium Format Horn	8.2
Compression Driver with Horn	
Consult Manufacturer's Data	25 to 30
Low-Frequency Enclosures	
12 in Woofer Ported Enclosure	3.7
15 in Woofer Ported Enclosure	5.0
Two 15 in Woofers Ported Enclosure	6.4
18 in Ported Woofer	4.0
Two 18 in Woofers Ported Enclosure	8.0
18 in Horn Enclosure	25.0

4.C Light Source Modeling of Loudspeaker Coverage

Small light sources can be used as a design aid to approximate the sound distribution in the audience seating area by a loudspeaker.

One of the first applications using light source modeling was developed jointly in 1964 by Ewart "Red" Wetherill and the late Wilfred A. Malmlund both of Bolt Beranek and Newman. The results of their study, "An Optical Aid for Designing

Loudspeaker Clusters," were published in the January 1965 *Journal of the Audio Engineering Society*.

Figures 4-B and 4-C show one of the remaining light projectors. A variable elevation "head" containing a 3 W light bulb with provisions for mounting a loudspeaker "template" are installed on a small post. The projector beams light through the template opening to produce a light pattern approximating the sound distribution of the loudspeaker. The evenness of coverage was evaluated by observing the light intensity projected on to scale models. Different templates representing various horn geometries were used to evaluate sound coverage. Direct read-out of elevation and loudspeaker aiming angles were obtained by scales on the light projector.

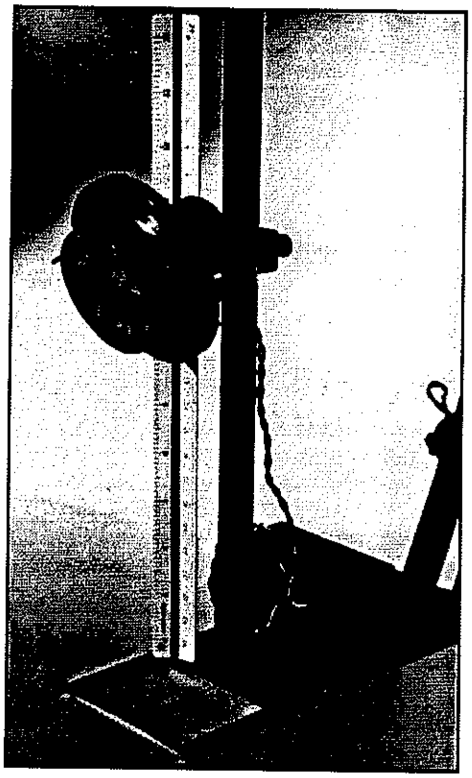


FIGURE 4-B. Light projector showing head, post, and electrical power supply. Photo courtesy of Red Wetherill, Paoletti Associates.

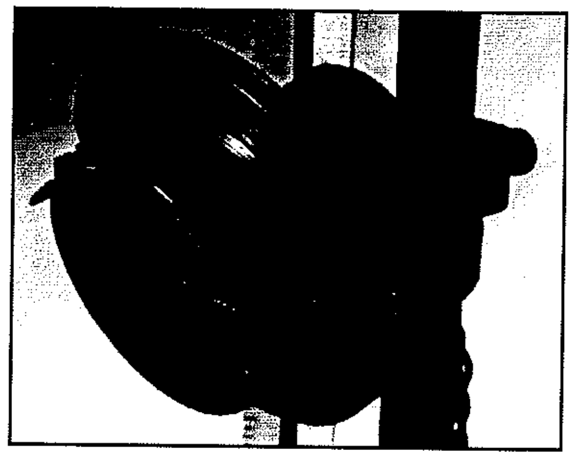


FIGURE 4-C. Close-up of the light projector head showing template for RCA 9594 loudspeaker with 60° horizontal by 35° vertical coverage pattern. Photo courtesy of Red Wetherill, Paoletti Associates.

4.D Sources of Computer-Aided Sound System Design Software

Commercially-available computer-aided sound system design software is available from several sources listed below. Many sound system equipment manufacturers offer programs specific to their products through their websites.

1. **Computerized Acoustic Theatre Technology (CATT-Acoustic®)** was developed by Dr. Bengt-Inge Dalenbäck of the Chalmers Institute of Technology in Sweden during the early 1990s. The program

initially performed room acoustics calculations only, but has been modified to import compatible data from other programs to perform a variety of sound system-related calculations. An optional program performs auralization. The current version of CATT-Acoustic® is 7.2 and is compatible with AutoCAD® for direct file import of room drawings. The program is available from: CATT, Mariagatan 16A, S-414 71 Gothenberg, SWEDEN, Tel: 46 31 145154, Web: www.netg.se/~catt.

2. **Distributed Loudspeaker Design Program** was developed by Joe Etrick, a sound system designer, to determine spacing and layout of distributed ceiling loudspeaker systems. The program is available from: Synergetic Audio Concepts, 8780 Rufing Road, Greenville, IN 47124 USA Tel: 812.923.0174, Web: www.synaudcon.com.

3. **Electro-Acoustic Simulator for Engineers (EASE)** was developed by Dr. Wolfgang Ahnert, Rainer Feistel, and Stefan Feistel of Germany. The program has undergone major revisions since its introduction in the early 1990s. The program offers an "open" database of loudspeaker electro-acoustical properties based upon 1/3 octave and 5° polar measurements. Because of its open database, EASE has become the de-facto standard sound system prediction program. A variety of room acoustics calculations are also supported by EASE, including auralization through a companion program EARS. One useful feature is the acoustical "probe" which permits access of various acoustical calculations through a point-and-click interface using a computer mouse. The current version is EASE 3.0. It is compatible with AutoCAD®. The program is available from: Renkus-Heinz, Inc., 19201 Cook Street, Foothill Ranch, CA 92610 USA, Tel: 949.588.9997, Web: www.renkus-heinz.com.

4. **GAINCALC PAG/NAG Calculator** was developed by Lectrosonics to calculate acoustical gain including PAG and NAG for single and multiple loudspeaker installations. The program permits changing the various parameters which affect acoustical gain. The program is available from: Lectrosonics, Inc., 581 Laser Road, Rio Rancho, NM 87124 USA, Tel: 515.785.6211, Web: www.lectro.com.

5. **JBL Distributed System Design (DSD™) Software** was developed by JBL Professional as a design aid for the Control Contractor series of ceiling loudspeakers. The software calculates and displays loudspeaker spacing and positioning of the Control Contractor series loudspeakers for rectangular rooms in addition to performing other electro-acoustical calculations. The program is available from: JBL Professional, 8500 Balboa Avenue, P.O. Box 2200, Northridge, CA 91329 USA, Tel: 818.894.8850, Web: www.jblpro.com.

6. **Meyer Acoustic Prediction Program (MAPP)** was developed by Meyer Sound Laboratories in the late 1990s. MAPP is a loudspeaker modeling program used to predict the direct sound coverage and interaction between adjacent loudspeakers in either the horizontal or vertical planes. All predictions include phase data and are based upon 1/36th-octave and 1° polar measurements. This data resolution is higher than EASE and as a result is useable only with Meyer products. A useful proprietary function is the ability to drag the "measurement" microphone anywhere within the modeled space and see the corresponding impulse and frequency response and in real time. This affords an effective way in to view comb filtering and lobing caused by the interaction with adjacent loudspeakers and/or room boundaries. The program can import data files from AutoCAD®. Future plans include development of three-dimensional predictions that will include field measurements of existing rooms utilizing a three-dimensional directional measurement system. The program is available from: Meyer Sound Laboratories, 2832 San Pablo Avenue, Berkeley, CA 94702 USA, Tel: 510.486.1166, Web: www.meyersound.com.

7. **Syn-Aud-Con Spreadsheet** was developed by Pat Brown of Syn-Aud-Con as a teaching aid to the sound system design seminars offered by Synergetic Audio Concepts. The spreadsheet performs a variety of acoustic, electro-acoustic, and electronic calculations useful for sound system design not offered by the above programs. The program is available from: Synergetic Audio Concepts, 8780 Rufing Road, Greenville, IN 47124 USA Tel: 812.923.0174, Web: www.synaudcon.com.

4.E Column Loudspeaker Characteristics for Different Room Reverberation Times

Table 4-B provides general guidelines on minimum column length versus coverage distance and room reverberation time for standard column loudspeakers.

TABLE 4-B. Minimum Standard Column Loudspeaker Length vs Coverage Distance and Room T_{60}

Minimum Column Length (ft)	Coverage Distance (ft)	Mid-Frequency T_{60} (s)
12	100	≥4.0
10	50	≥4.0
8	100	2.0 to 4.0
7	50	2.0 to 4.0
6	100	1.5 to 2.0
5	50	1.5 to 2.0
4	100	≤1.5
3	50	≤1.5

4.F Early Multi-Channel Sound Reinforcement

One of the first demonstrations of multi-channel sound occurred on 27 April 1933 when conductor Leopold Stowkoski adjusted the signal levels of the left, center, and right loudspeakers on stage at Constitution Hall in Washington, DC while the Philadelphia Orchestra played at the Academy of Music in Philadelphia. Three microphones were placed on stage to pick-up the Philadelphia Orchestra and the outputs were routed over telephone lines to Constitution Hall. The telephone line outputs were connected to three amplifiers each driving a dedicated loudspeaker. The loudspeakers were positioned on stage at Constitution Hall identical to the relative position of the microphones to the orchestra at the Academy of Music. The response from the public and music critics was overwhelmingly favorable to the reproduced sound quality of the Philadelphia Orchestra.

4.G Loudspeaker Cable Length for 0.5 dB Signal Level Loss

The approximate loudspeaker cable length for 0.5 dB signal loss can be calculated by rearranging Equation 4.17 as follows:

$$\mathbf{Z} = \frac{\mathbf{V}^2}{\mathbf{P}} \qquad \textbf{(4.A)}$$

where,

Z is the total load impedance, ohms

V is the constant voltage source, usually 70.7 volts

P is the sum of the calculated power drawn by all the loudspeaker transformers, watts

Using the calculated **Z** value, and an assumed loudspeaker cable gauge, look up the approximate cable length from Table 4-C. Interpolate from the table of values for different values of **Z** and the corresponding approximate cable length. The cable length values in the table are for the full loudspeaker circuit (positive and negative sides) which will be one half the physical distance between the loudspeaker and power amplifier.

TABLE 4-C. Approximate Loudspeaker Cable Length for 0.5 dB Loss

Cable Gauge, AWG	Load Impedance, Ohms				
	20	40	60	80	100
8	940	1900	2900	3800	4800
10	590	1180	1800	2400	3000
12	370	740	1120	1490	1860
14	230	460	700	930	1170
16	140	290	440	590	730
18	90	180	275	370	460
20	50	110	170	230	290
22	30	70	110	140	180

4.H Common Microphone Application Errors and Corrections

Use of too many microphones near the sound source is a common microphone application error. Figure 4-E illustrates two common errors and their solution.

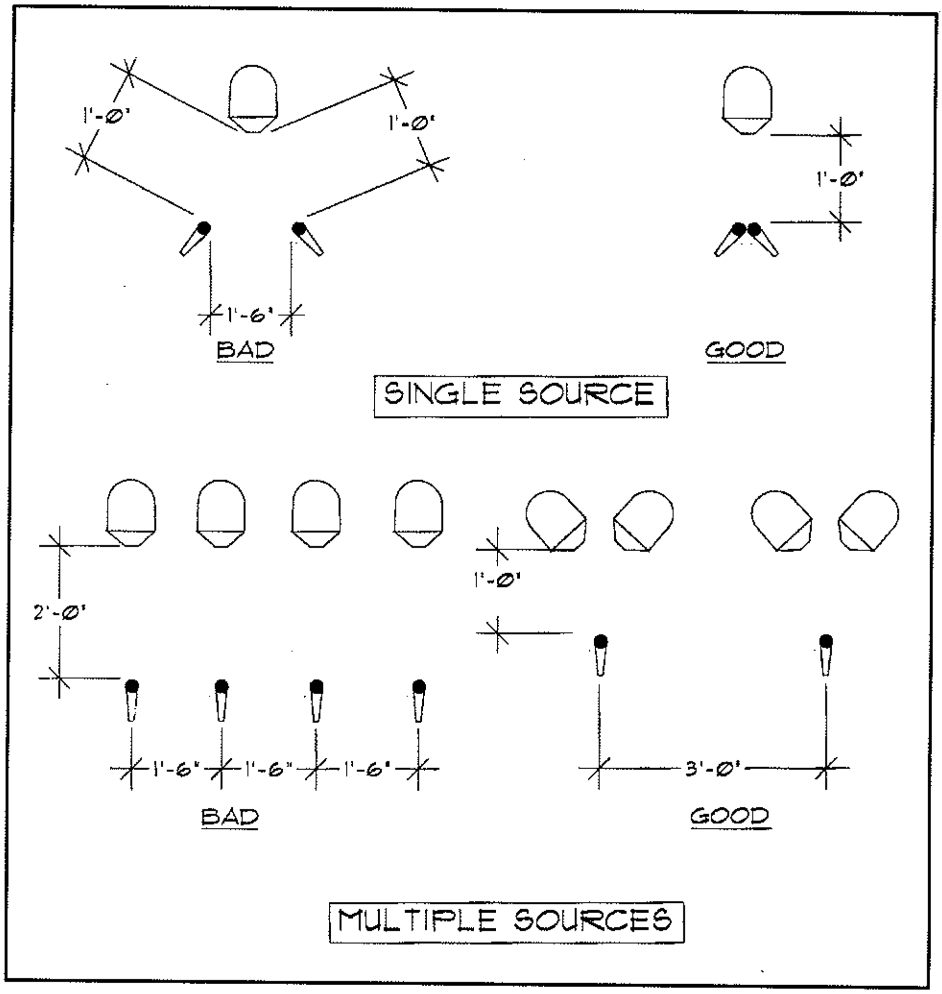


FIGURE 4-E. Single and multiple source pick-up with multiple microphones. Two spaced microphones (top left) similar to the distance between the source and the microphones, will result in comb filtering. As the talker moves relative to the microphones the character of the comb filtering will change making it more noticeable. Placing the microphone diaphragms close together (top right) will reduce the path length differential between the source and the microphones and reduce audible comb filtering. Multiple microphones and sources (bottom left) will result in comb filtering. Grouping the sources in pairs and using a single microphone per pair (bottom right) will improve the audible quality. The solutions for the single and multiple sources satisfies the three-to-one rule.

4.1 Levels of Various Audio Signals

Table 4-D provides data on different audio and video signal levels and their relative sensitivity to noise and interference.

TABLE 4-D. Characteristics of Different Audio and Video Signals

Classification	Signal Type	Voltage Range	Level Range	Relative Sensitivity
Audio	Microphone	100 to 500 mV	-17.8 to -3.8 dBu	Highly Sensitive
	Line Level	500 mV to 5 V	-3.8 to 16.2 dBu	Very Sensitive
Video	Baseband	500 mV to 2 V	-3.8 to 8.2 dBu	Very Sensitive
	Broadband	500 mV to 2 V	-3.8 to 8.2 dBu	Very Sensitive
Control	Digital	5 V	16.2 dBu	Moderately Sensitive
	Analog	24 V	29.8 dBu	Moderately Sensitive
Power	AC Mains	120 and 208 V	43.8 to 48.6 dBu	Not Sensitive

Chapter 5

Testing and Adjustment

"Somehow the arsenal of analytic tools and instrumentation never seems sufficient to win the battle of real-world performance evaluation; at least not to the degree of representing a universally accepted decisive victory."

Heyser, "Determination of Loudspeaker Arrival Times Part 1," *Journal of the Audio Engineering Society*, New York, NY (1971).

5

- Purpose of Testing and Adjustment
- Test Instrumentation
- Testing and Adjustment Procedures
- Chapter Summary
- Technical Notes

Until about 1982, sound system testing and adjustment was carried out in a haphazard manner using rudimentary equipment or almost approached a laboratory environment with the large amounts of complex equipment brought on-site. This changed in 1982 when the TECHRON Division of Crown International introduced a commercialized version of the late Richard Heyser's Time Delay Spectrometry (TDS) analyzer, the TEF®-10. This instrument gave acousticians, sound system designers, and sound system contractors the means to perform a variety of electro-acoustical measurements with a relatively inexpensive and portable instrument. Today a variety of sophisticated instrumentation is available to perform electro-acoustic measurements, many using portable computers, to automate testing and data analysis.

This chapter will provide the reader with information on the purpose of sound system testing and adjustment, test instrumentation, and an overview of subjective and objective electro-acoustical tests used to evaluate and optimize sound system performance.

5.1 Purpose of Testing and Adjustment

Sound system testing and adjustment is an integral part of the installation process. Testing should be undertaken at the individual equipment item, subsystem, and installed system levels. The purpose of the testing is to identify and correct problems before proceeding to the next installation step. A methodical testing procedure results in a system that performs and sounds better.

Individual equipment item testing has the objective to verify equipment is properly assembled and is not defective. Manufacturing errors can be as subtle as reversing polarity on the positive and negative terminals of a microphone connector or as obvious as a power amplifier self-destructing upon turn-on and use under full load. Another factor related to defective or inoperable equipment arises from hidden damage from mishandling during shipping. Verifying operational status upon receiving equipment can avoid delays in subsystem assembly should the defective condition not be discovered until the equipment is scheduled for installation in equipment racks.

Subsystem level testing includes operational verification of entire subsystems installed within the equipment racks. Commonly tested subsystems include: (1) wireless microphones; (2) signal mixers; (3) signal processing equipment; (4) power amplifiers; (5) recording equipment; and (6) ALS. The objective is to verify related equipment is correctly interconnected and functions properly as a complete subsystem prior to shipping equipment racks to the job site. Often contract documents require the acoustician or sound system designer to inspect the equipment racks prior to site delivery.

Installed system testing is normally carried out by the sound system contractor, however some acousticians and sound system designers provide this service. The measurement results are compared to specified performance criteria contained in contract documents and equipment is adjusted to meet these parameters. The sound system performance is the result of each component in the signal path being properly adjusted and operating within its design parameters. A properly adjusted sound system using lesser components will perform and sound better than a sound system of superior components not optimized in its performance potential.

5.2 Test Instrumentation

Test instrumentation can be grouped into electrical and acoustical types. Electrical instrumentation includes: (1) signal generators; (2) multimeters; (3) oscilloscopes; and (4) specialized audio test systems. Acoustical instrumentation includes: (1) acoustic calibrators; (2) microphones; (3) loudspeakers; (4) sound level meters (SLMs); (5) real time analyzers (RTAs); and (6) computerized test systems.

5.2.1 Electrical Test Equipment

Electrical test equipment is used to measure signal mixers, signal processing equipment, power amplifiers, and other electronic sound system equipment.

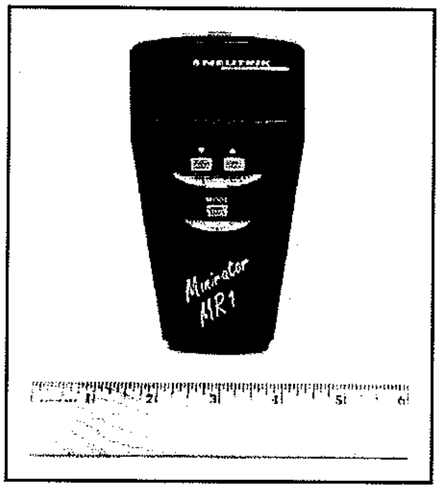


FIGURE 5-1. Portable signal generator (Neutrik Minirator MR1). Product courtesy of Neutrik USA.

5.2.1.1 Signal Generators

Signal generators provide an input stimulus signal which is used to excite the device under test (DUT). Signals can comprise fixed or swept sine waves, pink noise, *white noise*, or square (impulse) waves. An important signal generator characteristic is to have an adjustable output level so the input of connected equipment is not overloaded. The type of test signal depends upon the measurement being performed. Some test instruments have built-in signal generators, but separate signal generators are common. Figure 5-1 shows a multi-function signal generator.

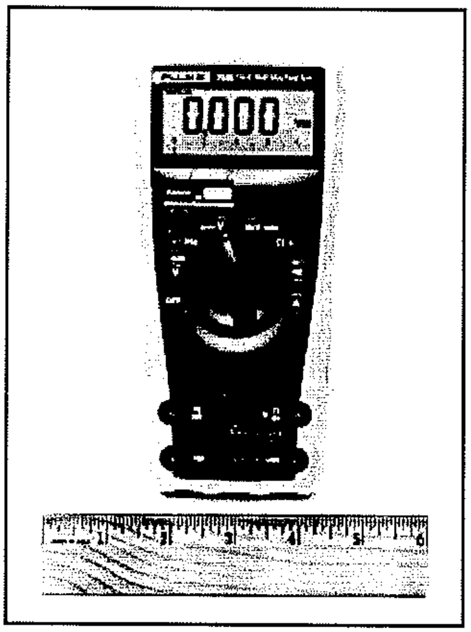


FIGURE 5-2. Basic multi-purpose DMM (Fluke 79 Series III). Product courtesy of Fluke Corporation.

5.2.1.2 Multimeters

The multimeter, sometimes called a VOM (volt-ohm-amp meter) or digital multimeter (DMM), is a multifunction electronic instrument designed to measure AC and DC voltage, current, and resistance. More sophisticated meters have the ability to measure *continuity*, capacitance, frequency, and levels using peak, peak-to-peak, and RMS scales, or directly in dB. Both analog style display meters (on VOMs) and direct digital numerical read-out display meters (on DMMs) are available, with the latter being the most common. Sophisticated DMMs provide the capability to record and store measured signals for recall or data output to a computer. An important multimeter characteristic is to have a very high input impedance which will not load the circuit under test and adversely affect the measurement results.

Common measurement applications of multimeters include: (1) AC line voltage; (2) adjusting signal levels of interconnected components for unity gain; and (3) output voltages of power amplifiers, used to calculate the power delivered to loudspeakers. Figure 5-2 shows a multi-function DMM.

5.2.1.3 Oscilloscopes

The oscilloscope, sometimes called an "O-scope" or "silly-scope," is an electronic instrument designed to measure the instantaneous voltage of a signal as a function of time. The measured signal waveform and its dynamics are displayed on an internal CRT screen. Oscilloscopes are available which work in both the analog and digital domains. The more sophisticated oscilloscopes provide the capability to record and store measured signals for recall or data output to a computer. An important oscilloscope characteristic is that it have adequate bandwidth, with 50 MHz or greater, to accurately measure RFI and other ultrahigh-frequency noise which might be present in the sound system.

Common measurement applications of oscilloscopes include: (1) voltage; (2) time, used to calculate signal frequency or evaluate phase and delay characteristics; (3) distortion; and (4) noise. Figure 5-3 shows a general purpose oscilloscope.

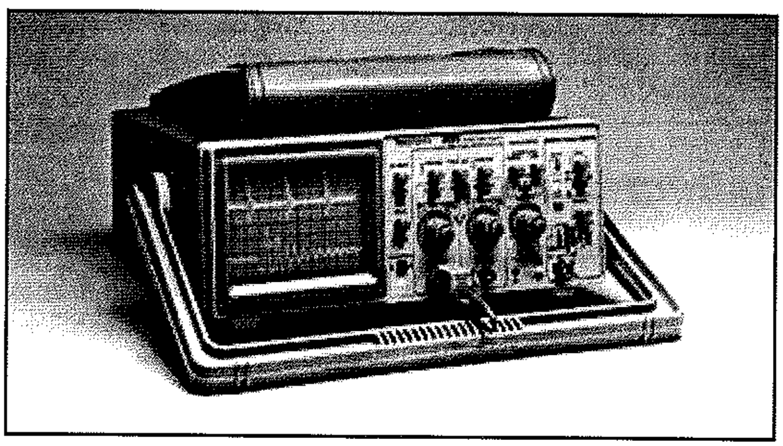


FIGURE 5-3. Oscilloscope for audio measurements (Tektronix 2225). Photo courtesy of Tektronix Corporation.

5.2.1.4 Specialized Audio Test Systems

Manufacturers such as Audio Precision, Neutrik, Rohde & Schwarz, and Tektronix produce dedicated instrumentation to measure audio equipment. Some performance parameters these devices can measure include: (1) frequency response; (2) total harmonic distortion (THD); (3) total harmonic distortion and noise (THD+N); (4) intermodulation distortion (IM); (5) S/N ratio; (6) phase and polarity; (7) crosstalk; (8) impedance; and (9) voltage level. Capabilities for storing measurement results, printing, and data transfer to computers are common.

These instruments can evaluate individual equipment items or completed subsystems to verify optimum performance and correct interfacing. Figures 5-4 and 5-5 show multi-purpose audio test systems.

FIGURE 5-4. Portable sophisticated multi-purpose audio test system (Audio Precision Portable One). Photo courtesy of Audio Precision.

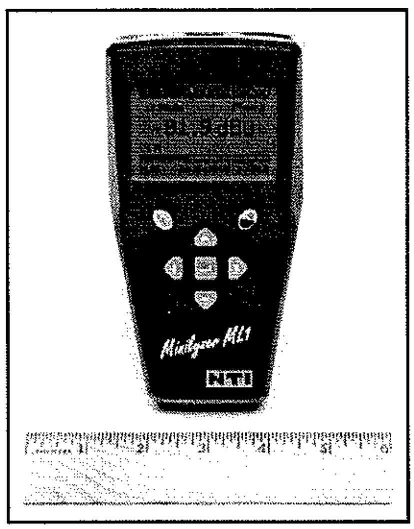


FIGURE 5-5. Handheld basic multi-purpose audio test system (Neutrik Minilyzer ML1). Product courtesy of Neutrik USA.

5.2.2 Acoustical Test Equipment

Acoustical test equipment is used to measure microphones, loudspeakers, and complete sound systems. Measurement analysis systems are classified as being single-port or two-port. Single-port systems measure the response of the DUT using an external signal generator. Two-port measurement systems generate their own unique test signal which is input to the DUT and returned back to the measurement system along with the DUT response. The analyzer then extracts the system response from the combined test and DUT response signals.

(See Technical Notes, Section 5.A, at the end of this chapter, for information on the precision of electro-acoustical measurement instrumentation.)

5.2.2.1 Acoustical Calibrators

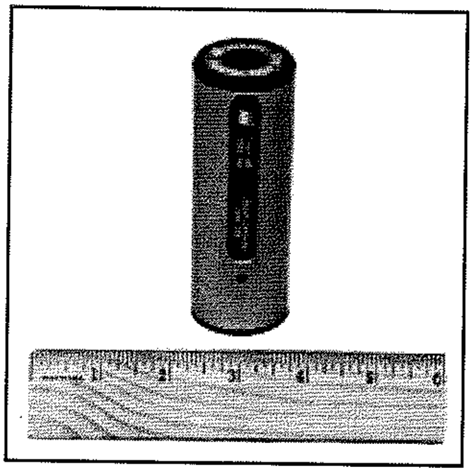


FIGURE 5-6. Type 1 acoustical calibrator (Norsonic 1251). Product courtesy of Norsonic A/S.

An acoustical calibrator is a device which is used to adjust the sensitivity of a measurement system to ensure absolute levels are measured. The calibrator consists of a signal generator and a miniature amplified loudspeaker within a robust housing. The measurement microphone is inserted into the calibrator and the unit is turned on. A reference sound pressure level is generated and amplified within the calibrator. The resulting sound level is displayed on a SLM, RTA, or other acoustical measurement system. The input preamplifier gain on the acoustical instrumentation is adjusted to correspond to the reference sound level generated by the calibrator. Common reference sound levels are 94 and 114 dB at 1,000 Hz. The 1,000 Hz frequency is used since it results in the same absolute level regardless of frequency weighting networks (A, C, or linear) which may be selected on the acoustical instrumentation. Figure 5-6 shows an acoustical calibrator.

5.2.2.2 Microphones

Measurement microphones convert acoustical test signals emitted from loudspeakers into an equivalent electrical voltage. They are classified as being free field, pressure, or random incidence types. Omnidirectional microphones having nominal ½ in diameter are commonly used because of their relatively high sensitivity and extended high-frequency response. One in microphones with higher sensitivity are used to measure low sound levels. A measurement microphone system includes a microphone capsule and a separate preamplifier to boost the microphone signal voltage level. Free field, pressure, and random incidence microphone capsules differ in both physical response to sound and measurement application. Figure 5-7 shows different microphones, preamplifiers, and windscreens.

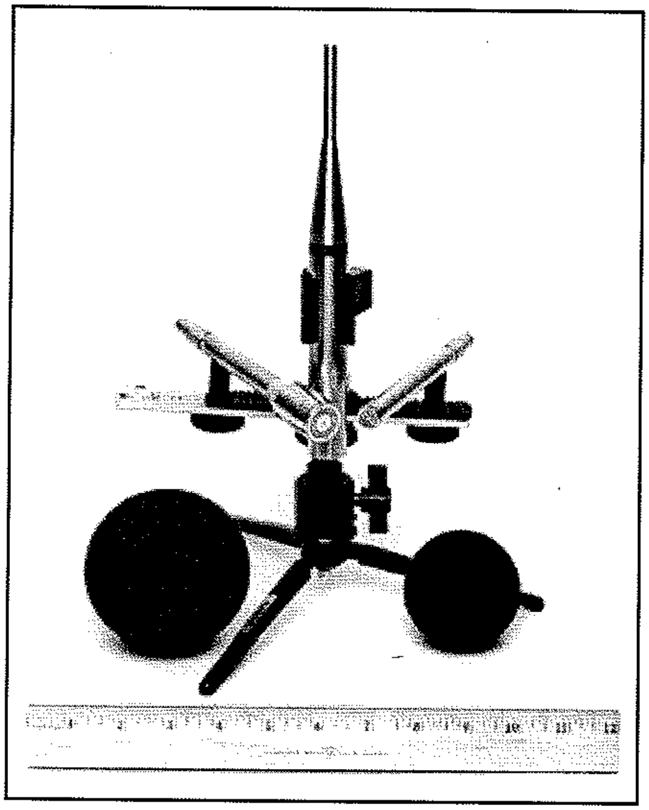


FIGURE 5-7. 1 in diameter pressure microphone (Microtech Gefell 112 microphone capsule with Norsonic 1221 preamplifier left), c in diameter random incidence microphone (Earthworks M30 center), and ½ in diameter pressure microphone and preamplifier (Norsonic 1230 microphone capsule and 1221 preamplifier right). Microphone windscreens for 1 in diameter microphone (left) and ½ in diameter microphone (right). Products courtesy of Earthworks, Inc., Microtech Gefell GMBH, and Norsonic A/S.

A free field microphone is designed to measure the sound pressure compensating for the influence of the microphone in the sound field. The microphone measures the sound pressure as it existed before the microphone was introduced in the sound field. It has a flat frequency response characteristic when pointed at a 0° angle of incidence to the sound source and when located in the free field. Free field microphones are used to measure close to the sound source or when outdoors.

A pressure microphone is designed to measure the sound pressure without compensating for the influence of the microphone in the sound field. The microphone measures the sound pressure that actually exists in front of the diaphragm. A pressure microphone will have a rising high-frequency response when pointed at a sound source. The response alteration is due to the sound field disturbance caused by the microphone at frequencies where the wavelength is smaller than the microphone diaphragm. High-frequency sound reflects from the microphone diaphragm and combines with the direct sound, resulting in a localized pressure increase. Pressure microphones are normally used to measure indoors at distant locations, where the rising frequency response compensates for high-frequency sound absorption by air, and in special acoustic calibration instruments. Measurements made using a pressure response microphone in a free field, with the microphone at 90° relative to the sound source, will approximate a free field measurement. The pressure microphone response characteristics are similar to the random incidence microphone and can be used where some sacrifice in measurement accuracy between 5,000 and 10,000 Hz is tolerable. For measurement accuracy beyond 10,000 Hz, a random incidence microphone is recommended.

A random incidence microphone, sometimes called a diffuse field microphone, is a special type of pressure microphone having a response characteristic which compensates for the influence of the microphone in the sound field. Like the free field microphone, it is designed to measure the sound pressure as it existed before the microphone was introduced in the sound field. The random incidence microphone is designed to respond uniformly to sounds arriving from all angles of incidence, as occurs in a reverberant field, and has a flat response extending beyond 8,000 Hz.

5.2.2.3 Loudspeakers

Loudspeakers are used for measuring room acoustical and speech intelligibility properties. Knowledge of room acoustical characteristics, such as the reverberation time, is needed prior to designing a sound system. Evaluating the speech intelligibility properties can indicate how well the installed sound system conveys the understanding of spoken words.

A dodecahedral (12-sided) loudspeaker is used for room acoustical measurements. The loudspeaker generates spherical sound waves resulting in greater reflected sound levels compared to a standard loudspeaker which has a rising directivity index versus frequency characteristic. The dodecahedral loudspeaker uses 4 in drivers to provide high-frequency response to approximately 8,000 Hz. At frequencies above 4,000 Hz some directional lobing occurs, with the pattern of the lobes resembling the individual driver placement. Low-frequency response to approximately 63 Hz results from mutual coupling of the drivers.

Measurements of speech intelligibility should be done using a small single driver loudspeaker approximately the size of the human head. Small multiple driver loudspeakers should not be used due to driver interactions in the crossover frequency region. The crossover for most small loudspeakers is between 1,500 and 3,500 Hz, which is in the middle of the frequency range where speech intelligibility is most affected. Some specialized speech intelligibility loudspeakers are available which more closely simulate human voice directional characteristics.

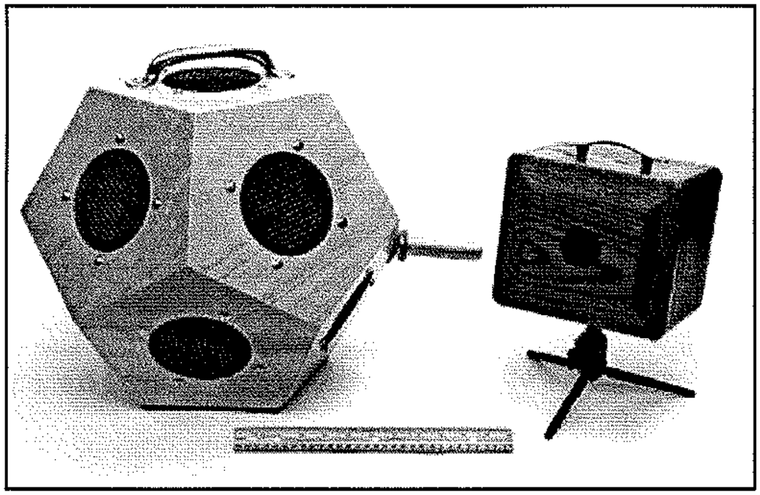


FIGURE 5-8. Dodecahedral loudspeaker (Norsonic 229 left) and voice directional loudspeaker (Acculab model 0687-H right). Products courtesy of Norsonic A/S and Acculab.

When measuring the sound system speech intelligibility, it is best to position the small loudspeaker about 1 to 2 ft from the sound system microphone and feed the input signal to the test loudspeaker. This ensures the sound system microphone response and voice directivity, simulated by the test loudspeaker, are included in the final speech intelligibility measurement. Figure 5-8 shows dodecahedral and voice test loudspeakers.

5.2.2.4 Sound Level Meters

The most basic electro-acoustical measurement instrument is the SLM. It detects pressure caused by a sound source, converts the pressure into an analogous voltage which is compared to a reference voltage, and displays a corresponding sound level in dB. The SLM comprises several component parts: (1) microphone; (2) preamplifier; (3) attenuator; (4) frequency weighting networks and optional octave and one-third octave filters; (5) logarithmic amplifier; (6) RMS detector; (7) adjustable time constants; and (8) display meter. SLMs are a single-port measurement system. Figure 5-9 shows a basic sound level meter.

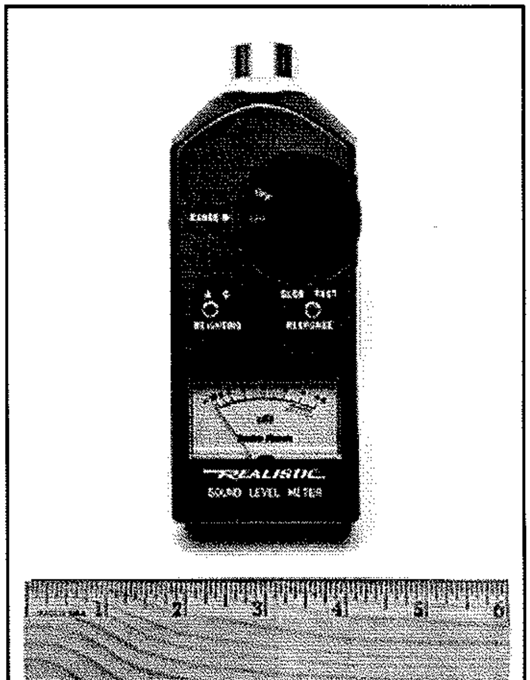


FIGURE 5-9. Basic sound level meter with 1 in microphone and A/C weighting networks (Radio Shack Model 33-2050). Product courtesy of Tandy Corporation.

The selection of microphone type and the purpose of the preamplifier have been discussed above. The attenuator is adjusted so the SLM does not overload enabling a wide signal dynamic range to be measured. Many SLMs are auto-ranging and do not require manual attenuator adjustment to match signal and meter dynamic ranges. The frequency weighting networks shape the meter response according to predefined standards. The A-weighting response is commonly used and closely correlates measured sound levels with perceived loudness. However, more detailed analysis may be required and one octave or one-third octave filters are used to subdivide the sound spectrum into smaller frequency bands. After frequency weighting, the signal is further amplified, rectified, and converted into an RMS signal making it suitable for analysis processing. The time constant enables the meter display response to approximate the changing signal dynamics. The "slow" response has a signal averaging time of about 1 second and is used for signals which do not change rapidly with time. The "fast" response has a signal averaging time of about ⅛ second and is used for rapidly changing signal dynamics.

5.2.2.5 Real Time Analyzers

The RTA, sometimes called a spectrum or *Fast Fourier Transform* (*FFT*) analyzer, is an extension of the SLM and performs signal frequency and level analysis using a series of parallel filters and the FFT. The frequency characteristics of the signal are instantaneously updated and displayed either as a continuously changing spectrum or are averaged over a specified time period to calculate an equivalent of the time varying signal. The frequency characteristics are displayed in a series of standardized frequency bands with frequency in Hz on the X axis and level in dB on the Y axis. Both single- and two-channel RTAs are available, with the latter providing more flexibility in data acquisition. RTAs are available as both single-port and two-port measurement systems. Figure 5-10 shows portable and full-featured RTAs.

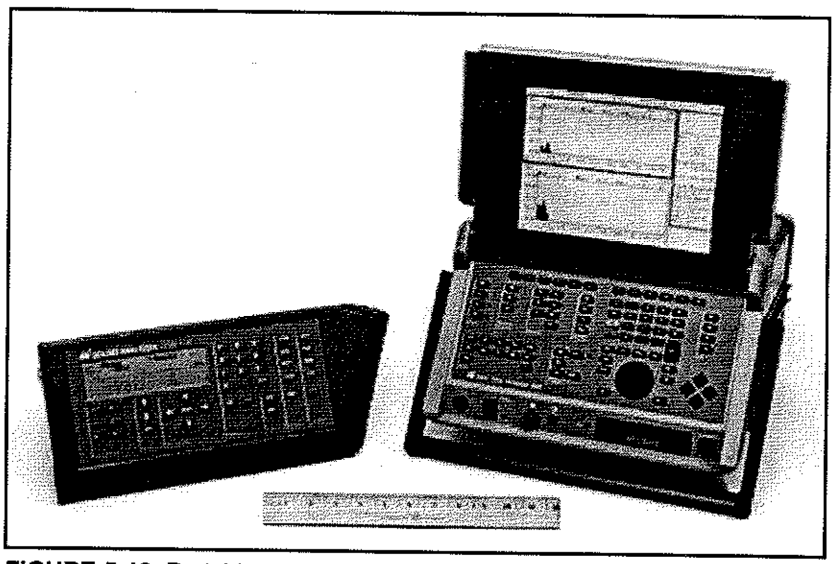


FIGURE 5-10. Portable real time analyzer (Norsonic NOR-110 left) and full-featured real time analyzer (Norsonic NOR-840 right). Products courtesy of Norsonic A/S.

The signal control and processing within the RTA shares many features of the SLM. The microphone signal is routed to a variable gain stage and then to a series of parallel *constant percentage bandwidth* filters spaced on one octave, one-third, one-sixth, or one-twelfth octave centers. The specific filter width varies between different analyzers. From here the signal passes through frequency weighting filters, rectifier stages, RMS detector, selectable time constant, and a display CRT. Most RTAs have built-in memory, computer interface, and direct printing capabilities. Additional features on more sophisticated RTAs include: (1) two data acquisition channels; (2) tabular display of frequency bands and sound levels; (3) linear and exponential averaging; (4) choice of data windows; (5) data analysis including signal differencing, *coherence*, and transfer function calculations; and (6) special features such as T60 analysis, sound intensity, and signal phase display.

(See Technical Notes, Section 5.B, at the end of this chapter, for additional information on the FFT.)

5.2.2.6 Computerized Test Systems

Computerized systems using FFT analysis techniques can perform a variety of acoustical measurements beyond those of SLMs and RTAs. Many of these analyzers perform identical measurements and share similar features, but handle and process data signals differently. At present, there is no one system which is clearly superior and selection can be based on user familiarity, measurement capabilities, or cost. Commonly used analyzers include: (1) Time Energy Frequency (TEF®) Analyzer; (2) SMAART® System; and (3) Maximum Length Sequence System Analyzer (MLSSA®).

5.2.2.6.1 TEF® Analyzer

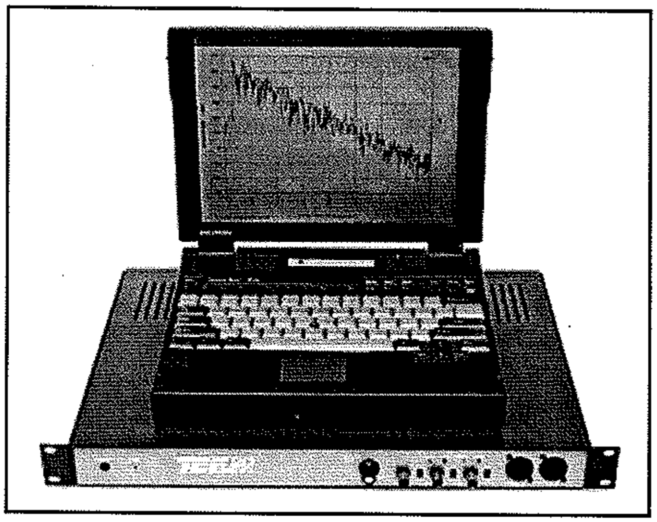


FIGURE 5-11. TEF®-20 analyzer (bottom) and Micron XPE laptop host computer (top). Product courtesy of Goldline.

The TEF® analyzer was an outgrowth of work by the late Richard Heyser who used TDS measurement techniques to evaluate audio and acoustic devices. The basic theory of TDS is described in his 1967 Audio Engineering Society paper, "Acoustical Measurements by Time Delay Spectrometry." The current version of the instrument is the TEF®-20, which is a two-port measurement system. A host computer runs companion software in conjunction with a separate analyzer module. Figure 5-11 shows the TEF®-20 analyzer.

The TEF® analyzer outputs a low *crest factor* linear swept sine wave signal which changes at a constant rate. The output from the DUT is compared with the original sweep giving DUT measurement results for the linear portion of the system response in the time and/or frequency domains. The complex frequency, phase, and amplitude response of the signals are preserved by the TDS process.

The TDS method permits a time delayed spectrum to be measured with the delay compensated for by tracking bandpass filters in the analyzer. The time delay is due to the sound propagation path between the DUT, such as a loudspeaker, and the measurement microphone. This concept can be extended to perform free field measurements in a room by windowing out sound reflections, enabling the direct sound field to be extracted from the total sound field.

The TEF® analyzer gathers data in the frequency domain, in contrast to standard FFT measurements. The test signal is applied over the entire measurement duration and inputs greater power to the DUT resulting in high noise immunity. Common TDS measurements performed by the TEF® analyzer include: (1) energy versus frequency (frequency response); (2) energy versus time (signal decay); (3) frequency versus time (frequency decay); and (4) energy versus frequency and time (three-dimensional signal decay). The analyzer is able to perform non-TDS measurements with optional software packages due to the built-in FFT analyzer.

5.2.2.6.2 SMAART® System Analyzer

The SMAART® system analyzer was developed by Sam Berkow in 1996 as a software-only measurement system designed to operate on Intel®-based PCs with Pentium® 166 or higher processors running Windows™ 95/98/NT. The software converts a standard PC into a measurement system and can be used for field measurements when installed on a laptop computer. Additional equipment necessary to perform measurements are a microphone, signal mixer, and signal source. The signal mixer is used to provide level control to the computer's internal sound card and to switch between measurement inputs. Signal sources can include test CDs or an external pink noise generator. The SMAART® software utilizes the A/D section of the host computer's sound card or an external A/D converter to transform the analog output from the signal mixer into the digital domain. The software uses 24 bit digital audio at a 48 kHz sampling rate to process data. Separate real time and analysis software modules enable a variety of acoustical measurements and analyses to be performed. The SMAART® system analyzer can function as both a single-port and two-port measurement system depending upon the measurement being performed. Figure 5-12 shows the SMAART® software running on a laptop computer.

FIGURE 5-12. SMAART® system analyzer running on laptop host computer with AudioControl Industrial, Inc. MP400 microphone preamplifier. Product courtesy of AudioControl Industrial, Inc.

The real time module is a two-channel FFT-based RTA which can measure the transfer function of the DUT in real time. Time domain data are transformed into frequency domain data by the FFT process. The transfer function performs and displays frequency and phase responses using fixed point per octave (FPPO) resolution, giving equal frequency resolution for the measurement data, in contrast to standard constant percentage bandwidth analyzers. With the FPPO resolution, most of the audio spectrum has 24 data points per octave except the lowest two octaves, which share 24 data points. Other features of the real time module enable remote control of different manufacturer's frequency equalizers and a delay locator, useful for loudspeaker signal alignment and adjustment of external signal delay lines.

The analysis module runs off line and reads standard Windows™ audio waveform (.wav) files as data input. Time and frequency characteristics of the stored data can be displayed, the latter using FFT processing. The primary function of the analysis module is to evaluate time-based parameters such as reverberation, early-late ratios, reflection audibility, and speech intelligibility calculations. Typical displays include: (1) energy versus time; (2) energy versus frequency; (3) energy versus time for a specific frequency range; and (4) energy versus time and frequency.

5.2.2.6.3 MLSSA® Analyzer

The MLSSA® analyzer was developed by Doug Rife in the late 1980s using the concept of a *Maximum Length Sequence* (*MLS*) as the test signal. The MLSSA® system comprises a full-length ISA computer card and dedicated software. The card plugs into an Intel®-based PC computer. The software is a DOS program that can run under Windows™ and permits measurements in the time and frequency domains. A microphone plugs into the dedicated card to complete the measurement system. Some of the unique measurement capabilities of the MLSSA® system include: (1) Thiele-Small loudspeaker parameters; (2) room acoustics properties in accordance with ISO 3382; and (3) speech intelligibility based on the RASTI and STI metrics. The MLSSA® system generates its own MLS test signal making it a two-port measurement system.

The MLS test signal is a pseudo-random signal similar to white noise, but having a repeating pattern of an adjustable defined length. The MLS test signal is sent out from the analyzer to the DUT and a return signal, comprising the response of the DUT and the MLS test signal, is picked-up by the microphone and input to the analyzer. The analyzer calculates the *impulse response* of the DUT by computing the cross correlation between the MLS and DUT output signals using the Fast Hadamard Transform (FHT). The correlation of the MLS signal with itself results in the impulse response of the DUT and contains the acoustical response of the DUT. The impulse response method suffers from poor S/N characteristics but can be offset by averaging a number of measurements. The MLS sequence length must be greater than the decay characteristics of the DUT to fully measure its acoustical properties.

5.3 Testing and Adjustment Procedures

Evaluation and adjustment of sound systems can be performed with a modest degree of accuracy using subjective testing procedures. However, objective measurement with acoustical instrumentation assures more accurate and repeatable results necessary to troubleshoot and correct a sound system which does not subjectively convey realistic reproduction.

5.3.1 Subjective Testing Procedures

A critical aspect of sound system testing is to listen to its audible qualities using a variety of speech and music material. Contrary to intuition, the human hearing mechanism is more revealing of system imperfections with speech as a program source than music. However, music provides greater dynamic range and frequency extension than speech. The more thorough testing procedure listed below and the brief testing procedure contained in Appendix B are suggested methods to subjectively evaluate installed sound systems.

1. **System Configuration:** The sound system should be configured in its most extensive operational mode with all microphones, line level sources, signal processing equipment, amplifiers, loudspeakers, and ALS connected and operating. Other noise-producing equipment which might adversely affect the subjective sound system response should be operating, such as HVAC equipment, lighting dimmers, video projectors, et cetera.

2. **Pink Noise Test:** Insert a pink noise source into the signal mixer and adjust the system to a comfortable listening level. Walk around the room and listen to the reproduced signal. Identify locations where the tonal quality or volume level changes compared to locations on-axis to the loudspeakers. Pay particular attention to locations where loudspeaker coverage patterns overlap.

3. **Prerecorded Speech Test:** Playback a prerecorded speech tape or CD, such as a "talking book," and adjust the sound system volume to a

comfortable listening level. The prerecorded material should be an unfamiliar topic to the listener in order to introduce unusual words and sentences. Walk around the room and listen to the reproduced signal. Identify locations where the spoken word does not seem as clear as other locations. Correlate these locations with locations identified above using the pink noise test. Once these locations have been identified go back to the locations and listen for a longer duration to confirm the initial observations.

4. **Live Talker Tests:** Using a microphone connected to the sound system, have a talker read unfamiliar source material while a listener evaluates speech intelligibility and other audible characteristics at the room locations identified as having poor response. Listen for voice naturalness, unclear speech consonants, bass and treble balance, distortion, perception of reverberation at farther distances from the loudspeakers, and ringing or hollow sounding quality indicative of the onset of feedback.

   Turn off HVAC equipment, lighting dimmers, video projectors, and other noise-producing equipment and repeat the live talker speech intelligibility testing. Note any changes in perceived audible quality. If turning off the noise-producing equipment improves the audible quality and speech intelligibility, sound masking is the likely cause, and the noise source(s) should be identified and corrected.

   Turn on all noise-producing equipment previously turned off. Disconnect the microphone, or turn off the sound system and repeat the live talker speech intelligibility testing. In larger rooms it may be necessary to use a small amplified loudspeaker to produce adequate sound levels at distant listening locations. Determine any changes in subjective speech intelligibility, naturalness, bass and treble balance, distortion, and reverberation perception without the sound system operating. If the perceived sound quality improves without the sound system, the cause(s) of the reduced sound system performance should be identified and corrected. Some things to check for are proper loudspeaker selection, correct loudspeaker aiming, and interference in loudspeaker overlapping coverage patterns.

5. **Other Program Source Tests:** Playback program sources through the various line level components to verify acceptable operational and subjective performance. Check ALS performance using a live talker with a microphone connected to the sound system. Determine that all seats are covered, have good speech intelligibility, and are free of noise and interference.

6. **Record Observations:** Record all subjective observations paying particular attention to those locations having reduced audible quality. These locations should be thoroughly evaluated using objective testing

procedures described below to determine if the sound system is performing to specification.

(See Technical Notes, Section 5.C, at the end of this chapter, for information on test materials for subjective evaluation.)

5.3.2 Objective Testing

Objective testing has goals to optimize sound system performance and identify causes of reduced performance using standardized measurement procedures. Corrective action can be taken once the causes have been identified. Measurements are performed at the equipment item, subsystem, and installed system levels.

5.3.2.1 Equipment Item Level Testing

Equipment item level testing is performed to verify equipment is in working order before assembly as part of a subsystem. The testing includes visual inspection, operation and listening tests, and limited electro-acoustic measurements. Equipment items suspected of being defective should be returned to the manufacturer for evaluation or replacement. The procedures outlined below are suggested to evaluate individual equipment items.

1. **Visual Inspection:** Visually inspect the equipment for any signs of damage. Look at packing materials for damage which could indicate rough handling in transit and result in hidden internal damage. Confirm all accessories, cords, software, and instruction manuals have been received.

2. **Operational Test:** Make a simplified sound system comprising a source, amplifier, loudspeaker, and the equipment item to be tested. Turn on the system and verify all buttons, knobs, and switches on the equipment item operate as intended.

3. **Listening Test:** Listen to the equipment item at different volume levels to confirm the absence of audible distortion and noise using a slow 50 to 15,000 Hz swept sine wave. For loudspeakers, check for rattles and buzzing at a level 20 dB below the maximum rated sound level output at 1 meter.

4. **Polarity Tests:** Verify microphones and loudspeakers have the correct polarity by using: (1) RTA with impulse response measurement; (2) impulse polarity tester; or (3) a battery to confirm transducer polarity. The RTA response should indicate a positive-going impulse followed by a negative-going impulse for loudspeakers, when the input signal is applied to the loudspeaker positive terminal, or when the impulse signal from microphone pin 2 is input to the RTA. The impulse polarity tester should result in a positive-going impulse shown on the receive

unit when a positive-going impulse is applied to the loudspeaker positive terminal or when the impulse signal from microphone pin 2 is input to the receive unit. The positive terminal of a 1.5 volt battery (for tweeters) or a 9 volt battery (for woofers) can be connected to the positive loudspeaker terminal causing the cone to move forward indicating positive polarity. Check that drivers have proper polarity at the crossover frequencies. Some loudspeaker systems purposely reverse the polarity on one driver at the crossover frequency to reduce the response dip in the crossover region and smooth the on-axis frequency response characteristics. Record the polarity of microphones and loudspeakers.

5. **Loudspeaker Impedance Tests:** Measure the impedance of each loudspeaker over its nominal passband using a constant current source or an impedance meter. The combined loudspeaker impedance and cable resistance should not be less than the rated load impedance of the associated power amplifier. Record the loudspeaker impedance ratings.

6. **Frequency Response Tests:** Verify loudspeaker and microphone frequency response characteristics. Measure the loudspeaker frequency response over its nominal passband. Measure the microphone frequency response between 50 and 15,000 Hz. Record the frequency response measurements.

5.3.2.2 Subsystem Level Testing

Subsystem level testing is performed to verify the system electrical components are properly connected, have little inherent distortion, noise, and interference, and work together as a subsystem. The testing consists of visual inspection, polarity, grounding, gain structure, **THD**, and frequency response measurements. The procedures outlined below are suggested to evaluate equipment subsystems.

1. **Visual Inspection:** Visually inspect equipment racks for correct equipment installation, cabling, and grounding. Confirm all specified equipment is installed and correctly positioned in equipment racks. Verify cables are properly terminated and grouped together by similar signal levels. Check that equipment grounding terminates at one equipment rack location.

2. **Polarity Tests:** Check the polarity of electrical equipment and cables after installation within the equipment racks. Testing should start at the power amplifier output and work backwards to the signal mixer input. Each equipment item and cable should be checked separately before moving to the next item. Check equipment in standard operating and bypass modes as some equipment inverts polarity with different settings. Verify polarity with an RTA impulse response measurement or with an impulse polarity tester with a positive pulse applied to the input connector pin 2. Cables can be checked with a VOM to verify

continuity for each pin connector or with the impulse testing methods. Record the polarity of electrical equipment.

3. **Audio Grounding:** Check balanced equipment inputs and outputs to verify proper audio grounding and eliminate potential for audio circuit *ground loops*. This can be checked through visual inspection by removing the equipment cover and looking at input and output connectors or by inserting 100 mAmp of AC electrical current into pin 1 of each connector. The internal equipment circuit ground connections should not make contact with pin 1 at balanced input and output connectors. The shields of the equipment input and output connectors should terminate at AC or chassis ground points on both ends of the connecting cables. If pin 1 problems are present, reroute the shield of balanced input and output connectors away from pin 1 and directly to the metal equipment chassis. This may require removing paint on the chassis to achieve a good electrical connection. Properly grounded equipment will not conduct ground current onto circuit boards; improperly grounded components will emit a low-level audible hum when the 100 mAmp current is applied to the input connector pin 1. Record any changes made to equipment pin 1 grounding.

4. **Gain Structure Tests:** Check the equipment gain structure of the electrical components to verify proper signal output levels. Start at the signal mixer input and work forward to the power amplifier output. Verify the input and output impedances of connected equipment have an impedance mismatch of at least 10. The output impedance of the source component should be a minimum of one-tenth the input (load) impedance of the connected component. This impedance convention will ensure the output voltage of each equipment item does not change when connected to the input of the next equipment item. Verify the input and output impedances of all equipment satisfy this criterion by checking published equipment specifications or by performing an impedance measurement.

   Set the signal mixer channel fader and master output level controls to read 0 VU on the meter when a microphone is plugged in and used at its normal working distance. Using a VOM, verify the signal mixer output voltage does not appreciably change when connected to the input of the next component. A large voltage change indicates excessive loading between the signal mixer output and the next component.

   Next adjust the output of the mixer by inserting a 1,000 Hz sine wave test signal into one of the inputs. Initially set the channel fader and master output level controls to zero. Bring up the channel fader and observe the output level on the meter. Bring up the master output level controls until the signal mixer output clips as observed on an oscilloscope. Adjust any signal mixer output trim controls to reduce the output level and permit the master output level controls to be brought up to maximum without clipping. If not, mark the maximum setting on the master output level controls. With the signal mixer output level controls set, connect the next sound system component to the signal mixer output and reduce its output level controls to zero. Connect the oscilloscope to its output. Advance the component output level controls until clipping is observed on the oscilloscope. If the output level controls are very low and clipping occurs, an attenuator pad is needed at the output of the signal mixer. Advance to the next equipment item and repeat this process for each and every equipment component item comprising the sound system until the power amplifiers are reached. Record the nominal gain settings of the equipment.

(See Technical Notes, Section 5.D, at the end of this chapter, for additional information on gain structure tests.)

5. **THD:** Verify the **THD** of the complete electronic subsystem from signal mixer input to power amplifier output is less than 2 percent for the frequencies of 50, 100, 250, 500, 1000, 2,500, 5,000, and 10,000 Hz. Connect the power amplifiers to dummy load resistors, turn on the electrical components, and adjust the system to full power output. Input a 50 Hz sine wave of less than 0.5 percent **THD** to the signal mixer input. Measure the fundamental, second, and third harmonics on an audio or real-time spectrum analyzer. Calculate the percent **THD** for the second and third harmonics. Repeat the measurements and calculations for the other frequencies. Record the **THD** measurement results.

6. **Frequency Response Test:** Verify the overall frequency response of the complete electronic system from signal mixer input to power amplifier output is a minimum of 50 to 15,000 Hz ± 1 dB as measured on an audio or real time analyzer using a pink noise or swept sine wave input. Adjust all frequency equalizers, low-pass, and high-pass filters to flat response prior to measuring the frequency response. Record the frequency response measurements.

5.3.2.3 Installed System Level Testing

Installed system level testing is performed to verify the sound system meets performance specifications and to identify site or installation conditions which adversely affect the sound system performance. The testing consists of visual inspections, extensive electro-acoustic measurements, and listening tests. The procedures outlined below are suggested to evaluate installed sound systems.

1. **Visual:** Verify that all components and cables are correctly and completely installed. Confirm initial equipment settings are correct to

include compressor/limiters, high- and low-pass filters, frequency equalizers, signal delay lines, crossovers, and equipment output levels. Record the normal equipment operation settings.

2. **Grounding:** Verify the sound system technical grounding points provide less than 0.10 W DC total resistance. Record the ground resistance.

3. **Loudspeaker Polarity and Impedance:** Measure the polarity of installed loudspeakers comprising separate components in loudspeaker clusters and distributed loudspeaker systems with an RTA analyzer or impulse polarity tester. Pole all loudspeakers identically with respect to color coding. Record the loudspeaker polarity.

   Measure the impedance of installed distributed loudspeaker systems over their nominal passband using a constant current source or an impedance meter. Verify values are within ± 10 percent of the value calculated for the circuit based upon the impedance of the connected loudspeakers and cable resistance. Record the loudspeaker system impedance.

4. **RFI/EMI Testing:** Check interconnected rack-mounted equipment to insure freedom from noise, oscillation, and RFI/EMI interference with and without an audio signal input. Disconnect the loudspeakers from the power amplifiers and connect an oscilloscope to each power amplifier output. With the grounding system operational and all electronic equipment connected, input a 1,000 Hz sine wave of less than 0.5 percent THD to the signal mixer. Adjust the power amplifier input attenuators to 10 dB below the full power amplifier output rating. Observe the oscilloscope for noise, oscillation, and RFI interference. Remove the audio signal input and observe the oscilloscope. Record observations of RFI/EMI interference.

   Check for interference from building electrical circuits by turning on and off nearby electrical equipment and lights and observe the oscilloscope. Reconnect the loudspeakers to the power amplifiers and listen in the room for audible pops or clicks as nearby electrical equipment and lights are turned on and off. Record observations of RFI/EMI noise and interference.

5. **S/N Ratio Testing:** Measure the S/N ratio of the of the complete electronic system from signal mixer input to power amplifier output. Disconnect the loudspeakers and terminate the power amplifier outputs with power resistors matching the rated loudspeaker impedance and power amplifier output. Input a 1,000 Hz sine wave of less than 0.5 percent THD to the signal mixer. Adjust the power amplifier input attenuators to 3 dB below the full power amplifier output rating. Measure the voltage level to establish a relative reference output level

for the signal input. Remove the signal source and terminate the system input with an equal load impedance. Remeasure the noise voltage level for the no input condition. Compute the S/N ratio. Alternately, use an audio test system which automatically measures the S/N ratio. Record the calculated S/N ratio.

6. **Power Amplifier Gain Structure:** Set the power amplifier gain level to produce adequate loudspeaker sound levels without clipping. Input pink noise to the signal mixer and adjust the signal mixer output level controls to read 0 VU on the signal mixer. Set the power amplifier input attenuators to zero and connect an oscilloscope to the power amplifier outputs. Carefully bring up the level of the power amplifier input attenuators until the loudspeaker sound level in the room is acceptable or the pink noise signal begins to clip on program peaks as observed on the oscilloscope. This input attenuator position should be the maximum level the power amplifiers operate at. Alternately, install an attenuator pad of appropriate value at the power amplifier input if it is desired to operate the power amplifier input attenuators "wide open" in their fully clockwise position. Record the nominal input attenuator settings.

7. **Loudspeaker Coverage and Aiming:** Adjust aiming of individual loudspeakers to optimize sound distribution coverage. Initially set the loudspeaker aiming angles as specified using a low-powered laser and protractor. Disconnect all loudspeakers except the one to be adjusted and input pink noise to the signal mixer. Maintain the power amplifier input attenuator settings determined above. Measure the sound level in octave bands between 125 and 8,000 Hz with a real time analyzer. Position the microphone 4 ft above the floor in the region covered by the loudspeaker and measure at the front, back, sides, and center of the coverage pattern. Disconnect the loudspeaker and repeat the procedure for all remaining loudspeakers. Compare the measurement results for all loudspeakers and adjust the loudspeaker aiming to achieve the most even coverage in the audience seating area. Power amplifier or crossover levels may require adjustment between low- and high-frequency drivers to achieve even sound levels. Record the loudspeaker aiming angles, power amplifier input attenuator, and crossover level settings. Upon completion of the individual loudspeaker testing, turn on all loudspeakers and measure the sound level distribution between 125 and 8,000 Hz with a real time analyzer at different locations in the room. Make minor adjustments as needed to optimize the response of the loudspeakers. Record any final adjustments to loudspeakers, power amplifiers, or crossovers.

8. **Signal Delay Setting:** Adjust the signal delay lines to provide the required time delay characteristics. Initially set the signal delay time to the physical path length difference between the main and delayed

loudspeakers. Use a TEF®, SMAART®, or MLSSA® analyzer to measure the arrival time and level of the direct sound from the main loudspeakers and the delayed sound from the secondary loudspeakers. Position the microphone 4 ft above the floor in the center of the region covered by the secondary loudspeakers. Adjust the signal delay time so the delayed sound from the secondary loudspeakers coincides with the sound from the main loudspeakers plus 15 ms for the prescedence effect. Adjust the secondary loudspeaker power amplifier input attenuators, or the signal delay line output level, so the level of the delayed loudspeakers relative to the main loudspeakers corresponds to the requirements of the Meyer and Schodder criteria for imperceptible secondary sources. Make final signal delay and power amplifier level adjustments by ear as needed for the most natural response. Record the signal delay time settings and the secondary loudspeaker power amplifier and signal delay line output level settings.

9. **System Equalization:** Adjust the frequency equalizers to provide the required equalization characteristics. Position the microphone 4 ft above the floor in the region covered by the loudspeakers and measure at the front, back, sides, and center of each loudspeaker coverage pattern. Set the initial frequency equalization using a real time analyzer to observe the combined direct and reflected sound fields. Adjust the final frequency equalization using a TEF®, SMAART®, or MLSSA® analyzer set to reject acoustic signals arriving at the measuring microphone later than 15 ms after the direct sound. Record the final frequency equalizer settings.

10. **Gain-Before-Feedback:** Measure the electro-acoustic gain of the sound system by placing a small amplified loudspeaker and cardioid microphone at the front of the room. Position each at a height of 4 ft separated by a distance of 2 ft. Connect the microphone to the signal mixer. Turn off noise sources in the room, such as HVAC systems, computers, and video projectors. Input pink noise to the small amplified loudspeaker and adjust the sound level to 65 dBA at the cardioid microphone. Position a measurement microphone about two-thirds from the front of the room, and with the sound system turned off, measure the sound level due to the amplified loudspeaker. Turn off the amplified loudspeaker and turn on the sound system, adjusting the input attenuators. Note the attenuator setting where a steady low-level feedback occurs and reduce the level by 6 dB. Turn on the amplified loudspeaker. Measure the sound level at the same room location with both the sound system and amplified loudspeaker operating. The sound pressure level increase with the sound system operating is the electro-acoustic gain. If the specified electro-acoustic gain is not met, the feedback frequencies should be determined by observing the equalized frequency response on a real-time spectrum analyzer. Notch filters should be inserted in the signal path until the feedback ceases and the

electro-acoustic gain specification is achieved. Readjust the power amplifier input attenuators to the values determined in 6., above. Record the gain-before-feedback value and the notch filter frequencies.

11. **Speech Intelligibility:** Measure the speech intelligibility of the sound system. Use the same amplified loudspeaker and cardioid microphone described in 10., above. Adjust the sound level of the small loudspeaker to 65 dBA at the cardioid microphone. Position a measurement microphone 4 ft above the floor in the region covered by the sound system loudspeakers and measure at the front, back, sides, and center of each loudspeaker coverage pattern. Use a TEF®, SMAART®, or MLSSA® analyzer set to measure the **%AL$_{CONS}$** at 2,000 Hz or the **RASTI** metric at 500 and 2,000 Hz. Record the speech intelligibility rating.

12. **Listening Tests:** Apply a slow swept sine wave from 50 to 15,000 Hz with the power amplifiers adjusted to the correct output level. Listen for buzzes, rattles, and objectionable distortion from loudspeakers and building architectural elements. Check for audible clicks or pops when operating equipment item controls and switches. Conduct listing tests using normal program material to verify quality reproduction as described under the subjective testing section above. Record observations of any distortion or noise.

5.3.3 System Acceptance Testing

System acceptance testing is performed to review the installed sound system and verify it is in accordance with specifications. The acoustician or sound system designer and owner are normally present for this testing. A "punch list" is prepared which identifies items requiring correction by the electrical and sound system contractors. The testing consists of visual inspections, operation of the entire sound system and subsystems, review of sound system contractor documentation, possible repeat of selected electro-acoustical measurements, and critical listening tests. The procedures outlined below are suggested for system acceptance testing.

1. **Visual:** The sound systems should be visually inspected for specification compliance, safety, workmanship, damage, and installation neatness. Inventory installed sound system equipment and compare to contract document equipment lists. Examine spare equipment and replacement parts inventory. Confirm dedicated conduits are provided for cables of different signal levels. Check for correct number and *ampacity* of power circuit assignments to the sound system equipment. Verify equipment, cables, connectors, audio device plates, and floor boxes are labeled as to function. Inspect equipment for safe installation practices including loudspeaker safety

cables. Check equipment rack layouts have access to frequently used controls and provide adequate equipment cooling. Confirm proper grounding for equipment in equipment racks. Check equipment, wall plates, and floor boxes are installed flush, square, and plumb. Verify proper quantity, location and aiming angles of loudspeakers. Note any damage or soiling to adjacent areas, including walls, floor, ceiling, furniture, casework, et cetera arising from installation of the sound systems. Inspect equipment installation for construction debris, dust, fingerprints, and general cleanliness.

2. **Systems Operation:** The sound system contractor should demonstrate the operation of the entire sound system and all subsystems, showing functions are as planned, specified, and desired by the owner. Check each subsystem independently and together as part of the entire system.

3. **Contractor Documentation:** Review documentation prepared by the sound system contractor to include electro-acoustic test results, as-built drawings, and operation/maintenance manuals. Test measurement results should show the performance of individual component items, subsystems, and installed systems. Note any tests which were repeated due to not meeting performance specifications and any corrective action taken. As-built drawings should indicate changes relating to substituted equipment and field conditions which resulted in the change of equipment, conduit routing, audio connectors, or audio device plates. Show changes on block and signal flow drawings, conduit drawings, and cable run and connector schedules.

   Inspect operation/maintenance manuals to include: (1) general operation procedures for the system; (2) schedule showing normal operating positions of equipment controls; (3) system block and signal flow drawings; (4) cable run and connector schedules; (5) maintenance and troubleshooting procedures; (6) manufacturer's equipment handbooks; (7) copies of sound system operating software; (8) equipment list with serial numbers; (9) spare parts list; (10) manufacturer and sound system contractor warranty items; (11) copies of certificates of use from authorities having jurisdiction; and (12) results of electro-acoustical tests.

4. **Repeat of Electro-Acoustical Tests:** The acoustician or sound system designer may request retesting of individual equipment items, subsystems, or the entire sound system, if the original testing did not meet specifications, or if modifications to the original system design by the sound system contractor would suggest retesting is necessary to validate system performance specifications.

5. **Listening Tests:** Critical listening tests should be made to evaluate the sound system in its various operational modes with all room noise

sources operating. A live talker should be used as described above in the subjective testing section. Sound systems intended for music reproduction should be evaluated with a variety acoustic and amplified music from CDs. Verify all subsystems, such as ALS and recording, provide subjectively acceptable results.

5.4 Chapter Summary

This chapter has covered the purpose, equipment, and methodology for testing, adjustment, and optimization of installed sound systems.

The human hearing mechanism is surprisingly accurate in evaluating sound system performance characteristics. Subjective listening tests can be undertaken by non-technical individuals to determine if problems exist that degrade sound system performance. Likely causes are due to defective equipment component items, system installation faults, or room acoustical conditions. Objective electro-acoustical measurements can provide the means to pinpoint the cause of problems. Methodical testing and adjustment of sound system equipment should be part of the system installation. All sound systems can benefit from testing and adjustment, which can make the system perform to its optimum potential. Often a new sound system may not be needed, only a thorough measurement adjustment "tune-up" of the existing equipment.

The next chapter will cover various documentation for sound systems, of which objective testing and adjustment results are normally included in the system manual for the users.

5.5 Technical Notes

5.A Precision of Electro-Acoustical Measurement Instrumentation

Acoustical calibrators, microphones, filter sets, SLMs, and RTAs vary in their measurement accuracy and precision. International standards organizations have classified acoustical instrumentation as Type 0, 1, 2, and 3, referred respectively as "laboratory", "precision", "general purpose", and "survey" grades. The accuracy of the instrumentation is approximately 0.5, 1, 2, and 4 dB for "laboratory" through "survey" equipment. Type 1 or 2 instrumentation are commonly used for measuring sound systems. One factor to consider when selecting measurement equipment is to choose an acoustical calibrator which is equal to, or one grade better, than the measurement equipment grade. Otherwise inaccuracies in the calibrator precision may affect the measurement equipment precision.

5.B The Fast Fourier Transform

The Fourier Transform evolved from work done by the French mathematician Jean Baptiste Joseph Baron de Fourier (1768 to 1830). Fourier determined a complicated waveform could be broken down into a series of component sine waves representing the fundamental frequency and its harmonics, each with a characteristic amplitude. Time domain data can be transformed into frequency domain data using the Fourier Transform. The Fourier Transform assumes the signal is periodic and continues infinitely, but only a portion of the signal is actually used in the analysis. Non-periodic signals have a data window superimposed to make the portion of the signal appear periodic. The windowed time signal is multiplied by sines, representing the imaginary part of the signal, and cosines, representing the real part of the signal. The complex (real and imaginary) numbers represent the frequency and phase properties of the original time signal. The resulting data set is very large and cumbersome to manipulate. A simplified process, the Fast Fourier Transform developed in 1965 by Cooley and Tukey, speeds up the calculations to show the frequency domain data in real time. Their technique reduces the number of required calculations by converting the time domain data into power of 2 samples [nlog2(n)] facilitating calculations in the digital domain by the FFT analyzer's internal computer.

5.C Test Materials for Subjective Evaluation

Suggested sources of material suitable for subjective listening tests are listed below.

Test CDs

Denon Audio Technical CD (CD number 38C39-7147) provides electronic test signals and recorded voice material.

Prosonus Intelligibility and Measurement Test Disc (CD number DIDX 010066) provides electronic test signals, spoken word lists, and modified rhyme lists.

Sound Quality Assessment Material (CD number 422 204-2) provides instrumental, singing, and spoken word material developed by the European Broadcasting Union.

Syn-Aud-Con Test CD for Sound Reinforcement Systems (CD number 1604) provides electronic test signals and recorded voice material.

Talking Books

Prerecorded audio tape or CD "talking books" are available from large booksellers and offer a variety of topics which can be useful for evaluating sound systems.

Books

The Dr. Seuss children's books provide nonsensical sentences using a variety of quasi-phonetically balanced words. Careful listening is required to distinguish between rhyming words and changing pentameters.

Narrative books by Buckminster Fuller, Richard Feynman, Steven Hawking, and Thomas Pynchon are excellent sources of material generally not familiar to most people. Careful listening is required to fully comprehend concepts and phrases.

The editorial and business sections of the daily newspaper provide a source of well-written and unfamiliar material which is readily obtainable.

5.D Alternate Gain and Level Testing Equipment

An alternate test tool for adjusting sound system gain was developed by Pat Brown of Synergetic Audio Concepts. The tool comprises a signal generator with a 400 Hz sine wave output and a piezoelectric transducer. The signal generator is connected to the DUT input and the transducer is connected to the DUT output. Increasing the DUT input or output level will cause the output signal to clip, resulting in harmonic distortion, which will be reproduced by the piezoelectric transducer. The DUT gain control should be reduced to make the clipping distortion inaudible which will maximize the electronic component gain. Each electronic component in the signal path should be checked to maximize its gain setting. Figure 5-A shows the gain adjustment system.

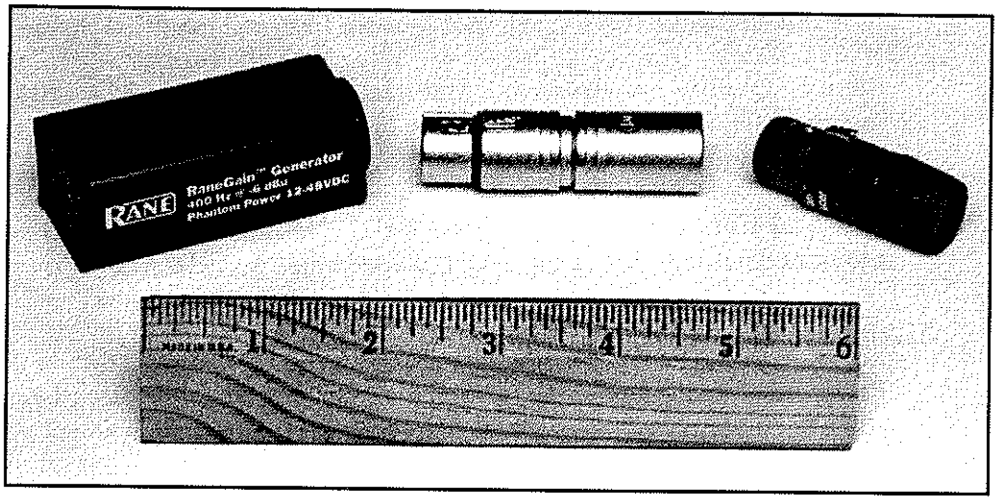


FIGURE 5-A. Portable test set for setting gain levels in sound system equipment (Rane RaneGain) comprising 400 Hz signal generator (left), male-to-female XLR adaptor (center), and piezoelectric transducer (right). Product courtesy of Rane Corporation.

Chapter 6

Design and Construction Documentation

"When the system design has been completed and the equipment list drawn up, a document must be produced which, when acted upon by an equipment installer will result in a system which fulfills the design. Most often, this document is in the form of a specification for the speech reinforcement system. This specification is the basis for a contract between the sound system installer and the building owner and must, therefore, state completely all factors pertinent to the installation which will affect the system performance, including required equipment and methods."

Kaye and Klepper, "Sound System Specifications," *Journal of the Audio Engineering Society*, New York, NY (1962).

6

- Documentation Overview
- Documents Prepared by Designers
- Specifications
- Design Drawings
- Documents Prepared by Contractors
- Miscellaneous Documentation
- Chapter Summary
- Technical Notes

Preparation of thorough construction documentation is an important responsibility for all parties involved in the design and installation of sound systems. More disputes, often resulting in unnecessary legal action, can be attributed to errors, both real and alleged, due to poorly prepared construction documentation.

All projects do not require the same level of documentation preparation. A large new performing arts center will have an extensive set of construction documents prepared by the architect/engineer (A/E) and specialty consultants with a similarly sized set of contract submittals prepared by the contractors. A simple renovation of an existing worship house sound system may only require a brief outline specification by the sound system designer, a list of supplied equipment prepared by the sound system contractor, and a one page form of agreement between the owner and sound system contractor outlining financial obligations.

This chapter will provide the reader with an overview of construction documentation to include design and contract documents prepared by the A/E and specialty consultants, submittals and shop drawings issued by the contractor, and miscellaneous construction documents.

6.1 Documentation Overview

Construction documentation is prepared by both the designers and the contractors. The designers indicate what work the contractor is to provide in the form of "bidding documents". The contractor provides documentation on equipment and systems it installs.

A common misconception exists that the project drawings and specifications comprise the "construction documents". These elements only define the scope of work the contractor is to provide. The bidding documents comprise, a set of additional contract documents intended to define the requirements for bidding, procurement, contract conditions, schedule, and outlining the responsibilities of the owner's representative, A/E, specialty consultants, general contractor, and specialty subcontractors. The bidding documents are legally binding between all parties involved in the design, installation, and purchasing of the specified equipment and systems.

During the course of construction it may be necessary to modify the design due to unforseen site conditions, unavailability of specified equipment, decisions on the part of the owner, or other unanticipated conditions. Revisions to the contract documents are formalized in the form of construction change directives. Work not fully understood by the contractor can be clarified by issuing a request for information (RFI) to the A/E or specialty consultant.

Contractors are required to provide shop drawings to show the equipment and systems to be installed are indeed equal or comparable to that specified. Upon approval by the A/E or specialty consultants, the contractor can install the selected equipment. Prior to final acceptance of the work, the contractor prepares operation and maintenance manuals for the equipment and systems and conducts acceptance tests, which are summarized in test reports. A warranty is issued by the contractors to the owner for materials and workmanship.

Many professional organizations including the American Institute of Architects (AIA), Associated General Contractors of America (AGCA), Construction Specifications Institute (CSI), and Engineers Joint Contract Documents Committee (EJCDC) have prepared sample contract forms for various phases of the construction procurement process. Where possible these documents should be used since they have been developed by consensus and are recognized as fair and legally binding contracts for all parties concerned.

6.2 Documents Prepared by Designers

The bidding documents prepared by the A/E and specialty consultants comprise the elements listed below. The objectives are to provide readable, fair, and enforceable documents which fully describe the contract conditions, standards, equipment, and installation procedures to be performed by the contractor.

1. **Contract Documents:** Comprise contract forms, contract conditions and modifications, drawings, and specifications.

2. **Project Manual:** Comprise the contract documents listed above and bidding requirement information.

3. **Addenda:** Comprise modifications to the contract documents and project manual issued during the bidding phase.

6.2.1 Contract Documents

The contract documents comprise forms to be completed by the contractor and graphical and narrative descriptions of the work to be performed. The forms include the legal agreement, performance bond, and payment bond.

The legal agreement outlines the responsibilities of the various parties in the construction process and makes use of industry standard documents. The performance bond in an insurance policy which provides financial protection to the owner should the contractor not complete the work. The payment bond is an insurance policy which protects the labor force and material suppliers from non-payment by the contractor and protects the owner from these parties demanding direct payment by the owner.

The contract conditions and modifications define the rights and responsibilities of the parties involved in the construction process and comprise general conditions and supplementary conditions. Industry standard documents are used to execute contract conditions. The general conditions address how a project is to be administered, including payment for work, changes in the scope of work, and dispute resolution. The supplementary conditions include insurance requirements and information on progress payments and liquidated damages clauses.

The drawings are the graphic representation of the work to be performed showing the relationship of materials and components to each other, sizes and shapes, locations, connections, and other information on how the elements are constructed. Notes, equipment schedules, and other information may be included on the drawings to clarify the design.

The specifications narratively describe the characteristics of particular items, products, materials, systems, and quality of workmanship which the design is based on. The specifications are subdivided into divisions which cover a specific type of construction. Each division is further subdivided into sections which address particular materials or products.

6.2.2 Project Manual

The project manual includes the contract documents and bidding information. The contract documents have been described above. The bidding information includes an invitation to bid, instructions and information for bidders, bid forms, and bid bond.

The invitation to bid provides information on the project scope and schedule, locations where bidding documents can be viewed, bid security requirements, contractor pre-qualifications, and information on applicable laws or regulations.

The instructions and information for bidders outlines the requirements for preparation and submitting bids, along with general information, planning studies, and investigative reports relating to the project.

The bid forms are documents which the bidder signs confirming acceptance of the terms of the bid and outlines the bid price for the work.

The bid bond is a payment guaranteeing owner that the bidder will enter into a formal contract with the owner should the owner accept the bid.

6.2.3 Addenda

Addenda are documents issued during the bidding period which modify the construction documents and may include revising, adding, or deleting work the contractor is to perform. Questions raised by the contractor during the bidding which are clarified by the A/E and specialty consultants are also issued as addenda.

6.3 Specifications

Specifications prepared by the A/E and specialty consultants describe the work to be performed by the contractors including the performance characteristics of the equipment and the quality of workmanship, both at the equipment item and system levels. Specifications can be prepared using industry accepted specifications or custom specifications prepared by the A/E or specialty consultants. Methods of specifying include (1) proprietary; (2) reference standard; (3) performance; (4) descriptive; and (5) custom.

6.3.1 Industry Standard Specifications

Industry standard specifications include (1) MASTERSPEC prepared by the AIA; (2) SPECTEXT prepared by CSI; and (3) Federal Construction Guide Specifications prepared by various government agencies. The advantage with the industry standard specifications is they are thorough in scope, particularly with regard to defining contractor responsibilities, which can help limit the designer's liability exposure. The disadvantage with these specifications is the general content may not reflect the latest technology associated with quality sound systems. Thus, the specifier may need to develop custom language to describe the equipment and installation.

Equipment and systems can be specified in a variety of industry standard methods as described below.

1. **Proprietary:** Specifies particular brand names, equipment model numbers, or exclusive equipment features.

2. **Reference Standard:** Specifies materials, equipment, and systems by reference to established standards without identifying particular products.

3. **Performance:** Specifies required product results and criteria by which the performance can be verified.

4. **Descriptive:** Specifies exact properties of materials, equipment, and systems with methods of installation without using proprietary names.

Proprietary specifications ensure the desired materials, equipment, and systems meet the requirements of the specifier and owner. The problem with this specification method is the contractor may not be franchised to sell or install such products. This can result in situations where a bidding contractor has to purchase the product directly from a franchised contractor, which can result in a higher cost to the owner, and potential questions on responsibility for product warranty. Government agencies normally do not permit proprietary specifications unless it can be proved that there is only one source which can provide the material, equipment, or systems which satisfy the unique technical requirements. Government procurement is based on the other specification types.

The reference standard, performance, and descriptive specifications are methods by which singular product names are not directly mentioned. These specifications are intended to define a quality of product and installation. A problem with these specifications is greater effort is required by the specifier to fully describe the product in a non-proprietary manner using a "basis of design" as a standard of performance. These specifications can often result in disputes between the specifier, who fully knows what is to be provided, and the contractor interpreting the specifications.

6.3.2 Custom Specifications

Custom specifications are prepared by the A/E or specialty consultants using some combination of the above specifications. These specifications are often the best to use for specialty materials, equipment, and systems, where the industry standard specifications may be too generic.

One feature of the custom specification is the use of the "or equal" clause or listing three known products which provide a level of performance equal to the "basis of design." The advantage with this specification method is it opens up the bidding to a variety of contractors franchised to distribute and install different manufacturer's products, resulting in the lowest installed cost to the owner.

6.3.3 Specification Sections

Specifications have a defined format regardless of the specification type selected. The specification is subdivided into three major sections: PART 1 – GENERAL; PART 2 – PRODUCTS; and PART 3 – EXECUTION. Each part is subdivided into articles and then further divided into paragraphs and subparagraphs.

The **PART 1 – GENERAL** section describes administrative, procedural, and temporary requirements of the work to be provided, related specification sections, industry standard references, quality control, acceptance, and warranty.

The **PART 2 – PRODUCTS** describes materials, equipment, and systems to be supplied, both manufactured and contractor-fabricated.

The **PART 3 – EXECUTION** describes preparatory actions, installation methods, field quality control, repair, clean-up, and demonstration of materials, equipment, and systems.

A complete specification based on the CSI format will include the sections and articles described below. Tailoring the specification to the work may result in not all of the sections and articles being used. This will require reformatting to retain sequential numbering.

6.3.3.1 PART 1 Sections and Articles

The **PART 1 – GENERAL** sections and articles include the items listed below (items in parentheses are optional).

SUMMARY
1.1 Section Includes
(1.2 Products Supplied but not Installed)
(1.3 Products Installed but not Supplied)
1.4 Related Sections

REFERENCES
1.5 Industry Standard References

(DEFINITIONS)
1.6 Definitions Unique to the Work

SYSTEM DESCRIPTION
1.7 Design Requirements
1.8 Performance Requirements

SUBMITTALS
1.9 Product Data
1.10 Shop Drawings
1.11 Samples
1.12 Certificates, Design Data, and Test Reports

QUALITY ASSURANCE
1.13 Qualifications
(1.14 Regulatory Requirements)
(1.15 Certificates)
(1.16 Field Samples and Mock-ups)
(1.17 Pre-Installation Meetings)

DELIVERY, STORAGE, and HANDLING
(1.18 Packing and Shipping)
(1.19 Acceptance at Job Site)

PROJECT OR SITE CONDITIONS
1.20 Requirements
(1.21 Existing Conditions)

SEQUENCING AND SCHEDULING
(1.22 Contractor Coordination)

WARRANTY
1.23 General Warranty
1.24 Manufacturer's Equipment Warranty

OWNER'S INSTRUCTIONS
1.25 Operation and Maintenance Manuals
1.26 Owner Training

COMMISSIONING AND ACCEPTANCE TESTS
1.27 Test Procedures
1.28 Test Reports
(1.29 Corrective Action)

MAINTENANCE
1.30 Extra Materials
(1.31 Maintenance Service)

6.3.3.2 PART 2 Sections and Articles

The **PART 2 – PRODUCTS** sections and articles include the items listed below (items in parentheses are optional).

MANUFACTURERS
(2.1 Pre-Approved Manufacturers)

EXISTING PRODUCTS
(2.2 Existing Equipment)
(2.3 Owner Furnished Equipment)

EQUIPMENT AND COMPONENTS
2.4 Equipment and Components

FABRICATION
2.5 Shop Assembly

6.3.3.3 PART 3 Sections and Articles

The **PART 3 – EXECUTION** sections and articles include the items listed below (items in parentheses are optional).

ACCEPTABLE INSTALLERS
3.1 Contractor Requirements
(3.2 Pre-Approved Contractors)

EXAMINATION
3.2 Site Verification and Conditions

PREPARATION
3.3 Preparation of Work Areas
3.4 Protection of Adjacent Areas

INSTALLATION
3.5 Installation Requirements
3.6 Interface with Other Work

REPAIR and CLEANING
3.7 Repair to the Work
3.8 Repair of Adjacent Areas

ADJUSTING
3.9 Adjustments and Calibrations

FIELD QUALITY CONTROL
3.10 Site Testing
3.11 Inspection
3.12 Manufacturer's Field Services

DEMONSTRATION
3.13 Owner Demonstration

PROTECTION
3.14 Protection of the Work

6.3.4 Sound System Specification Numbers

The CSI specification numbering format has established several specification numbers which are assigned to sound and related systems. The two broad number classifications include 11130 Audio Visual Equipment and 16770 Communications. Specific division and section numbers are based on the type of equipment and systems to be provided as described below.

There is a trend with some specifiers to group special systems, including sound systems, in a 17000 division number. The intent is to retain autonomy for these systems separate from other broader specification sections which comprise a variety materials, equipment, and systems.

6.3.4.1 11130 Audio Visual Equipment

Section 11130 in the equipment division, includes the applicable specification numbers below.

11132 Projection Screens: Includes fixed and moveable projection screens for film and video systems.

11134 Projectors: Includes film and video projector systems.

11136 Learning Laboratories: Includes modular language training laboratories complete with booths, cassette/CD playback, sound amplification, headphones, and loudspeakers.

6.3.4.2 16770 Communications

Section 16770, in the electrical division, includes the applicable specification numbers below.

16720 Alarm and Detection Systems: Includes fire alarm, smoke detection, gas detection, intrusive detection, and security access systems.

16730 Clock and Program Systems: Includes timing and control systems.

16740 Voice and Data Systems: Includes telephone, paging, call, data, local area network, door answering, microwave, radio, and intercommunication systems.

16770 Public Address and Music Systems: Includes voice reinforcement, music reinforcement, production intercom, assistive listening, program playback, and recording systems.

16780 Television Systems: Includes master antenna, video telecommunications, video surveillance, and broadcast video systems.

16785 Satellite Earth Station Systems: Includes satellite communication and data transmission systems.

16790 Microwave Systems: Includes microwave communication and data transmission systems.

6.3.5 Related Specification Numbers

Several specification numbers are part of the CSI specification numbering format which are necessary for sound and related systems. The two broad number classifications include 16050 Basic Electrical Materials and Methods and 16400 Service and Distribution. Specific section numbers are described below.

6.3.5.1 16050 Basic Electrical Materials and Methods

Section 16050 in the electrical division, includes the applicable specification numbers below.

16110 Raceways: Includes cable tray, conduit, surface *raceways*, underfloor ducts, and underground ducts.

16120 Wires and Cables: Includes fiber optic, low voltage, 600 volt and less, medium voltage, and under carpet cables.

16130 Boxes: Includes floor, outlet, pull, and junction boxes.

16140 Wiring Devices: Includes low voltage switching.

16190 Supporting Devices: Includes methods of support and attachment for electrical systems.

16195 Identification: Includes methods of identifying electrical raceways, wire, cable, and boxes.

6.3.5.2 16400 Service and Distribution

Section 16400 in the equipment division, includes the applicable specification numbers below.

16450 Secondary Grounding: Includes grounding systems.

16470 Panelboards: Includes branch circuit and distribution panelboards.

6.4 Design Drawings

The design drawings prepared by the A/E and specialty consultants graphically depict the work to be performed by the contractors. Sound system design drawings are often included in the electrical drawing section, or if a large system, may have their own drawing section. Often there are sound system design coordination items which are shown on the architectural and electrical drawings.

A thorough sound system drawing package will include: (1) block and signal flow diagrams; (2) conduit riser diagrams; (3) audio box, device plate, and connector schedules; (4) schematic symbols; (5) notes; and (6) detail drawings.

6.4.1 Block and Signal Flow Diagrams

Block and signal flow diagrams depict the interconnection of individual equipment items to comprise either a subsystem or complete system. These drawings are typically single line drawings which use schematic symbols to represent the different equipment items. Convention has the input signal sources (microphones and line level sources) at the drawing left side with the drawing right side showing the output (loudspeakers or line level connections). A narrative description or notes on the schematic drawing are keyed to other drawings to provide additional information.

6.4.2 Conduit Riser Diagrams

Conduit riser diagrams show conceptually the location, routing, interconnection, sizes, and quantity of back boxes, pull boxes, and conduit. Good design practice has the conduit riser diagrams referenced to the block and signal flow diagrams and the audio box, device plate, and connector schedules. Convention has the input and outputs, respectively, on the left and right sides of the drawing. The conduit riser drawing is used by the sound system contractor to coordinate locations of device plates and audio connectors based on the conduit and back boxes installed by the electrical contractor.

6.4.3 Audio Box, Device Plate, and Connector Schedules

The audio box, device plate, and connector schedule shows the audio function, location, device plate size, audio connector types, and plate label engraving. This information is necessary so the sound system contractor can properly fabricate device plates and terminate cables with the correct connector. Notes in the schedules can clarify specific installation requirements, such as plate label engraving and finish appearance.

Typical services include control switches, loudspeaker and ALS output plates, remote wireless microphone antenna plates, volume controls, and wall and floor mounted microphone input plates.

6.4.4 Schematic Symbols

Schematic symbols are graphic representations showing conceptually what function a device is to provide. No industry standard sound system equipment schematic symbols list is available, although several industry standards groups are working to correct this.

(See Technical Notes, Section 6.A, at the end of this chapter, for additional information on schematic symbols.)

6.4.5 Notes

Notes are provided on the drawings to clarify the design or to describe specific conditions which would affect the installation or operation of the sound systems. The notes are tailored to the sound system design and may address: (1) general information; (2) installation methods; (3) electrical system; (4) equipment interconnection and grounding; and (5) equipment adjustment. Typical drawing notes are described below.

1. **General Information Notes:** Outline common conditions for all aspects of the work to be performed and refer the contractor to other administrative or technical information contained elsewhere in the project manual.

2. **Installation Methods Notes:** Describe general or unique architectural or electrical requirements which are to be performed as part of the sound systems installation. Often this information is best included on the drawings rather than the specifications since the drawings are more commonly used by the installer.

3. **Electrical System Notes:** Clarify information relating to the location, interconnection, sizes, and quantity of back boxes, pull boxes, conduit, electrical power, and circuit requirements for the sound systems.

4. **Equipment Interconnection and Grounding Notes:** Describe cable and interconnection between different audio equipment items, audio device plates, and system grounding, both life safety and technical.

5. **Equipment Adjustment Notes:** Provide information on settings of adjustable controls or switches on equipment items and on modifications internal to the electrical components to be performed by the contractor.

6.4.6 Detail Drawings

Detail drawings show unique installation requirements for the sound systems. They may include: (1) equipment rack layouts; (2) audio device plates and connectors; (3) loudspeaker installation and aiming; and (4) other unique drawings necessary to illustrate the system installation.

1. **Equipment Rack Layout Details:** Show the location of individual equipment items, ventilation panels, and other accessories within the equipment racks.

2. **Audio Device Plate And Connector Details:** Show the sizes of audio device plates, the connector types, and label engraving as to function. These details are often drawn to full size for fabrication purposes.

3. **Loudspeaker Installation Details:** Show the support of loudspeakers, anchorage to the building structure, and any other factors which might affect installation or life safety. Aiming of the loudspeakers can be shown either in a schedule for the loudspeakers listing aiming angles or by showing the angular positioning for each loudspeaker in an isometric-type drawing.

6.4.7 Coordination with Architectural and Electrical Drawings

Architectural drawings (plan, section, and elevation) are marked-up by the sound system designer to show locations of ALS panels, loudspeakers, wall and floor plates, and other sound system items for design coordination purposes. The architect includes this information on the final architectural design drawings.

The sound system designer will normally mark-up a set of electrical drawings showing back boxes, pull boxes, and conduit, including routing, sizes, and quantity. Information provided by the sound system designer is incorporated by the electrical engineer on the final electrical design drawings. The electrical contractor normally will install all conduit, including that associated with the sound systems, and all similar work should be shown on a common drawing.

6.5 Documents Prepared by Contractors

The documents prepared by the contractors comprise the elements listed below. The objectives of the contractor submittals are to substantiate the proposed equipment meets the design intent, is installed and operates as planned, and aids the owner through defined maintenance procedures and a warranty period.

1. **Shop Drawings:** Comprises product catalog sheets, dimensioned drawings, and other information describing the equipment and systems to be provided.

2. **Operation and Maintenance Manuals:** Comprises information on equipment and systems operation along with maintenance procedures.

3. **Test Reports:** Comprises the results of proof-of-performance acceptance testing on equipment and systems.

4. **Record Drawings:** Comprises installation drawings and marked-up drawings noting changes in construction that differ from the original design.

5. **Warranties:** Provides both manufacturer's warranty for equipment and systems and contractor's warranty for installation and workmanship.

6.5.1 Shop Drawings

Shop drawings are normally required as part of the approval process for materials and equipment the contractor proposes. Submittals can take the form of extensive dimensioned drawings prepared by the equipment manufacturer or contractor to catalog sheets describing the product. Specifics for shop drawings are outlined in the relevant project specification sections.

6.5.2 Operation and Maintenance Manuals

Operation and maintenance manuals assist the user with the installed systems. A well documented operation and maintenance manual will include the following elements: (1) table of contents; (2) equipment manufacturer's specification sheets, operating instructions, and service information; (3) operating instructions for each subsystem describing the functions, operation, and maintenance written for comprehension by non-technical people; (4) troubleshooting guide for equipment and systems listing the procedures to follow in the event of equipment failure, written in logical outline form; (5) illustrative drawings to include system block diagram, connector and cable location plan and log for all tagged connectors and cables, and as-built/record drawings showing all cable numbers, construction details, and other modifications reflecting differences from the original design; (6) list of settings and adjustments for semi-fixed controls on equipment used during normal operation; (7) copies of sound systems operational software and documentation; and (8) results of electro-acoustical performance tests and measurements.

6.5.3 Test Reports

Test reports may be required to document the performance of equipment and systems. These test reports may be prepared by the contractor, equipment manufacturer, or an independent testing authority.

6.5.4 Record Drawings

Record drawings, often called "as-built" drawings reflect changes in construction which deviate from the original design due to site conditions, changes in design, or differences in equipment. Commonly a set of drawings are "red-lined" by the contractor to indicate changes.

6.5.5 Warranties

Copies of warranties from the contractors and the specific equipment manufacturers should be provided. The warranties normally obligate the contractor to adjust, repair, or replace equipment which fail in materials and workmanship for a stipulated time period. Longer warranty periods than the contractor's warranty may be offered from equipment manufacturers. Some warranty provisions may require the contractor to execute a service agreement to maintain equipment on-site, provide operation assistance for performances, assist the owner with questions and system operation, and provide loaner equipment to temporarily replace equipment being repaired.

6.6 Miscellaneous Documentation

Miscellaneous documentation is often issued by the A/E, specialty consultants, or contractors during the course of the bidding or construction phases to clarify, modify, or question the scope of the work. Common documents include: (1) construction change directives (CCD); (2) construction bulletins; and (3) requests for information (RFI). Standard forms have been developed by professional and industry groups for these documents and should be used due to legal ramifications.

Modifications of the work to be performed by the contractor after execution of the contract is covered in a CCD, commonly called a "change order." The CCD may address addition or deletion of work, or an adjustment in the contract cost or time, as originally defined in the contract documents. The owner, A/E, and contractor sign the CCD to modify the contract.

Construction bulletins are issued by the A/E or specialty consultants to clarify or modify the original design based on interpretation of the drawings and specifications, a change in the scope of work, or unforeseen field conditions. Drawings and narrative description are typically provided.

The contractor may issue an RFI during the bidding or construction phases to request clarification by the A/E or specialty consultants on specifics of the design contained in the drawings and specifications which are not fully understood. A written response is provided by the A/E or specialty consultants and may be issued as an addendum during the bidding phase or as a construction bulletin during the construction phase.

6.7 Chapter Summary

This chapter has covered the basics of construction documentation prepared by the A/E, specialty consultants, and contractors.

Construction documentation is an important aspect of the design process since it forms part of a legally binding contract between the owner and the contractors. The A/E and specialty consultants prepare construction documentation which describes the work and binding requirements the contractor is obliged to execute when undertaking the work. The contractors provide documentation to the A/E and specialty consultants to support their scope of work, and operation documentation to the owner, to aid in their use of the systems.

The next chapter will cover various architectural, electrical, and HVAC considerations in the design of sound systems.

6.8 Technical Notes

6.A Schematic Symbols

Schematic symbols are used to create sound system block and signal flow drawings. No single standard has been established to represent sound system electrical components. The schematic symbols shown below are compiled from a variety of sources and are suggested for use in creating sound system drawings. Recommended schematic symbols are shown in Figures 6-A.1 through 6-A.5 below.

Figure 6-A.1 Signal sources and loudspeakers

Figure 6-A.2 Electrical components, audio plates and boxes, miscellaneous audio devices, and mixers

Figure 6-A.3 Amplifiers

Figure 6-A.4 Signal processing

Figure 6-A.5 Connectors

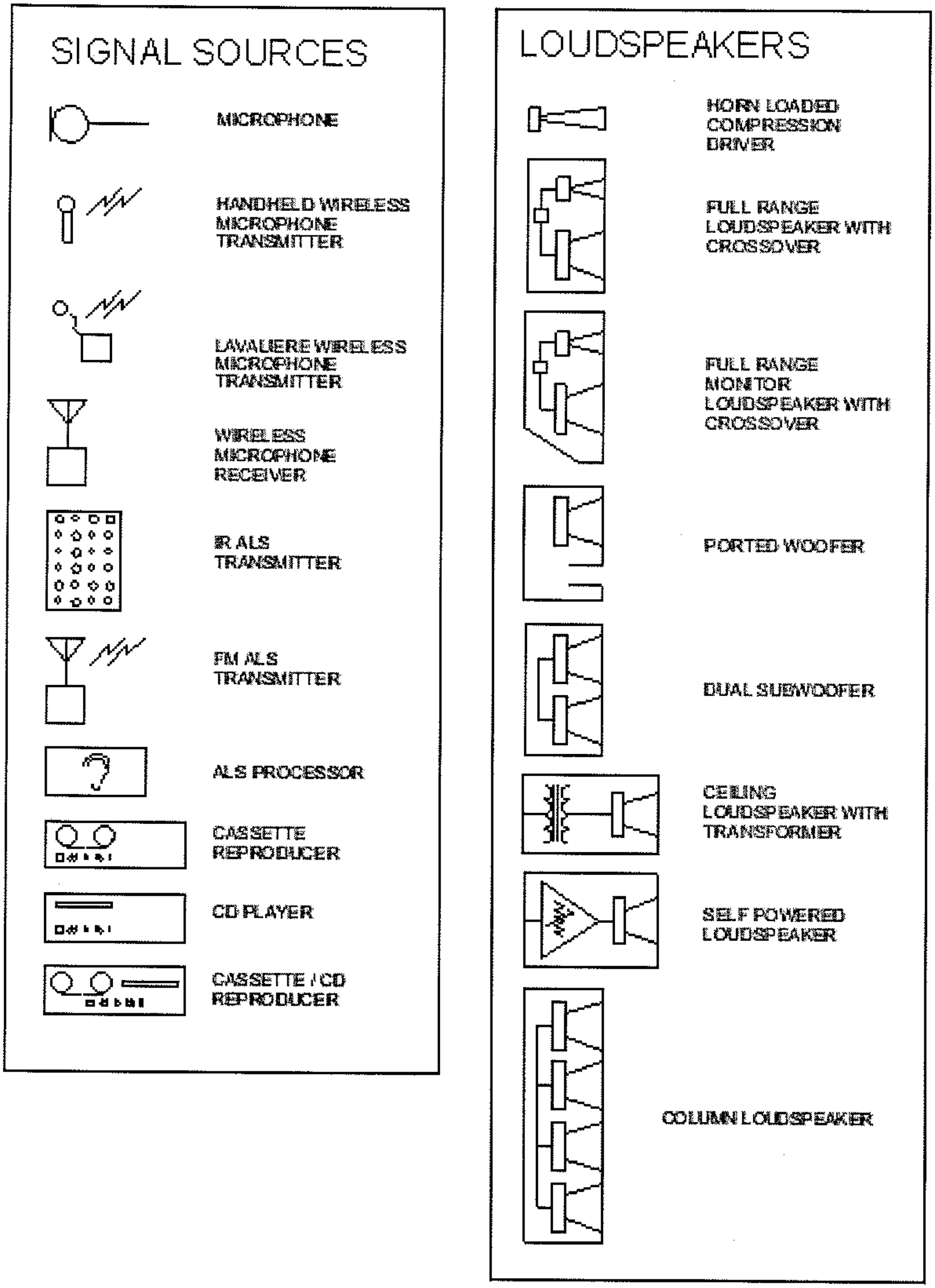

SIGNAL SOURCES
MICROPHONE
HANDHELD WIRELESS MICROPHONE TRANSMITTER
LAVALIERE WIRELESS MICROPHONE TRANSMITTER
WIRELESS MICROPHONE RECEIVER
IR ALS TRANSMITTER
FM ALS TRANSMITTER
ALS PROCESSOR
CASSETTE REPRODUCER
CD PLAYER
CASSETTE / CD REPRODUCER
LOUDSPEAKERS
HORN LOADED COMPRESSION DRIVER
FULL RANGE LOUDSPEAKER WITH CROSSOVER
FULL RANGE MONITOR LOUDSPEAKER WITH CROSSOVER
PORTED WOOFER
DUAL SUBWOOFER
CEILING LOUDSPEAKER WITH TRANSFORMER
SELF POWERED LOUDSPEAKER
COLUMN LOUDSPEAKER

FIGURE 6-A.1

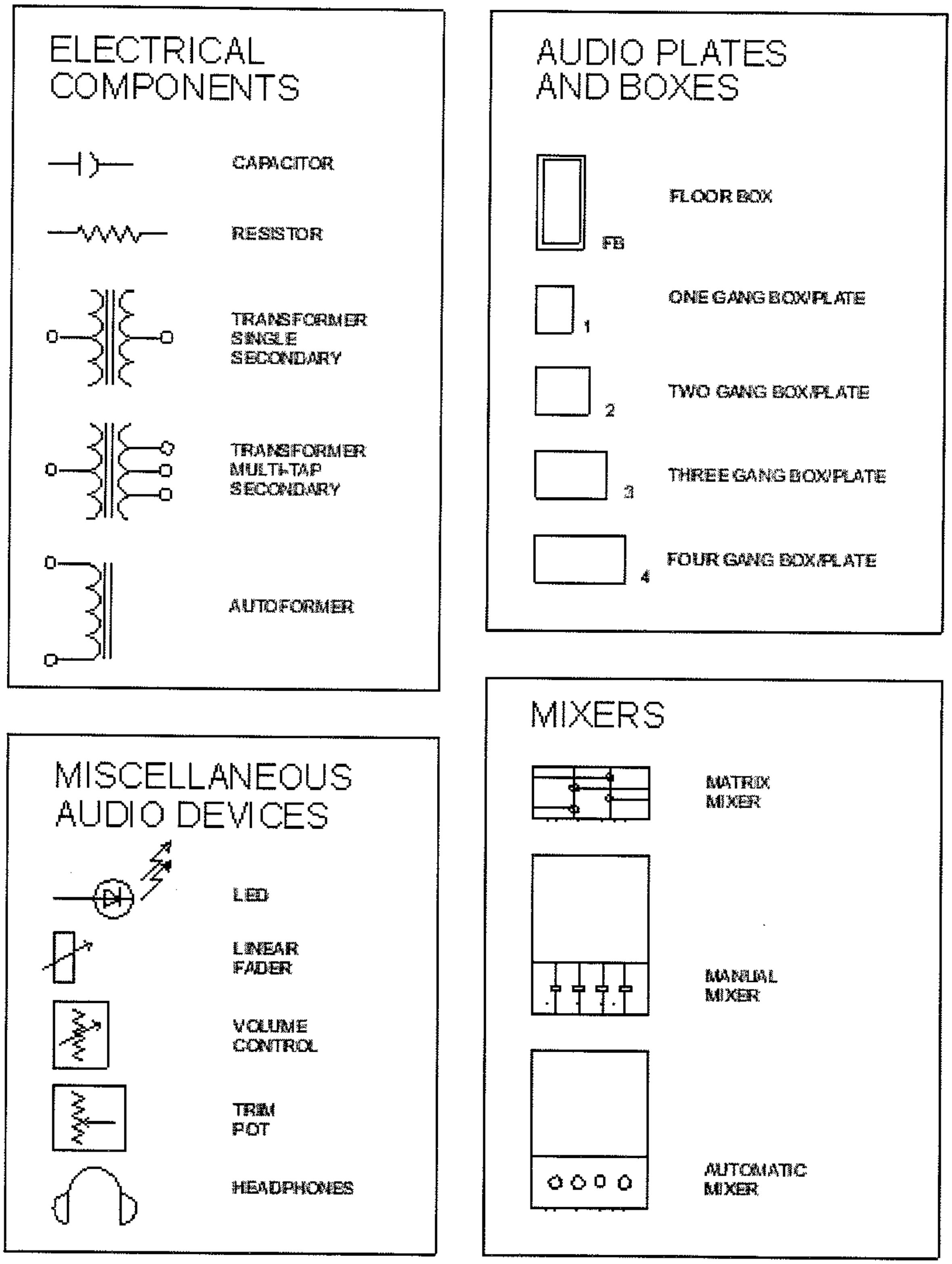

ELECTRICAL COMPONENTS
CAPACITOR
RESISTOR
TRANSFORMER SINGLE SECONDARY
TRANSFORMER MULTI-TAP SECONDARY
AUTOFORMER
AUDIO PLATES AND BOXES
FLOOR BOX
FB
ONE GANG BOX/PLATE
1
TWO GANG BOX/PLATE
2
THREE GANG BOX/PLATE
3
FOUR GANG BOX/PLATE
4
MISCELLANEOUS AUDIO DEVICES
LED
LINEAR FADER
VOLUME CONTROL
TRIM POT
HEADPHONES
MIXERS
MATRIX MIXER
MANUAL MIXER
AUTOMATIC MIXER

FIGURE 6-A.2

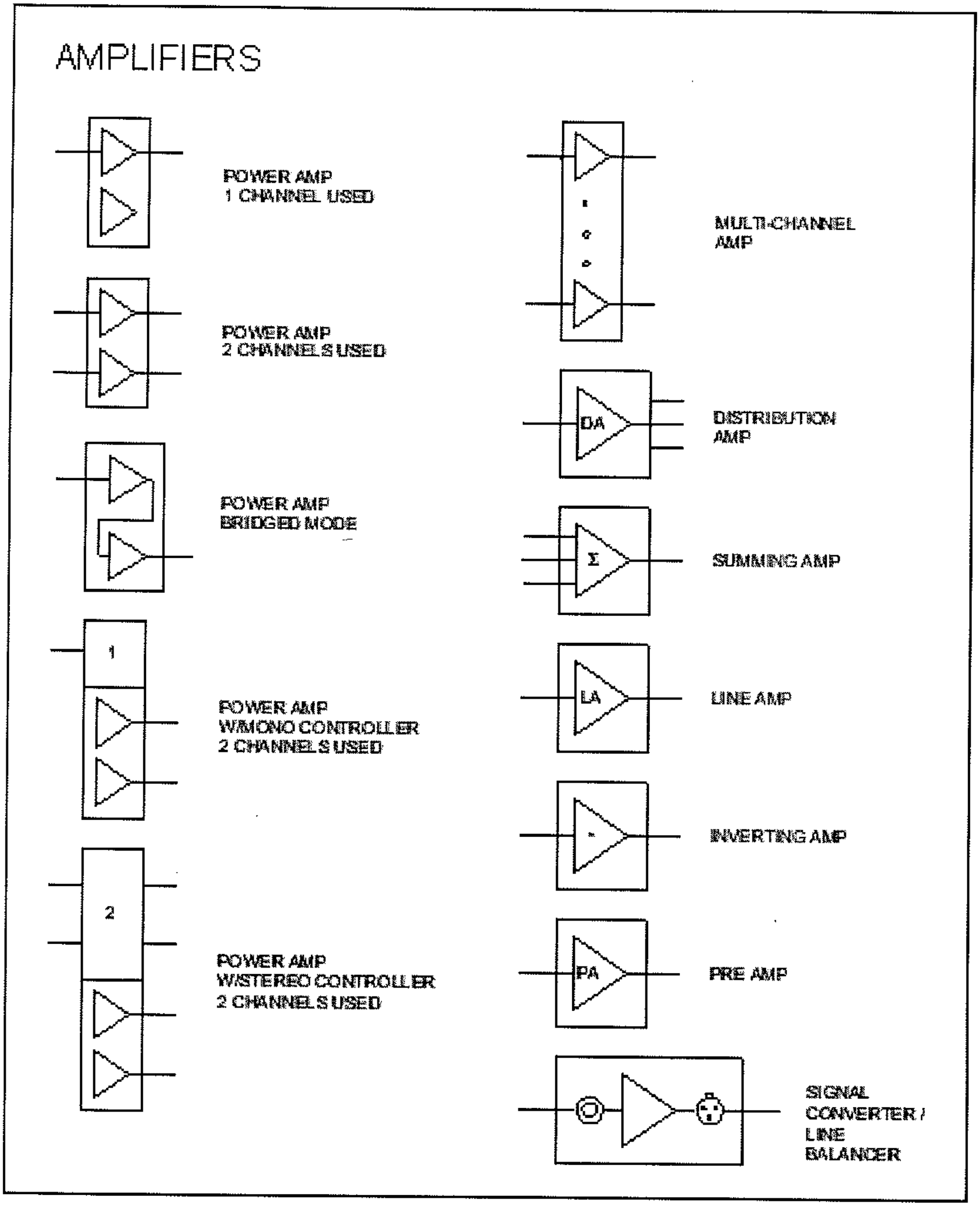

AMPLIFIERS
POWER AMP
1 CHANNEL USED
POWER AMP
2 CHANNELS USED
POWER AMP
BRIDGED MODE
1
POWER AMP
W/MONO CONTROLLER
2 CHANNELS USED
2
POWER AMP
W/STEREO CONTROLLER
2 CHANNELS USED
MULTI-CHANNEL
AMP
DA
DISTRIBUTION
AMP
Σ
SUMMING AMP
LA
LINE AMP
INVERTING AMP
PA
PRE AMP
SIGNAL
CONVERTER /
LINE
BALANCER

FIGURE 6-A.3

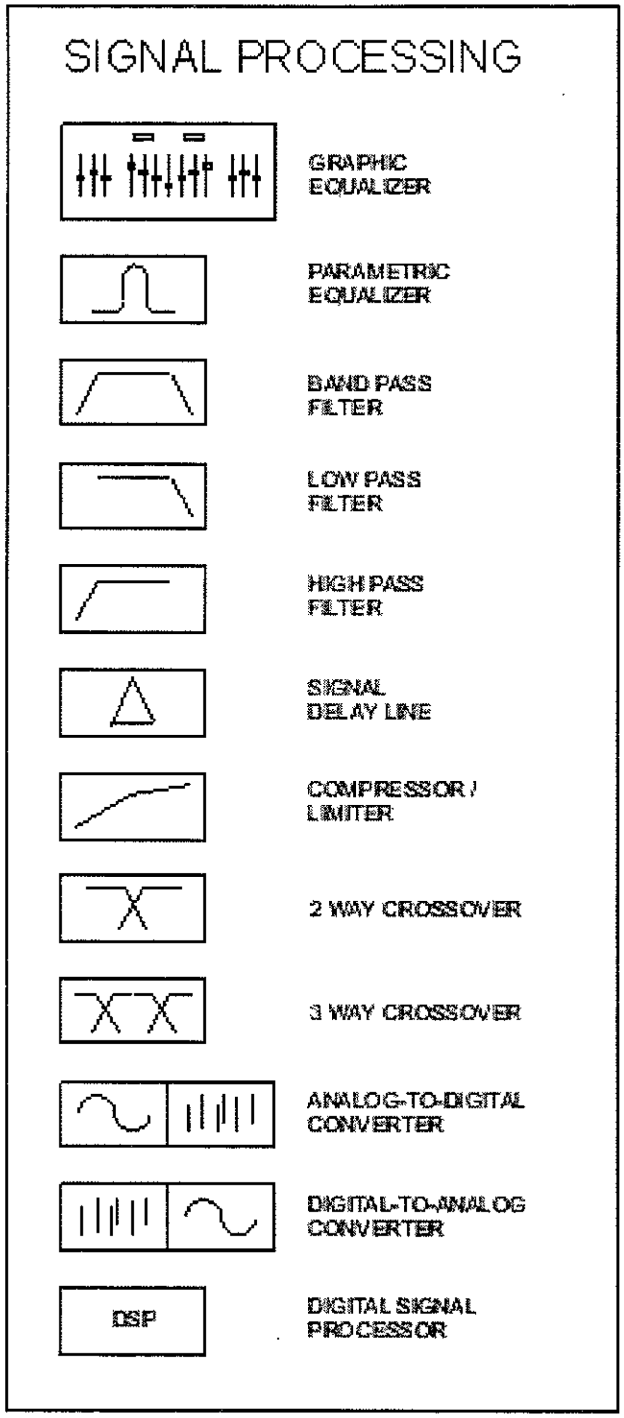

SIGNAL PROCESSING
GRAPHIC EQUALIZER
PARAMETRIC EQUALIZER
BAND PASS FILTER
LOW PASS FILTER
HIGH PASS FILTER
SIGNAL DELAY LINE
COMPRESSOR / LIMITER
2 WAY CROSSOVER
3 WAY CROSSOVER
ANALOG-TO-DIGITAL CONVERTER
DIGITAL-TO-ANALOG CONVERTER
DSP
DIGITAL SIGNAL PROCESSOR

FIGURE 6-A.4

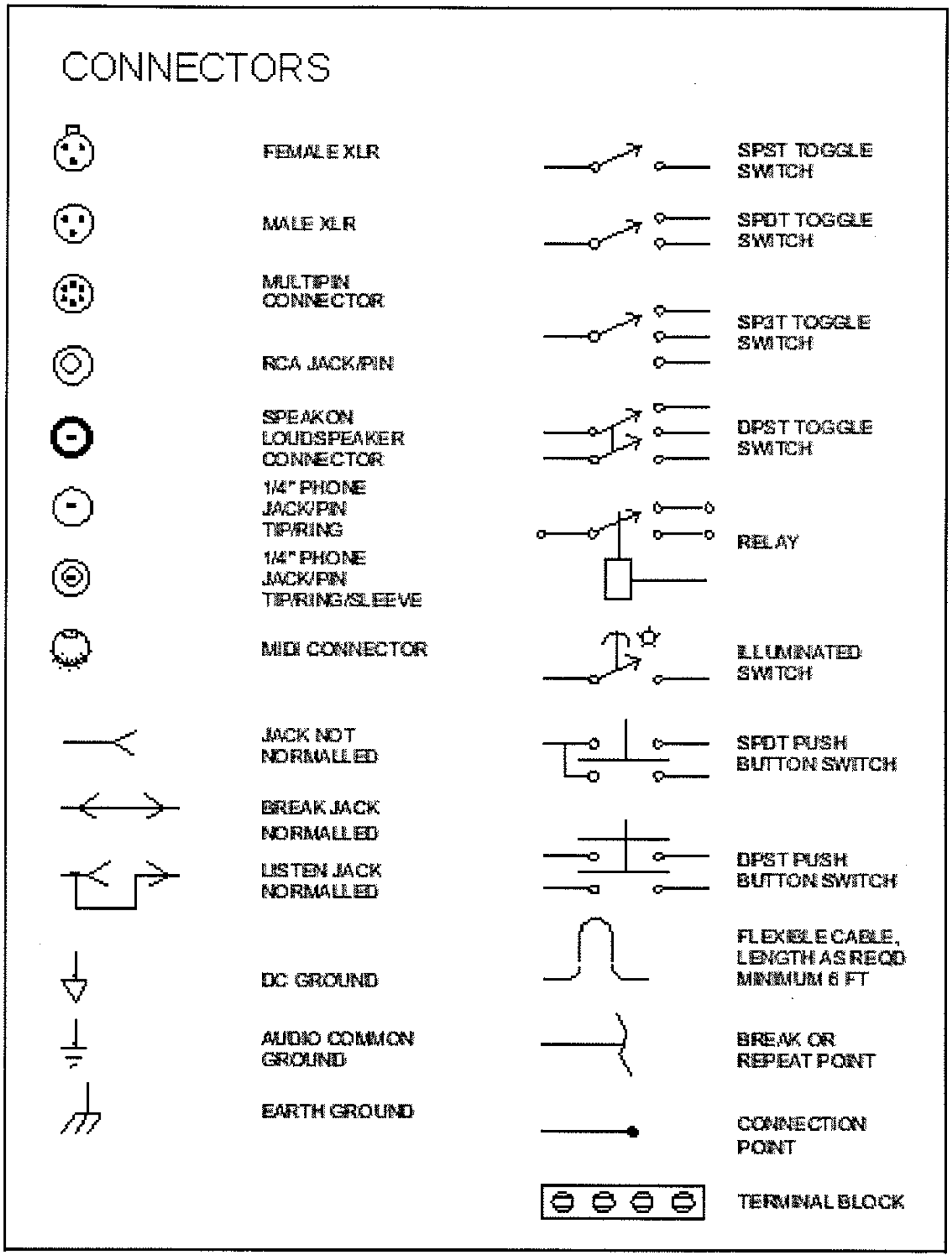

CONNECTORS
FEMALE XLR
MALE XLR
MULTIPIN CONNECTOR
RCA JACK/PIN
SPEAKON LOUDSPEAKER CONNECTOR
1/4" PHONE JACK/PIN TIP/RING
1/4" PHONE JACK/PIN TIP/RING/SLEEVE
MIDI CONNECTOR
JACK NOT NORMALLED
BREAK JACK NORMALLED
LISTEN JACK NORMALLED
DC GROUND
AUDIO COMMON GROUND
EARTH GROUND
SPST TOGGLE SWITCH
SPDT TOGGLE SWITCH
SP3T TOGGLE SWITCH
DPST TOGGLE SWITCH
RELAY
ILLUMINATED SWITCH
SPDT PUSH BUTTON SWITCH
DPST PUSH BUTTON SWITCH
FLEXIBLE CABLE, LENGTH AS REQD MINIMUM 6 FT
BREAK OR REPEAT POINT
CONNECTION POINT
TERMINAL BLOCK

FIGURE 6-A.5

Chapter 7

Architectural, Electrical, and Mechanical Considerations

"You mean we have to program space for this stuff!" "That big? Can't it be smaller?" "It has to be invisible because it will distract from the architectural integrity of the space." "I never thought we would have to provide supplemental cooling for amplifiers." "I'll have to check with the structural engineer about hanging those microphones and loudspeakers." "I don't care about sound, just make it work so we don't see it!" "Those loudspeakers will look like hanging Samsonite luggage." "Are custom colors available?"

Various comments from the author's project experience by architects grasping to integrate sound systems within the architectural design of various spaces.

7

- Dedicated Sound System Rooms
- Installation of ALS, Loudspeakers, and Microphones
- Electrical System Services
- Grounding
- Cables and Connectors
- Equipment Heat Loads and Cooling
- Codes and Life Safety Issues
- Chapter Summary
- Technical Notes

A major issue facing the architect and sound systems designer is integrating the sound system equipment within the spaces it is to serve. This planning should consider the equipment itself, sound system operator requirements, electrical system infrastructure, equipment heat loads, Codes and life-safety issues, along with attendant architectural, electrical, mechanical, and structural design criteria.

Often dedicated building areas need to be programmed to accommodate sound system equipment. Installation of ALS, loudspeakers, and microphones can pose both technical and aesthetic challenges. Electrical system services, including power, conduit and other cable routing systems, may be required and need to conform to Codes and life-safety requirements. Certain equipment produces a large sensible heat load which may require supplemental cooling to protect the equipment or provide a comfortable thermal environment for nearby occupants.

This chapter will provide the reader with general information on architectural elements (equipment rooms, control rooms, and installation of sound system equipment), electrical systems (power, conduit, grounding, cable, and connectors), equipment cooling, and basic Code and life-safety considerations related to sound system design.

7.1 Dedicated Sound System Rooms

Dedicated sound system rooms can be broadly classified as being: (1) equipment rack rooms; (2) control rooms; and (3) miscellaneous support spaces. The equipment rack rooms can be as small as a single closet or a dedicated room depending on the complexities of the sound system. Control rooms vary in size depending on the number of equipment racks, technical production staff size, and whether the space serves other uses, such as a stage manager's office, lighting control, or video/film projection room. Miscellaneous support spaces include audience and stage sound mixing locations, language translation booths, and broadcast media rooms. Figure 7-1 shows a typical installation of different dedicated sound rooms.

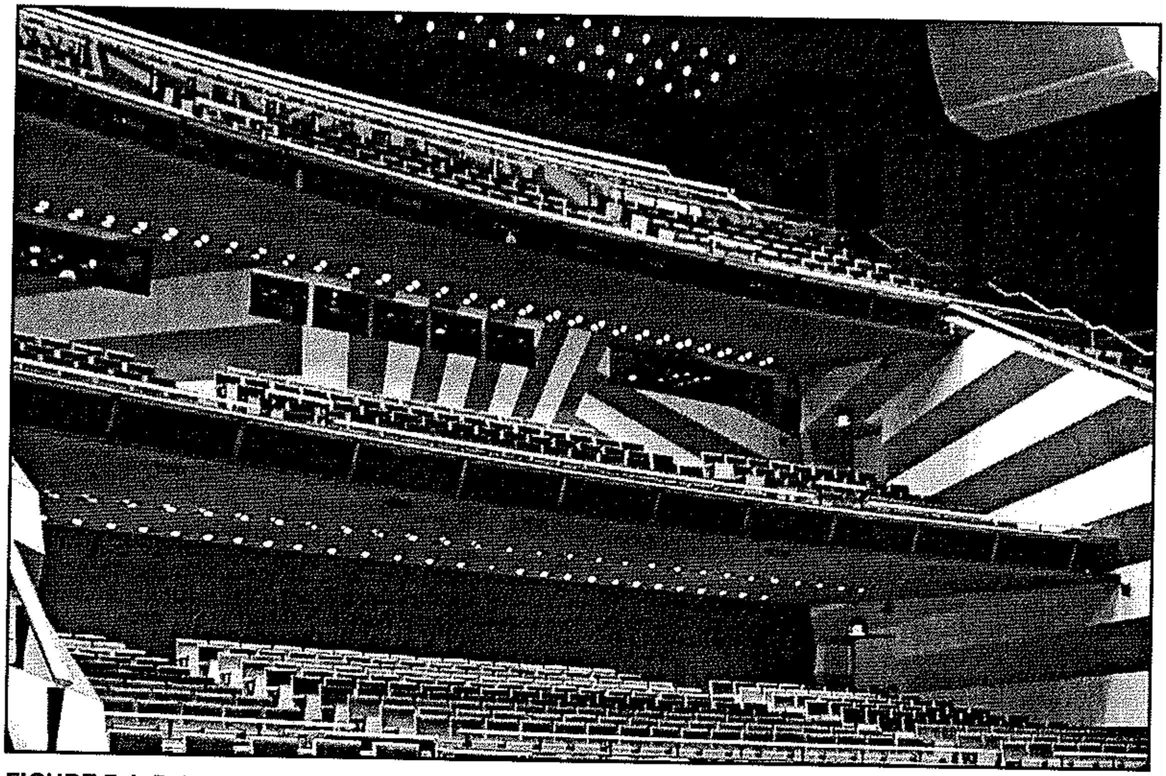


FIGURE 7-1. Balcony level rear wall in the Auditorium de Dijon in Lyon, France showing audio control room (left), language translation master control room (center), and lighting control room (right). System design by Tom Young of ARTEC Consultants, Inc. Photo courtesy of ARTEC Consultants, Inc.

The designers need to know the number, size, and location of sound system rooms at the outset of the design process. Often there is a tendency for excessive space requirements voiced by technical production staff in proportion to the size of the facility. Their needs have to be balanced with the construction budget and considerations for future technological equipment, much of which is becoming physically smaller.

As a minimum, all sound system spaces should have the following features: (1) location in close proximity to the areas they serve; (2) dedicated AC power of sufficient capacity; (3) dimmable lighting; (4) HVAC to include heating, air conditioning, and ventilation; (5) security and restricted access from the public; (6) sound-isolated construction; (7) clear unobstructed view to the stage; (8) intercommunication with other technical production spaces; (9) adequate working surfaces for permanent and temporary equipment; and (10) ADA-compliance.

7.1.1 Equipment Closets and Rack Rooms

Equipment closets and rack rooms house and protect equipment rack enclosures from damage and unauthorized tampering. These spaces are sized to permit installation of the equipment racks, clearance for servicing, electrical power, conduit, and cooling or ventilation services.

7.1.1.1 Equipment Closets

An equipment closet to contain a single equipment rack requires a nominal space of 4 ft by 4 ft clear interior dimensions. These dimensions permit approximately 12 in of space around a single equipment rack. Separate electrical pull boxes for loudspeaker and microphone cables, electrical power receptacles, a ceiling light, and a locking louvered door are recommended. Figure 7-2 shows a suggested layout of a small equipment closet to house a single equipment rack and Figure 7-3 shows an installation example.

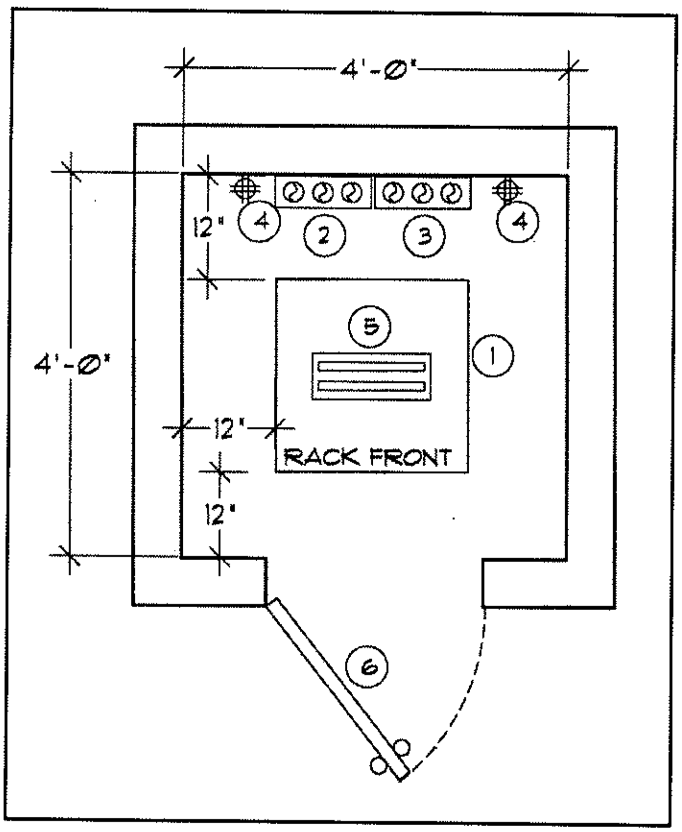


FIGURE 7-2. Small equipment rack closet. Features include: (1) equipment rack on castors (typical dimensions 77 in high by 24 in wide by 18, 24, or 30 in deep); (2) 12 in high by 12 in wide by 4 in deep electrical pull box for loudspeaker cables 84 in above floor with EMT conduit to loudspeaker locations and equipment rack; (3) 12 in long by 12 in wide by 4 in deep electrical pull box for microphone cables 84 in above floor with EMT conduit to microphone locations and equipment rack; (4) quadplex 120 VAC electrical power outlet 18 in above floor; (5) ceiling-mounted fluorescent light fixture; and (6) 30 in wide locking door with ventilation louver. Dimensions shown are clear interior minimum dimensions.

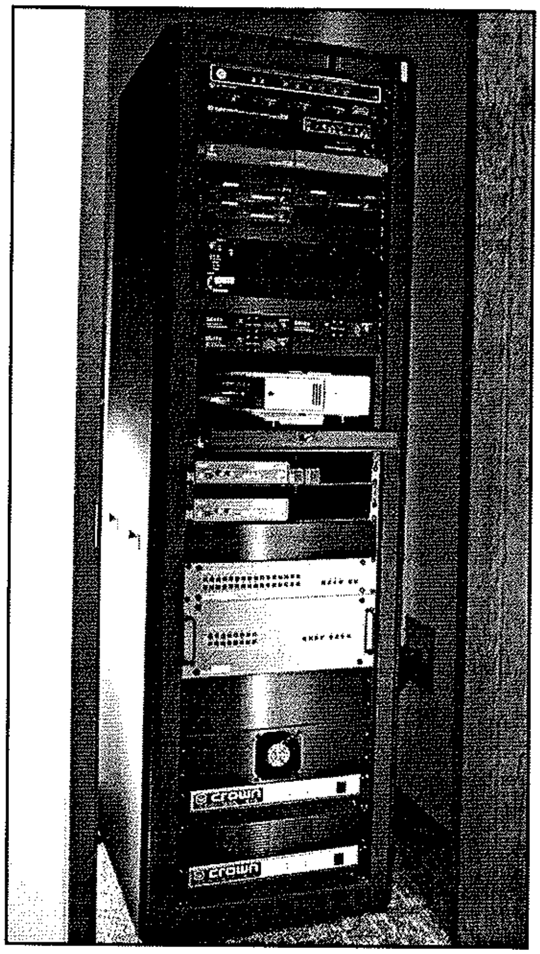


FIGURE 7-3. Single equipment rack installed in a closet near a lecture hall at the James C. Kirkpatrick Library at Central Missouri State University in Warrensburg, MO. System design by Paul Corraine of Convergent Technologies Design Group, Inc. Photo courtesy of Convergent Technologies Design Group, Inc.

7.1.1.2 Equipment Rack Rooms

Equipment rack rooms vary in size depending on the number of equipment racks to be housed and whether the room is to be ADA-compliant, which requires a 62 in radius around obstructions to permit wheelchair clearance. Some large sound system installations may have satellite amplifier rooms located near loudspeakers where operator access is normally not necessary. These rooms typically do not require ADA compliance. The equipment rack room has features similar to the equipment closet, except conditioned air is delivered to the space and a working surface is often provided. Figure 7-4 shows a suggested layout of an equipment rack room to house five equipment racks and Figure 7-5 shows an installation example. Rooms to containing other equipment rack quantities can be scaled up or down based on the information in Figure 7-4.

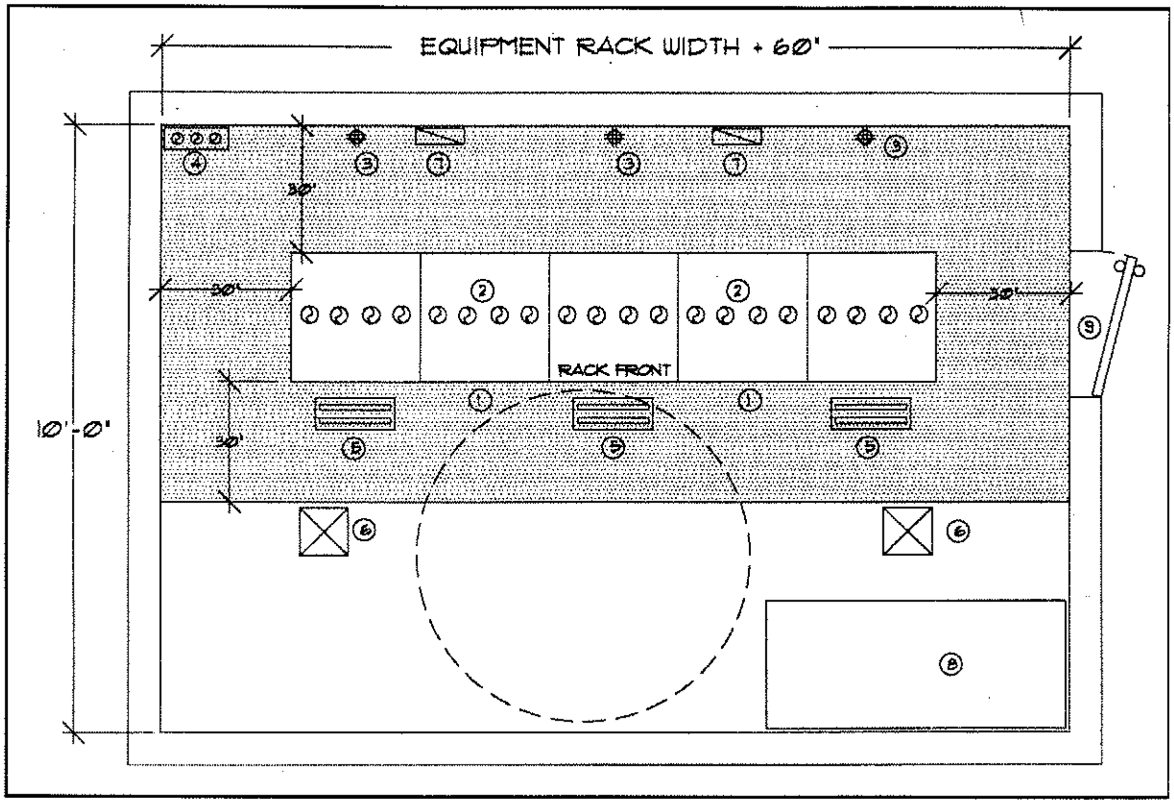


FIGURE 7-4. Equipment rack room. Features include: (1) equipment racks (typical dimensions 77 in high by 24 in wide by 18, 24, or 30 in deep); (2) separate EMT conduits for microphone, line level, loudspeaker, and electrical power from ceiling down to equipment racks and back to connected services; (3) quadplex 120 VAC electrical power outlet 18 in above floor at every other equipment rack; (4) AC power sequencing system panel mounted at 60 in above floor for rack-mounted sound system equipment (typical dimensions 30 in high by 20 in wide by 4 in deep) with EMT conduits to panelboard and equipment racks; (5) ceiling-mounted fluorescent light fixtures; (6) ceiling-mounted supply air diffusers; (7) return air grilles at 12 in above floor; (8) 60 in long by 30 in wide hinged countertop for equipment repairs; and (9) 30 in wide locking door with ventilation louver. Shaded area indicates clearance for equipment maintenance. Dimensions shown are clear interior minimum dimensions. Area at front of equipment racks is ADA-compliant.

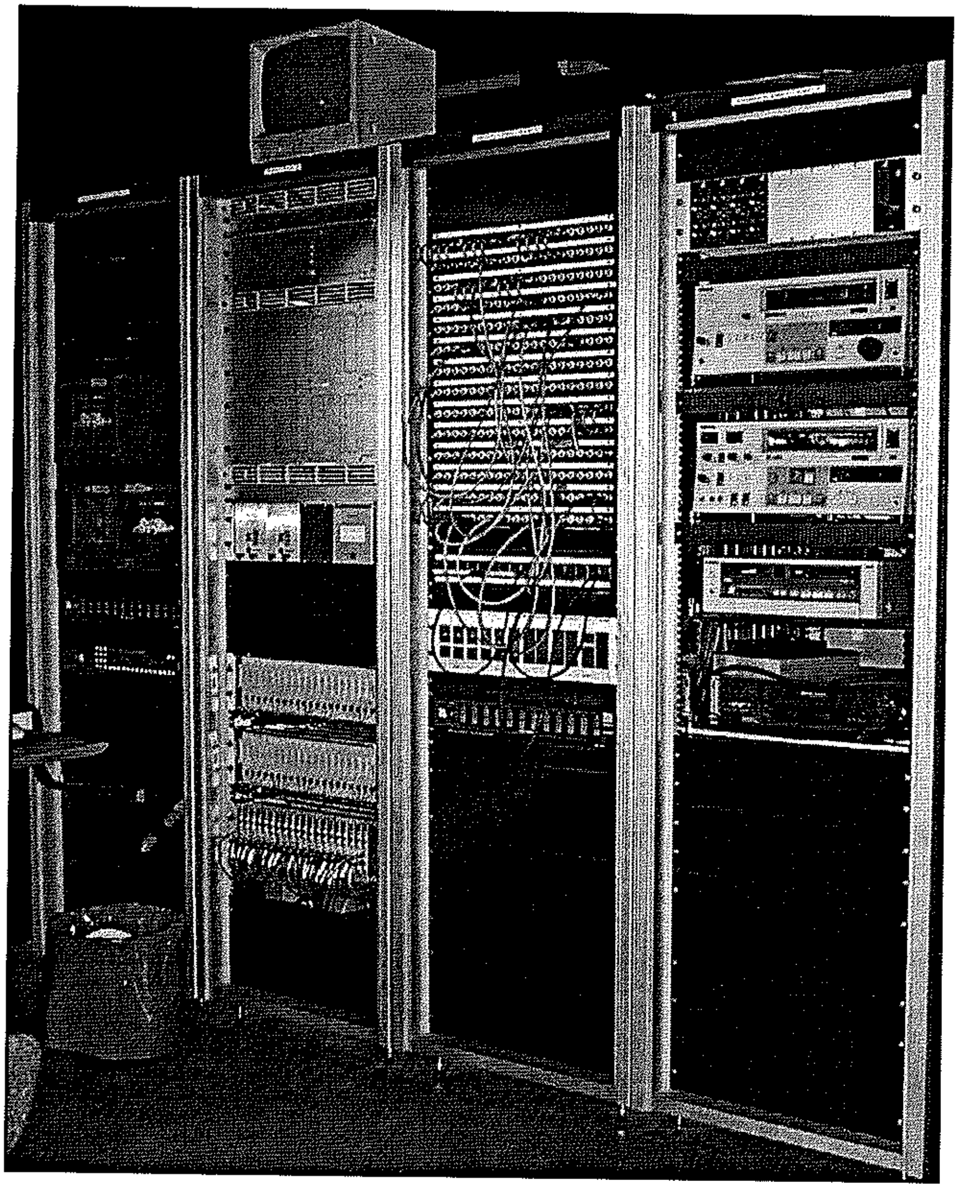


FIGURE 7-5. Four equipment racks housing various audio and video production equipment installed in the Auditorium de Dijon in Lyon, France. System design by Tom Young of ARTEC Consultants, Inc. Photo courtesy of ARTEC Consultants, Inc.

7.1.2 Control Rooms

Facilities use a variety of rooms for audio, video/projection, and lighting control. Smaller facilities will often combine functions into one control room. For audio control rooms, the sound system operator must be able to hear the program as the audience experiences it. Often this is not the case when the control room is not located in the same acoustic space as the audience or the control room window is not operable. In these cases a binaural head or boundary layer microphone system can be positioned in the audience acoustic space to provide an audio pick-up which is routed to an amplified loudspeaker in the control room. Audio control rooms contain the mixing console and may include some or all of the sound system equipment racks.

Control rooms are normally located at the rear of the auditorium along the room centerline. Often the audio control room is at the center flanked by a video/projection room and/or lighting control room. In spaces having a balcony, the control rooms are often located at the upper rear balcony level, provided the visual sightlines to the stage, wing, and borders are acceptable and not blocked by the heads of the audience in the last seating row. An alternate location is on the main floor behind the last row of audience seats. Access to the control rooms from outside the auditorium is essential, preferably separate from any patron access. Direct entry to the auditorium from the control rooms is helpful for rehearsals, but is not used during performances.

Room surfaces are normally painted a dark color and the window may have blackout curtains. Track lighting is located above the mixing console and is used during performances. Fluorescent lighting is used during rehearsals and for equipment maintenance. Electrical service of approximately 100 amperes is recommended for the control room lighting and convenience outlets. Additional electrical power capacity may be required when equipment racks are located in the room, particularly if they contain power amplifiers. An AC power sequencing system is often connected between the equipment racks and the panelboard to sequentially turn on/off equipment and avoid in-rush currents and tripped circuit breakers.

Conditioned air and ventilation are required for the control room and should be sized based on the equipment and lighting sensible heat load and number of occupants. Separate filtration for supply air is recommended to minimize dust contamination in the control room.

Control room windows should be operable and have a minimum STC-35. Additionally, the windows need to be angled to prevent illuminated reflections from the control room being visible to the sound system operators. On the auditorium side, light reflections can be distracting to performers on stage. Angling of the window will also limit sound reflections back to the stage which might be audible.

Partitions should extend structure-to-structure and have a minimum STC-45 rating.

7.1.2.1 Small Audio Control Room

A small audio control room requires a minimum space of 10 ft by 12 ft clear interior dimensions, assuming several equipment racks will be contained in the space. These dimensions permit access by a wheelchair to comply with ADA requirements. Audio control rooms which do not have equipment racks require a minimum space of 7 ft by 9 ft. A ceiling height of 8 ft with an acoustically absorptive finish is recommended regardless of the control room size.

The front wall is recommended to have a full length countertop to install the mixing console and provides a work surface. Equipment racks, if present, are located at the rear of the room with conduit between the racks, the connected services, and the mixing console. A raised computer access floor can be of benefit in routing cables within the room. Figure 7-6 shows a suggested layout of a small audio control room and Figure 7-7 shows an example.

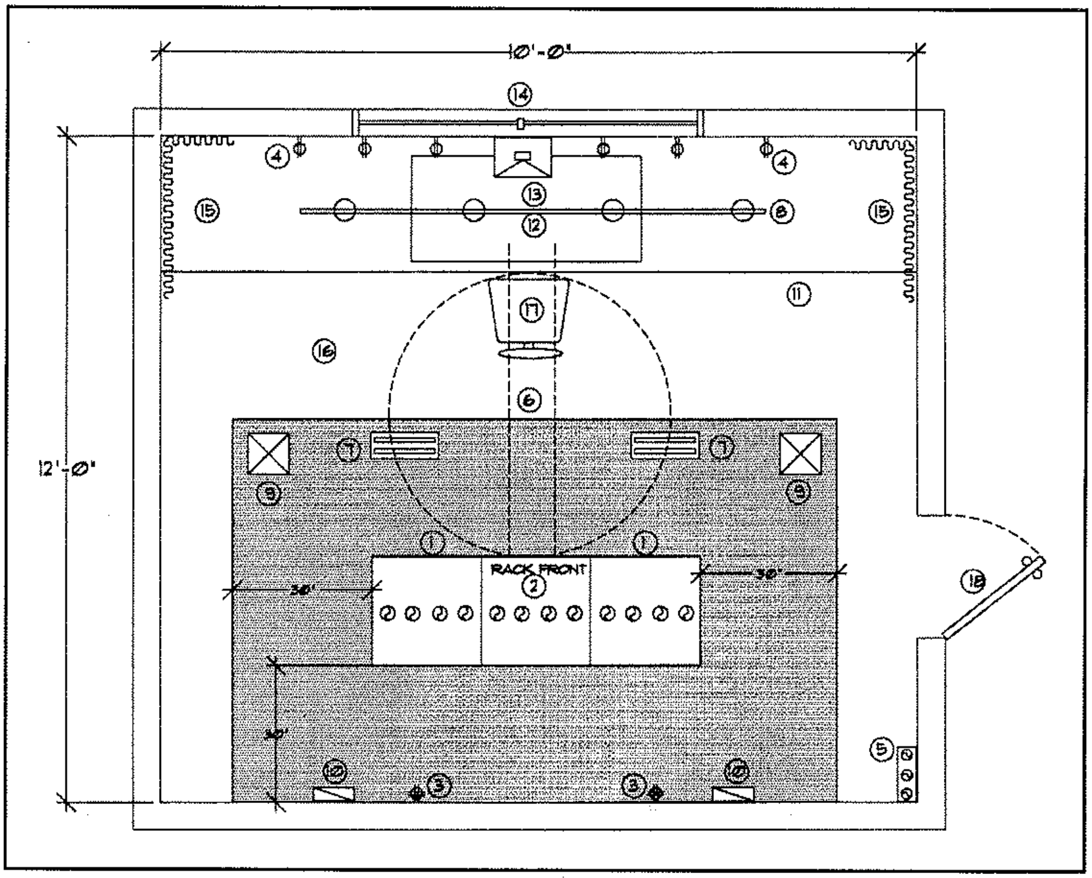


FIGURE 7-6. Small audio control room. Features include: (1) equipment racks (typical dimensions 77 in high by 24 in wide by 18, 24, or 30 in deep); (2) separate EMT conduits for microphone, line level, loudspeaker, and electrical power from ceiling down to equipment racks and back to connected services; (3) quadplex 120 VAC electrical power outlet 18 in above floor at every other equipment rack; (4) duplex 120 VAC electrical power outlet 18 in above floor at 18 in on center below countertop; (5) AC power sequencing system panel mounted at 60 in above floor for rack-mounted sound system equipment (typical dimensions 30 in high by 20 in wide by 4 in deep) with EMT conduits to panelboard and equipment racks; (6) cable tray routed under raised computer floor between equipment racks and mixing console; (7) ceiling-mounted fluorescent light fixtures; (8) incandescent track lighting above mixing console; (9) ceiling-mounted supply air diffusers; (10) return air grilles at 12 in above floor; (11) 30 in deep fixed countertop full length of room; (12) small mixing console (typical dimensions 48 in long by 24 in wide by 9 in high); (13) monitor loudspeaker on wall above window; (14) sliding operable viewing window with 1 in insulated glass (minimum dimensions 72 in wide by 48 in high) at 30 in above floor; (15) window blackout curtains on track; (16) raised computer access floor with carpet; (17) sound system operator's chair; and (18) 30 in wide locking door with ventilation louver. Shaded area indicates clearance for equipment maintenance. Dimensions shown are clear interior minimum dimensions. Area at front of equipment racks is ADA-compliant.

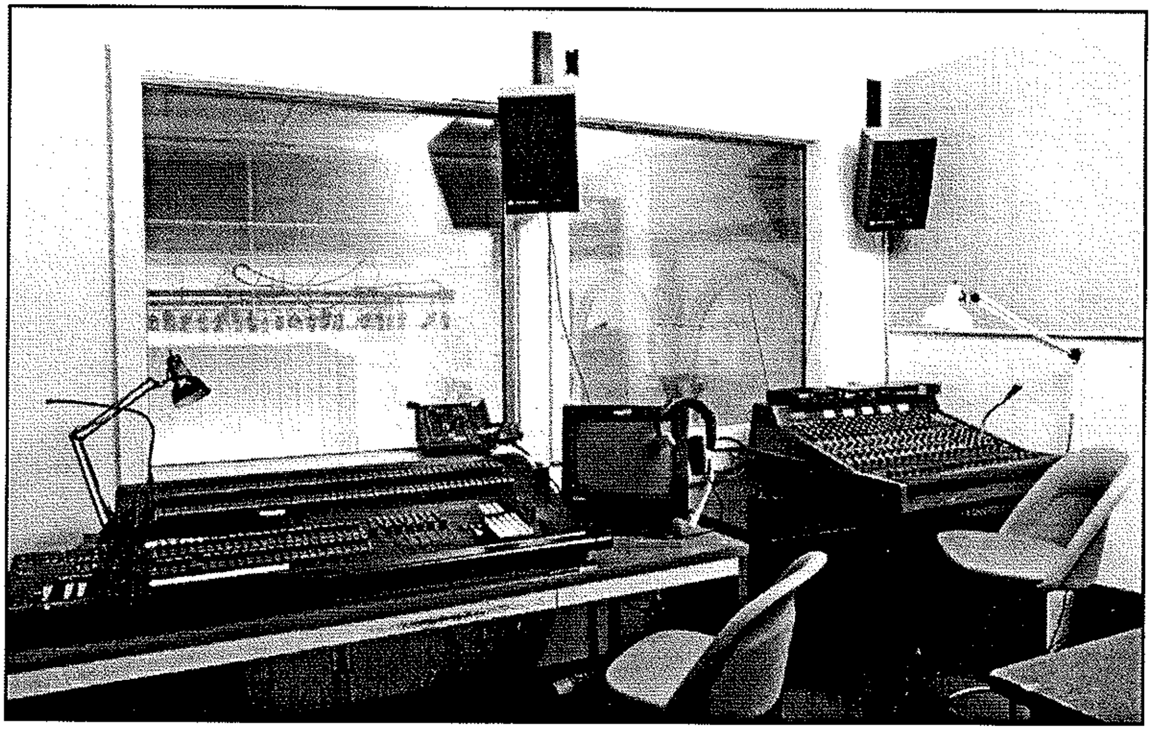


FIGURE 7-7. Small audio control room at the upper level balcony of the Friedberg Concert Hall at the Peabody Conservatory of Music in Baltimore, MD. Note large control room window with direct line-of-sight to the stage, separate mixing consoles for sound reinforcement and recording, and monitor loudspeakers.

7.1.2.2 Large Audio Control Room

A large audio control room is similar in layout and features to the small audio control room. Minimum clear interior dimensions of 14 ft by 14 ft are recommended. These dimensions will permit housing up to five equipment racks with wheelchair access inside the room. Audio control rooms which do not have equipment racks should have minimum dimensions of 12 ft by 10 ft. An acoustically absorptive ceiling of 8 ft height is recommended for all large audio control rooms. Figure 7-8 shows a suggested layout of a large audio control room and Figure 7-9 shows an example.

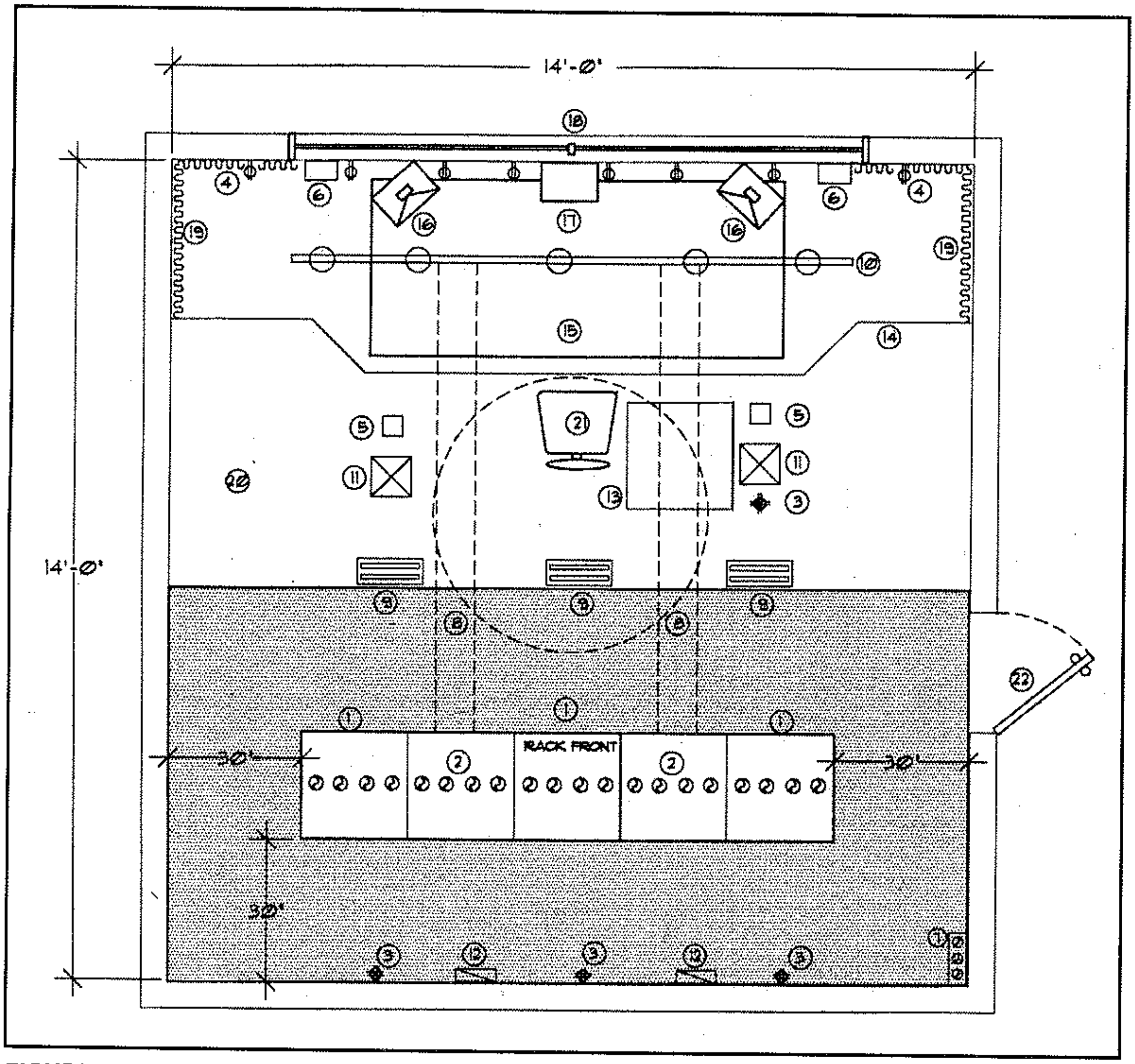


FIGURE 7-8. Large audio control room. Features include: (1) equipment racks (typical dimensions 77 inches high by 24 inches wide by 18, 24, or 30 inches deep); (2) separate EMT conduits for microphone, line level, loudspeaker, and electrical power from ceiling down to equipment racks and back to connected services; (3) quadplex 120 VAC electrical power outlet 18 inches above floor at every other equipment rack; (4) duplex 120 VAC electrical power outlet 18 inches above floor at 18 inches on center below countertop; (5) 12 inch long by 12 inch wide by 4 inch deep audio floor box with duplex 120 VAC electrical power outlet and conduit to equipment racks and mixing console; (6) intercom headset station mounted at wall above countertop with conduit to intercom master station; (7) AC power sequencing system panel mounted at 60 inches above floor for rack-mounted sound system equipment (typical dimensions 30 inch high by 20 inch wide by 4 inch deep) with EMT conduits to panelboard and equipment racks; (8) cable tray routed under raised computer floor between equipment racks and mixing console; (9) ceiling-mounted fluorescent light fixtures; (10) incandescent track lighting above mixing console; (11) ceiling-mounted supply air diffusers; (12) return air grilles at 12 inches above floor; (13) portable equipment rack on wheels (typical dimensions 46 inches high by 22 inches wide by 25 inches deep); (14) 30 inch deep fixed countertop full length of room except 48 inch deep at mixing console; (15) large mixing console (typical dimensions 72 inches long by 36 inches wide by 11 inches high); (16) monitor loudspeakers on wall above window in equilateral triangle with center of mixing console; (17) television video monitor on wall above window; (18) sliding operable viewing window with 1 inch insulated glass (minimum dimensions 84 inches long by 48 inches wide) at 30 inches above floor; (19) window blackout curtains on track; (20) raised computer access floor with carpet; (21) sound system operator's chair; and (22) 30 inch wide locking door with ventilation louver. Shaded area indicates clearance for equipment maintenance. Dimensions shown are clear interior minimum dimensions. Area at front of equipment racks is ADA-compliant.

FIGURE 7-9. Large audio control room in the Auditorium de Dijon in Lyon, France. Note MIDAS XL4 mixing console on rolling table (left), Yamaha 02R digital mixer (right), Genelec monitor loudspeakers, and Panasonic video monitor all above operable control room window. Control room walls and ceiling are painted black. System design by Tom Young of ARTEC Consultants, Inc. Photo courtesy of ARTEC Consultants, Inc.

7.1.3 Miscellaneous Spaces

Miscellaneous dedicated sound system spaces include audience and stage mixing positions, language translation booths, and broadcast media rooms.

7.1.3.1 Audience and Stage Mixing Positions

Sound mixing for amplified concerts, musicals, and productions with music accompaniment is best served by a central position within the audience seating area. The objective is for the sound system operator to experience the same sound the audience and performer hears. The correct location of the sound mixing position is critical to the sound quality perceived by the operator and therefore the audience and performers. Two sound mixing applications are commonly encountered with music: (1) front-of-house sound reinforcement and (2) performer monitor/foldback. The front-of-house sound reinforcement is used to mix the sound routed to the main loudspeakers which the audience hears. The performer monitor/foldback is used to provide a separate sound mix which is tailored to the needs of the performers and is not heard by the audience. The front-of-house mixing position is usually within the audience area. The monitor/foldback mixing position is usually located at stage left and the stage director position at stage right.

The option of having an audio control room or an audience-located sound mixing position needs to be addressed early in the design. A permanent audience area can be allocated or a temporary area within the audience seating can be used. This latter concept requires seats which can be removed when necessary to install the sound mixing console. Both locations are included in larger multi-purpose spaces.

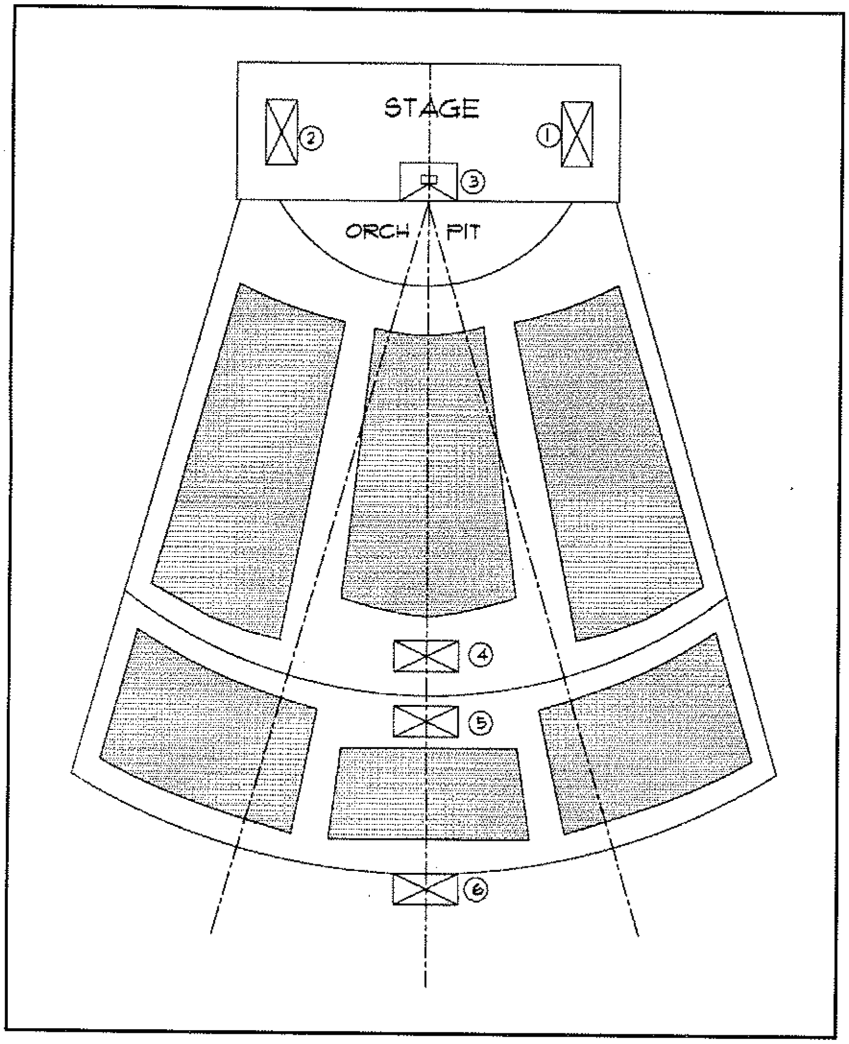


FIGURE 7-10. Sound mixing positions in an auditorium. Common locations include: (1) monitor/foldback mix position at stage left; (2) stage director position at stage right; (3) central cluster system location; (4) front-of-house sound reinforcement mix position forward of balcony approximately one-half to two-thirds back from stage (preferred position); (5) front-of-house sound reinforcement mix position on balcony level (alternate position); and (6) audio control room on main floor or upper balcony levels. Horizontal limits of sound reinforcement mix positions and audio control room are within 15° either side of the room centerline shown by dashed lines.

The sound mixing position should minimize audience distraction due to the presence of the equipment and sound system operator. The mixing position needs a flat floor surface with a low-rise perimeter partition to separate the technical equipment from the audience. A minimum floor area of 7 ft by 7 ft is necessary for a sound mixing console, single equipment rack, and two sound system operators. Larger floor areas, up to 12 ft by 9 ft, are necessary to accommodate a sound mixing console, fixed and portable equipment racks, and up to three sound system operators. Permanent electrical power and audio receptacles need to be provided at the audience-located sound mixing position. Figure 7-10 shows possible locations for sound mixing positions within an auditorium. Figures 7-11 and 7-12 show suggested layouts of small and large sound mixing positions and Figure 7-13 shows an example.

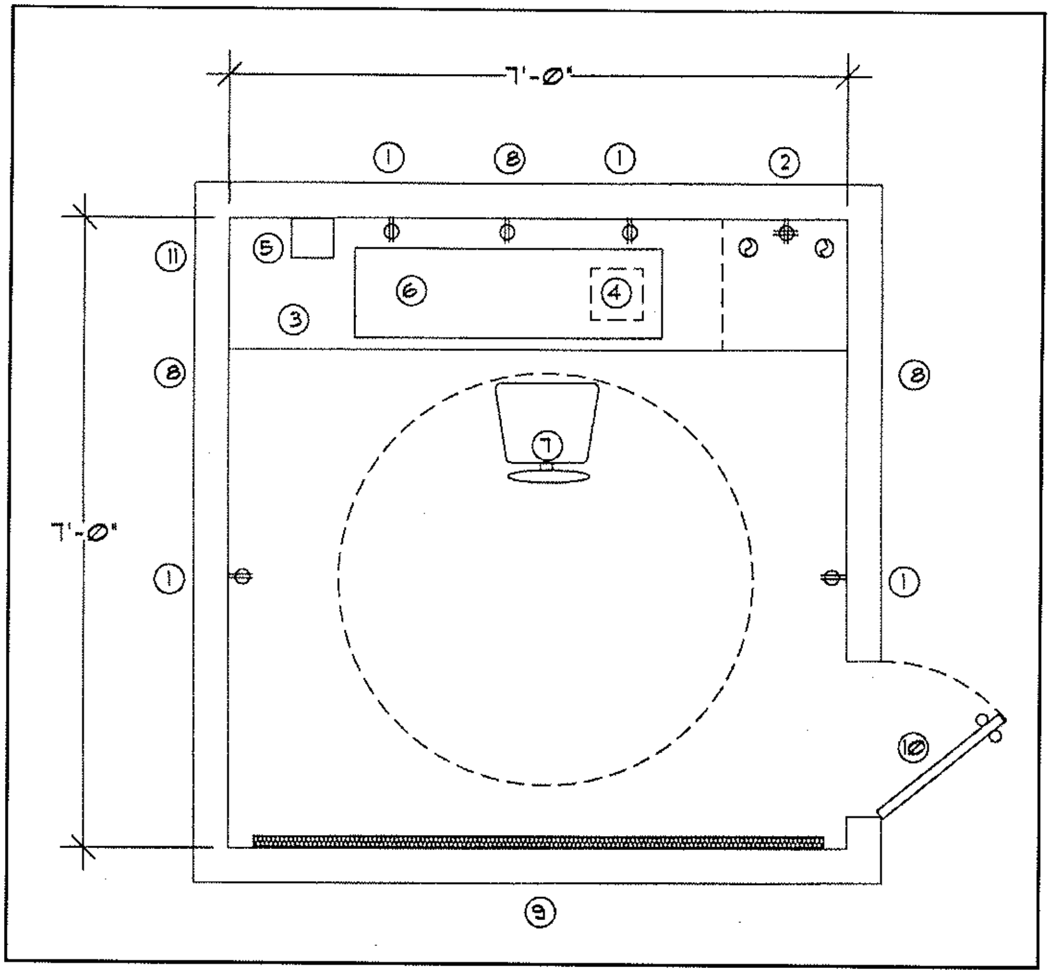


FIGURE 7-11. Small mixing position. Features include: (1) duplex 120 VAC electrical power outlet 18 in above floor at 18 in on center below countertop and along side walls; (2) quadplex 120 VAC electrical power outlet 18 in above floor at under counter equipment rack; (3) 30 in deep fixed countertop full length of space; (4) 12 in long by 12 in wide by 4 in deep audio floor box with conduit to equipment racks at other locations; (5) intercom headset station mounted at wall above countertop with conduit to intercom master station; (6) small mixing console (typical dimensions 48 in long by 24 in wide by 9 in high); (7) sound system operator's chair; (8) 42 in high partition around front and sides; (9) 42 in high rear partition (if in audience area) or full height rear wall (if at back of auditorium with 1 in thick sound absorptive panels from 48 in to 96 in above floor); and (10) 30 in wide by 42 in high locking door. Dimensions shown are clear interior minimum dimensions. Area behind mixing console is ADA-compliant.

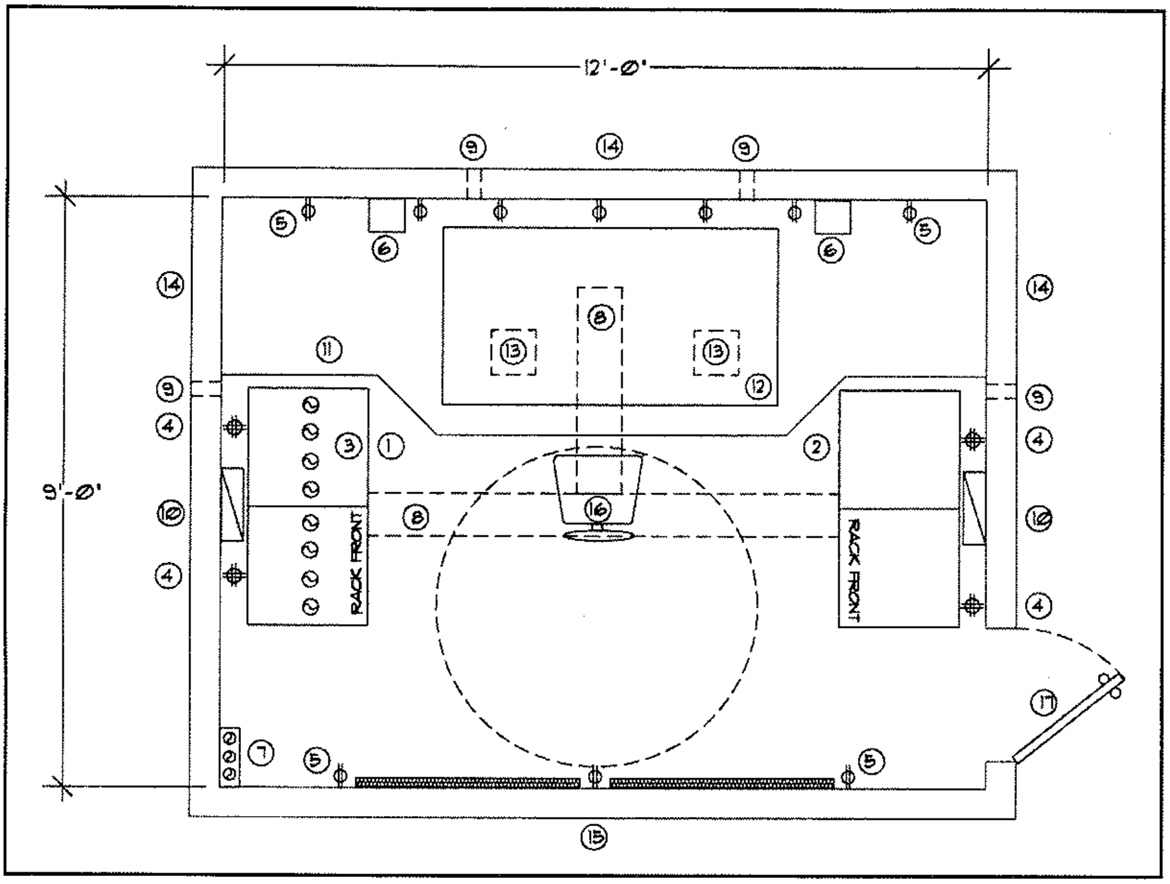


FIGURE 7-12. Large mixing position. Features include: (1) fixed equipment racks (typical dimensions 77 in high by 24 in wide by 18, 24, or 30 in deep); (2) portable equipment racks (size varies); (3) separate EMT conduits for microphone, line level, loudspeaker, and electrical power from ceiling down to fixed equipment racks and back to connected services; (4) quadplex 120 VAC electrical power outlet 18 in above floor behind each equipment rack; (5) duplex 120 VAC electrical power outlet 18 in above floor at 18 in on center below countertop; (6) intercom headset station mounted at wall above countertop with conduit to intercom master station; (7) AC power sequencing system panel mounted at 12 in above floor for rack-mounted sound system equipment (typical dimensions 30 in high by 20 in wide by 4 in deep) with EMT conduits to panelboard and equipment racks; (8) cable trough with metal access cover routed under floor between equipment racks and mixing console; (9) cable pass through opening in wall at 18 in above floor; (10) exhaust air diffuser at equipment racks 12 in above floor; (11) 30 in deep fixed countertop full length of space except 48 in deep at mixing console; (12) large mixing console (typical dimensions 72 in long by 36 in wide by 11 in high); (13) 12 in long by 12 in wide by 4 in deep audio floor box with conduit to equipment racks at other locations; (14) 42 in high partition around front and sides; (15) 42 in high rear wall (if in audience area) or full height rear wall (if at back of auditorium with 1 in thick sound absorptive panels from 48 in to 96 in above floor); (16) sound system operator's chair; and (17) 30 in wide by 42 in high locking door. Dimensions shown are clear interior minimum dimensions. Area behind mixing console is ADA-compliant.

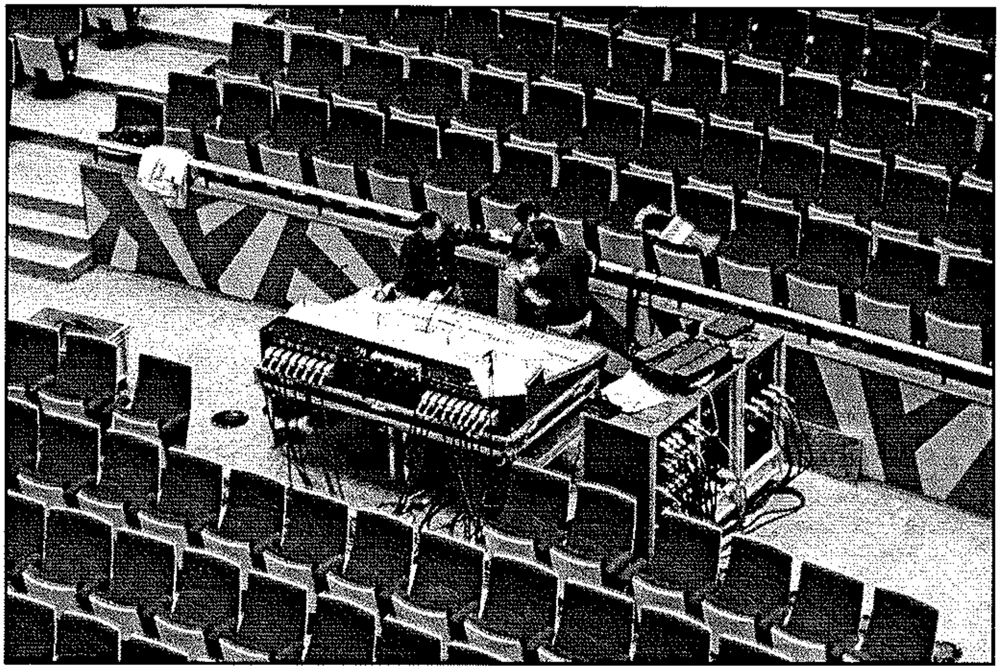


FIGURE 7-13. Front-of-house sound mixing position in the Auditorium de Dijon in Lyon, France. Note MIDAS XL4 sound mixing console on rolling table and portable rolling equipment racks. System design by Tom Young of ARTEC Consultants, Inc. Photo courtesy of ARTEC Consultants, Inc.

7.1.3.2 Language Translation Booths

Language translation booths are normally located at the rear or sides of the auditorium in close proximity to the audio control room. The booths should be grouped side-by-side and provided with small windows to permit communication between the interpreters. Line-of-sight conditions from the language translation booths to the stage is critical particularly if the auditorium uses video projection. The size of the booths should accommodate two interpreters with minimum dimensions of 7 ft by 6 ft necessary. A fixed 3 ft by 3 ft window is at the front of the booth. The glazing should be angled to eliminate visual glare from lighting and detrimental sound reflections.

A countertop at the front of the booth is sized for scripts, texts, and other papers. A table-top mounted microphone, audio control switching, and a loudspeaker with volume control are required, although the interpreters use headphones to listen to the floor language from which they are translating. Tie-lines connect the audio control room and language translation booths.

7.1.3.3 Broadcast Media Rooms

Facilities used for broadcast origination should include provisions for cable, radio, and television media organizations. Several options for broadcasting include: (1)

dedicated media room; (2) sharing audio or lighting control rooms; and (3) dedicated input plates at locations in the facility.

A separate media room requires line-of-sight to the stage and should be located on the main floor level to permit easy load-in and load-out of broadcast equipment. Minimum room dimensions of 7 ft by 9 ft are suggested. The room should have good sound isolation properties. Input boxes for connecting audio and video recording feeds from other facility sources are required as are multiple 120 VAC electrical power outlets. Audio and video tie-lines between the media room and an exterior location are necessary to route signals to the broadcast truck.

Sharing audio or lighting control rooms by broadcast personnel is an option if a separate media room is not possible. This will require increasing the size of the control room, often by as much as 50 percent, to accommodate the extra personnel. Dedicated input boxes for audio and video recording feeds are necessary as are tie-lines to the exterior.

Locations within the audience area are a last resort for broadcast camera and audio positions. The presence of the media personnel will be a distraction to the audience and locations at the rear audience area are preferred. Audio and video input boxes and tie-lines to the exterior are required at these locations.

7.2 Installation of ALS, Loudspeakers, and Microphones

Planning for the installation of ALS, loudspeakers, and microphones in the spaces they serve requires coordination between the architect, electrical engineer, structural engineer, sound system designer, and sound system operators. Manufacturers have developed a variety of products to simplify installation of sound system equipment and interconnection of audio cables and connectors.

Numerous aspects related to the installation of ALS, loudspeakers, and microphones are fundamental to their operation and have been covered in Chapters 3 and 4. Outlined below are other installation requirements which affect the designers.

7.2.1 ALS Requirements

Individuals with hearing impairments are protected by civil rights legislation established by the ADA on 26 July 1990 and subsequently modified on 21 December 1992 and 1 July 1994. Regulations applicable to sound systems are covered in the Title III of the ADA which addresses 12 categories of public accommodation and assembly spaces which require an ALS. These spaces include all governmental facilities and facilities operated by a private entity whose operations affect commerce. Examples of such facilities include: (1) education, entertainment, exhibition, lodging, and public assembly buildings; (2) exercise, recreation, and transportation facilities; (3) judicial, legislative and social service centers; and (4) factories, hospitals, offices, retail stores, and warehouses. The ADA regulations are applicable to public and private facilities providing goods or services to the public and include existing buildings, alterations made to existing buildings, and new building construction.

Facilities exempt from ADA regulations include private schools, private clubs, and religious facilities, except portions of those facilities made available to the general public. Also exempt are facilities where it can be proved that compliance would cause undo burden, be excessively costly, or result in a fundamental alteration to the facility.

Permanently installed or portable ALS can be used depending on specifics of the facility. Regardless of the type of ALS used, it must provide adequate coverage to permit hearing impaired individuals to sit anywhere in the space. General guidelines are noted below.

1. New construction of public facilities must have a minimum quantity of receivers equal to four percent of the room occupancy load.

2. Existing public facilities seating more than 50 people must have a minimum quantity of receivers equal to four percent of the room occupancy load.

3. Existing public facilities seating less than 50 people must have a minimum quantity of receivers equal to four percent of the room occupancy load, but no less than two.

4. Existing public facilities under renovation require a minimum quantity of receivers equal to four percent of the room occupancy load if the cost of making it accessible does not exceed 20 percent of the renovation cost. Otherwise compliance is not required.

Visible signage is required to notify individuals that ALS is available for their use.

7.2.1.1 Requirements for Permanently Installed ALS

A permanently installed ALS is required in places where audible communication is integral to the use of the space and the space: (1) accommodates at least 50 people; (2) has fixed seating; and (3) has a sound reinforcement system. Spaces covered under this ruling would include assembly, education, judicial, legislative, regulatory, and similar facilities

Judicial, legislative, and regulatory facilities, such as court houses and other governmental facilities which have multiple common spaces, are required to have a permanently installed ALS in 50 percent, but not less than one, of each type of space. This would include chamber rooms, courtrooms (circuit, district, juvenile, magistrate, et cetera), hearing rooms, jury deliberation rooms, juror orientation rooms, and meeting rooms. Separate chambers (house and senate) for bicameral legislation are required to each have a permanently installed ALS.

The location of a permanently installed ALS will depend on the system type. A small area FM ALS having a transmitter and antenna can be within an equipment rack. Large audience areas which require coverage from an FM ILS need the antenna within the space to be covered necessitating an antenna remote from the transmitter. IR ALS which use a separate modulator and emitter panel have the modulator installed in the equipment rack and the emitter panel installed in the room it serves.

7.2.1.1.1 FM ALS Installation

Antennae covering different size areas are available from FM ALS manufacturers. Some antennae are intended to be used with a compatible ground plane supplied by the manufacturer. The ground plane increases the effective radiating area of the antenna system. The antenna element is mounted perpendicular to the ground plane at the center of the plate. The antenna or ground plane should not be physically bonded to any metal building surface. Coaxial cable with the proper impedance is used to connect the remote antenna and transmitter.

The location of a remote FM antenna should be selected to provide line-of-sight to the antenna from all audience seating areas. The antenna can be installed behind a curtain or other non-solid material or painted to match the surrounding surfaces to minimize its visibility. In rooms with apses, niches, or other recessed spaces it may be necessary to try several locations to optimize the antenna coverage within the audience areas.

Care should be taken to locate the antenna away from any electrical power lines and service drops within the building. This is necessary to prevent the antenna from physically touching the power lines and to minimize possible high voltage arc-over from the power cables to the antenna.

(See Technical Notes, Section 7.A, at the end of this chapter, for additional information on remote antenna installation.)

7.2.1.1.2 IR ALS Installation

The simplest installation of IR ALS emitter panels is at the front of the room facing the audience. The panels should be elevated between 15 and 30 ft above the floor and aimed down towards the audience to "flood" the seats with IR light. The emitter panel LEDs should not be covered with any material, otherwise the radiated IR energy into the room will be reduced. Likewise, the LEDs on the receivers worn by the audience members should not be concealed. Figure 7-14 provides installation guidelines for fixed IR emitter panels.

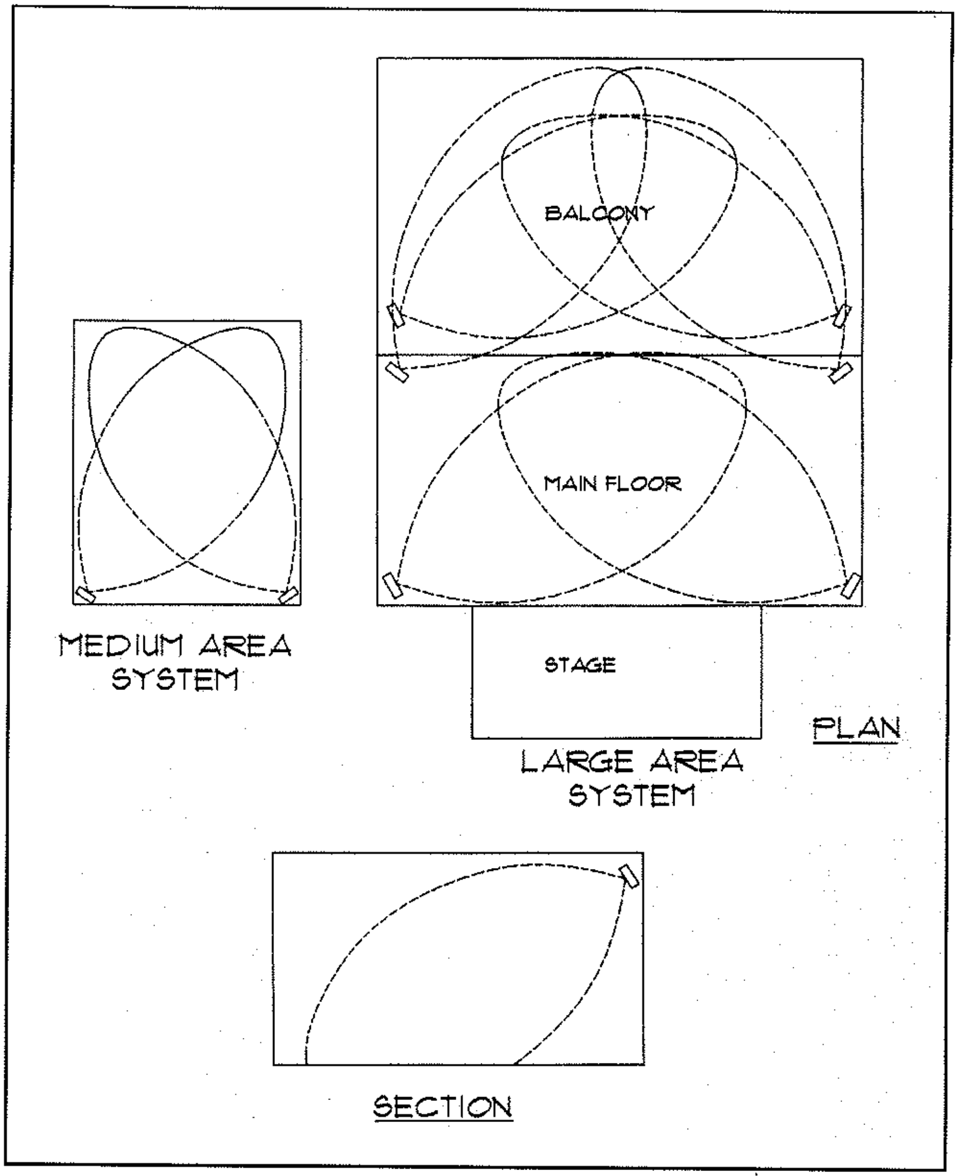


FIGURE 7-14. IR emitter panel installation guidelines. "Large area" systems may require multiple emitter panels to cover the main, underbalcony, and balcony level seating areas. "Medium area" systems may also require multiple IR emitter panels. Note IR emitter panels are arranged to cross-fire and are aimed down on the audience seating areas.

The pattern of the IR signal is conical shaped with approximately 50° to 150° horizontal and vertical coverage angles (both equal) dependent on manufacturer's different products. The transmission distance will depend on the emitter panel output. The area covered by the emitter panel is a function of the number of channels the emitter panel radiates and decreases with increasing number of channels. The area covered by the emitter panel can be calculated using the following equation:

$$A_M = \frac{A_R}{C} \tag{7.1}$$

where,

A_M is the area covered by emitter panel for number of channels radiated, ft^2

A_R is the area covered by emitter panel for one channel, ft^2

C is the number of channels in IR system

The number of IR emitter panels can be calculated using the following equation:

$$N_R = \frac{A_F C}{A_R} \tag{7.2}$$

where,

N_R is the number of emitter panels required

A_F is the audience floor area, ft^2

A_R and C are as above

The above equations assume good **S/N** ratio for the received IR signal. The **S/N** ratio will decrease with greater distances from the IR panel. A high quality audio signal with minimum background noise will result when the **S/N** ratio exceeds 40 dB. Useable audio signals, but having considerable background noise, will result for a **S/N** ratio as low as 26 dB. Levels below this **S/N** ratio will be dominated by background noise and may not be useable.

Figure 7-15 provides polar coverage data for "medium area" and "large area" single unit systems. Figure 7-16 provides polar coverage data when multiple units are used side-by-side to increase their coverage area.

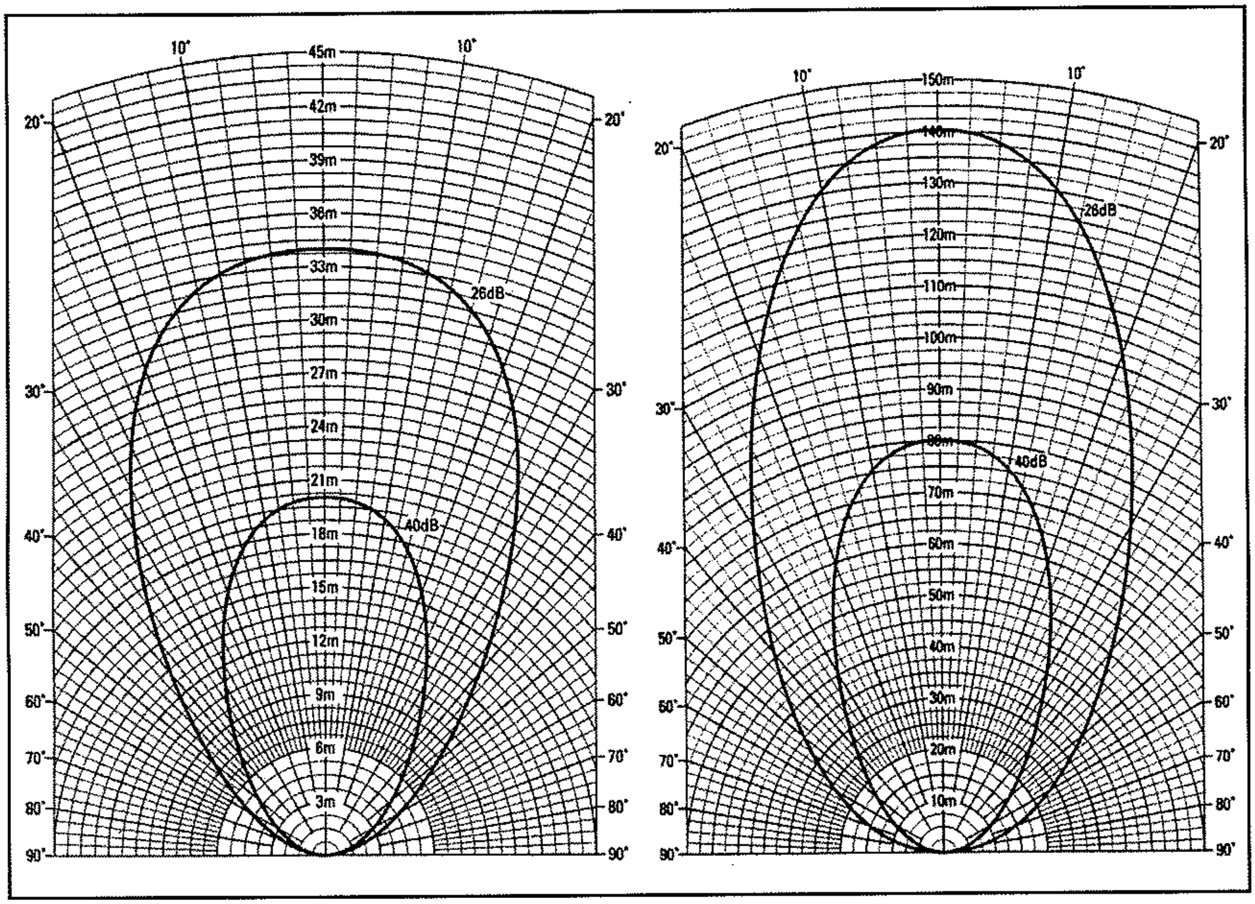


FIGURE 7-15. Polar coverage pattern for "medium area" (left) and "large area" (right) IR emitter panels for single units showing 40 dB and 26 dB **S/N** ratio limiting distances. Data courtesy of Sennheiser Electronic Corporation.

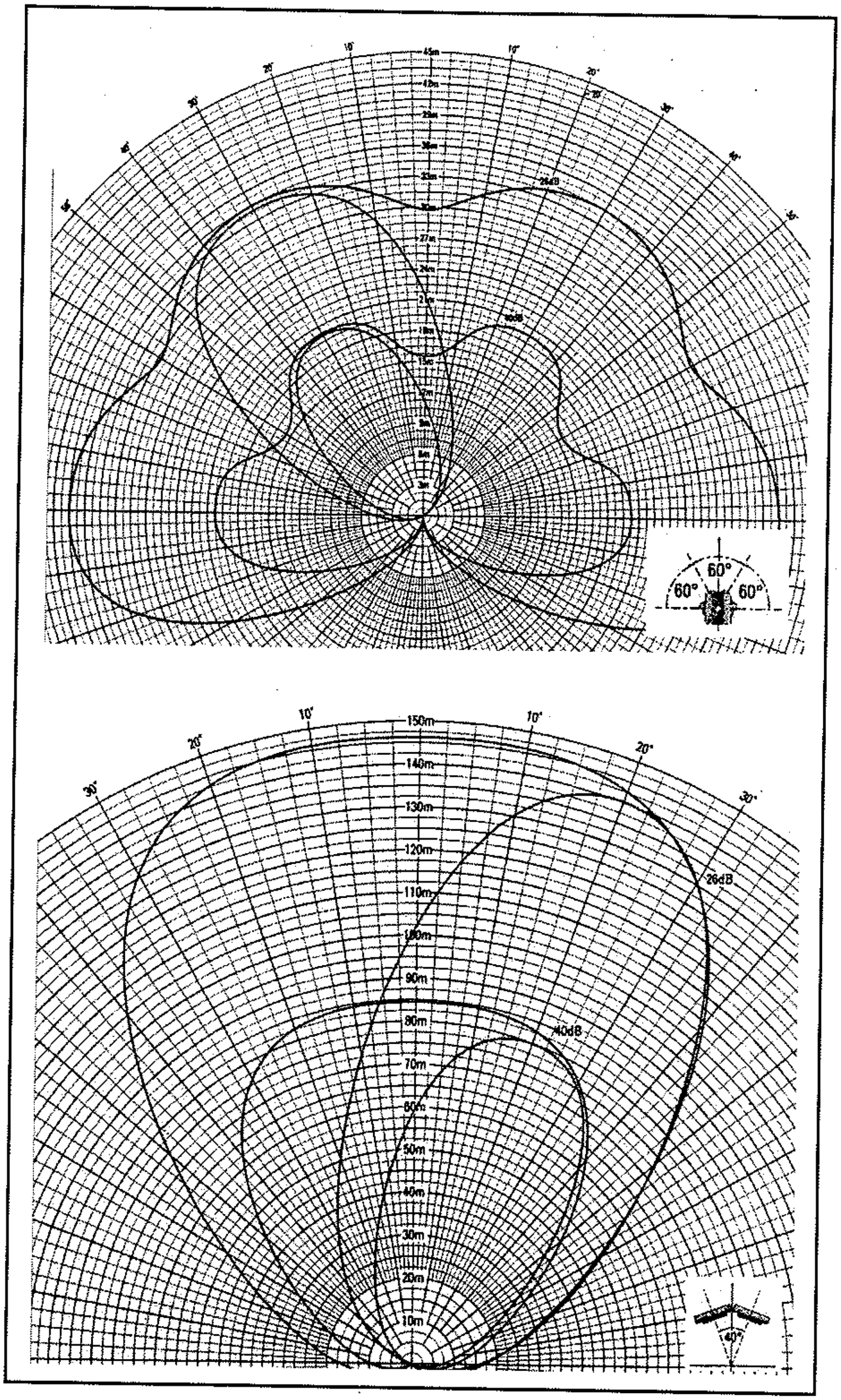


FIGURE 7-16. Polar coverage pattern for four "medium area" IR emitter panels at 60° to each other (left) and two "large area" IR emitter panels at 40° to each other (right) showing 40 dB and 26 dB **S/N** ratio limiting distances. Note wider coverage areas for multiple units compared to Figure 7-15. Data courtesy of Sennheiser Electronic Corporation.

The transmission coverage quality can be checked in an emitter panel installation by walking through the audience area wearing a receiver and positioning one hand six in in front of the receiver LEDs. While splaying the fingers, move the hand in and out in front of the receiver LEDs and note if the background noise level increases. The absence of increased background noise indicates the IR field strength is sufficient. Should the background noise levels increase provide additional emitter panels equal to 50 percent of the original installation quantity.

The IR spectrum in sunlight can interfere with the radiated IR signal from the emitter panels. Direct sunlight on the receiver LEDs will make reception of the emitter panel IR signal difficult. In cases where sunlight is a problem installation of a tinted film, such as ScotchTint™ by 3M Corporation, on windows or skylights can be helpful.

Some fluorescent light systems may cause interference due to the strong IR emission caused by the mercury used in the fluorescent light bulbs. Energy efficient electronic lighting ballasts can modulate the mercury-generated IR emissions, which makes the operation of IR ALS system difficult or impossible at the 95 kHz transmission carrier frequency.

7.2.1.2 Requirements for Portable ALS

Judicial, legislative, or regulatory facilities which are used on a sporadic basis are required to have fixed conduit, wiring, audio connectors, and an electrical power receptacle to support a portable ALS where a fixed ALS is not required or installed. Some applicable spaces include courtrooms, hearing rooms, jury deliberation rooms, jury orientation rooms, town halls, council chambers, and all meeting rooms intended for public use. Portable ALS can be used in historically designed spaces where a fixed ALS would detract from the architectural character of the space.

An IR ALS is used almost exclusively when a portable system is necessary. The installation guidelines described above for the fixed IR emitter panels are generally applicable for the portable system. The one exception is the emitter panel is normally installed on a tripod about 6 ft above the floor and positioned close to the listeners.

Manufacturers produce portable IR ALS with different output levels for various room sizes. These systems typically include both the modulator and emitter panel in one assembly. Line and microphone level inputs permit direct connection of the audio signal to the IR ALS.

7.2.2 Loudspeaker Requirements

Loudspeaker installation involves life-safety, electro-acoustical performance, and design aesthetics. Central cluster, line source, and ceiling distributed systems have unique installation requirements. Architects, sound system designers, and sound system contractors should be familiar with proper mounting hardware application and safe rigging practices. The guidelines below are intended to be general in nature. Specific installations may require unique solutions and the design assistance of a structural engineer should be sought. Some local jurisdictions require loudspeaker rigging to be designed by a licensed structural engineer.

Manufacturers provide a variety of proprietary mounting hardware to simplify loudspeaker installation. This hardware is often pre-engineered for a specific loudspeaker, but knowledge of the building structure is required in order to safely attach the mounting hardware. Other mounting hardware is general in nature suitable for a variety of loudspeaker types and installation applications.

In new building design, the sound system designer should advise the architect of all locations where loudspeakers are to be hung from the building structure, including the center-of-gravity, dimensions, and weight. Using this information the architect or structural engineer can design appropriate structural reinforcement to support the loudspeakers. For sound system renovation or upgrade projects, there is often no architect or structural engineer involved. Architectural drawings may not be available or the location and load capacity of structural members may not be known. The sound system designer or contractor may be unprepared to competently design safe loudspeaker mounting and the assistance of an architect, structural engineer, or rigging subcontractor is strongly recommended.

7.2.2.1 Central Cluster

Central cluster systems require the greatest safeguards for installation since the loudspeakers are suspended above the audience or stage by as much as 40 ft. Rigging can be by custom or proprietary hardware. Other installation considerations include vibration isolation, control of sound reflections from nearby room surfaces, access for servicing, and aesthetics.

7.2.2.1.1 Installation Hardware

Rigging hardware is available in a variety of sizes and quality. Selection of hardware is not an insignificant decision as the installation safety depends on the ultimate breaking strength, referred to as tensile strength, of each hardware item. The use of guaranteed load-rated hardware will add an insignificant cost to the total installation. Load ratings are specified by the Society of Automotive Engineers (SAE) with SAE grade 5 (120,000 psi tensile strength) and grade 8 (150,000 psi tensile strength) hardware commonly used in sound system rigging. Load ratings can be identified by raised lines on bolt heads (grade 5 with three lines at 120° apart and grade 8 with six lines at 60° apart) or direct stamping of the grade rating on each item. Hardware load ratings are based on the load applied along the central axis to the item. Loads applied at angles other than axial will exert other forces which may exceed the hardware axial load rating. Installation hardware includes: (1) shackles; (2) bolts; (3) eye bolts; (4) metal angles or U-channel sections; (5) wire ropes and chains; and (6) slings. Typical installation hardware is shown in Figure 7-17.

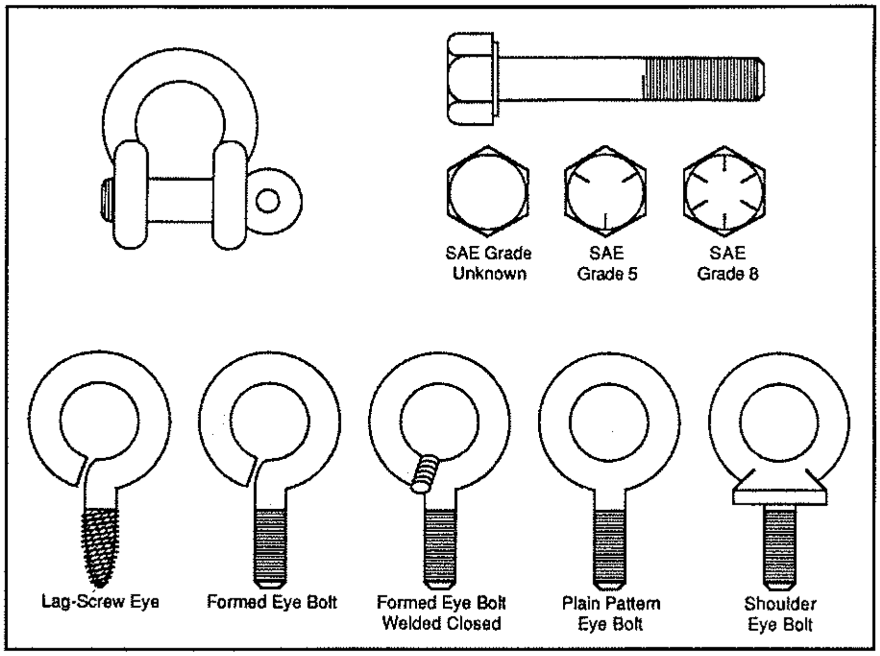


FIGURE 7-17. Typical mounting hardware screw pin anchor shackle (top left), straight bolts (top right), and eye bolts (bottom). Drawing courtesy of JBL, Inc.

Shackles are U- or anchor-shaped metal fittings with a pin used to attach wire rope slings to eye bolts. The screw pin anchor shackle is used for sound system rigging. Sizes range between ½ to 1¼ in diameter with load ratings between 2,800 and 16,400 lbs. The load rating is stamped on the shackle body. Forged carbon steel shackles provide the greatest durability and strength. Shackles should be loaded pin-to-end and never on the sides.

Bolts are straight metal rods having a head at one end and a screw thread at the other end secured by a nut. Bolts are normally of steel construction of various alloys having different tensile strengths. Both load-graded and ungraded bolts are available. Load-graded bolts should only be used for sound system installation. Sizes range between ½ to 1 in diameter with grade 5 load ratings between 2,160 and 9,050 lbs and grade 8 load ratings between 2,700 and 11,800 lbs.

Eye bolts are similar to straight bolts except have a loop for attaching other hardware. Eye bolts are available in four types: (1) formed eye bolt (open and close welded loop); (2) lag screw eye bolt; (3) plain pattern eye bolt; and (4) shoulder eye bolt. The formed eye bolt is a steel rod bent into an eye pattern which may be left open where it meets the shank or welded at this point. Both of these eye bolt types are not load-rated and should not be used for rigging due to their uncertain structural characteristics. The lag screw eye bolt is screwed directly into wood surfaces. The strength and surface area of the wood threads determines the bonding strength. Lag screw eye bolts should not be used due to unreliable anchorage from hygroscopic effects, causing expansion and contraction of the wood. Plain pattern eye bolts are load-rated forged construction and can be used for straight vertical loading. They should not be used for angled loading. Shoulder eye bolts are load-rated forged construction appropriate for rigging, and can be used for angled loading, assuming the force is not oriented more that 45° degree from the axial direction. The bolt needs to be properly oriented when angled loads are suspended. Sizes of shoulder eye bolts range between ¼ and 1¼ in diameter with axial load ratings between 500 and 15,200 lbs. The load ratings decrease between 125 and 3,800 lbs for the same size shoulder eye bolts when the load is applied at 45° degree.

Metal angles or U-channel sections used to reinforce loudspeakers should be of forged or welded construction of minium ⅛ in thick steel with a width-to-length aspect ratio equal to one-to-twelve. These structural elements are normally not load-rated.

The most suitable rope type for sound system rigging is wire rope. A variety of wire rope is available, based on diameter, material (plow steel, improved plow steel, or extra improved plow steel), number of strands, number of wires in each strand, direction of stranding (left or right lang lay and left or right regular lay), and core material. Steel is used because of the high tensile strength, with individual wires (typically 19) wound into strands (typically six), and the strands wound around a central core. Lang lay rope is more flexible and abrasive damage resistant than regular lay rope which has greater crushing strength. The primary purpose of the central core is to support the strands and prevent the outer strands from crushing under load. Independent wire rope core (IWRC), wire strand core (WSC), or fiber core (FC) are available. Wire rope for sound system rigging usually employs improved plow steel of ⅜ or ½ in diameter in 6 by 19 IWRC stranding. The load rating for these ropes are 13,120 and 23,000 lbs, respectively. Care should be taken to minimize bending wire rope as this will reduce its strength since a higher portion of the load bears on a fewer number of wires and strands.

Chains can be used as a substitute for wire rope with SAE grade 8 or higher alloy chain recommended. The alloy chain is more resistant to shock loading and will stretch before it breaks, in contrast to regular steel chain which is brittle and subject to shattering due to shock loading. Care should be taken not to suspend a load from a single chain link.

Wire rope is commonly made into slings either factory- or field-fabricated. Factory-fabricated slings have an integral zinc-cast or swaged metal eye fitting equal in strength to the wire rope itself. Factory-fabricated polyester and natural rope slings are available but are not recommended for permanent sound system installations due to durability and fire-rating concerns. Field-fabricated slings use forged load-rated clips to anchor the ends of the wire rope. Commonly used clips include J-bolts ("fist-grip" clips) and U-bolts ("Crosby" clips). The number of clips, spacing between clips, and the applied screw torque will determine the strength of the field-fabricated sling. Periodic inspection and retightening the screws is necessary with field-fabricated slings. Figure 7-18 shows installation guidelines for field-fabricated wire rope slings.

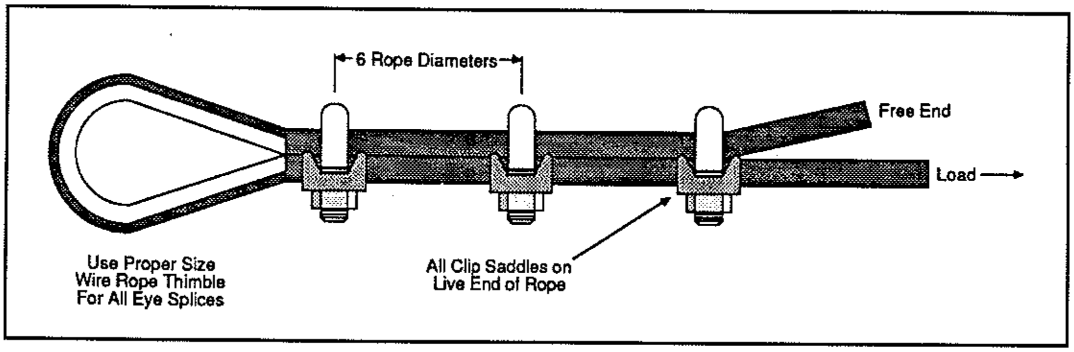


FIGURE 7-18. Installation guidelines for field-fabricated wire rope slings. Drawing courtesy of JBL, Inc.

7.2.2.1.2 General Installation Practices

Rigging and installation hardware should be installed as it is designed. Central cluster installation guidelines for new and existing building construction are noted below.

1. Safety is the most important consideration and installation precautions are necessary to protect the public. The potential for human injury and property damage due to unsafe rigging or installation practices should always be considered a possibility. Never downgrade installation hardware or rigging from what has been designed.

2. When in doubt seek professional help from an architect, structural engineer, or rigging subcontractor.

3. Know the structural integrity of beams, joists, and columns. Hidden damage can be present which can weaken the structural member.

4. Securely attach equipment only to structural members. Do not make direct attachment to wall or ceiling surfaces.

5. Do not overload any rigging point.

6. Do not use wood screws, nails, or rivets to install mounting hardware. Only use load-rated straight bolts and plain pattern or shoulder eye bolts. All hardware should be purchased from knowledgeable sources specializing in rigging hardware.

7. Fabric or nylon-type ropes should not be used due to potential damage from stretching, vermin, wearing, or weather, and its low fire resistance.

8. Sharp or abrasive objects should not installed be near wire rope or chains to avoid cutting and nicking.

9. Provide a safety chain in case the primary rigging suspension hardware fails.

10. Welding structural members or rigging hardware can result in weakness unless done properly. Only certified welders should be used.

11. Grilles and grille cloth frames should be screw-mounted to the cabinet. Adhesives, hook-and-loop fabrics, or magnets should not be used as these materials may fail causing the grille cloth frame to fall.

12. Loudspeakers should be located close together so the acoustic centers are physically near each other.

13. All loudspeakers should have adjustable mounting hardware to permit approximately a 5° variation in loudspeaker aiming angles.

14. Carefully inspect all attachment and anchorage points at installation completion.

Loudspeakers can be suspended using either wire rope or chain. Attachment to structural members can be by forged load-rated eye bolts or a sling wrapped around the structural member. In the first installation method, wire rope or chain is attached with forged load-rated shackles to the eye bolts. The other end of the wire rope or chain is attached with forged load-rated shackles and carabiners connected to integral loudspeaker mounting hardware. The shackle should carry two separate chains for safety. The other installation method is similar except a wire rope or chain sling wraps around the structural member 1½ times with soft padding between the sling and the structural member. Both sides of the chain should wrap around back to the shackle when chains are used to prevent the full load from being carried by one chain link. Snap hooks or "quick links" should not be used. The former are usually not load-rated and "quick links" have a screw threaded sleeve which can vibrate loose. With these installation techniques, the loudspeaker hangs from three suspension points on the cabinet top. Two are located at the cabinet top front edge and one is at the cabinet top rear edge. Adjusting the relative cable lengths sets the loudspeaker aiming angle.

Many loudspeaker cabinets are engineered to have a specified load rating. These cabinets can be identified by integral mounting hardware, usually "universal mounting hardware" offering 12 stepped adjustment points or three fixed fly points at the cabinet top. Attachment of wire rope or chain slings can be made directly to these attachments. Loudspeakers without factory-installed mounting hardware should have structural reinforcement if the cabinet weighs in excess of 40 lbs and is to be hung. An unreinforced wood cabinet will be no stronger than the materials and joinery methods used in manufacture. Many cabinets are made from particle board, medium density fiberboard (MDF), and other wood fiber composites. These cabinets should only be used indoors due to the hygroscopic nature and expansion/contraction of the wood fiber from humidity and wide temperature variations. These extreme environmental conditions can weaken the cabinet material. Quality cabinets use 9 ply

"Finnish" plywood which offers greater resistance to humidity and weather damage. Lightweight particle board or fiber board cabinets can have mounting hardware directly installed to the cabinet provided ⅛ in steel load distribution plates of nominal 2 in by 2 in dimensions are attached to the exterior and interior wood surfaces which load-rated shoulder eye bolts, locking washers, and nuts attach to.

Loudspeaker cabinet reinforcement can be either internal or external, depending on the cabinet material. Internal reinforcement is suitable only for plywood cabinets. Nominal ⅛ in thick steel plates and corner brackets are attached to the internal and external cabinet surfaces at all load-bearing points. The disadvantage with this reinforcement method is the drivers have to be removed from the cabinet to gain access to the cabinet interior. External reinforcement can be used for particle board and other fiber board, as well as, plywood cabinets. Steel angles or channels are attached externally with load-rated bolts, locking washers, and screws to all four cabinet sides at 8 in centers. The reinforcement should anchor dadoed front baffles which carry the weight of the drivers. See Figure 7-19 for internal and external loudspeaker reinforcement methods.

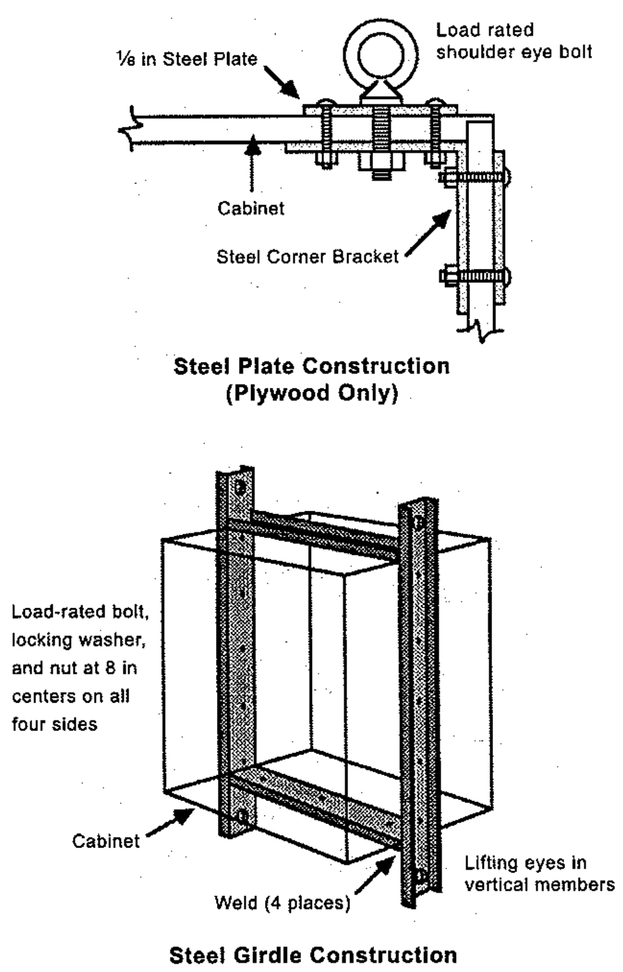


FIGURE 7-19. Reinforcement of loudspeaker cabinets with steel plate internal reinforcement for plywood cabinets (top) and external reinforcement for particle board, fiber board, or plywood cabinets (bottom). Drawing courtesy of JBL, Inc.

(See Technical Notes, Sections 7.B and 7.C, at the end of this chapter, for additional information on strength and corrosion of rigging hardware.)

7.2.2.1.3 Proprietary Systems

Proprietary hanging hardware and systems are available from a variety of manufacturers, including strut-type hardware requiring contractor-assembly or complete ready-to-install hanging systems. Strut frame hardware is marketed under different trade names such as B-Line®, Elcen®, Globe®, Kindorf®, Powerstrut®, Superstrut®, and Unistrut®. Preassembled hanging systems are available from ATM Group and Polar Focus®.

Strut hardware is available in carbon steel, stainless steel, and aluminum channel sections in various sizes. Both solid and prepunched channels with regular hole spacings are available. Carbon steel channels are fabricated from ASTM A500 Grade B or ASTM A36 steel which is continuously roll formed by a cold working process to maintain the material structural properties. The metal can be finished in a variety of coatings to inhibit corrosion. Stainless steel channels are fabricated from ANSI Type 304 or 316 materials in a cold rolled process. The Type 316 material offers better corrosion resistance in harsh environments. Aluminum channels are extruded from either 5052-H32 or 6063-T alloys. Steel hardware can be finished with zinc coatings to include: (1) electro-plate; (2) pregalvanized; (3) mechanical galvanizing; and (4) hot-dip galvanized.

Strut frame hardware is assembled into a grid, called a hanging bar or bumper to which wire rope or chain is attached to support the loudspeakers. The hanging bar holds the loudspeakers in place at the desired aiming angles and permits the array to be lifted into place as one unit. Grid fabrication can be either by welding or installing manufacturer-supplied fittings. The grid suspension points, typically two to four, are attached to the building structure to support the loudspeaker array. The grid is designed to transfer the loads from each loudspeaker attachment point on the strut frame hardware to the suspension points. The strut frame hardware is subject to bending loads by the loudspeakers and compression or tension loads by the aiming angles of the wire rope or chain which supports the grid. Common practice for grid design is to use radial front-to-back framing to anchor the loudspeakers with front and rear cross members for stabilization. Figure 7-20 shows a representative strut frame channel and Figure 7-21 shows an installation example.

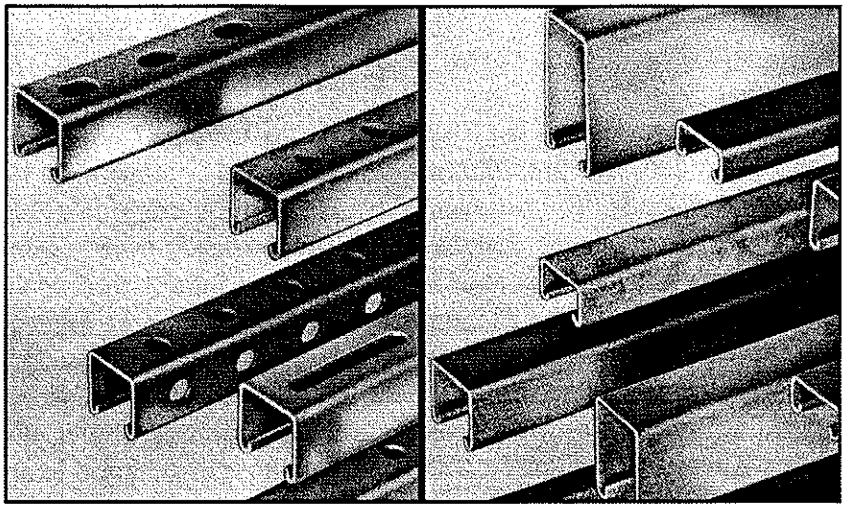


FIGURE 7-20. Strut frame channel sections prepunched (left) and solid (right). Photo courtesy of B-Line®.

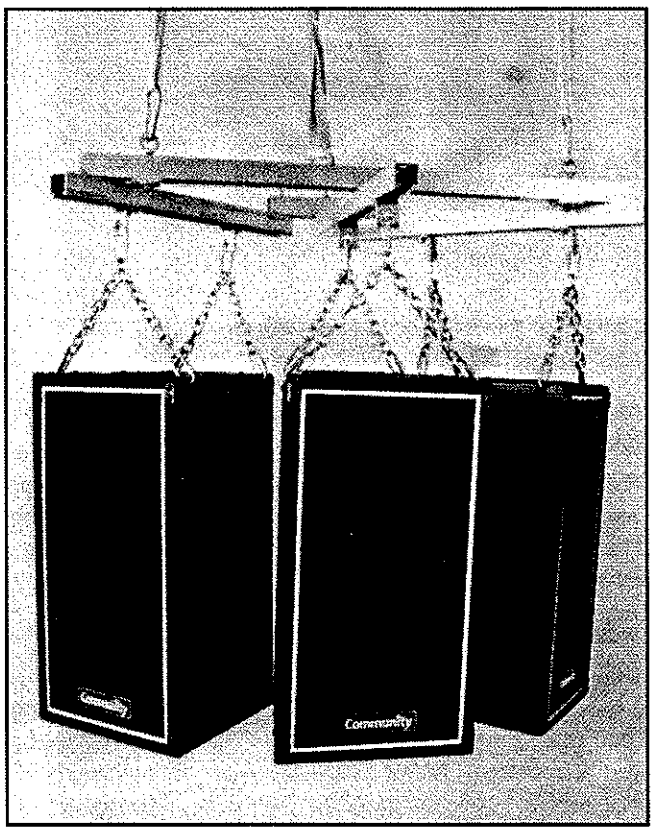


FIGURE 7-21. Frame-type loudspeaker suspension grid. Photo courtesy of Community Light and Sound, Inc.

Preassembled hanging systems include a suspension grid and associated hanging hardware accessories. The contractor provides the wire rope or chain sling, shackles, and shoulder eye bolts to attach the loudspeakers to the grid and to the building structure.

One unique product is the Zbeam® from Polar Focus® which permits loudspeakers to be suspended and provides rotational control around a vertical axis for balanced suspended loads. The product is available in a variety of configurations suitable for different loudspeaker installations. The basic Zbeam® comprises two aluminum extrusions joined by an axle and friction bearings with four vernier scales to adjust the aiming angles. The friction bearing holds the aiming angles once the loudspeaker is adjusted. The structural members have a load capacity of 800 lbs and are prepunched to permit direct attachment of shackles. Accessory tilt cable kits permit rapid adjustment of loudspeaker downward aiming angles. Figure 7-22 shows the basic Zbeam® and Figures 7-23 and 7-24 show installation examples.

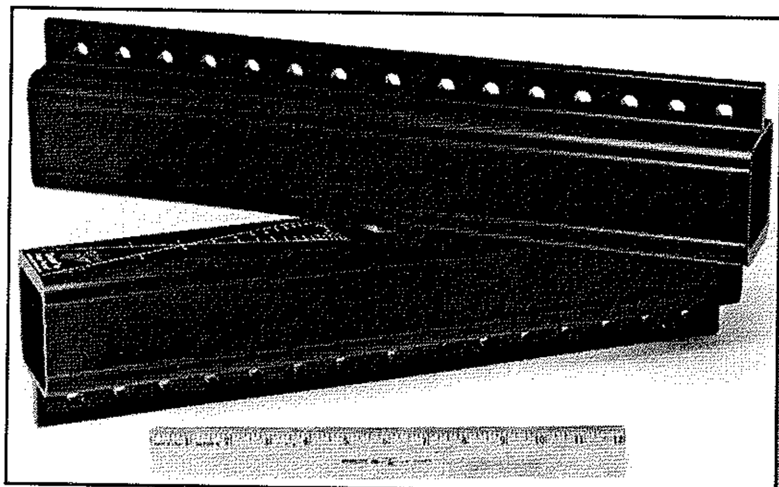


FIGURE 7-22. Polar Focus® Model ZB-20-880-B Zbeam®. Product courtesy of Polar Focus®, Inc.

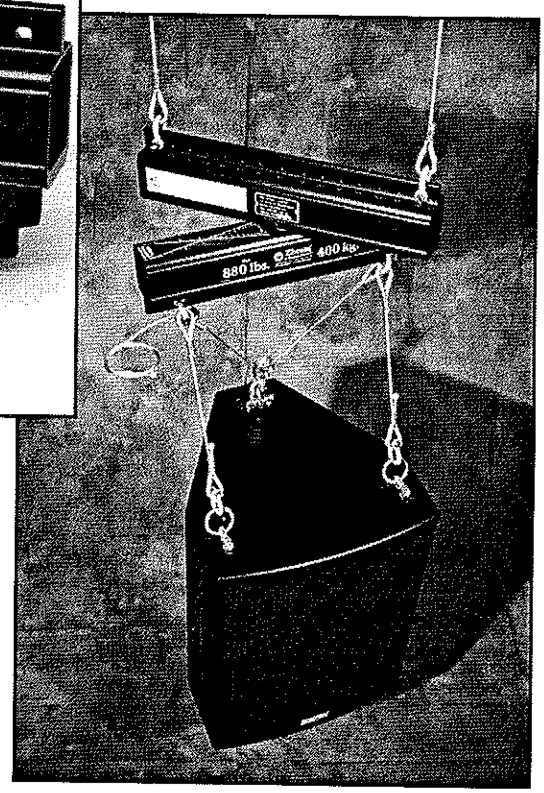


FIGURE 7-23. Polar Focus® Model ZB-20-880-B Zbeam® supporting a single full-range loudspeaker. Note integral three-point mounting hardware on loudspeaker top, eye rings, shackles, and wire rope suspension hardware. Photo courtesy of Polar Focus®, Inc.

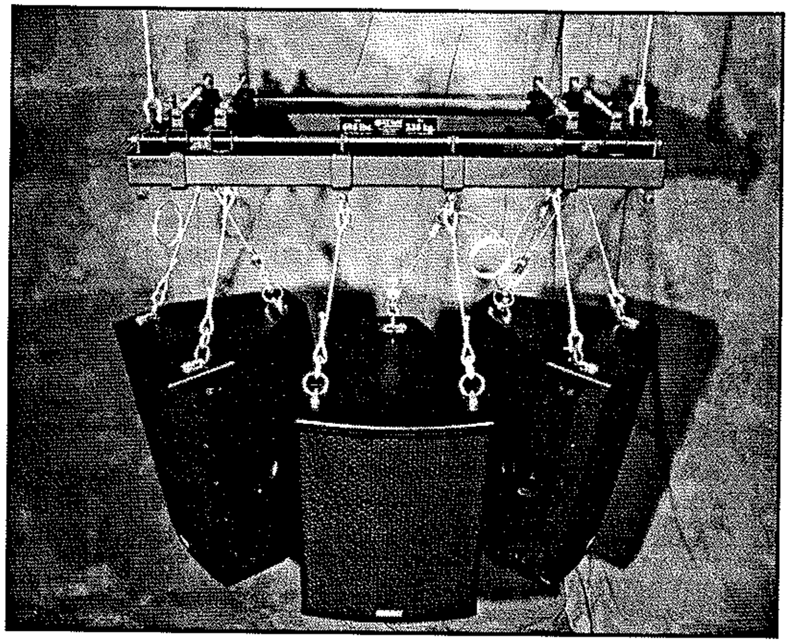


FIGURE 7-24. Polar Focus® Model XY Grid™ XY-6622-495 supporting multiple full-range loudspeakers. Loudspeakers can be positioned about the X-Y plane. Photo courtesy of Polar Focus®, Inc.

7.2.2.1.4 Loudspeaker Vibration Isolation

Loudspeaker installations may require vibration isolation to limit structureborne sound from being transmitted into the building which can radiate as unintended airborne sound at distant locations. The vibration isolator type will depend on whether the loudspeaker is ceiling-suspended or floor-mounted and the woofer low-frequency cutoff. Vibration isolators comprise separate neoprene-only elements or combination steel springs with neoprene elements.

Neoprene-only elements are available as: (1) pads, usually ¼ to ½ in thickness; (2) truncated cone-shaped mounts with attachment studs; and (3) truncated cone-shaped mounts with attachment studs integral within a hanger assembly. Pads and mounts can be used for floor-mounting or direct attachment to the structure above. Isolated hangers are for overhead suspension. Neoprene is available in a variety of hardness ratings (*durometer* or shore hardness). Durometer ratings vary between 30 and 70, with higher numerical values suitable for supporting greater loads or providing less vibration isolation. Pads come in sheets, typically 24 in by 24 in, and the sheets are cut to fit the supported item. Mounts and hangers come in different prefabricated sizes. Figure 7-25 shows neoprene pad and mount vibration isolators and Figure 7-27 shows a neoprene hanger vibration isolator.

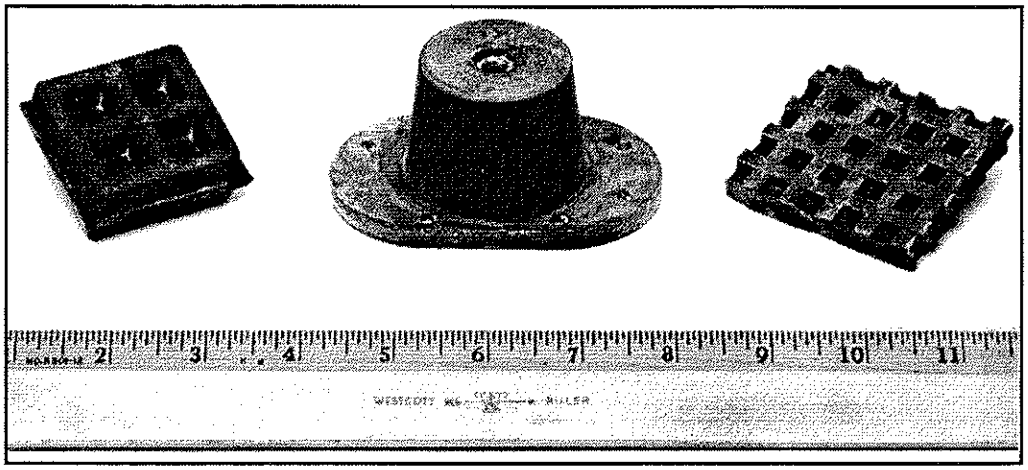


FIGURE 7-25. Neoprene pad vibration isolators (Mason Industries Super W left and Type W right) and neoprene mount vibration isolator (Mason Industries Type ND center). Products courtesy of Mason Industries, Inc.

Steel spring with neoprene element vibration isolators are intended either for floor-mounting or hanging from the structure above. The spring element provides vibration isolation for low-frequencies; the neoprene element provides vibration isolation for high-frequencies. The spring element is available in a variety of *spring constant* ratings, with higher numerical values suitable for supporting greater loads. Different prefabricated sizes are available for mounts and hangers. Figures 7-26 shows floor-mounted steel spring with neoprene element vibration isolators and Figure 7-27 shows ceiling-mounted steel spring with neoprene element vibration isolators.

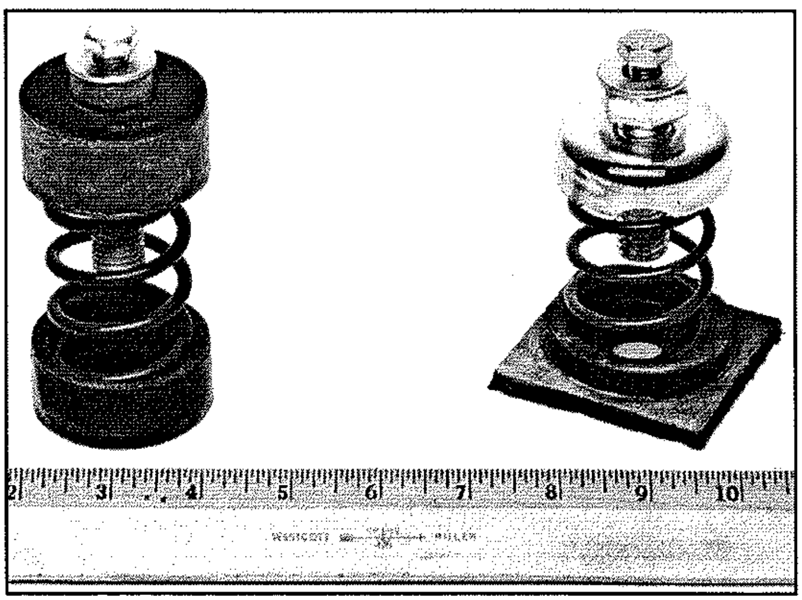


FIGURE 7-26. Combination steel spring and neoprene element vibration isolators (Mason Industries Type SLF left and Kinetics Noise Control Model FDS right). Products courtesy of Kinetics Noise Control and Mason Industries, Inc.

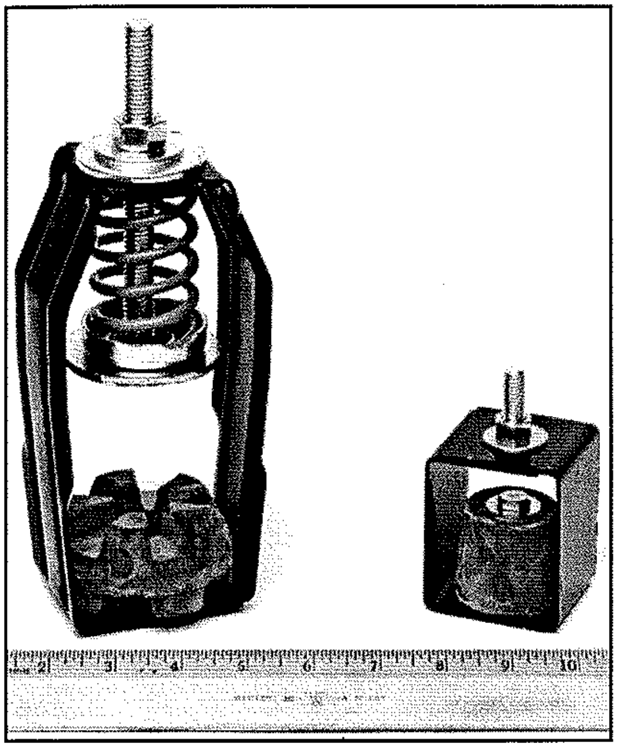


FIGURE 7-27. Combination steel spring and neoprene element hanger vibration isolator (Kinetics Model SRH left) and neoprene element hanger vibration isolator (Mason Industries Type HD right). Products courtesy of Kinetics Noise Control and Mason Industries, Inc.

The key to vibration isolation is to provide a resilient element between the vibrating source and the building structure which reduces the vibration *transmissibility* into the structure. Numerical values of transmissibility less than one indicate vibration isolation. A value of one indicates no vibration isolation. Values greater than one indicate the vibration is amplified.

The forcing frequency (**f_f**) is the lowest frequency the loudspeaker reproduces and is normally taken at the -10 dB down point in the loudspeaker response. The natural frequency (**f_n**) is the rate at which the supported loudspeaker moves about the vibration isolator zero displacement equilibrium position. Vibration isolators should be selected to have an **f_n** value which is one-tenth of **f_f** to achieve good vibration isolation. This implies vibration isolation results for all frequencies at or above **f_f**.

The vibration isolator static deflection (**SD**) measured in inches is the difference between the vibration isolator free height without load and the compressed height with load. Table 7-1 provides information on the range of static deflection, **f_n**, and **f_f** for different vibration isolators.

TABLE 7-1. Characteristics Of Vibration Isolators

Vibration Isolator Type	SD Range, in	f_n Range, Hz	f_f Range, Hz
Steel Spring and Neoprene Element Mount or Hanger	0.25 to 4.0	6.3 to 1.6	63 to 16
Neoprene Mount or Hanger	0.10 to 0.50	10.0 to 4.4	100 to 44
Neoprene Pad	0.02 to 0.25	22.0 to 6.3	220 to 63

The table indicates that steel spring and neoprene element vibration isolators will be necessary for most full-range loudspeakers having woofers greater than about 10 in and all subwoofers. Neoprene mounts, hangers, or pads are suitable for most other loudspeakers.

The vibration isolator SD requirements can be calculated using the following equation:

$$\mathbf{SD} = \frac{9.8}{f_n^2} \tag{7.3}$$

where,

SD is the static deflection, in

f_n is the natural frequency of the system, Hz

The SD will depend on the vibration isolator durometer or spring constant (**k**) and the supported or hung load at each attachment point. To determine the required durometer or spring constant, first divide the loudspeaker weight by the number of isolation points, to arrive at a load per isolator (**W**). Then **k** can be calculated using the following equation:

$$\mathbf{k} = \frac{\mathbf{W}}{\mathbf{SD}} \tag{7.4}$$

where,

k is the spring constant, lbs/in

W is the load per isolator, lbs

SD is as above

Installation guidelines for vibration isolators and loudspeakers are noted below.

1. Vibration isolators work best when attached or supported at a rigid or heavy surface such as a concrete slab.

2. Increase the calculated **SD** by a factor of two to account for the flexibility of the structure when vibration isolators are attached or supported on lightweight surfaces.

3. Provide a steel channel base with mounting brackets for attaching vibration isolators when supporting floor-mounted loudspeakers.

4. Attach vibration isolator hangers as close to the structure as possible. Threaded rod or steel aircraft cable can be used to connect the loudspeaker to the vibration isolator.

5. Do not attach rigid conduit or maintain taut cables at loudspeakers, otherwise vibration will transmit via these paths. Provide flexible "BX" style armored conduit or use a full 360° degree loop in the loudspeaker cable.

6. After installation, rock the loudspeaker system back and forth to verify the vibration isolators are free to move and the spring coil elements do not compress to solid. Verify rods or wires are not touching hanger-type vibration isolators and that no construction debris remains under mount-type vibration isolators.

7.2.2.1.5 Loudspeaker Sound Reflection Control

Loudspeakers positioned near ceiling and wall surfaces can cause specular sound reflections which result in comb filtering. Assuming the loudspeakers are aimed properly, the comb filtering tends to be controlled by mid- to low-frequency sound where the loudspeaker starts to lose directional control. As a rule-of-thumb reflecting surfaces which are within 5 to 10 ft of the loudspeaker should have some acoustical treatment to suppress reflected sound. Figure 7-28 shows applications of sound absorptive material near a ceiling loudspeaker.

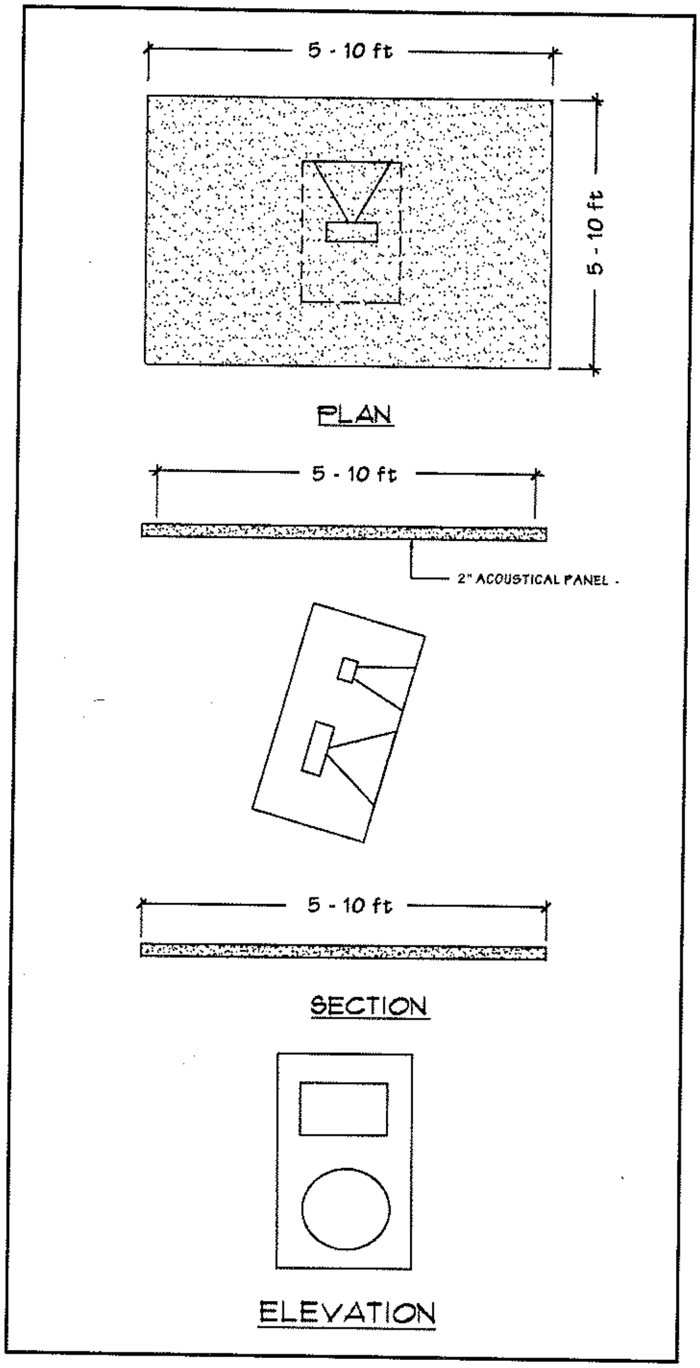


FIGURE 7-28. Application of sound absorptive material near a loudspeaker. Material coverage around the loudspeaker is between 5 to 10 ft from the center of the cabinet or cluster.

The comb filtering tends to be more of a problem affecting musical timbre than speech. At a 5 ft path length difference the first null in the comb filter response will be at 113 Hz. A 10 ft path length difference will correspond to a null at 56 Hz. The loss in the low-frequency range can give the perception "weak bass" in music.

Applying sound absorptive materials around the loudspeaker can be an effective technique to reduce reflected sound. The material should be sufficiently thick to reduce reflected sound in the 125 Hz octave band by a minimum of 6 dB. This would correspond to a material having a α value of 0.75. A fiberglass board of 6 lb/ft^3 density in 4 in thickness, such as Owens Corning Type 705 is recommended. Thicker fiberglass is necessary for reflection control at lower frequencies, but this is often aesthetically not practical unless the loudspeakers are recessed into a wall or ceiling. In these installations, the use of standard R-30 fiberglass insulation can be used as shown in Figure 4-55. The reflected sound level reduction (**LR**) due to a sound absorbing material can be calculated for a given frequency using the following equation:

$$\mathbf{LR} = \mathbf{10\log_{10}(1-\alpha)} \tag{7.5}$$

where,

LR is the sound level reduction due to the absorptive material, dB

α is as above

For ceilings, an alternate to installing sound absorptive material is to hang the loudspeaker upside down with the woofer closer to the ceiling and the horn closer to the audience. At lower frequencies the woofer will have a half space radiation which will reduce the path length difference and shift the first comb filter null to a higher frequency.

7.2.2.1.6 Access for Servicing

Provisions to service loudspeakers should be planned for. Large cluster systems often have a dedicated winch to lower the loudspeakers to the stage. Adding a winch can be an expensive proposition and is normally warranted only for larger systems. Unless the loudspeaker cable is disconnected prior to lowering, the winch needs a take-up reel for the loudspeaker cable slack to permit the entire assembly to be lowered. The winch also requires a mounting platform, a source of electrical power, and a catwalk for service access. As an alternate to a winch, many owners invest in a telescoping man-lift for loudspeaker servicing.

7.2.2.1.7 Aesthetics

Cluster loudspeaker systems should be exposed in the room and not be hidden behind architectural elements for best electro-acoustic performance and ease of servicing. This recommendation is often in conflict with aesthetic considerations. Loudspeakers can be painted in a color to match the surrounding surfaces which minimizes the visual impact. Tight-packing the loudspeakers will also reduce the physical cluster size and help align the acoustic centers to minimize comb filtering. In spaces where exposing the central cluster is unacceptable, the loudspeakers can be installed behind a cut-out above the stage with a transondent grille cloth in front. The guidelines in Section 4.4.6 should be followed.

7.2.2.2 Line Source Loudspeakers

Line source loudspeakers are generally physically smaller and weigh less than cluster-type loudspeakers. Installation to wall and ceiling surfaces can be made using commercial universal mounting assemblies which have an adjustable ball-and-clamp to permit rotating and aiming the loudspeaker in both the horizontal and vertical planes. One end of the mounting assembly attaches to the wall or ceiling surface and the other end attaches to the loudspeaker. A variety of sizes, configurations, and colors are available with load capacities between 5 and 225 lbs. Figures 7-29 and 7-30 show different universal mounting assemblies.

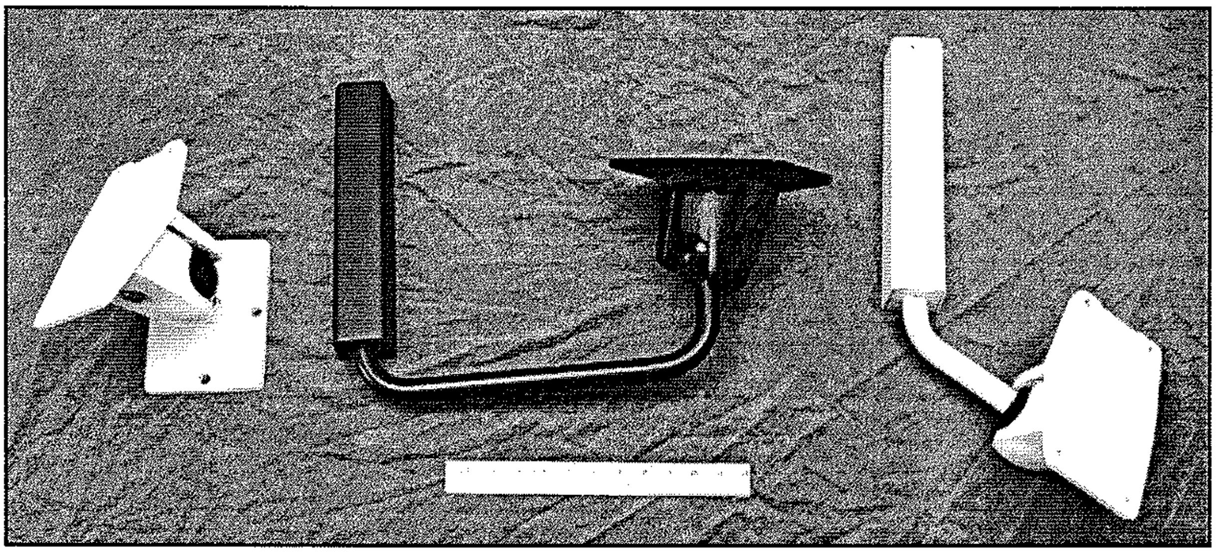


FIGURE 7-29. Large universal mounting assemblies in black and white colors for 120 lb loads (Omnimount® 300ST-MP, left, 300WBX, center, and 300WA, right). Note size compared to mounting assemblies in Figure 7-30. Products courtesy Omnimount® Systems, Inc.

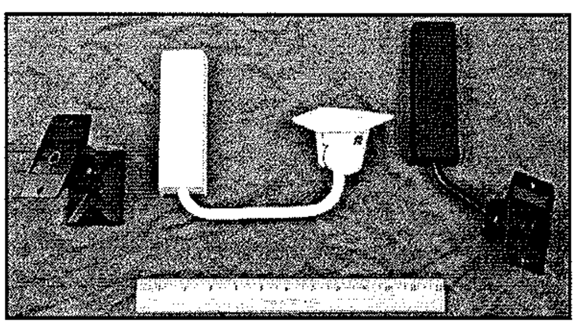


FIGURE 7-30. Small universal mounting assemblies in black and white colors for 15 lb loads (Omnimount® 50ST-MP left, 50WB center, and 50WA right). Products courtesy of Omnimount® Systems, Inc.

Standard column loudspeakers need to be vertically mounted and angled down towards the audience to limit unintended sound reflections off room surfaces. The column loudspeaker should be aimed at a point about two-thirds to three-fourths of the distance between the loudspeaker and the last seating row. Good coverage can be achieved in spaces where the coverage distance is no greater than seven times the column height above the floor. The horizontal spacing between column loudspeakers should be no greater than 45 ft apart, with the length spacing between column rows determined by the loudspeaker height above the floor. Electronic signal delay is recommended to preserve localization with multiple loudspeaker rows.

Some installations can recess column loudspeakers into the wall surface where it is desired conceal their presence. The above installation guidelines are applicable. Figures 7-31 and 7-32 shows an installation example of an in-wall mounted column loudspeaker.

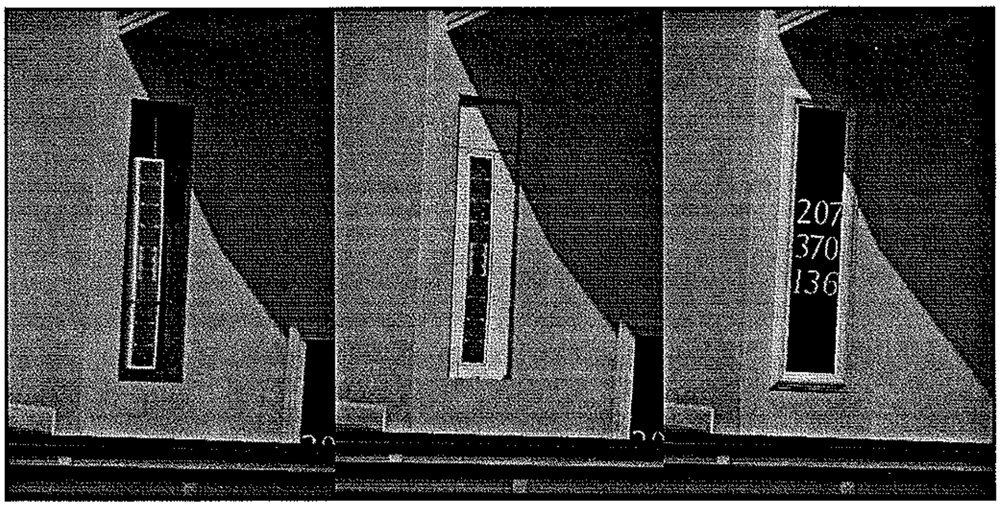


FIGURE 7-31. Close-up view of in-wall installation of EAW LS832 column loudspeaker at the Third Church of Christ Scientist in New York, NY showing infill around loudspeaker and grille doubling as hymn notification board. System design by Tom Young of Electroacoustic Design Services. Photo courtesy of Electroacoustic Design Services.

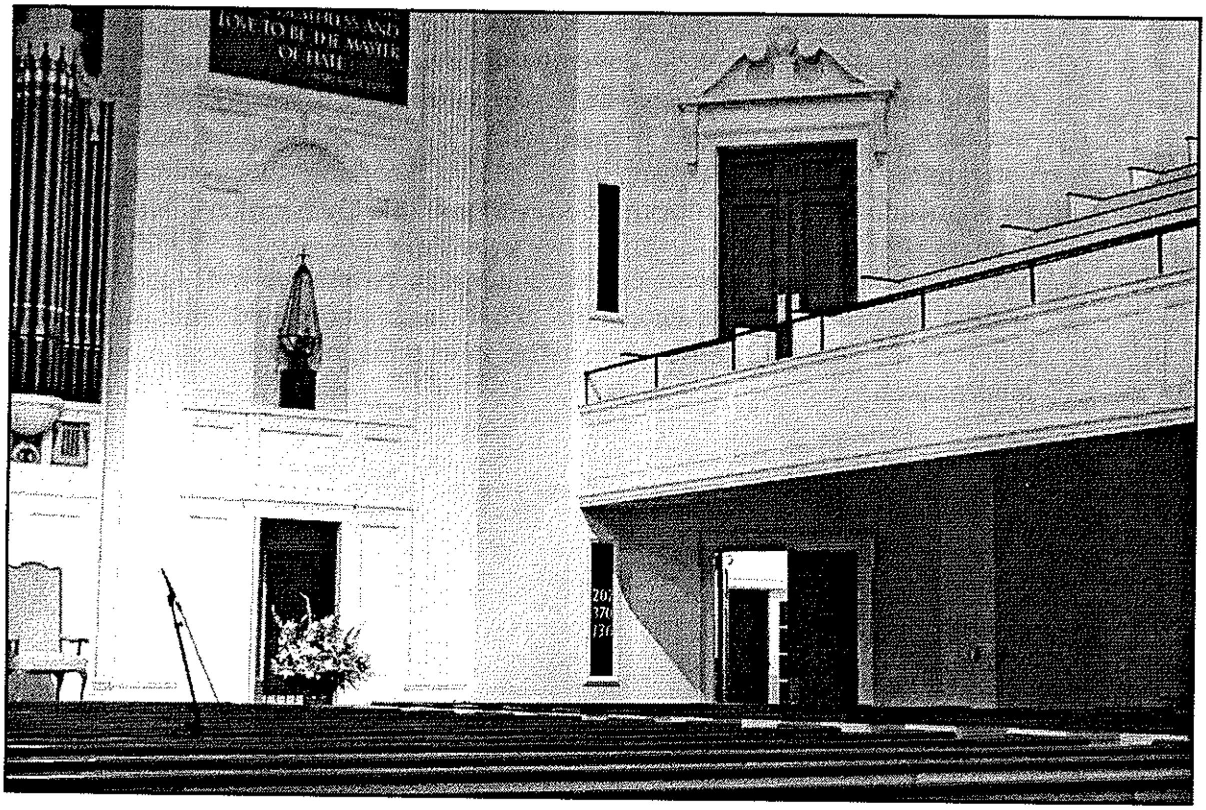


FIGURE 7-32. Distant view of column loudspeaker shown in Figure 7-31. Note low elevation of loudspeakers relative to main floor and balcony pews to maintain high **D/R** ratio. System design by Tom Young of Electroacoustic Design Services. Photo courtesy of Electroacoustic Design Services.

Horizontal coverage by column loudspeakers is very wide and can present gain-before-feedback problems. The column loudspeaker should be placed as far from the microphone as possible. In some feedback-prone installations this can require placing the microphone at the left and the column loudspeaker at the right which sacrifices auditory localization.

7.2.2.3 Ceiling Distributed Systems

The loudspeakers in ceiling distributed systems are installed in one of three methods: (1) flush mounted; (2) surface mounted; and (3) freely suspended. The most common installation is to flush mount the loudspeaker grille to an acoustical tile ceiling with the back box enclosure recessed within the ceiling return air plenum. Surface mounting attaches the back box directly to the ceiling and is used with non-accessible ceiling surfaces such as concrete, GWB, or plaster. Freely suspended loudspeakers are often used in worship houses where the loudspeakers are integral within chandelier light fixtures or use free-hanging back boxes similar to those shown in Figure 4-45.

Coordination with HVAC diffusers, lights, smoke detectors, and sprinklers is necessary for ceiling loudspeakers. The sound system designer has to accept compromises in loudspeaker positioning due to other building services at the ceiling which have Code-defined spacing requirements. Another coordination issue is to verify the size of the back box is compatible with the return air plenum depth and the position relative to HVAC ducts.

The weight of flush mounted loudspeaker enclosures needs to be assessed for acoustical ceiling tiles and the supporting suspension grid system due to the potential for sagging. The sagging resistance will be a function of the acoustical ceiling tile material and the suspension grid type. Suspension grids are normally classified as: (1) light duty (commercial or residential construction), designed to carry acoustical ceiling tile and few HVAC diffusers and lights; (2) intermediate duty (normal commercial construction), designed to carry acoustical ceiling tile and ordinary HVAC diffusers and lights; and (3) heavy duty (heavy commercial or institutional construction), designed to carry acoustical ceiling tile and numerous HVAC diffusers and lights.

Installation guidelines for flush mounted loudspeakers are noted below.

1. Thicker and heavier acoustical ceiling tiles, such as mineral fiber types, will have greater load carrying capacity than thinner or lightweight ceiling tiles, such as fiberglass types.

2. Suspension grid load carrying capacities are: (1) light duty 6 to 12 lbs/lin ft; (2) intermediate duty - 12 to 16 lbs/lin ft; and (3) heavy duty - greater than 16 lbs/lin ft. These classification ratings refer to the load carrying capacity of the suspension system main runner.

3. Suspension grid systems with wider profile sections have greater load carrying capacity than smaller narrower profile sections.

4. The maximum uniform load a grid component can carry prior to the maximum allowable deflection per ASTM C 635 is 1/360 of the span.

5. Reducing the hanger wire spacing for the main runner and other ceiling suspension members can significantly increase the load carrying capacity. A typical reduction from 5 ft on center spacing to 4 ft on center spacing can double the load carrying capacity.

6. Humid environmental conditions or intermittent seasonal use of the facility where the HVAC system is turned off can accelerate sagging. Some ceiling tiles are formulated from high sag resistance materials.

7. Individual loudspeaker fixtures which weight in excess of 50 lbs should be supported from the structure above and not supported on the acoustical tile ceiling or suspension grid.

Loudspeaker manufacturers offer a variety of installation accessories for distributed ceiling systems including ceiling tile bridges and mounting rings. Ceiling tile bridges comprise 24 gauge rust resistant sheet metal with precut holes for round and rectangular shaped back boxes and grilles. The ceiling tile bridge installs directly behind the acoustical ceiling tile and distributes the loudspeaker and back box weight to the grid system to help minimize ceiling tile sagging. Mounting rings are pre-drilled metal or plastic units intended to directly mount the loudspeaker and grille in acoustical tile, gypsum board, or plaster ceiling surfaces which do not require back boxes. Both ceiling tile bridges and mounting rings are available in various sizes for 4 and 8 in loudspeakers. Figures 7-33 and 7-34 show ceiling tile bridges and mounting rings.

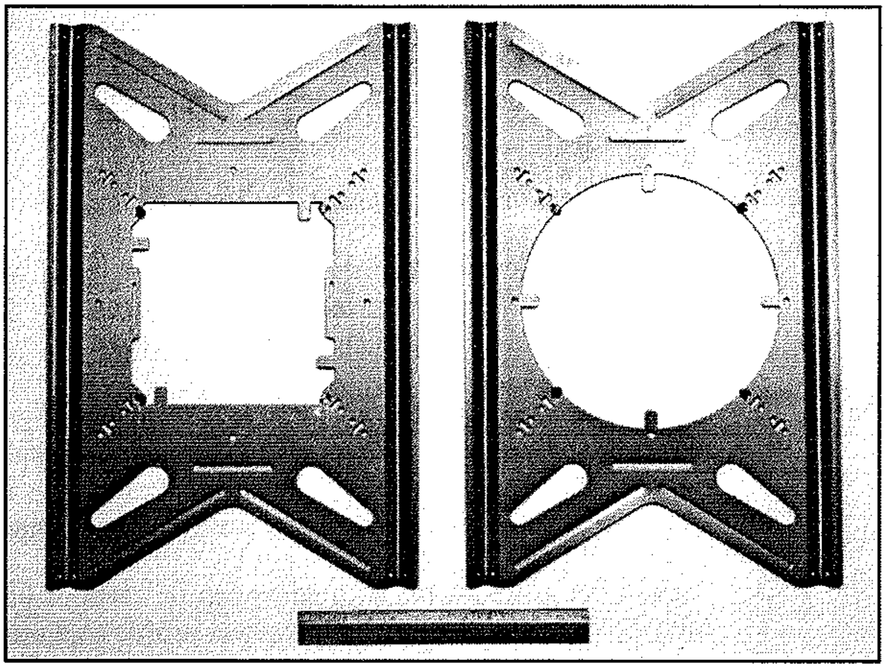


FIGURE 7-33. Ceiling tile bridges (rectangular, left, and round, right) for installing 8 in loudspeakers in acoustical ceiling tile. Products courtesy of Atlas/Soundolier, Inc.

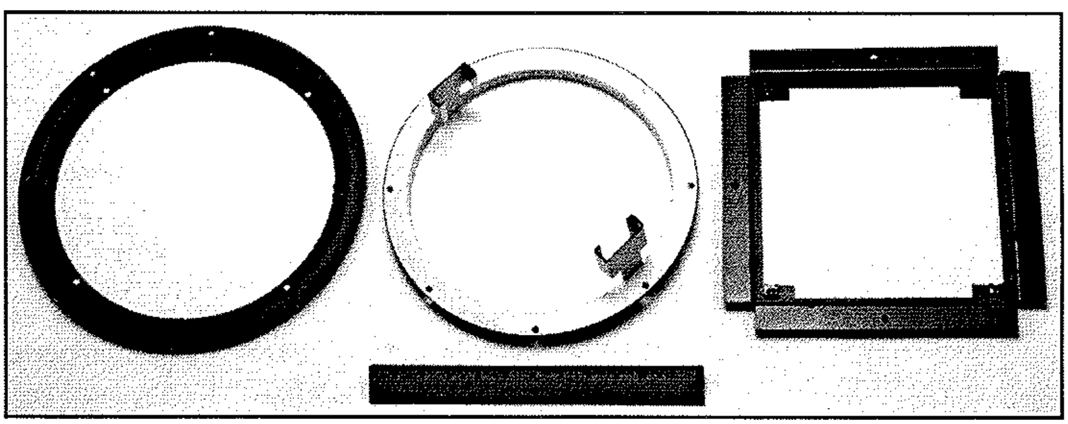


FIGURE 7-34. Mounting rings for installing 8 in loudspeakers in acoustical ceiling tile, gypsum board, or plaster ceilings. Products courtesy of Atlas/Soundolier, Inc.

Loudspeaker grilles are available in square and circular shapes fabricated from perforated metal or plastic. Sizes are available for 4, 6½, 8, 12, and 15 in loudspeakers. Flush and recessed profile grilles are the most common, although contoured shapes are available. Colors are normally white or silver, but can be painted to match surrounding surfaces. Manufacturers make general purpose grilles, suitable for a variety of different loudspeakers and back boxes, and loudspeaker or back box-specific grilles. To minimize off-axis high-frequency attenuation, grilles should be selected with the largest percent open area. Square grilles tend to have a higher perforation area than circular grilles and are therefore recommended. Note that the grille size will be much larger than the loudspeaker size due to the perimeter mounting flange. Figure 7-35 shows different square and circular loudspeaker grilles.

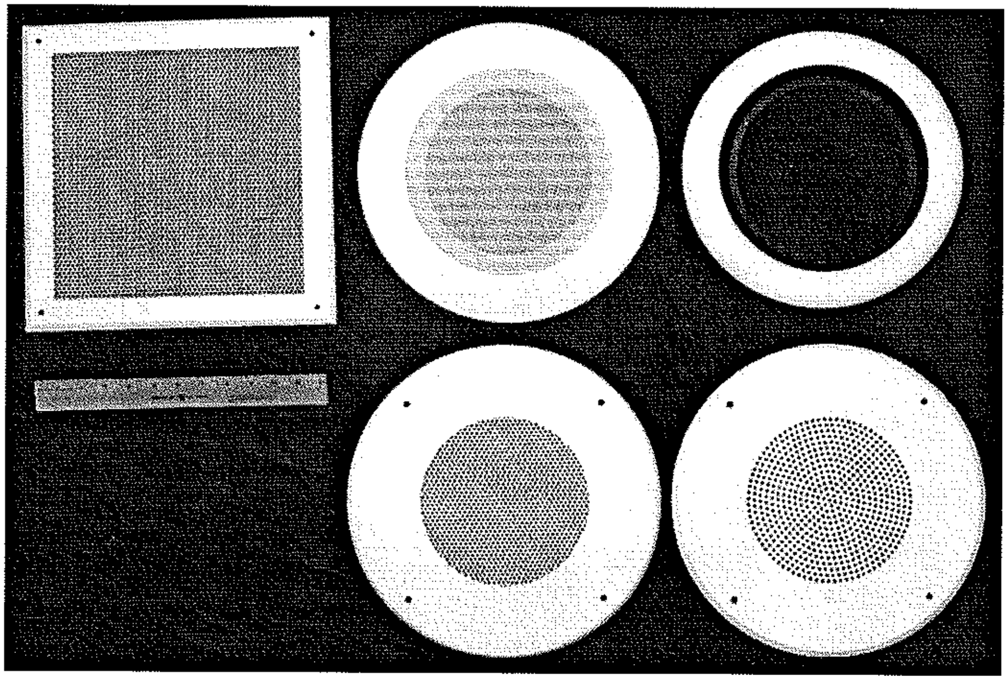


FIGURE 7-35. Square and circular loudspeaker grilles for 8 in ceiling loudspeakers. Note larger relative size of open area in square grille compared to round grilles. Products courtesy of Atlas/Soundolier, Inc.

Fire-rated ceiling systems may be required in some installations. Openings, such as ducts, lights, and loudspeakers in the ceiling that do not exceed 0.7 percent of the total ceiling area will have little affect on the ceiling fire rating performance. Larger opening areas will require modification to maintain the ceiling fire rating. One common modification is to enclose the opening with a mineral fiber or mineral wool board "tent" which will retard heat transmission from the fire. Figure 7-36 shows an installation of a tent around a ceiling loudspeaker.

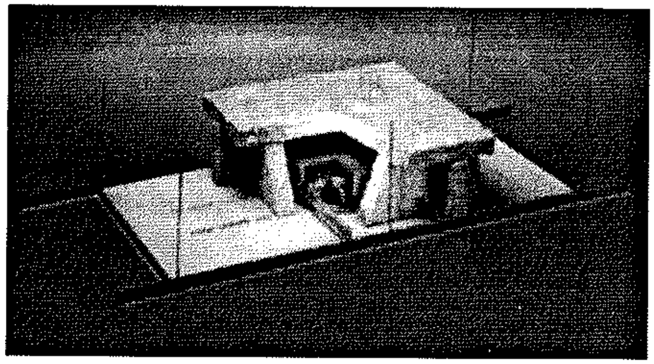


FIGURE 7-36. Cut-away view of a ceiling loudspeaker back box with 2 in thick mineral fiber tent enclosure around the four sides and the top to retain the ceiling fire rating. Photo courtesy of Atlas/Soundolier, Inc.

7.2.3 Microphone and Line Level Requirements

Microphone and line level connector installation needs to satisfy the following requirements: (1) provide convenient location and user access; (2) protect connectors from physical damage, dirt, and electrical interference; and (3) match surrounding architectural elements. Wall and floor surfaces are common locations for microphone and line level connectors. Manufacturers provide a variety of proprietary wall and floor boxes, intended for recessed or surface mounting, to satisfy the above requirements. Simpler installations can use less costly standard electrical boxes with cover plates. The sound system designer should advise the architect and electrical engineer of all locations where wall and floor boxes are to be installed.

Common locations for wall and floor boxes and typical audio functions are listed below.

1. **Stage Floor:** Microphones, line level, and monitor loudspeakers.

2. **Front Proscenium Wall:** Front-of-house loudspeakers and ALS emitter panels.

3. **Rear Proscenium Wall:** Microphones, line level, supplemental front-of-house loudspeakers, monitor loudspeakers, and production cuing.

4. **Side and Upstage Walls:** Microphones, line level, monitor loudspeakers, and production cuing.

5. **Orchestra Pit:** Microphones, line level, monitor loudspeakers, and production cuing.

6. **Rear Auditorium Wall:** Wireless microphones and ALS antennae.

7. **Ceiling:** Front-of-house loudspeakers and microphones.

8. **Audience Floor:** Microphones.

9. **Audience Mix Position:** Microphones, line level, and production cuing.

10. **Control Room:** Microphones, line level, monitor loudspeakers, and production cuing.

11. **Backstage:** Program monitoring and production cuing.

12. **Green Room and Dressing Rooms:** Program monitoring and production cuing.

13. **Technical Shops:** Program monitoring and production cuing.

14. **Ticket Booth:** Program monitoring.

15. **Public Areas and Lobby:** Program monitoring.

Different signal levels such as microphone, line level, loudspeaker, production intercom, and electrical power are best installed in separate wall or floor boxes for each signal type or in custom boxes which have separate sections for the different signal levels. Signals of the same type can use standard 1-, 2-, 3-, or 4-gang electrical back boxes depending on the number of input and output connectors. The stage and orchestra pit areas have the greatest number of wall and floor boxes as shown in Figure 7-37.

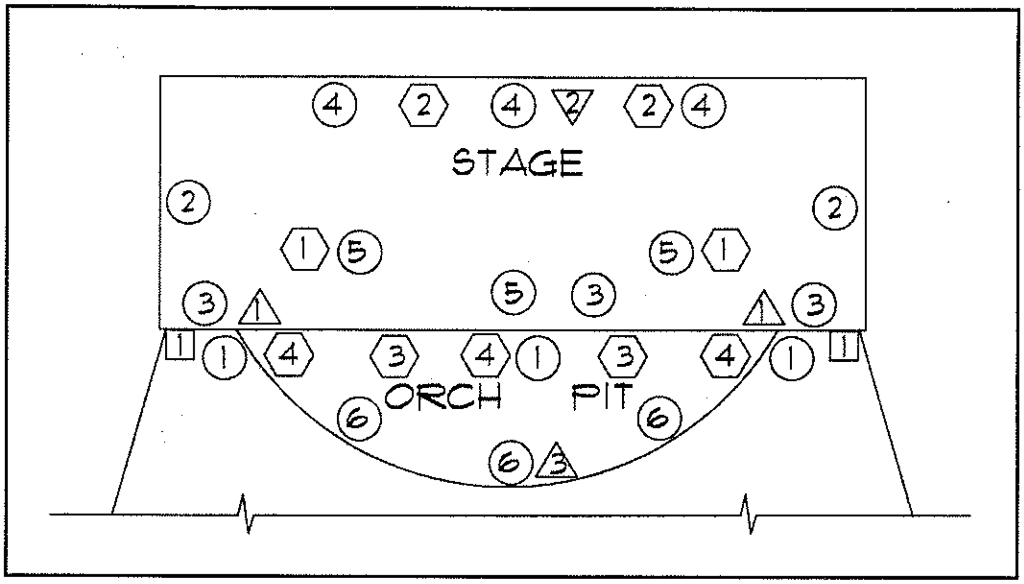


FIGURE 7-37. Recommended locations for wall and floor boxes at stage and orchestra pit areas. Circles show microphone and line level boxes: (1) left, center, and right stage lip 6 in below stage edge; (2) stage left and stage right walls 18 in AFF; (3) stage left and stage right rear proscenium wall 18 in AFF; (4) left, center, and right upstage wall 18 in AFF; (5) left, center, and right stage floor; (6) left, center, and right orchestra pit front wall 18 in AFF. Diamonds show loudspeaker boxes: (1) monitor or supplemental front-of-house loudspeakers left and right stage floor; (2) monitor loudspeakers left and right upstage wall 18 in AFF; (3) monitor loudspeakers orchestra pit left and right rear wall 18 in AFF; and (4) left, center, and right front-of-house loudspeakers at upper proscenium wall. Triangles show production intercom boxes: (1) stage left and stage right rear proscenium wall 48 in AFF; center upstage wall 48 in AFF; and (3) center orchestra pit front wall. Squares show ALS boxes: (1) upper left and right stage proscenium wall.

Proprietary wall boxes comprise a separate back box pre-punched for different conduit sizes which is recessed and anchors to the building structure. An insert-type basket slips over the back box to mount the audio connectors which terminates in a perimeter mounting ring. Basket sizes will match standard 3- and 4-gang back boxes. Some manufacturers provide an optional hinged locking door. Colors include black, white, and unfinished prime metal suitable for custom painting. Figure 7-38 shows a wall box for recessed mounting.

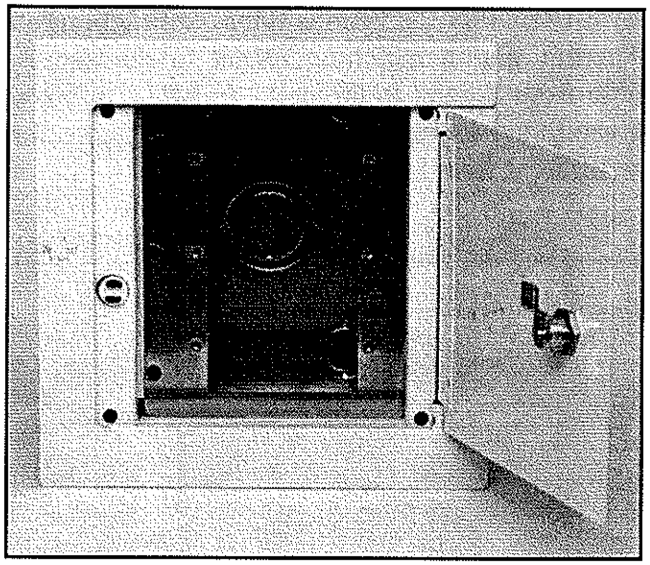


FIGURE 7-38. Three-gang audio wall box (FSR WB-3G) for recessed mounting with locking door and factory-applied white painted finish. Product courtesy of FSR, Inc.

Surface mounting wall boxes is necessary for masonry wall surfaces or where it is not possible to gain access to the wall cavity to recess the wall box. In these cases, standard 1-, 2-, 3-, and 4-gang electrical back boxes can be used. Some manufacturers provide pre-painted back boxes pre-punched for different conduit sizes as shown in Figure 7-39.

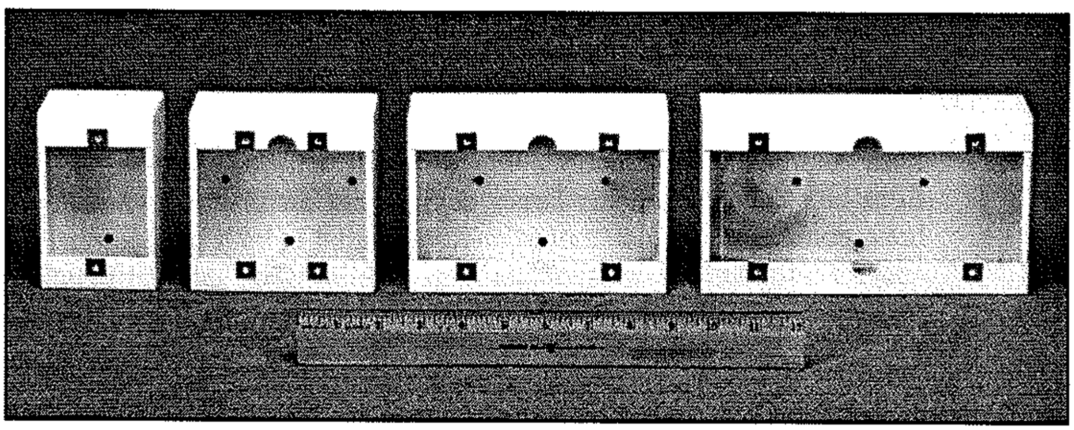


FIGURE 7-39. Wall boxes of various sizes for surface mounting: 1-gang (Lowell P1X left), 2-gang (Lowell P1X-2 left center), 3-gang (Lowell P1X-3 right center), and 4-gang (Lowell P1X-4 right) in factory-applied white painted finish. Products courtesy of Lowell Manufacturing Company.

Proprietary floor boxes comprise either a separate or integral back box pre-punched for different conduit sizes which is recessed into and rests on the floor surface. Some products have leveling feet to provide flush mounting. An integral basket installed on the back box mounts the audio connectors which terminates in a perimeter mounting ring with a hinged or removable access door. Floor boxes are available for installation in concrete slabs, wood cavity, and raised computer floors. Each box is slightly different for the various floor constructions and finishes. Optional carpet trim rings permit gluing matching carpet to the access door. A variety of sizes are available from 2-gang to larger 12 in by 12 in boxes. Colors include black, white, brass, sand, and unfinished prime metal suitable for custom painting. Figure 7-40 shows different floor boxes.

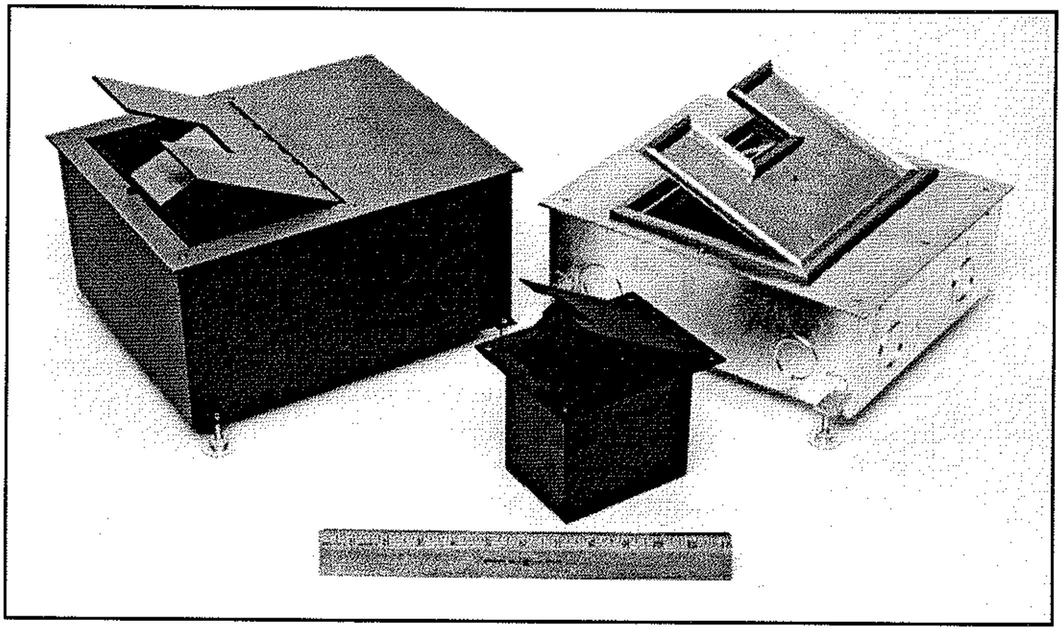


FIGURE 7-40. Floor boxes of various sizes and types for different floors: 10 in by 12 in by 6 in deep back box with brass access door (FSR FL-1000 left) for concrete slab floors with tile or thin wood finish; 4⅜ in by 3⅜ by 4½ in deep back box with black access door (FSR FL-1200 center) for wood cavity or raised computer floors; and 10 in by 12 in by 4 in deep back box with unfinished access door with carpet trim rings (FSR FL-540P right) for raised computer floors. Products courtesy of FSR, Inc.

Less complicated installations which require only one or two audio connectors per floor box do not need the larger devices discussed above. Smaller floor boxes of brass construction which mount on a standard 4 in by 4 in octagonal back box can be used. This assembly is complete with leveling feet, integral matching brass cover plate, and optional carpet trim ring. Figure 7-41 shows a small floor box.

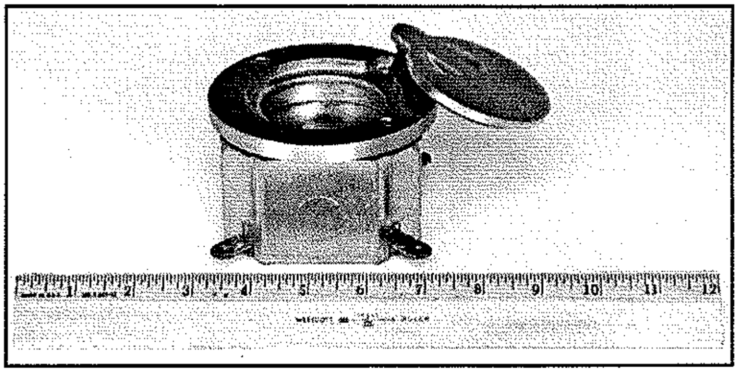


FIGURE 7-41. Small brass floor box with back box and brass cover plate (Lowell Series MO). Product courtesy of Lowell Manufacturing Company.

Microphones can be suspended from the ceiling using either proprietary microphone cable hangers or custom hanging bars. The microphone cable hanger uses the microphone cable as the primary load support with guy wires to stabilize the microphone array. The hanger rotates in the vertical plane to angle the microphone as necessary. Custom hanging bars use high-strength nylon cable or thin wire rope fitted with matching swages to support the microphone array. The weight of the microphone, cable, connectors, hanging bar, and windscreen needs to be determined prior to selecting the load capacity of the nylon cable or wire rope. The materials listed below can be used.

1. **Nylon Cable:** $^{1}/_{32}$ in monofilament; rated capacity 80 lbs and breaking strength 8 lbs.

2. **Stranded Core Wire:** $^{1}/_{32}$ in 7 by 7 stranding; rated capacity 140 lbs and breaking strength 14 lbs.

3. **Stranded Core Wire:** $^{3}/_{64}$ in 7 by 7 stranding; rated capacity 270 lbs and breaking strength 27 lbs.

Floor or tabletop stands are the most common method of installing microphones. The stands have a weighted base for stability. Floor stands are available as adjustable upright or boom styles. The latter style is useful to position the microphone close to the source. Microphone stands are available in matte black or chrome finishes. The matte black finish has the advantage of not reflecting light which can result in visual glare. Figure 7-42 shows tabletop and floor microphone stands.

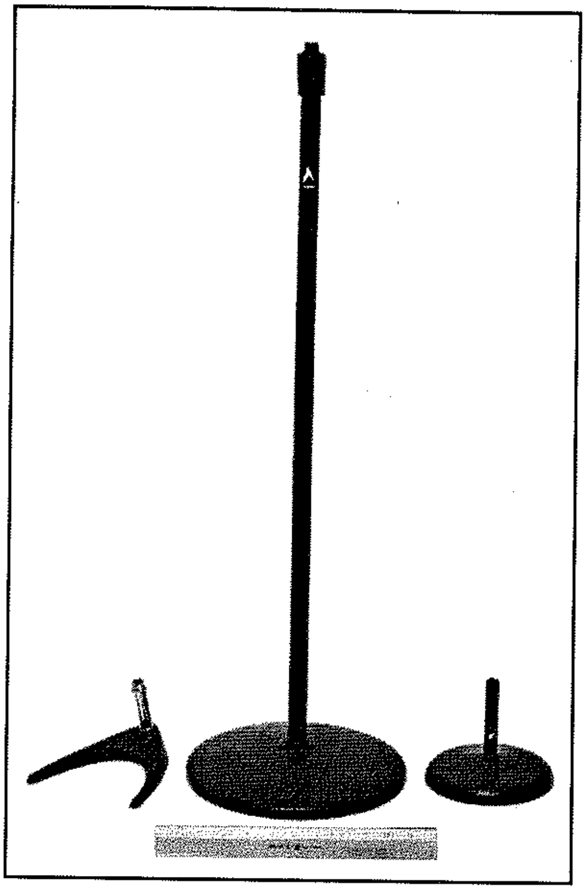


FIGURE 7-42. Tabletop microphone stands (Atlas DS-14 left and Atlas DS-5 right) and floor microphone stand (Atlas MS-12CE center). Products courtesy of Atlas/Soundolier, Inc.

7.3 Electrical System Services

With electricity, a little knowledge is sufficient to make equipment function, but a much higher knowledge level is necessary for safety. Electrical power installation is regulated by local and national Codes, most notably the National Electrical Code (NEC®). The Codes are intended to protect the public from electrocution and fire resulting from faulty electrical system installation practices. Normally, designers and contractors must answer to an authority having jurisdiction (AHD) for interpretation of Code requirements, particularly those which are applicable at the local level. Often in conflict with the Codes are common sound system installation practices intended to minimize ground loops and reduce EMI/RFI noise pick-up.

The electrical engineer, sound system designer, and sound system contractor should have a basic knowledge of electrical system services applicable to sound systems including: (1) power and distribution; (2) calculating sound system power loads; (3) power protection devices; (4) electrical interference and correction; and (5) cable protection. The important topic of grounding is covered in Section 7.4.

Electrical power requirements for permanent and temporary power are different and only permanent power installations will be covered here.

7.3.1 Electrical Power and Distribution Services

The principal design requirements for electrical power and distribution services include: (1) power requirements; (2) cable sizing; (3) *overcurrent* protection devices; (4) switching and isolation; (5) ground fault detection; and (6) working clearances.

7.3.1.1 Electrical Power

The building electrical system provides a source of power to the sound systems without which they can not operate. The reliance on computers and DSP-based signal processing equipment in sound systems increases the demands on the electrical power system. Both safety and technical performance need to be designed into electrical power systems. Standard electrical power is found in all buildings but recently the concept of balanced power distribution has been introduced.

7.3.1.1.1 Standard Electrical Power

Power in North America is nominally 120 VAC (range 115 to 120) at a 60 Hz line frequency. Power companies maintain a constant line frequency, but the voltage at a particular location can vary due to local demand conditions. Most electrical equipment will operate between 110 and 120 VAC, but a limit of 105 to 130 VAC is necessary to prevent equipment damage. Some local power lines will experience a voltage drop (power line sag) due to either power line resistance or high current demands.

Electrical power is delivered to equipment by standard three-prong outlets, with duplex (two three-prong outlets) and quadplex (four three-prong outlets) being common. The purpose of the three-prong outlet is to provide an effective ground path, via the center round prong, to eliminate the potential for electrical shock and fire. Two-prong outlets are not recommended for sound system use due to the possibility of not establishing a good ground connection. Replacing two-prong outlets with three-prong grounded outlets is recommended where this condition occurs.

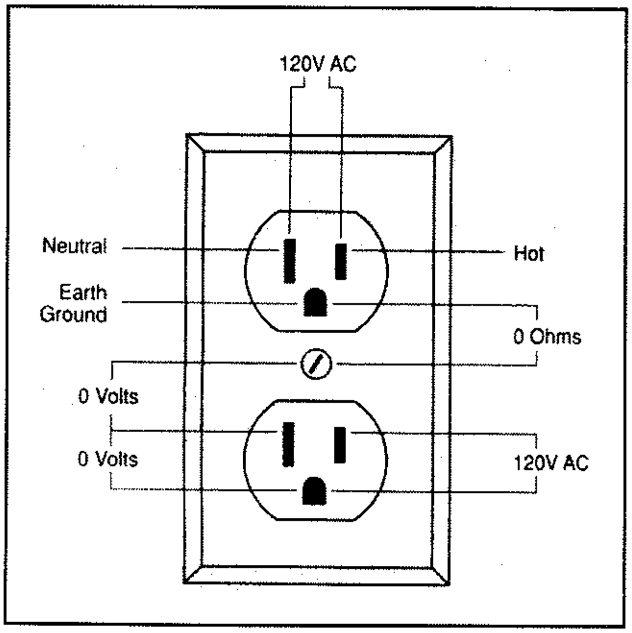


FIGURE 7-43. Duplex 120 VAC three-prong power outlet. Drawing courtesy of Furman Sound, Inc.

The standard wiring practice, when looking directly at a three-prong outlet, is as follows: (1) hot 120 VAC black wire connects to the brass or copper screw at the right (smaller slot opening); (2) neutral 0 VAC white wire connects to the silver or chrome screw at the left (larger slot opening); and (3) ground/earth green wire connects to the green screw at the side or between the hot and neutral connectors (circular opening). Figure 7-43 shows a typical duplex 120 VAC power outlet.

Sound systems should have dedicated circuit breaker power panels preferably located near the equipment racks or otherwise near the building power panels. Utilities such as lighting and convenience outlets should not be powered from the dedicated sound system power panels. Facility planning should include a sufficient number of electrical outlets for equipment in audio control rooms, amplifier rack rooms, audience mix position, stage, and orchestra pit areas. The outlets should be connected to a sufficient number of circuit breakers to avoid nuisance tripping due to circuit overload.

Emergency paging systems which operate on a battery back-up system use rechargeable batteries. The batteries can drive the equipment directly or power a converter to provide a source of AC voltage. Acid-filled batteries require proper ventilation and should not be in the same location as the sound system equipment to prevent contamination by the battery acid or fumes.

7.3.1.1.2 Balanced Power

One electrical power distribution method, developed by Martin Glasband of Equi=Tech Corporation, is called "balanced power", often referred to as symmetrical AC power. The concept is similar to balanced audio signal lines, using two separately derived 60 VAC lines with respect to ground which are out-of-polarity with each other, and summed to provide 120 VAC. The neutral conductor polarity is reversed in the summing process which reduces radiated EMI and RFI noise. This powering concept is in contrast to normal AC power which can be considered "unbalanced" since the hot conductor is 120 VAC above the neutral conductor at ground potential. EMI and RFI noise radiated from the power line can couple to the signal ground conductor because the equipment ground and neutral conductors can be connected together in systems having multiple equipment components. Balanced power can reduce noise by 5 to 15 dB in certain installations. NEC® Article 530 - "Motion Picture and Television Studios and Similar Locations" and Article 640 - "Audio Signal Processing, Amplification, and Reproduction Equipment" allow the use of balanced power. A proposed NEC® Article 647 - "Sensitive Electronic Equipment" would permit balanced power systems where power line noise is a problem. Figure 7-44 schematically depicts the balanced power concept.

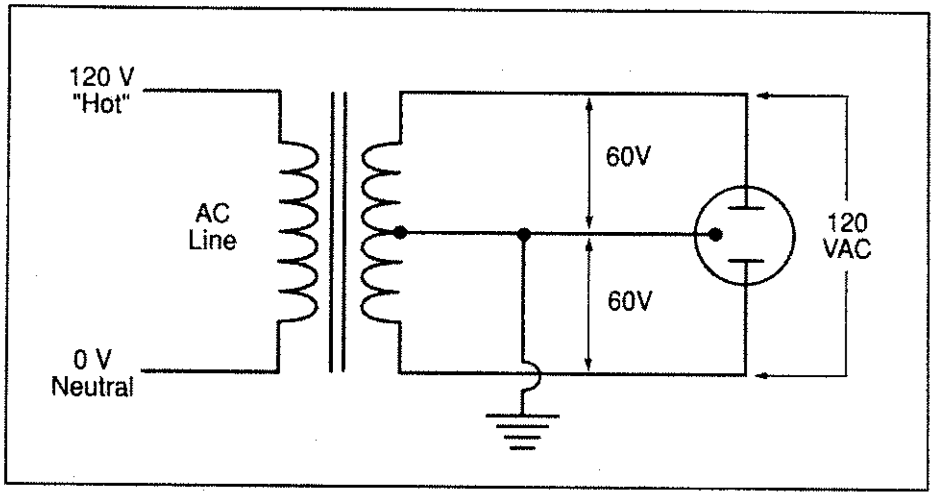


FIGURE 7-44. Schematic of balanced power. Drawing courtesy of Furman Sound, Inc.

The basic balanced power system uses a toroidal power transformer with both 60 VAC hot lines connected to a center-tapped ground. Rack-mount and wall cabinet balanced power systems are available. The rack-mount balanced power system connects to a standard 120 VAC source and delivers balanced power to chassis-mounted receptacles. The wall cabinet system provides centralized power distribution for hard wiring dedicated electrical circuit breakers and outlets in the facility. Figure 7-45 shows a rack-mounted balanced power system and Figure 7-46 shows a wall cabinet balanced power system.

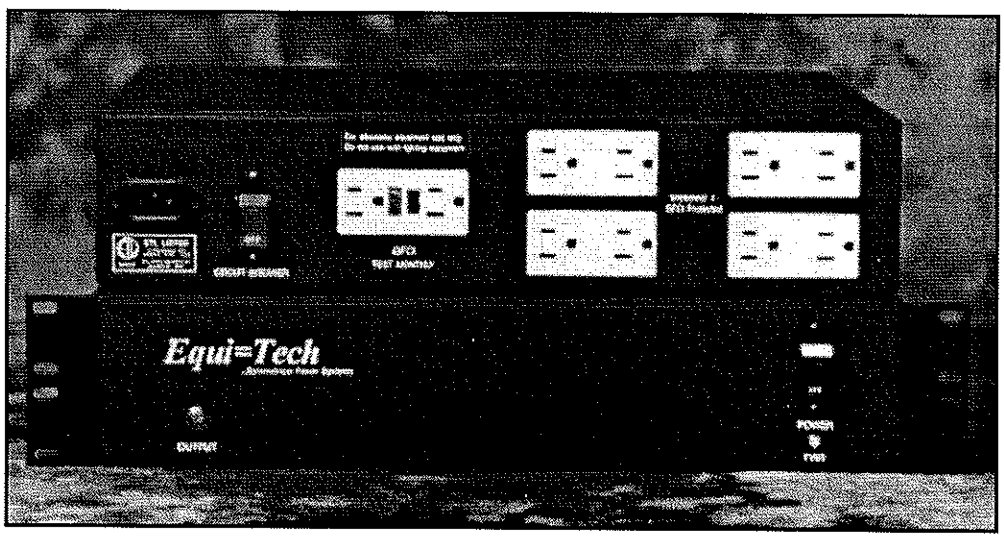


FIGURE 7-45. Rack-mounted balanced power system (Equi=Tech ETR Series) showing front (bottom) and back (top). Photo courtesy of Equi=Tech Corporation.

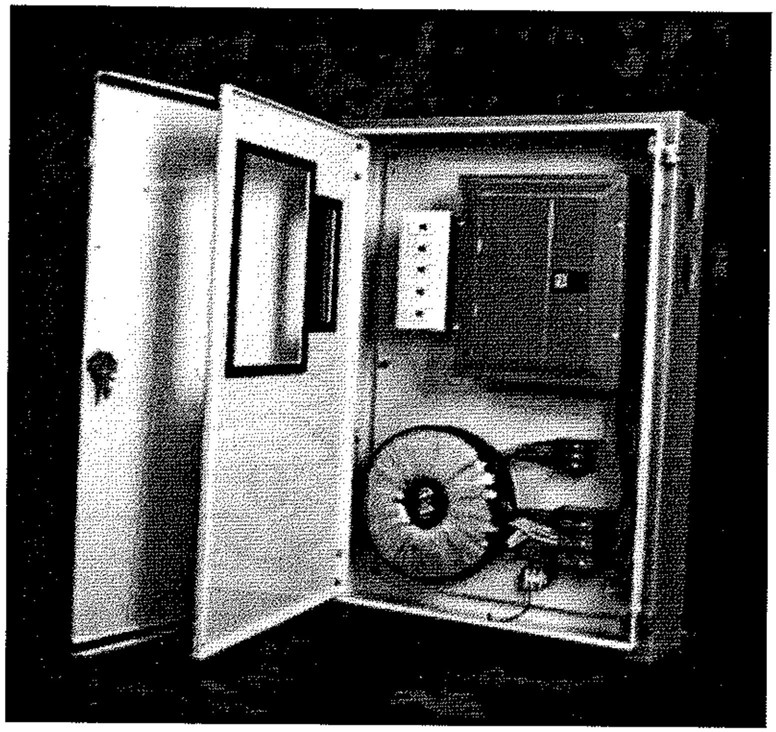


FIGURE 7-46. NEMA-style wall cabinet balanced power system (Equi=Tech ETW Series) for 208/240 VAC service at building electrical service panel showing toriodal power transformer (at bottom), circuit breaker box (at top), and access doors (at left). Photo courtesy of Equi=Tech Corporation.

Per NEC® requirements, using balanced power requires modification to overcurrent protection, ground fault current interrupt (GFCI) circuits, and power receptacles since both power conductors are hot. Overcurrent protection must use two-pole circuit breakers where both poles simultaneously open and close. GFCI protection must be used on balanced power circuits with each outlet labeled identifying it as having 60/120 V single-phase AC technical power. Regular parallel blade 15 and 20 amp power receptacles can be used, but only where accessible to authorized personnel, since there may be possible equipment damage when plugging unbalanced equipment into a balanced power receptacle.

Balanced power may not provide the most effective noise reduction in all installations, particularly when equipment comprises a mix of professional balanced audio lines and consumer unbalanced audio lines. An isolation transformer inserted between the output and input of unbalanced and balanced audio components can be helpful to separate the ground conductor from the signal conductor. One assumption with balanced power is that the interconnected equipment components have matched capacitances on each power line leg to the equipment chassis. To achieve this would require equipment manufacturers to use power transformers having the primary with capacitively balanced windings which is costly. Hence, the actual noise reduction with most equipment connected to a balanced power source can be less than this idealized condition would suggest.

7.3.1.2 Cable Sizing

The electrical power conductor cable size is based on the ampacity determined by: (1) the electrical load; (2) number of current carrying conductors; (3) environmental temperature; (4) overcurrent protection; and (5) conductor insulation. Requirements are contained in NEC® Article 310 - "Conductors for General Wiring" and NEC® Appendix B.

Another factor in sizing electrical conductors is the voltage drop across the power distribution circuit. Feeders and branch circuits are recommended to be sized to limit the voltage drop to less than three percent at the farthest electrical outlet and the maximum voltage drop across all feeders and branch circuits should not exceed five percent. This recommendation results in larger conductor sizing than based on ampacity alone. Note that an excess voltage drop can adversely affect equipment performance or damage sensitive electrical components.

7.3.1.3 Overcurrent Protection Devices

Overcurrent protection devices prevent electrical circuits and equipment from receiving excessive current by interrupting the current flow at a predetermined value. The most common overcurrent protection devices are circuit breakers and fuses. Both are sized at different amperage ratings. Modern electrical power panels use circuit breakers which can be reset when tripped. Fuses contain a sacrificial element, which breaks when exposed to overcurrent, requiring fuse replacement. Common circuit breaker values are 15, 20, 25, 30, 35, 40, 45, 50, 60, 70 A and higher. Circuit breakers for overcurrent protection are covered in NEC® Section 240-80 - "Circuit Breakers Method of Operation".

One problem with large power amplifiers or sound systems using multiple power amplifiers is the possibility of *inrush* currents when amplifiers are turned on. The inrush current can exceed the circuit breaker ampacity causing inadvertent tripping. Typical power amplifiers will draw full amperage at turn on but will operate at lower amperage. One solution is to use a power sequencer to separately turn on individual power amplifiers to limit exceeding the circuit breaker ampacity. Some power amplifier manufacturers have included a "soft start" circuit within the power supply to limit inrush current at turn on.

7.3.1.4 Switching and Isolation

Switching sound system equipment on and off can be handled by several methods depending on the system complexity and facility staffing. For simple systems, or facilities having a sound system operator, equipment components can each be activated by the chassis on/off switch. Line level sources, such as signal mixers and signal processing equipment, should be turned on first followed by the power amplifiers to avoid loud power-on transients which can damage loudspeakers. The reverse order should occur when turning off equipment. Alternately, the equipment components can be plugged into a power strip within the equipment rack which when activated simultaneously turns on all equipment. This method may trip circuit breakers if there is a large inrush current.

The best solution is to use a power sequencing system to automatically turn on and turn off the equipment at timed intervals. The power sequencing system connects between the electrical power panel and the sound system equipment racks. Remote on/off switches can be used with power sequencing systems and can be located outside the audio control room. The power sequencing system will be appreciated in facilities with limited technical staff since it reduces unintentional operator error and the chance of tripped circuit breakers due to inrush current.

Separating electrical power cables from low level audio signal and computer data cables is necessary to prevent signal contamination primarily from EMI radiation by the power cables. Table 7.2 lists minimum recommended separation distances for an electrical power source of less than 480 VAC.

TABLE 7-2. Minimum Recommended Separation Distances (in) Between Electrical Power Cables And Audio Signal And Computer Data Cables

Condition	Apparent Electrical Power		
	< 2,000 VA	2,000 to 5,000 VA	> 5,000 VA
Unshielded power lines to open or non-metal conduit	5	12	24
Unshielded power lines to grounded metal conduit	2.5	6	12
Shielded grounded conduit power lines to grounded metal conduit	0	6	12
Transformers and electric motors	40	40	40
Fluorescent lighting	12	12	12

Use of a "ground isolator" can help reduce noise induced in unbalanced audio cables by acting as a differential responding device having high common-mode rejection. Ground isolation can be achieved by using a transformer in the unshielded audio signal path. This breaks the ground connection at the cable shield between connected audio components and eliminates the noise current flow in the shield. To be effective the ground isolator transformer has to be installed between the two audio components where the noise coupling originates.

7.3.1.5 Ground Fault Detection

Ground fault detection using a GFCI protects personnel against electrocution from a source of AC power by detecting current flow in the ground conductor via a partial short circuit. A human touching the ground conductor or "live" metal equipment chassis will complete the circuit to ground resulting in shock, or worse, electrocution. The GFCI shuts off the current flow in the AC power source or receptacle when a difference of 5 mA ±1 mA between the hot and neutral conductors is detected. This value is sufficient to prevent shock where moisture may be present, but may be too low to prevent nuisance tripping. Note that some GFCI units have trip currents in excess of 30 mA which are installed to prevent accidental short circuits to ground from overheating and starting a fire. This rating is too high for personal protection from shock or electrocution. A GFCI should always be used in locations where there is the potential for moisture, as occurs in outdoor performance venues or locations where standing water is present.

GCFI protection can be within individual AC power outlet receptacles or within panelboards. Caution should be exercised in selecting a panelboard fitted with a GFCI to ensure the GFCI current rating is not less than the maximum current which could pass through the GFCI. One disadvantage with the GFCI within panelboards is the GFCI monitors several branch circuits and will disconnect the current flow to all branch circuits when only one branch circuit trips.

Ground fault protection for power distribution equipment is covered in various portions of the NEC®. Power distribution equipment Protection is covered in NEC® Section 110-9 - "Interrupting Rating". This section requires equipment intended to break current at a predetermined fault level have an interrupt rating sufficient to the current that is available at the equipment line terminals. NEC® Section 215-9 - "Ground-Fault Circuit-Interrupter Protection" for Personnel covers use of GFCI protection for electrical panelboards which feed 15 and 20 A power receptacle branch circuits. NEC® Section 640-10 - "Audio Systems Near Bodies of Water" addresses sound system equipment near a fountain, hot tub, pool, or spa. Sound system equipment should not be placed within 5 ft of water to prevent shock and must be provided with a GFCI, unless the equipment is contained within a separate room remote from the standing water. NEC® Section 680-23 -"Underwater Audio Equipment" covers requirements for underwater audio equipment including loudspeakers, wiring, and grounding.

7.3.1.6 Working Clearances

Clearances for electrical power equipment rated below 600 V are specified in NEC® Section 110-26 - "Spaces about Electrical Equipment" and Article 384 - "Switchboards and Panelboards". Equipment has front and back clearance requirements between 3 and 4 ft and minimum headroom of 6½ ft. Section 384-8 addresses dedicated space in the vicinity of electrical equipment for the installation of conduit and physical protection of equipment. The dedicated space is equal to the equipment footprint extending from the floor to the structural slab above, or 25 ft, whichever is less. Also covered are restrictions on pipes or ducts into these spaces.

7.3.2 Equipment Electrical Power Loads

The sound system electrical power load needs to be calculated to properly size circuit breakers. The electrical power drawn by equipment from the AC power receptacles is used to determine the required circuit breaker amperage. Ideally, circuit breakers will not trip, due to excess equipment current draw, nor will they be oversized which unnecessarily increases the electrical system costs. There is no one standardized calculation procedure to determine circuit breaker amperage. The method described below is conservative and accounts for anticipated current demands based on program type. One calculation is to be used for line level sources. The other more detailed calculation is to be used for power amplifiers.

Line level electrical components do not draw as much power from the AC power receptacles compared to power amplifiers. For line level sources the power drawn (in watts) is usually stamped on the equipment rear. The total power drawn by line level equipment ($\mathbf{P'}_{LL}$) can be obtained by summing the power consumed by all equipment and multiplying by 1.25 to account for a safety margin. Table 7-3 lists typical power consumption values in watts for different line level equipment components.

TABLE 7-3. Typical Line Level Sound System Equipment AC Power Consumption

Equipment Item	Power Consumption, W
Analog Signal Processing Equipment	15 to 30
DSP-Based Signal Processing Equipment	25 to 50
Computer-Controlled Multi-Function Equipment	150 to 300
CD or Cassette Reproducer	15 to 30
ALS or Wireless Microphone Transmitter	15 to 30
Automatic Microphone Mixer	20 to 40
Manual Mixer, 16 Inputs	150 to 300
Manual Mixer, 32 Inputs	300 to 600

The power consumed by a power amplifier depends on multiple factors listed below.

1. **Number of Amplifier Channels:** The power draw will be greater for multi-channel amplifiers than single channel amplifiers since the total power output is greater.

2. **Output Power:** Higher amplifier power output will draw greater power from the AC power receptacle.

3. **Amplifier Efficiency:** The efficiency rates the conversion of electrical power drawn by the amplifier from the AC power receptacle to the power output delivered to the loudspeaker by the amplifier. Amplifier efficiencies range between 40 and 65 percent for standard amplifiers and 75 to 85 percent for switch mode power amplifiers.

4. **Program Crest Factor and *Duty Cycle*:** The power amplifier will not have a continuous output due to the program crest factor and duty cycle. Duty cycles for different programs are: (1) short duration paging - 1 percent; (2) continuous speech - 10 percent; (3) background music - 20 percent; (4); rock music - 40 percent; (5) acoustic/chamber music - 60 percent.

The power drawn by the power amplifier from the AC power receptacle can be calculated using the following equation:

$$\mathbf{P'_A} = \left[\frac{\mathbf{(P)(DC)}}{\eta}\right] + \mathbf{P'_Q} \qquad \textbf{(7.6)}$$

where,

P'_A is the AC power drawn by the power amplifier, watts
P is the amplifier output power with all channels operating, watts
DC is the duty cycle, dimensionless
η is the amplifier efficiency, dimensionless
P'_Q is the amplifier quiescent power draw, watts

The *quiescent* power draw for power amplifiers range between 50 and 150 watts depending on amplifier efficiency and power rating.

The calculated power drawn by the power amplifiers (**P'_A**) is added to the power drawn by the line level sound system components (**P'_{LL}**) calculated above to arrive at a total system power requirement (**P'_T**).

The circuit breaker current rating for the line level equipment components and power amplifiers can be calculated for each using the following equation:

$$\mathbf{I} = \frac{\mathbf{P'}}{\mathbf{(V_L)(0.83)}} \qquad \textbf{(7.7)}$$

where,

I	is the current drawn by the power amplifiers or line level sound system components, amperes
P'	is the AC power drawn by the line level sound system components or power amplifiers, watts
$\mathbf{V_L}$	is the AC line voltage, normally 120 VAC
0.83	is a constant to correct for the power factor in the AC line

The power factor accounts for the phase difference between the AC voltage and current.

It is necessary to calculate multiple circuit breakers for large sound systems with multiple power amplifiers. Experience suggests that most line level equipment can be handled by one or two 20 A circuit breakers and each 100 to 400 W/channel power amplifier requires a dedicated 20 A circuit breaker.

Power amplifiers can present a significant load to the building electrical system. In large sound systems a power sequencing system is recommended to control inrush currents to reduce the potential for tripping circuit breakers.

7.3.3 Power Protection Devices

The integrity of the AC power supply is critical to sound system equipment especially DSP-based signal processors and computers. A variety of power line protection products are available, each designed to solve one or more power line problems. The major issues with AC power line integrity include: (1) transient spike and surge protection; (2) voltage regulation; (3) back-up power; (4) noise filtering; and (5) power sequencing. Some manufacturers make products which provides several of the power protection functions described below.

7.3.3.1 Spike and Surge Protection

Power line voltage spikes, up to 6,000 V and lasting up to several ms, and surges, up to 35 percent of the nominal line voltage and lasting up to several minutes, can occur in the AC power line. The cause of the spikes and surges are due to equipment, such as air conditioners, appliances, and photocopiers cycling on and off, and electrical storms. The spikes and surges can have a cumulative degradation on sensitive electrical components, result in audible noise, or irreparably damage equipment.

The most common protection method for spikes and surges is to use a metal oxide varistor (MOV) in a surge protection device to which the sound system components plug into. At normal line voltage the MOV has very high resistance. As the voltage increases and reaches the clamping voltage, the MOV activates and the resistance decreases letting the MOV absorb the momentary voltage increase. The energy handling of MOV devices is rated in joules, with devices ranging between 80 and 500 joules. Quality MOVs can last up to a minimum of 500 surges with a 6,000 V, 500 A current of 20 ms duration. Many MOV-based surge protectors route the surge voltage to the building ground conductor, which can damage unprotected equipment on the same circuit. Additionally, MOVs have a finite life duration and tend to lose effectiveness as time increases.

A different approach to surge protection is the series mode technology developed by New Frontier Electronics, Inc. This technique used an inductor in series with the hot conductor as the first element to restrict the surge current and voltage. A voltage limiter bridge circuit tracks the surge voltage and clamps at 2 V above peak power line voltage. The residual surge energy is slowly dissipated to the neutral conductor and not to the ground conductor. Advantages with surge mode technology include the surge protector not having to handle large surge currents and not contaminating the power line ground conductor with the surge current.

Both rack mounted and wall mounted surge protectors are available. Rack mounted surge protectors are designed to have equipment power cords plug directly into the unit. Wall mounted surge protectors connect between the building inlet power service panel and the local electrical panelboard. Figure 7-47 shows rack-mounted series mode surge protectors and Figure 7-48 shows a series mode surge protector in NEMA-style wall cabinet.

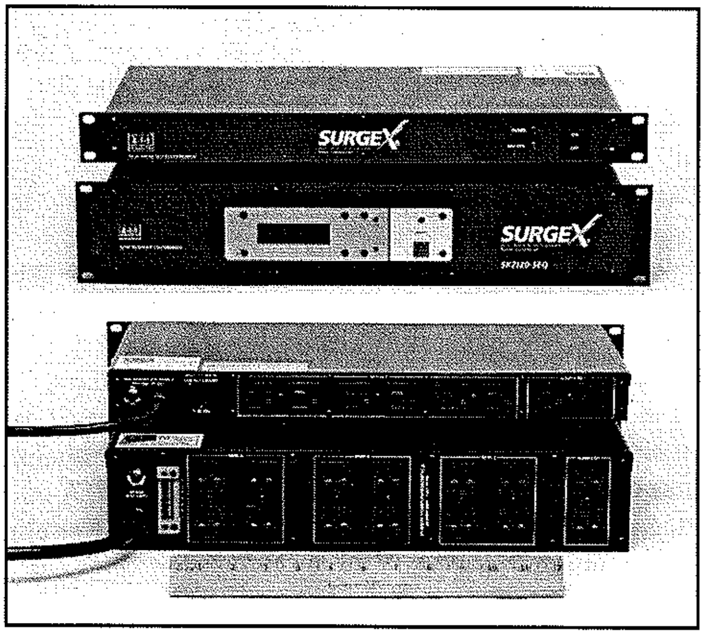


FIGURE 7-47. Rack mounted series-mode surge protector (SurgeX® SX115R, top) and combination surge protector and rack mounted power sequencer (SurgeX® SX2120-SEQ, bottom). Products courtesy of New Frontier Electronics, Inc.

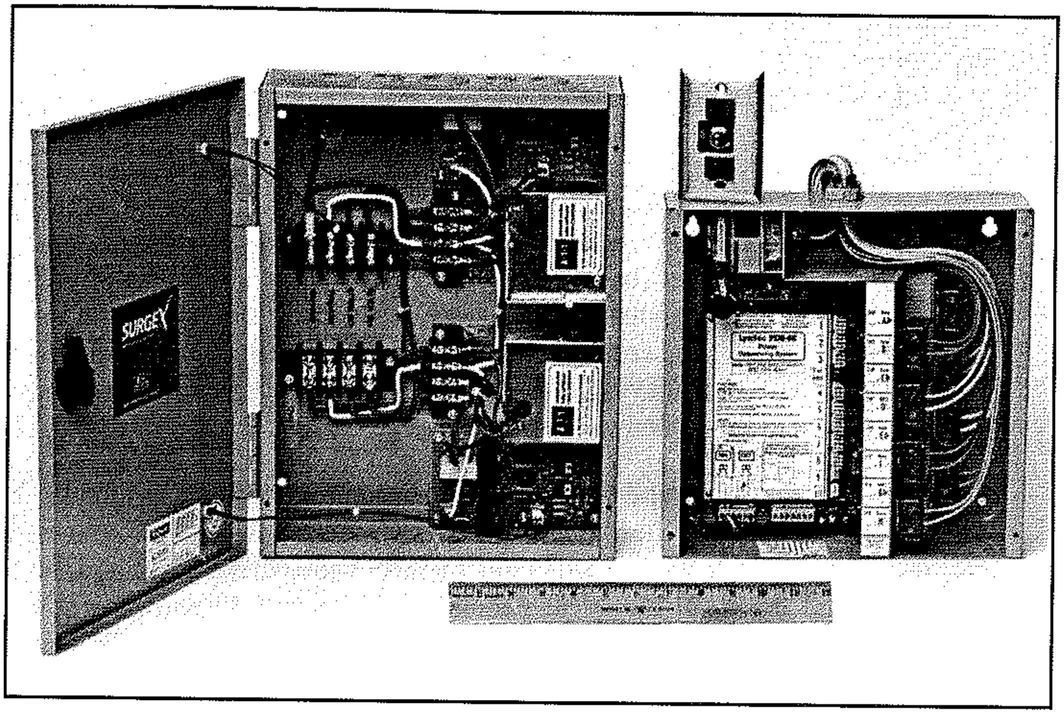


FIGURE 7-48. Series-mode surge protector (SurgeX® SXN240 left) for 220 VAC service at building electrical service panel showing surge suppression modules for each 110 VAC leg (at right) and line and load connectors (at left) along with power sequencer (LynTec PDS-8 right) showing high voltage relays (at right), low voltage timing sequencer (at left), and remote on/off switch (at top). Both products are enclosed in NEMA-style wall cabinets. Products courtesy of New Frontier Electronics, Inc. and LynTec, Inc

7.3.3.2 Voltage Regulation

Variations in the voltage supply can result in under- and over-voltage conditions at the AC power receptacle. "Brownouts" result when the local power grid is overburdened by excessive electricity demand resulting in lower voltage levels which can last for several minutes to several days. Voltage sags are shorter in duration and last for a fraction of a second to several seconds. A common cause of voltage sags is inrush current demands when equipment is turned on. Over-voltages are less common and last for up to several seconds. A common cause of over-voltage is when large loads are removed from the local power circuit.

Sound system equipment using ICs and DSP circuits are particularly sensitive to low voltage which can cause low-level power supplies that drive integrated circuits to go out of regulation. Excessive voltage can destroy equipment. The use of a voltage regulator is recommended at locations where the electrical power source voltage is unreliable.

A variety of voltage regulators are available. The most suitable type for audio equipment is the tap-switching voltage regulator which measures the line voltage by a voltage sensing circuit. If the voltage varies more than a predetermined amount the voltage sensing circuit activates a multi-tap transformer to adjust the voltage. A greater number of tap "steps" permits the voltage to be more closely regulated to the desired line voltage value. Quality voltage regulators will have toroidal auto-transformers, for less hum noise generation, and a minimum of eight tap steps.

7.3.3.3 Back-Up Power

A "blackout" is the sudden loss of electrical power. Blackouts can cause problems with sound systems, such as losing settings in micro-processor controlled equipment and computers, not to mention system inoperation. Unless a standby electrical generator is available, the latter consequence has to be accepted. Use of an uninterruptible power supply (UPS) unit can prevent micro-processor controlled equipment and computers from losing settings and provides the sound system operator time to orderly turn the equipment off.

A UPS unit is a back-up battery power source which uses an inverter circuit to convert DC to AC line current at the line voltage. The most suitable UPS unit for computer and audio components is the online type which is always on line and continuously provides current to the connected equipment. The batteries are recharged by the line current delivered by the AC power receptacle. The online UPS unit synthesizes the AC line voltage and serves also as an effective voltage regulator. When the AC power fails a tripping circuit activates the UPS unit to provide a source of voltage to the connected equipment. The run time which the UPS unit will operate is a function of the battery size and the power drawn by the connected equipment. The power drawn from the connected equipment should not exceed the UPS power rating. UPS units will commonly provide between 5 and 15 minutes of run time.

7.3.3.4 Noise Filtering

Electrical noise from EMI and RFI sources can be present in power lines. The voltage levels are considerably less than voltage spikes and surges, but are more continuous, compared to the transient spikes and surges. Little damage potential can result from EMI and RFI interference but it can result in audible noise. Common sources of EMI and RFI noise include appliances, electric motors, lighting dimmers, radio and television transmitters, and switching power supplies in computers and some power amplifiers.

EMI and RFI interference can be minimized through proper grounding techniques, using balanced audio connectors, twisting audio cables, enclosing cable in metal conduit, and in some cases, by low-pass filtering. Many surge protectors, voltage regulators, and UPS units contain low-pass filter circuits to reduce the transmission of EMI and RFI noise through the power lines. Good filters will provide up to 60 dB noise reduction between 1 to 100 MHz.

7.3.3.5 Power Sequencer

Power sequencer systems provide user convenience, prevent circuit breakers from tripping, and limit signal transients from damaging loudspeakers. Turning on and off sound system equipment is simplified with a power sequencer as only one control needs to be activated. Inrush currents are reduced as equipment is sequentially turned on which prevents circuit breakers from tripping. Low-level signal processors can result in signal transients as power is applied to the circuit. The transient can be amplified by the power amplifier resulting in damaged loudspeakers.

The power sequencer should be either rack mounted or installed in a separate NEMA-style wall cabinet and have either 15 or 20 A rating. A minimum number of three delay groups with two duplex AC power receptacles per delay group is recommended along with adjustable delay times. Some units have an unswitched AC outlet, which always has power activated, even if the unit is turned off.

Both rack mounted and wall mounted power sequencers are available. Rack mounted power sequencers are designed to have equipment power cords plug directly into the unit. Wall mounted power sequencers connect between the local electrical panelboard and the equipment rack. Figure 7-47 (above) shows a rack mounted combination surge protector/power sequencer and Figure 7-48 (above) shows a power sequencer in a NEMA-style wall cabinet.

7.3.4 Electrical Interference and Correction

Certain types of lighting ballasts and dimmers can result in audible noise and electrical interference with sound systems.

Noise from lighting fixtures manifests itself as an audible "buzz" which is the result of the lighting filament vibrating due to the dimming system. This noise is common with lighting ballasts, especially silicon controlled rectifier (SCR) types, and with dimming systems using thyristor or triac control. Solid state SCR dimmers phase controls the electrical AC sine wave which generates turn on/off transients with associated harmonic noise components. Noise can be audible in the space and the radiated EMI from the lighting ballast or dimmer can be pick-up by sound system cables which act as antennae. High-quality theater lighting dimmers have inductive filters (chokes) with rise times in the 500 to 800 μs range to suppress generated noise. Dimmer costs increase with filtering capacity and faster response times. The type of lighting filament affects the audible noise. Stiffer filaments are more resistant to vibration and have less audible noise.

Low noise and energy-efficient fluorescent lighting ballasts can radiate high levels of VHF noise and modulated IR emissions. The former can interfere with some wireless microphones and the latter can interfere with remote controllers and IR ALS operating on the 95 kHz bandwidth.

For small rooms such as conference rooms, video teleconference and meeting rooms, auto transformer wall box dimmers can be used. Sizes range between 450 and 1,800 W and are suitable for dimming incandescent lighting. One such dimmer manufacturer is Warner Electric/Superior Electric.

7.3.5 Cable Protection

Conduit is essential to protect cable from damage and provide EMI, EMC, and RFI shielding. Where possible EMT, IMT, or RSC conduit should be used. NMT conduit should only be used when the other conduit types are not permitted by Code. Many building Codes do not permit EMT or IMT conduit to be used within concrete slabs or for in-ground burial due to the potential for rusting. RSC or NMT conduit must be used for these installations. Penetrations up from the slab are made with standard EMT conduit with fittings to either RSC or NMT conduit within the slab.

The following alternates to NMT conduit can be used when EMI and RFI exists and EMT, IMT, or RSC conduit can not be used: (1) aluminum conduit; (2) smaller EMT conduit within larger NMT conduit; and (3) twisted power and audio cables within separate NMT conduit. Twisting cables will help reduce EMC but do little for the EMI and RFI pick-up, for which metal conduit shielding is necessary.

(See Technical Notes, Section 7.D, at the end of this chapter for information on software to calculate the effects of EMI.)

7.4 Grounding

Grounding electrical equipment provides safety and audio signal integrity. Safety is necessary to prevent fire, shock, or electrocution and is provided by the rounded pin on the AC power cord and receptacle. Audio signal integrity is necessary to minimize noise pick-up in low-level signal carrying cables or prevent ground loops and is provided by the shield conductor on the interconnecting audio cables. Grounding is applicable to individual electrical components and all components which comprise the sound system.

Conflicts with grounding can occur between what is required by NEC® and recommended or mythical practices for sound system installations. Additionally, some grounding problems result directly from equipment manufacturing practices where the pin 1 of balanced input and output connectors is tied directly to the internal ground within the audio circuit and not to the equipment chassis ground.

7.4.1 Equipment Component Grounding

There are three types of ground references with audio equipment components: (1) earth ground; (2) chassis ground; and (3) signal ground. Earth (safety) ground is referenced to zero potential voltage with respect to the earth and results when the rounded third prong on the electrical power receptacle or power cord makes a true earth ground connection. Chassis ground is the ground point on an electrical equipment component with the chassis ground connected to earth via the three-prong power cable. Signal ground is a common connection point in the audio circuit to which signal voltages within the circuit are referenced and makes connection to earth ground at the electrical component power supply.

7.4.2 Safety Ground

The safety ground connection is necessary to prevent an electrical device from developing a fault where the hot power line inadvertently connects to the equipment chassis. Someone touching the chassis will act as effective ground path, resulting in shock. When properly grounded, high currents from the hot power line are diverted to the safety ground, which opens the circuit breaker and removes the power from the branch circuit the faulty electrical device is connected.

A dedicated ground wire, identified by the color green, connects the metal chassis to the safety ground via the round pin at the AC power receptacle. This ground connection is routed back to the panelboard containing circuit breakers, either through a green ground wire or metallic conduit, and is tied to the neutral conductor. The green wire and conduit carry only the safety fault currents. Both grounding practices, while adequate for life safety, have multiple connection points which can result in noise pick-up in the ground conductor and be induced as noise currents in low-level audio circuits.

Equipment grounding is covered in NEC® Section 250-130 - "Equipment Grounding Conductor Connections" and ground conductors sizes are covered in NEC® Section 250-122 -"Size of Equipment Grounding Conductors" for equipment racks and wire raceways.

7.4.3 Technical Ground

Many electronic systems, including sound systems, often require an isolated ground connection to reduce noise pick-up or ground loops. This "technical system ground" is not connected to any other ground in the building and can be in conflict with strict interpretation of the NEC®. The technical system ground routes noise currents circulating in the shield conductor of balanced audio cables directly to a true earth ground to prevent noise from being induced into the signal carrying conductors. Technical system ground connections make use of the "star" interconnection concept.

The earth safety ground and technical ground should not contact each other otherwise the benefit of the dedicated technical ground can be lost. Equipment components should have separate safety and technical earth connection points. A separate ground conductor from each power receptacle, called a "home run", is routed to a single remote ground location creating a star grounding system. The technical ground makes a true earth connection using either a horizontal plate or a vertical copper grounding rod driven deep into the soil. The technical grounding rod, when physically close to the safety earth ground rod, results in minimum impedance between the technical system ground and the earth safety ground. This will help maintain a minimum voltage above ground on the technical system signal ground regardless of the ground conductor current level. NEC® Section 640-7 - "Grounding" addresses isolated ground receptacles for technical power systems. Technical power systems, covered in NEC® Section 250-146 - "Connecting Receptacle Grounding Terminal to Box" permits isolating the green ground terminal on electrical power outlets from its physical mounting connection.

7.4.4 Ground Loops

Connecting several equipment components to the safety ground conductor can cause circulating currents in the signal cable shield (ground loops) that result in an audible "buzz" or "hum" through the loudspeakers. Multiple ground connection paths between equipment components can result in duplicate ground paths which act as a loop antennae and effectively pick-up undesired EMI noise. A common ground loop occurs when the earth ground path and a shield cable signal ground path forms a loop. Figure 7-49 shows this common ground loop condition.

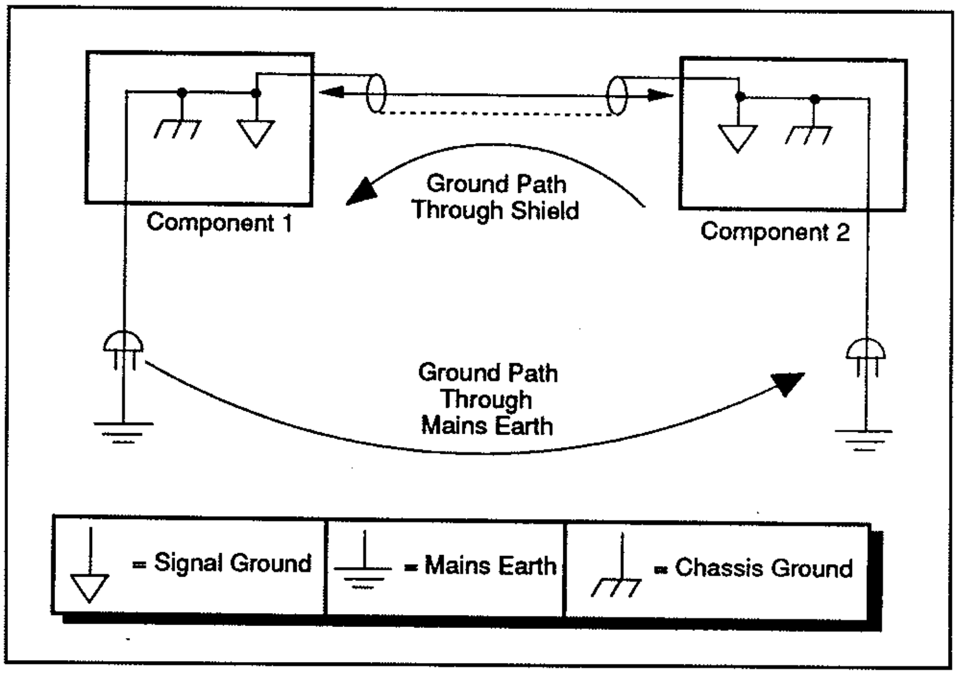


FIGURE 7-49. Typical ground loop path between two interconnected equipment components showing ground paths through the audio shield conductor and the main earth ground conductor. Drawing courtesy of Yamaha Corporation.

Ground loops can be minimized if the interconnection between components is balanced or "floating" and the signal shield is separate from the equipment chassis ground. Noise currents can circulate within the ground loop and couple to signal carrying circuits if one or both of the above conditions are not met. It is essential to maintain the safety earth ground conductor for all power cables. Ground loops can often be eliminated by disconnecting the earth ground connection at the power receptacle by using a two-to-three prong adapter plug or by cutting the round third prong on the power cord. This practice should never be done because it exposes users to risk of fire, shock, or electrocution. Additionally, this practice can invalidate equipment warranties. A better solution is to disconnect the shield connector at one end of the audio cable for the equipment causing the ground loop.

(See Technical Notes, Section 7.D, at the end of this chapter for information on software to calculate the effects of EMI.)

7.5 Cables and Connectors

Cable and connectors are essential for routing signals between the various interconnected equipment components comprising the sound system. Purpose-designed cables and connectors have been designed to fulfill a variety of functions. Audio cable and connector types include: (1) loudspeaker; (2) microphone; (3) line level; (4) video and RF; and (5) power. Cable and connectors are constructed in a variety of materials and methods to fill the need for different functions and services.

The term "wire" refers to one conductor; "cable" refers to multiple wires bundled together around a common protective jacket. A less confusing designation is "conductor" which describes the purpose of the wire or cable.

7.5.1 Cable Types

Different audio cables are necessary for system interconnection, each with unique electrical and physical characteristics. Sound systems do not need "esoteric" audiophile cables. There are significant non-linearities in sound systems that will mask the alleged claims of superior performance for these cables.

7.5.1.1 Loudspeaker Cables

Loudspeaker cables are classified as low-impedance and high-impedance based on the loudspeaker (load) impedance. The major concern in selecting a low-impedance loudspeaker cable is to limit signal losses in the cable to less than 1 dB. Heavier and thicker gauge cables will have less characteristic resistance resulting in less signal loss. Both untwisted and twisted-pair cables are available, with typical sizes between 10 and 16 AWG commonly used. Heavier gauge cable is necessary for longer routing distances. Twisted-pair cable is recommended for superior resistance to EMI, EMC, and RFI noise pick-up. The signal loss in the cable ($\mathbf{L_C}$) can be calculated using the following equation:

$$L_C = 20\log_{10}\left[Z + \frac{R_D}{Z}\right] \qquad (7.8)$$

where,

L_C is the signal loss in the cable, dB

Z is the nominal loudspeaker impedance rating, ohms

R_D is the resistance of the cable for a given length calculated from equation 4.18, ohms

An alternate to computing the signal loss is to limit the cable resistance to less than 0.1 or one-tenth the nominal loudspeaker impedance rating, whichever is less.

High-impedance loudspeaker cables are twisted-pair cable in sizes between 14 and 20 AWG. The signal loss in the cable is limited to 0.5 dB to due the added insertion loss of the line matching transformer. Maximum cable length as a function of load impedance and cable gauge to limit signal level loss to 0.5 dB is summarized in Table 4.C. Figure 7-50 shows different loudspeaker cables.

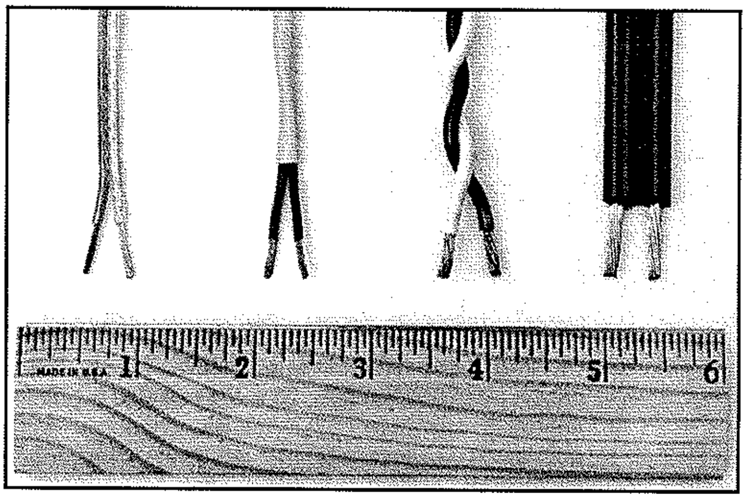


FIGURE 7-50. Different types of loudspeaker cables: 18 AWG silver and copper colored "speaker wire" (left), 16 AWG jacketed twisted-pair (left center), 14 AWG unjacketed twisted-pair (right center), and 14 AWG untwisted-pair (right). Products courtesy of West Penn Wire/CDT and Naim Audio.

7.5.1.2 Microphone Cables

Microphone cables are balanced cables comprising a twisted-pair of conductors surrounded by a braided or foil shield. The capacitance between conductors is important to limit high-frequency signal loss where cables are routed for distances greater than 250 ft. Capacitance values less than 30 picofarads (pf)/ft are recommended. Microphone cable size ranges between 20 to 24 AWG. Special "quad"-type cables comprising four conductors are available for use in higher EMI environments but have the disadvantage of higher capacitance. Their use should be limited to short cable runs or portable cables. Microphone cable intended for permanent installation normally have a foil shield; cables for portable use normally have a braided shield. Figure 7-51 shows different microphone cables.

7.5.1.3 Line Level Cables

Line level cables are similar to microphone level cables in construction but can be either balanced or unbalanced. The balanced cable configuration is preferred because longer routing distances, up to 1,000 ft, can be used without risk of picking-up undesired EMI or RFI. For longer routing distances, the characteristic impedance of the cable is important, with values between 60 and 70 being optimum. Characteristic impedance is not a concern for short cable lengths. Figure 7-51 shows different line level cables.

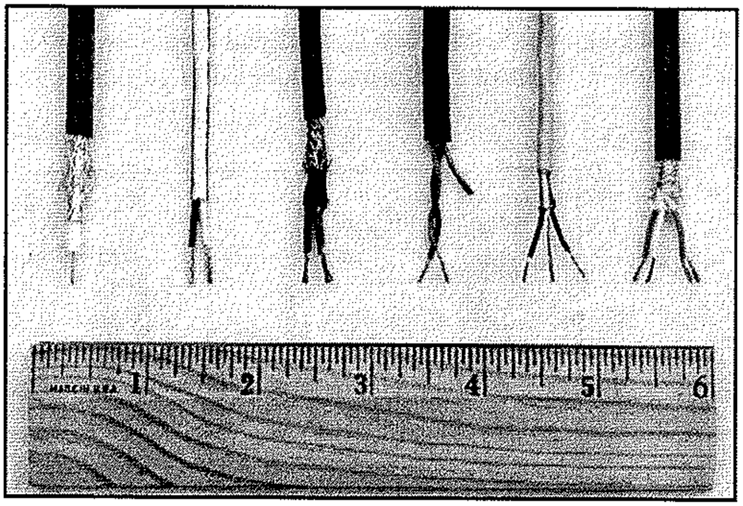


FIGURE 7-51. Different types of microphone, line level and coax cables: RG/U-59 (far left), 20 AWG jacketed two conductor line level untwisted pair (left), 22 AWG two conductor shielded line level untwisted-pair (left center), 24 AWG two conductor shielded microphone twisted-pair (right center), 24 AWG two conductor shielded (drain wire and foil shield) microphone twisted-pair (right), 22 AWG four conductor shielded microphone twisted-pair (far right). Products courtesy of West Penn Wire/CDT and Canare Cable, Inc.

7.5.1.4 Video and RF Cables

Video and RF cables use coaxial (coax) cable types having a characteristic impedance between 50 and 75. Coax cables are used to transmit high-frequency signals since the cables produce no external EMI and RFI fields and are little affected by external EMI and RFI fields. The cable has a single solid or multi-strand conductor with a shield, either foil, braid, or combination. One important factor with coax cables is to match the cable impedance with conductors having identical impedance characteristics. The most common coax cable is the Radio Guide Utility (RG/U) type. Figure 7-51 shows a coax cable.

7.5.1.5 Power Cables

Power cables are used for fixed routing between panelboards and AC power receptacles. The most common cable used for fixed installations is type THHN cable. Code requirements need to be followed based on ampacity and grounding requirements. Detachable power cables which connect to electrical components to AC power receptacles are between 20 and 16 AWG based on the current drawn. The power cable supplied with the equipment should be used since it has been appropriately sized.

7.5.2 Cable Construction

Cables are constructed from different materials to satisfy electrical and mechanical strength properties. The basic parts of a cable comprise: (1) conductors; (2) shield; (3) insulation and dielectric; and (4) protective jacket.

7.5.2.1 Conductor

The conductor is the metal portion of the cable which transmits the electrical signal. By convention the positive (+) signal, in unbalanced cables, and the positive (+) and negative (-) signals, in balanced cables are transmitted by the conductor. Conductors vary from multiple small stranded wires, for flexibility, to single monofilament type wire, for strength. Smaller and more numerous stranded wires make the conductor more flexible which is desired where the cable must be pulled in conduit or is subject to frequent handling. Monofilament conductors, while strong, can break if flexed. Copper is the most commonly used material for the conductor due to low cost and is available as bare, tinned, or combined with steel. Conductors are sized based on the AWG scale, with physically larger conductors having a smaller AWG number.

7.5.2.2 Shield

The shield prevents radiation and signal loss at high frequencies as well as reducing the pick-up of EMI/RFI interference. By convention, the shield is connected to the ground return for unbalanced and balanced cables. Three types of shield are used: (1) foil; (2) braided; and combination foil/braid. The foil shield is a thin layer of aluminum which is wrapped around the cable conductors and provides 100 percent coverage. A drain wire physically contacts the foil shield and serves as the ground connection. Foil shields provide good RFI resistance but poor EMI resistance. Braided shields comprise woven wire which covers between 95 to 98 percent of the cable conductors. In contrast to foil shields, braided shields offer good EMI resistance but poor RFI resistance. Combination foil/braided shields have an aluminum foil wrapped around the conductors with a braid over and in contact with the foil. The braid is used to make the ground termination. This shield type provides the best EMI and RFI noise rejection.

7.5.2.3 Insulation and Dielectric

Insulation and dielectric materials are plastic materials which are applied around the wire conductors in the manufacturing process to resist current flow between conductors. The insulation and dielectric materials also provide physical strength and fire resistance to protect the conductors. Different materials are used including: (1) thermoplastics; (2) thermosets; (3) fluoropolymers; and (4) elastomers.

7.5.2.4 Protective Jacket

The protective jacket is wrapped around the conductors and shield using one of the materials described in the insulation and dielectric materials above. The primary purpose of the jacket is to protect the cables from abrasion, chemicals, ozone, and other contaminants.

7.5.2.5 Electrical Characteristics

The primary electrical characteristics of cable include: (1) resistance; (2) capacitance; and (3) impedance.

The resistance is a measure of the opposition to current flow in the cable. Ideally, the resistance will be low to minimize signal loss. Cable resistance is usually tabulated in terms of ohms/1000 ft and increases as cable length increases.

The capacitance measures the storage of an electrical charge between conductors when a voltage exists across the conductors. As the cable length increases the capacitance increases and can result in high-frequency signal loss for cables greater than 1,000 ft. Ideally, the cable capacitance will be low to minimize high-frequency signal loss. Cable capacitance is usually tabulated in terms of pf/ft.

The impedance measures the combined effect of resistance and impedance which opposes the flow of alternating current in the conductors and is normally tabulated only for coax cables. Impedance becomes more critical for very high-frequency signals such as video and RF and is less critical for audio frequencies. Ideally, the cable impedance will match the input and output impedances of the connected electrical components. Cable impedance is usually tabulated in terms of ohms.

7.5.2.6 Mechanical Characteristics

The primary mechanical characteristics of cable relate to its pulling tension and breaking strength. Cable which is pulled in conduit needs to be pulled at a minimum tension, to prevent it from stretching or breaking, both of which can change the electrical characteristics. Table 7-4 lists maximum pulling tensions for cable as a function of AWG.

TABLE 7-4. Maximum Pulling Cable Pulling Tension Versus AWG

Gauge, AWG	Maximum Pulling Tension, lbs
24	4
22	7
20	12
18	19
16	30
14	48
12	77

Cables should never be jerked when installing as this will place extra stress on the cable which may exceed the maximum pulling tension.

7.5.3 Connector Types

Connectors mate the cable to the equipment component and are installed on wall, floor, or ceiling mounting plates and directly to the equipment component, usually at the rear. Special connectors have been designed for different equipment services and the types include: (1) loudspeaker; (2) microphone; (3) line level; and (4) video and RF. Connectors are usually classified by sex, with male connectors (plugs) sending signals and female connectors (jacks) receiving signals. Figure 7-52 shows different connectors installed on mounting plates.

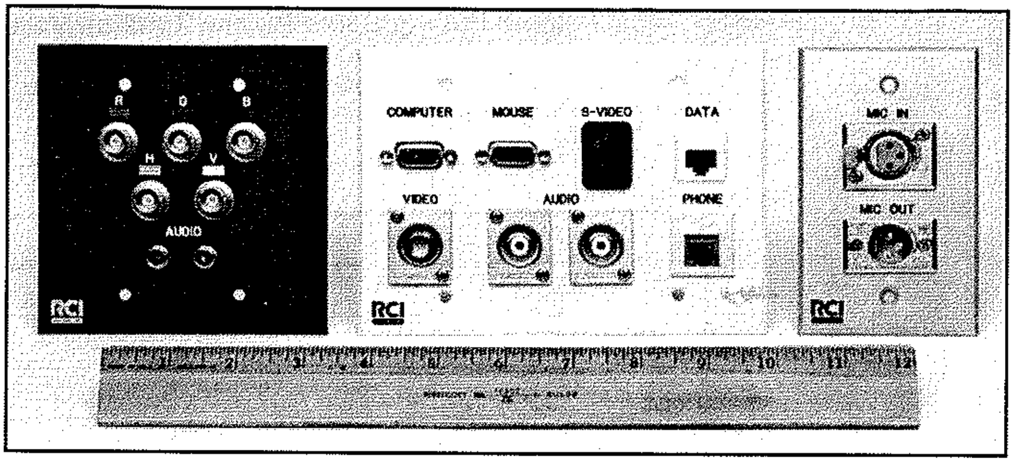


FIGURE 7-52. Different connectors installed on mounting plates: two-gang black anodized panel with five female BNC connectors for video and two female RCA-phono connectors for audio (left), three-gang painted white panel with 9-pin serial connectors for computer and mouse, RJ-11 connectors for data and telephone, female BNC connector for video, and two female RCA-phono connectors for audio (center), and single-gang brushed anodized aluminum panel with male and female XLR connectors for microphones (right). Products courtesy of RCI Systems, Inc.

7.5.3.1 Loudspeaker Connectors

Loudspeaker connectors are designed for permanent or portable application, although most loudspeakers have portable connectors installed. Permanent connectors are usually barrier strips which attached to the loudspeaker cabinet or amplifier chassis. Screw terminals anchor the loudspeaker cable to the connector. Portable connectors include the ¼ inch phone plug and "Speakon" connectors. The "Speakon" connector is preferred due to its locking collar which mates the connector to the loudspeaker or amplifier to prevent unintended disconnection. For the ¼ inch phone plug, the tip is connected to the (+) conductor and the sleeve is connected to the (-) conductor. The "Speakon" connector has two pair of (+) and (-) pins for connection. Pins and spade lugs are available in different sizes which connect directly to cables and mount onto panel mounted binding posts. Less commonly used in professional application is the GR/banana connector due to the small cable size that can be fitted. The left pin connects to the (+) conductor and the right pin (with tab) connects to the (-) conductor. Figure 7-51 shows different loudspeaker connectors.

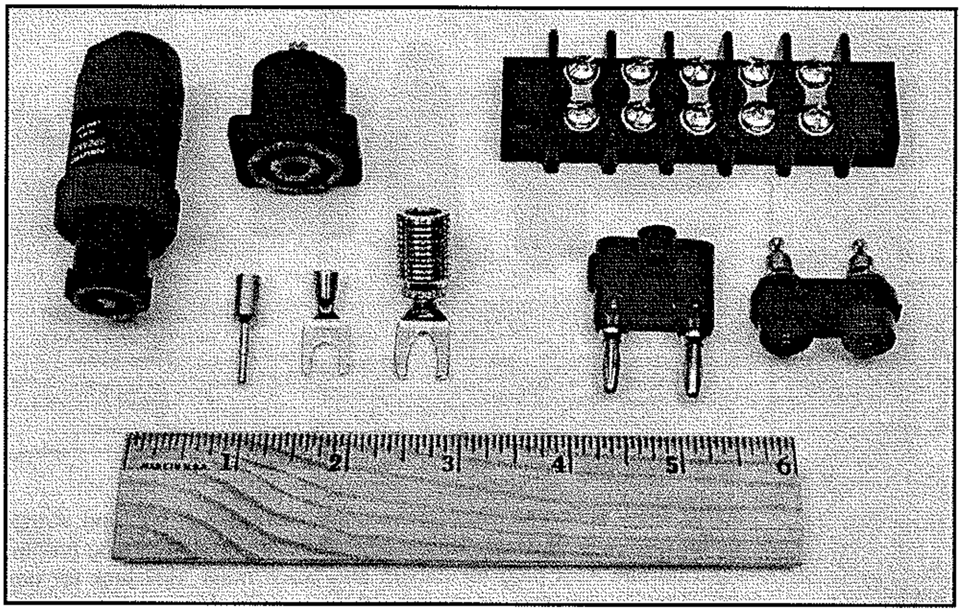


FIGURE 7-53. Different loudspeaker connectors: male "Speakon" for cables and female "Speakon" jack for panel mounting (upper left), 10 pin barrier strip for panel mounting (upper right), pin, small spade lug, and large spade lug for cables (lower left), and male GR/banana plug for cables and female GR/banana jack for panel mounting (lower right). Products courtesy of Neutrik USA, Pomona Electronics, Inc., and Esoteric Audio.

7.5.3.2 Microphone Connectors

The universal standard for microphone interconnection is the three pin XLR connector. It is available in straight and 90° configurations in black and nickel finishes. Pin 1 is connected to the shield conductor, pin 2 is connected to the (+) conductor, and pin 1 is connected to the (-) conductor. Figure 7-54 shows different XLR microphone connectors.

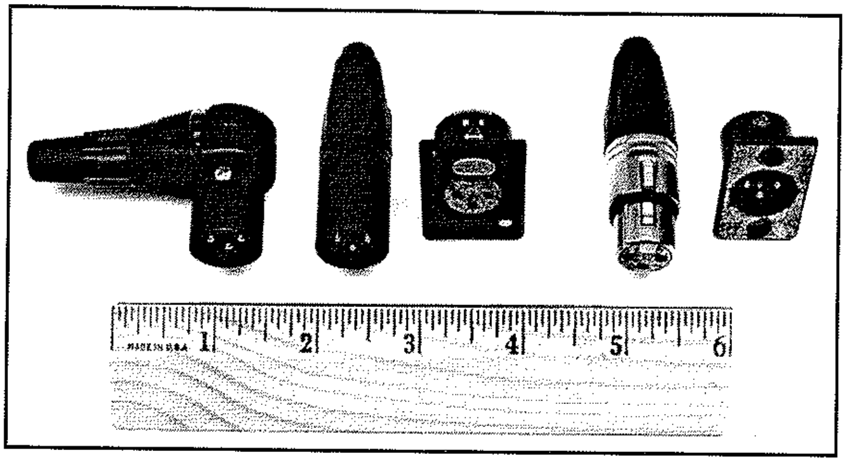


FIGURE 7-54. Different microphone connectors: black 90° and straight male XLR plugs for cables (left), black female XLR jack for panel mounting (center), and nickel straight female XLR plug for cables and black male XLR jack for panel mounting (right). Products courtesy of Neutrik USA.

7.5.3.3 Line Level Connectors

Line level connectors include balanced and unbalanced types. For balanced connections, the XLR connector is used. Unbalanced connections use ¼ inch phone plugs, RCA-phono, or DIN connectors. The XLR connector is wired as described above. The ¼ inch phone plug and RCA-phono plugs are wired identically with the tip connected to the (+) conductor and the sleeve connected to the shield conductor. The DIN connector is wired with pins 1, 3, 4, and 5 connected to the (+) conductors and pin 2 connected to the shield conductor. Figure 7-55 shows different line level connectors.

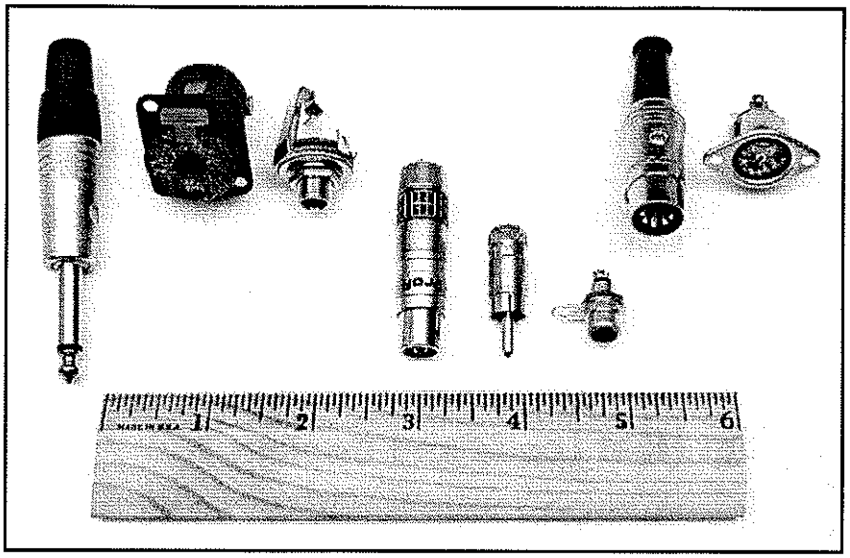


FIGURE 7-55. Different line level connectors: nickel male ¼ inch phone plug for cables, black female and nickel female phone jacks for panel mounting (left), male "quiet type" and standard RCA-phono plugs for cables and female phono jack for panel mounting (center), and five pin male DIN plug for cables and five pin female DIN plug for panel mounting (right). Products courtesy of Neutrik USA and Swichcraft.

7.5.3.4 Video and RF Connectors

Video and RF connectors use either BNC or "F" connectors. The center pin connects to the (+) conductor and the sleeve connected to the shield conductor. Most video and RF connectors use a crimping tool to make connection between the cable anc connector. Figure 7-56 shows different video and RF connectors.

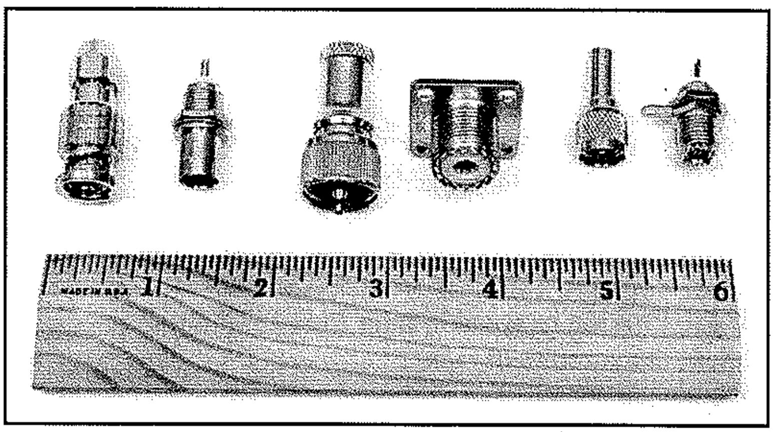


FIGURE 7-56. Different video and RF connectors: nickel male BNC plug for cables and nickel female BNC jack for panel mounting (left), large nickel male "F" plug for cables and large nickel female "F" jack for panel mounting (center), and small nickel male "F" plug for cables and small nickel female "F" jack for panel mounting (right). Products courtesy of Tandy Corporation.

7.5.4. Connector Construction

All connectors have common parts including: (1) body; (2) pins or terminals; (3) retaining chuck; and (4) strain relief. The body protects the internal pins or terminals. Plastic and metal bodies are available for most connectors. Metal bodies are preferred for strength. The pins or terminals are either screws or solder cups attached to the connector to which the cables attach. Internal to the body is a retaining chuck or clamp to hold the cable and take excess stress off the terminals. Many connectors have an external strain relief which takes excess stress off the cable as it enters the connector body.

7.6 Equipment Heat Loads and Cooling

Electronic equipment produces sensible heat, which is the byproduct of current flowing through the circuits. The heat needs to be removed from the sound system equipment to prevent malfunction and prolong equipment life. Studies have shown that each 50°F temperature reduction within the electrical circuits will double the electrical component life.

The sensible heat generated by the sound system equipment should be calculated to determine the need to provide cooling. Supplemental cooling can be provided at equipment racks or the equipment room can be cooled with conditioned air from the building HVAC system.

Equipment rack cooling can be by convection, where the hot air rises out through openings in the equipment rack, or by small pressurized fans installed within the equipment racks. Many equipment racks do not need fan cooling, provided adequate natural convection is present, which can release up to 500 W of sensible heat. Natural convection can be enhanced by ventilation panels around equipment components and in equipment rack side panels, front, and rear doors.

Larger equipment racks, or ones with power amplifiers, often require greater cooling efficiency only achieved with forced air fans. Installation guidelines for equipment rack fans are noted below.

1. Locate cooling fans at the equipment rack bottom.

2. Provide removable inlet filters at cooling fans to limit dust intake into the equipment rack.

3. Exhaust air openings should be sized to be approximately two times the cooling fan size.

4. Locate exhaust air openings above and below equipment components producing the greatest heat (power amplifiers and computer-controlled equipment) and at the equipment rack top.

5. Verify installation and clearance requirements for equipment components which have built-in cooling fans (power amplifiers and computer-controlled equipment).

6. Deeper equipment racks will have less constricted air flow.

7. Equipment rack side panels, front, and rear doors should have open ventilation louvers.

8. Periodically clean the cooling fan inlet filters to remove accumulated dust which can create a large static pressure drop decreasing the fan efficiency.

Supplemental equipment rack cooling fans can be sized based on the equipment sensible heat output and the desired cooling temperature reduction. The electrical power consumed from the AC power receptacles for the line level equipment ($\mathbf{P'_{LL}}$) as described above. Equation 7.6 is used to calculate the electrical power consumed by the power amplifiers ($\mathbf{P'_A}$). The total electrical power consumption ($\mathbf{P'_T}$) described above results in a conservative estimate to determine the dissipated sensible heat load. The required air volume for the cooling fans operating at normal room temperature and sea level conditions can be calculated using the following equation:

$$\mathbf{AV} = \frac{\left[(3160)\left(\frac{\mathbf{P'}}{1000}\right)\right]}{\Delta \mathbf{T}} \qquad \textbf{(7.9)}$$

where,

AV is the fan air volume, cfm

P' is as above

ΔT is the cooling temperature reduction, °F

3160 is a constant, dimensionless

1000 is a constant to convert W to kW, dimensionless

A cooling temperature reduction of 20°F is typical for equipment racks located in spaces having normal room temperature. Once the fan air volume is determined the designer can use catalog information to select the cooling fan. Most equipment rack manufacturers offer accessory cooling fans.

The sound system equipment sensible heat load is calculated when equipment rack cooling is not used and cooling is provided by the building HVAC system. The equipment heat load is calculated separately for the line level equipment components and the power amplifiers. The electrical power drawn from the AC power receptacles by the line level equipment ($\mathbf{P'_{LL}}$) is used to calculate the sensible heat load using the following equation:

$$\mathbf{HL} = (\mathbf{P'_{LL}})(3.415) \tag{7.10}$$

where,

HL is the sensible heat load, BTU/hr

$\mathbf{P'_{LL}}$ is as before

3.415 is a constant to convert consumed electrical power to heat load, dimensionless

Power amplifiers require more detailed calculations to determine sensible heat load to account for the efficiency and program duty cycle and can be calculated using the following equation:

$$\mathbf{HL} = \left[\frac{(\mathbf{P})(\mathbf{DC})(1-\eta)}{\eta} + \mathbf{P'_Q}\right](3.415) \tag{7.11}$$

where,

$\mathbf{HL}$, $\mathbf{P}$, $\mathbf{DC}$, and $\mathbf{P'_Q}$ are as above

The sensible heat load of the line level equipment components and power amplifiers should be summed to arrive at a total heat load which the mechanical engineer will use to determine the necessary supplemental cooling.

7.7 Codes and Life Safety Issues

Building Codes are laws which address life safety issues relating to building construction.

Most building Codes are based on "model codes" developed in cooperation between building officials, insurance companies, equipment manufacturers, and design professionals. Local jurisdictions adopt a model building code and customize it to suit local requirements.

The major building Code organizations include: (1) Building Officials and Code Administrators International (BOCAI), who publish the National Building Code (NBC); (2) Southern Building Code Congress International (SBCCI), who publish the Standard Building Code (SBC); and (3) International Council of Building Officials (ICBO), who publish the Uniform Building Code (UBC). These three Code organizations work together under the aegis of the International Code Council (ICC), who publish the International Building Code.

Other organizations whose standards are recognized by building Codes include: (1) The National Fire Protection Association (NFPA), who publish the Life Safety Code (LSC) and the National Electric Code (NEC®), also referred to as NFPA 70; (2) Underwriter's Laboratories, a testing agency; and (3) Factory Mutual, a testing agency and insurance company. The ADA is not a Code but civil rights legislation. Guidelines for barrier-free access, including ALS in places of public assembly, are developed by the Architectural Transportation Barriers Compliance Board, part of the US Department of Justice.

Of the various Codes governing building construction the NEC® is the most pertinent to sound systems design. The major articles in the NEC® applicable to sound systems are listed below.

1. Article 250 - "Grounding"
2. Article 300 - "Wiring Methods and Materials"
3. Article 310 - "Conductors for General Wiring"
4. Article 520 - "Theaters, Audience Areas of Motion Picture Theater and Television Studios and Similar Spaces"
5. Article 530 - "Motion Picture and Televison Studios and Similar Locations"
6. Article 640 - "Audio Signal Processing, Amplification, and Reproduction Equipment"
7. Article 725 - "Class 1, Class 2, and Class 3 Remote-Control, Signaling, and Power-Limited Circuits"
8. Article 760 - "Fire Alarm Systems"
9. Article 800 - "Communication Circuits"

7.8 Chapter Summary

This chapter has covered architectural, electrical, and mechanical serviceswhich are necessary for sound systems. Facilities require dedicated sound system rooms and may include audio control rooms, amplifier rooms, front-of-house, and stage mix positions. These spaces should be designed with consideration for both the sound system operators and the equipment. Sound system equipment, including ALS, loudspeakers, and microphones, are hung from the building structure. Design criteria and installation techniques have been developed to safely support this equipment. Loudspeakers are particularly critical with regard to safe rigging practices and maintaining proper orientation to achieve the desired electro-acoustical performance. Sound systems require electrical power and must be delivered safely to the equipment. Noise and interference can be picked-up by the sound system or unreliable electrical power can damage sound system equipment. Specialized equipment has been developed to reduce noise pick-up and minimize damage to the equipment from the electrical service. The most critical factor is proper grounding, necessary for life safety, signal integrity, and eliminating ground loops. Cable and connectors route the audio signal between the different sound system components. Optimizing the audio signal transfer requires good cable and connectors which are properly installed. The sound system designer needs to determine the electrical power drawn by the sound systems and the heat output and cooling necessary for the equipment. Lastly, sound systems need to be installed in accordance with applicable Codes and regulations as a matter of life safety.

7.9 Technical Notes

7.A Remote Antennae, Interference, Transmission, and Installation

Remote antennae either radiate or receive radio frequency signals. An FM ALS transmitter antenna radiates an audio signal which is picked-up by an individual wearing a receiver. A wireless microphone receiver antenna picks-up the signal radiated by the microphone transmitter.

Manufacturers have developed different antennae for receiving and transmitting radio frequency signals. These often are available as options to the standard quarter wave antenna provided with FM ALS and wireless microphones. Antenna types include: (1) quarter wave (half dipole); (2) half wave (full dipole); quarter wave with ground plane (full dipole); and (4) directional.

The quarter wave antenna is sized to be one-fourth the wavelength of the received or transmitted radio carrier frequency signal. It is the standard antenna provided with most FM ALS and wireless microphone systems. The larger half wavelength antenna will provide 3 dB greater gain resulting in better reception with a higher **S/N** ratio and up to 50 percent greater transmission distance. A quarter wave antenna should not be remotely mounted from the receiver as the antenna impedance will change resulting in degraded signal transmission or reception. When remotely mounted, a quarter wave antenna requires a ground plane. The ground plane is a metal plate which is one-quarter the wavelength of the carrier frequency. The ground plane is electrically connected to the shield of the antenna cable and provides a return path between the antenna and the receiver. The ground plane acts as an antenna "mirror" which reflects the other half of the signal received or transmitted by the quarter wave antenna. The ground plane has a hemispherical pattern of 360° by 180°. A wall-mounted ground plane antenna will have equal reception or transmission from all angles to the antenna. A quarter wave antenna with a ground plane will have performance similar to the half wave antenna.

Directional antennae can provide increased sensitivity to receiving or transmitting radio frequency signals compared to the omnidirectional pattern antennae discussed above. The most common directional antennae have a cardidoid pattern resulting in up to 10 dB additional gain in received signal strength or four times greater transmission distance when the antenna is properly aimed.

Either a half wave or a quarter wave antenna with a ground plane is the best choice for an FM ALS due to the greater coverage which minimizes audience areas with poor signal reception. Wireless microphones should use a half wave, quarter wave with ground plane, or directional antenna to maximize the signal strength received.

Remotely locating the antennae from the transmitter or receiver closer to the audience area can improve the radiated or recieved audio signal. The efficiency of a remote antenna is affected by its installation and local environment. The signal transmission distance can decrease for a radiating antenna. A receiving antenna can pick-up interference or multiple signals (multi-path distortion) resulting in weaker useable signal strength. The antenna should not be mounted near or parallel to electrically conductive objects because the antenna will capacitively couple to the conductive object and will be detuned. The amount the antenna is detuned will be a function of the receiving or radiating frequency and the distance between the antenna and the metal object. The detuning can approach 50 percent for some installations. Gypsum board wall construction using metal studs is recommended to have wood blocking installed between the metal studs to mount the antenna back box with the antenna spaced equidistant from the metal studs.

Many ALS and wireless microphone systems can provide acceptable performance in small rooms with the normally supplied quarter wave antenna. Care should be taken to ensure the antenna, if connected directly to the transmitter or receiver, does not physically contact the metal surfaces of the equipment rack for reasons described above.

7.B Strength of Rigging Hardware

The design factor (**DF**) is a theoretical reserve capacity of the supporting element and is selected to be appropriate for the particular application or as mandated by local Codes. Typical design factors can range between 3 and 10 with 5 being common for sound system installations. The rated capacity (**RC**) of a structural element when hanging or lifting is based on the nominal strength (**NS**) and the **DF**. The **RC** value is for axial pull directly along the centerline of the structural element. The **RC** can be calculated using the following equation:

$$\mathbf{RC} = \frac{\mathbf{NS}}{\mathbf{DF}} \tag{7.A}$$

where,

RC is the rated capacity of the structural element, lbs
NS is the nominal strength of the structural element, lbs
DF is the design factor, dimensionless

A fixed loudspeaker is normally subjected to static (dead) loading due to gravity. If suddenly moved, the loudspeaker can be subjected to dynamic loading resulting in an increase in the live load (sum of static and dynamic loads) imparted to the loudspeaker rigging and structural members.

Stresses and live loads should be limited to several times smaller than the loudspeaker rigging and building structure breaking strengths.

The center of gravity (**CG**) of an object is the point within the body which the weight appears to originate from and can be supported by a single force. For most regularly-shaped objects, the center of gravity is at the physical center, assuming relatively equal weight distribution. Irregularly-shaped objects have a more complicated center of gravity dependent on the weight distribution and object shape. A suspended load hangs from an attachment point through its center of gravity.

The load angle **(LA)** is the angle between the load and the supporting cable or chain sling. The tension on the supporting sling is affected by the load angle. Decreasing the load angle results in greater tension on the sling and greater forces on the mounting hardware, particularly for integral loudspeaker mounting hardware. All mounding components will experience the same tensile loading as that on each sling leg and will need to be increased in size for the non-axial loading conditions.

The load angle efficiency **(LAE)** is equal to the sin(load angle). For example, a 45° load angle will have a **LAE** of 71 percent implies the sling tension will be 1.4 times the load of the sling were the load axially applied.

7.C Rigging Hardware Corrosion

Rigging hardware is subject to corrosion which can weaken the material or make it inflexible. Different metal alloys have varying degrees of corrosion resistance. Corrosion is the oxidation of a metal which accelerates with a greater number of oxygen molecules as occurs in humid environments. Another form of oxidation occurs when dissimilar metals, having different electronic potentials, contacting each other form a galvanic cell and creates an oxide deposit. Grade 5 and grade 8 hardware are made from steel alloys which can rust. Stainless steel alloys do not rust and are appropriate for corrosive, humid, or wet environments. Three types of stainless steel are available: (1) non-magnetic; free-machining; and (3) corrosion proof.

7.D Grounding and EMI Software

Software developed in cooperation between the Georgia Institute of Technology and the Steel Tube Institute, "Grounding and Electromagnetic Interference Analysis (GEMI)", provides designers a way to calculate and size equipment grounding conductors, calculate magnetic field intensity at power circuits, and analyze new or existing wiring installations with linear and non-linear loads. The software is available from: Steel Tube Institute, 8500 Station Street, Suite 270, Mentor, OH 44060. Tel: 440.974.6990; Fax: 440.974.6994; email: sti@apk.net.

Appendix A

Electronic Sound System Identification Checklist

Objectives

Students are to visit a building such as an auditorium, lecture hall, or worship space to identify the type of sound system and its major audio components. Ask building owners which aspects of the sound system they believe to be good or bad. Use this guide to identify sound system functions and equipment. Some survey items may have multiple responses. Perform a subjective listening test of the sound system and describe any recommended improvements.

References

Prior to site visit, review material on sound systems from the following sources: 1. M. D. Egan, *Architectural Acoustics*, McGraw-Hill, 1988, pp. 356-386; 2. W. J. Cavanaugh and J. A. Wilkes (eds), *Architectural Acoustics Principles and Practice*, Wiley, 1999, pp. 187-232; and 3. M. Mehta, J. Johnson, and J. Rocafort, *Architectural Acoustics Principles and Design*, Prentice-Hall, 1999, pp. 341–362.

Room Characteristics

1. Room dimensions: _________ Length _________ Width _________ Height
2. Under balcony or transept seating? ❑ Yes ❑ No
3. Separate room for sound system controls? ❑ Yes ❑ No
4. Sound system operator in the same room as loudspeakers? ❑ Yes ❑ No
5. Reverberation time: _____ seconds (clap hands, estimate decay time by counting).

Draw small-scale floor plan and section in space below or on separate sheet. Identify presenter/performer location, audience seating, and loudspeaker locations.

Sound System Function and Type
(interview building owners or sound system contractor)

1. Function(s): ❑ Voice ❑ Music ❑ Playback/recording of audio media

   ❑ Other
   (list):____

2. Estimate cost of sound system installation: $____________________

3. Sound system requires dedicated operator? ❑ Yes ❑ No

4. Location of equipment rack (houses electronic audio equipment):

   ____

5. Equipment rack conveniently located to operate sound system? ❑ Yes ❑ No

6. Adequate clearance at equipment rack for service and ventilation? ❑ Yes ❑ No

7. Number of electrical power circuits:________ Total amperage rating:________

Equipment Components

(survey the individual equipment items which make up system)

Microphones

(convert acoustical signals into electrical audio signals)

1. Microphone type(s): ❑ Podium ❑ Handheld ❑ Lavaliere ❑ Boundary layer ❑ Wireless handheld ❑ Wireless lavaliere ❑ Ceiling hung

2. Total number of microphones:_________

3. Type(s) of connections: ❑ Wall plate ❑ Casework plate ❑ Wall box with cover ❑ Floor plate ❑ Floor box with cover

4. Connections labeled or identified? ❑ Yes ❑ No

5. Distance between microphone and closest loudspeaker (use Pythagorean theorem to get hypotenuse distance):_________ft

Assistive Listening System

(enables hearing-impaired persons to hear audio program)

1. Assistive listening system used? ❑ Yes ❑ No

2. Type of assistive listening system: ❑ Infrared ❑ FM ❑ Induction loop

3. Visible signage indicating an assistive listening system available? ❑ Yes ❑ No

4. Number of assistive listening headsets equal 4 percent of room occupancy? ❑ Yes ❑ No

5. Instructions for using equipment readily available? ❑ Yes ❑ No

6. Staff person responsible for distributing assistive listening headsets? ❑ Yes ❑ No

Special Features

(provides audio signal processing, level adjustment, and other functions)

1. Type of signal mixer used: ❑ Automatic type ❑ Manual type

2. Type(s) of electronic signal processing used: ❑ Frequency equalizer ❑ Signal delay ❑ Active crossover ❑ Compressor/limiter ❑ Multi-function device

3. Computer control as part of sound system operation? ❑ Yes ❑ No

Loudspeakers

(convert electrical audio signals into acoustical signals radiated into room)

1. Loudspeaker type(s): ❑ Single loudspeaker enclosure above stage

   ❑ Multiple horn loudspeakers above stage

   ❑ Single loudspeaker both sides of stage

   ❑ Split/distributed small cluster loudspeakers

   ❑ Multiple small ceiling loudspeakers

   ❑ Other (list):________________________

2. Total number of loudspeakers:_________

3. Loudspeaker mounting height appropriate for system type? ❑ Yes ❑ No

4. Loudspeakers visible in room? ❑ Yes ❑ No

5. Listeners have line-of-sight to loudspeakers covering their seats? ❑ Yes ❑ No

6. Obstructions in loudspeaker path? ❑ Yes ❑ No

7. Distance from loudspeaker to farthest listener (use Pythagorean theorem to get hypotenuse distance):_________ft

8. Describe support method used to mount loudspeakers:

   ____

   ____

   ____

   ____

Subjective Listening Evaluation Test

First, request permission from the building owners to perform this test. Then obtain instructions or assistance on using the sound system. Set up a microphone 1 to 2 ft away from a talker and turn on the sound system. Listeners should be positioned along room centerline, approximately one-fourth the room length away from the talker. The talker should read material not familiar to the listeners, for 1 to 2 minutes duration. After listening to the material, listeners should reposition themselves farther away and off-axis from the talker. The talker should read the same material again. Next, turn off the sound system or disconnect the microphone, and read the same material without sound amplification. The listeners should be at their original locations. Finally, repeat for the second listener locations. Compare listening experiences at different locations with and without sound amplification. Record the following observations:

1. Uniform sound level. (Sound level should not be noticeably lower as you move away from loudspeakers to perimeter or to under balcony locations).
2. Clarity of individual consonant sounds and words.
3. Frequency balance between bass and treble tones.
4. Any audible feedback (such as ringing tones or hollow-sounding voices).
5. Natural quality of vocal reproduction.
6. Amplified sound should be sufficiently loud with microphones 1 to 2 ft from talker.

Recommendations

List below modifications to the sound system you feel would improve its electro-acoustical performance or operational convenience for the users.

Appendix B

References and Sources of Additional Technical Information

Numerous sources are available which provide more detailed information on specific aspects of sound system design than is possible to include in a book of this scope. Summarized below in different categories are publications the reader can consult for additional information. Many publications are out-of-print, but may be available in university libraries or antiquarian bookstores. The author's candid comments on the utility of the information contained in these sources follows each entry.

Architectural Design Data

The books noted below provide architectural design data on building systems and different building types which can be helpful in planning sound systems.

DeChiara, J. and Callender, J. (Editors). 1990. *Time Saver Standards for Building Types - 3rd Edition*. New York, NY: McGraw-Hill Publishing Company.

General planning information on most building types is provided which is useful in helping to outline acoustical and sound system requirements.

Hoke, J. (Editor). 1998. *Architectural Graphic Standards - 9th Edition*. New York, NY: John Wiley & Sons.

General graphical and tabular data on all building system elements with brief sections on architectural acoustics, noise control, and sound systems.

Kinzey, B. and Sharp, H. 1963. *Environmental Technologies in Architecture*. Englewood Cliffs, NJ: Prentice-Hall, Inc.

A general architectural text covering all aspects of building environmental systems with three chapters on acoustics. One of the few architectural sources outlining acoustical requirements for pipe organs. (Out-of-print).

Stein, B., Reynolds, J., and McGuinness, W. 1986. *Mechanical and Electrical Equipment for Buildings*. New York, NY: John Wiley & Sons.

The standard issue textbook for architecture schools covering building environmental systems. The book has an excellent section on architectural acoustics, noise control, and sound systems.

Watson, D., Crosbie, M., Callender, J. (Editors). 1997. *Time Saver Standards for Architectural Design Data - 7th Edition*. New York, NY: McGraw-Hill, Inc.

General planning and design criteria for different building elements and systems with brief sections on acoustics and audio/visual systems.

Architectural Acoustics Design

The books noted below cover topics related to sound systems design including general acoustics, room acoustics, sound isolation, equipment noise control, and vibration isolation.

Apfel, R.E. 1998. *Deaf Architects & Blind Acousticians?*. New Haven, CT: Apple Enterprises Press.

Don't let the humorous title fool you. This is a brief non-technical overview of room acoustics, sound isolation, and noise control with several case studies.

Egan, M. D. 2000. *Instructor's Architectural Acoustics Workbook*. Lincoln Center, MA: The Robert Bradford Newman Student Award Fund.

A compilation of various nomograms, design charts, case studies, and field exercises to enable the student to obtain an overview of room acoustics, sound isolation, and environmental acoustics.

Hoover, A. 1993. *An Appreciation of Acoustics*. Lincoln Center, MA: The Robert Bradford Newman Student Award Fund.

Basic physical and architectural acoustics information applicable to sound systems design is contained in this concise volume.

Architectural Acoustics Books with Chapters on Sound System Design

The books noted below primarily cover architectural acoustics design but have non-technical chapters on sound systems.

Cavanaugh, W. and Wilkes, J. (Editors). 1999. *Architectural Acoustics Principles and Practice*. New York, NY: John Wiley & Sons, Inc.

A nicely balanced book presenting all aspects of architectural acoustics with a chapter on sound systems design by J. Jacek Figwer. Lots of good illustrations and case studies.

Cremer, L, Müller, H, and Schultz, T. 1982. *Principles and Applications of Room Acoustics, Volume 1 and Volume 2.* Barking, Essex, England: Elsevier Science Publishers, Ltd.

Probably one of the most important and thorough books on room acoustics summarizing acoustical research and design aspects through the early 1980s. Brief chapters on sound reinforcement and electronic architecture systems are included. Highly recommended. (Out-of-print).

Doelle, L. 1972. *Environmental Acoustics.* New York, NY: McGraw-Hill Book Company.

The title is a little deceiving as the book focus is on room acoustics. One brief chapter on sound system design is included. (Out-of-print).

Egan, M. D. 1988. *Architectural Acoustics.* New York, NY: McGraw-Hill, Inc.

Great book, great graphics, no math, with a short chapter on sound systems, covering all basic acoustical concepts which architects need to know. Sound system chapter was written by David Kaye.

Irvine, L. and Richards, R. 1998. *Acoustics and Noise Control Handbook for Architects and Builders.* Malabar, FL: Krieger Publishing Company.

A nice book covering all aspects of architectural acoustics and noise control including a short chapter on sound systems design. The book was written based on the author's lectures presented to architectural students. The sound system chapter covers speech and music systems.

Kuttruff, H. 1991. *Room Acoustics - 3rd Edition.* Barking, Essex, England: Elsevier Science Publishers, Ltd.

Room acoustics presented in a theoretical context with a moderate degree of mathematics. The chapter on sound systems provides one of the few theoretical descriptions of acoustical feedback.

Maekawa, Z. and Lord, P. 1994. *Environmental and Architectural Acoustics.* London, England: E. & F.N Spon.

A book with a moderate mathematical treatment of room acoustics, sound isolation, HVAC noise control, and sound systems design. One of the better books that combines acoustical theory and applications without going overboard on the mathematics. Sound systems for auditoria are covered. (Out-of-print).

Mehta, M., Johnson, J., and Rocafort, J. 1999. *Architectural Acoustics Principles and Design*. Upper Saddle River, NJ: Prentice-Hall, Inc.

An excellent book covering all aspects of architectural acoustics with a chapter on sound systems design. The material is presented in a concise narrative supported by elementary mathematics, good illustrations, case studies, and design rules-of-thumb. Highly recommended.

Parkin, P., Humphreys, H, and Cowell, J. 1979. *Acoustics, Noise and Buildings - 4th Edition*. London, England: Faber & Faber.

A classic text on architectural acoustics written by one of Britain's pre-eminent acousticians, Peter Parkin. The book has an excellent chapter on theory and application of line source array loudspeakers. (Out-of-print).

Salter, C and Associates. 1998. *Acoustics Architecture Engineering The Environment*. San Francisco, CA: William Stout Publishers.

A beautifully prepared and bound "coffee-table" book describing aspects of acoustics and audio systems in moderate detail with many case studies by one of San Francisco's larger acoustical consulting firms. Brief chapters on sound systems and audio/video systems are included.

Templeton, D. (Editor). 1993. *Acoustics in the Built Environment - Advice for the Design Team*. Jordan Hill, Oxford, England: Butterworth Architecture.

An expanded edition of an earlier book by Templeton with a good chapter on sound systems design by Peter Mapp, one of England's leading sound system designers.

General Acoustics

The books noted below primarily cover theoretical aspects of acoustics with some material on transducers.

Blackstock, D. 2000. *Fundamentals of Physical Acoustics*. New York, NY: John Wiley & Sons, Inc.

An excellent book on physical acoustics with chapters on horn theory, waveguides, radiation from baffled pistons, and arrays. Very technical.

Fahy, F. 2001. *Foundations of Engineering Acoustics*. San Diego, CA: Academic Press.

Great overview of physical acoustics and simple horn and waveguide theory, presented with a moderate degree of mathematics. Highly recommended.

Hall, D. 1987. *Basic Acoustics*. New York, NY: John Wiley & Sons.

An abbreviated approach to physical acoustics and its applications with good chapters on microphones, dynamic cone drivers, and loudspeaker enclosures. The use of numerous variables which are not fully noted make the book a difficult read. (Out-of-print).

Raichel, D. 2000. *The Science and Applications of Acoustics*. New York, NY: Springer-Verlag New York.

Good overview of physical acoustics and its numerous applications, but has little information on electro-acoustics and sound systems. Very technical.

General Sound System Design

The books noted below cover general design for a variety of sound system types including equipment components and application.

Ahnert, W. and Steffen, F. 1999. *Sound Reinforcement Engineering*. London, England: E&FN Spon.

An excellent technical resource describing equipment and sound systems design with a particular emphasis on European design philosophies with a moderate degree of mathematics. Highly recommended.

Burris-Meyer, H., Mallory, V., and Goodfriend, L. 1979. *Sound in the Theatre*. New York, NY: Theatre Arts Books.

This book mostly covers special effects generation for theatrical performances, but does have information on theater acoustics and sound reinforcement systems. The book is very dated. (Out-of-print).

Capel, V. 1992. *Public Address Systems*. Oxford, England: Focal Press.

A very poorly written text on sound systems which fails to cover many basic concepts, but does have useful information on line source array loudspeakers. (Out-of-print).

Collison, D. 1982. *Stage Sound*. London, England: Cassell.

Nice book covering equipment functional features and operation of sound system equipment with an emphasis on theatrical and special effects written by one of the founding members of Theatre Projects. (Out-of-print).

Davis, D and D. 1975. *Sound System Engineering*. Indianapolis, IN: Howard W. Sams Publishing Company.

The first book to fully describe the application of sound system components and the design of sound systems using a systems engineering approach. Highly recommended. (Out-of-print).

Davis, D and D. 1986. *Sound System Engineering - 2nd Edition.* Indianapolis, IN: Howard W. Sams Publishing Company.

A disappointing follow-up to the original first edition due to numerous typographical errors, poorly written text, and equations and variables appearing with little explanation or references to previously presented material. The book does provide useful information on applications of TDS acoustical measurements. The newer Focal Press edition reportedly has corrected many typographical errors but the overall poor presentation of the material remains.

Eargle, J. 1990. *Music Sound and Technology.* New York, NY: Van Nostrand Reinhold.

An emphasis on basic acoustics of musical instruments is featured with several brief but concise chapters on audio equipment, sound recording, and sound systems. A very good non-mathematical presentation. (Out-of-print).

Eiche, J. 1990. *Guide to Sound Systems for Worship.* Milwaukee, WI: Hal Leonard Publishing Corporation.

A non-technical overview of basic theory, audio equipment, sound systems, and installation techniques intended for houses of worship, but applicable to any assembly space. The majority of the material is taken from the Davis and Jones book, *Sound Reinforcement Handbook.*

Harley, R. 1998. *The Complete Guide to High-End Audio - 2nd Edition.* Albuquerque, NM: Acapella Publishing.

A nice narrative description of audio equipment albeit intended more for home high-fidelity systems, but providing good fundamentals applicable to any type of sound system.

Herrick, C. 1974. *Audio Systems.* Reston, VA: Reston Publishing Company, Inc.

Simple descriptions of different audio systems in use: stereo, telephone, broadcasting, public address, theater sound, and electronic music systems. (Out-of-print).

Johnson, K., Walker, W., and Cutnell, J. T*he Science of High-Fidelity - 2nd Edition.* Dubuque, IA: Kendall/Hunt Publishing Company.

Different high-fidelity components are used as examples to explain elementary concepts of physics and the underlaying physical principles of how audio equipment works. (Out-of-print).

Moir, J.1961. *High Quality Sound Reproduction - 2nd Edition.* London, England: Chapman & Hall, Ltd.

An excellent, if somewhat dated, book covering all the essentials of audio system design and application written by one of England's leading acousticians. (Out-of-print).

Olson, H. 1966. *Music, Physics, and Engineering - 2nd Edition.* New York, NY: Dover Publications, Inc.

Another magnum opus from Olson covering all aspects of basic acoustics, musical instruments, audio equipment, and sound systems. Much material is essentially repeated from Olson's earlier books, but with a particular emphasis on musical instruments.

Olson, H. 1972. *Modern Sound Reproduction.* New York, NY: D. Van Nostrand Company, Inc.

Probably the most disappointing of Olson's books, featuring material covered in his earlier books. (Out-of-print).

Rossing, T. 1990. *The Science of Sound - 2nd Edition.* Reading, MA: Addison-Wesley Publishing Company.

This is a physics textbook written by one of the preeminent acoustics educators describing most aspects of sound production, musical acoustics, room acoustics, noise control, and sound systems in an easy to understand manner with a minimum of mathematics. Highly recommended.

Wadsworth, H. 1983. *Basics of Audio and Visual System Design.* Indianapolis, IN: Howard W. Sams Publishing Company.

The title says it all, although there is more emphasis on video systems than audio systems design. The video system chapters are somewhat dated given current technology. (Out-of-print).

Engineering Aspects of Sound System Design

The books noted below are more mathematically based and cover transducers and sound systems design at the engineering level.

Beranek, L. 1954. *Acoustics.* New York, NY: McGraw Hill Book Company, Inc.

One of the classic textbooks covering most aspects of acoustics, with excellent chapters on transducers, written by one of the foremost acousticians of this century. Highly recommended. The original edition is out-of-print, but a reprint edition is available through the Acoustical Society of America.

Briggs, G. 1955. *Loudspeakers The How and Why of Good Reproduction.* Idle, Bradford, Yorkshire, England: Wharfedale Wireless Works, Ltd.

A cute and quirky book describing aspects of loudspeaker design with an emphasis on cone drivers, written by one of the early pioneers of British loudspeaker design. A reprint edition is available through Audio Amateur Press in Peterborough, NH.

Cohen, A. 1989. *Audio Technology Fundamentals*. Indianapolis, IN: Howard W. Sams Publishing Company.

Good book on basics of audio system components, gain calculations, operational amplifiers, filters, transformers, and basic audio circuits. (Out-of-print).

Duncan, B. 1996. *High Performance Audio Power Amplifiers*. Oxford, England: Newnes.

The only book you will ever need covering all aspects of power amplifier design written in a concise, witty English kind of way.

Giddings, P. 1990. *Audio System Design and Installation*. Indianapolis, IN: Howard W. Sams Publishing Company.

An excellent nuts and bolts book on the minutia of sound system installation with an emphasis on proper electrical grounding. This book is one of the few sources of material available on sound system installation at the wire and connector level. Highly recommended.

Lampen, S. 1998. *Wire, Cable, and Fiber Optics for Video and Audio Engineers*. New York, NY: McGraw Hill, Inc.

Excellent summary on interconnection materials and methods for audio and video equipment.

Leach, W. 1999. *Introduction to Electroacoustics and Audio Amplifier Design*. Dubuque, IA: Kendall/Hunt Publishing Company.

A thorough coverage of basic acoustics, electro-acoustical analysis, and transducer design is presented with complete mathematical derivations. Very technical. Could become a classic.

Merhaut, J. 1981. *Theory of Electroacoustics*. New York, NY: McGraw Hill, Inc.

A concise mathematical treatise on electro-acoustical systems with emphasis on lumped and distributed system analysis techniques. Very technical. (Out-of-print).

Olson, H. 1947. *Elements of Acoustical Engineering*. New York, NY: D. Van Nostrand Company, Inc.

The Holy Grail of electro-acoustics. More sound system designers and manufacturers have "stolen" ideas from this book than probably any other one source. As they say, "the fundamentals never change" and "the ancients are stealing our ideas." A must have for anyone in sound system design. (Out-of-print).

Olson, H. 1957. *Acoustical Engineering*. New York, NY: D. Van Nostrand Company, Inc.

An updated and expanded version of Olson's earlier book. Highly recommended. A reprint edition is available from Professional Audio Journals, Inc.

Olson, H. 1958. *Dynamical Analogies*. New York, NY: D. Van Nostrand Company, Inc.

A unique book which uses analogies to describe acoustical, mechanical, electrical, and magnetic principles. These analogies are used throughout all of Olson's books, but are collected here in one edition. (Out-of-print).

Ott, H. 1988. *Noise Reduction Techniques in Electronic Systems - 2nd Edition*. New York, NY: John Wiley & Sons, Inc.

Everything you could want to know about noise reduction of electronic equipment down to the component level.

Pohlman, K. 1985. *Principles of Digital Audio*. Indianapolis, IN: Howard W. Sams Publishing Company.

Basics of digital audio theory and applications presented in a non-mathematical fashion with emphasis on recording technologies. Newer book edition covers more current technology.

Morrison, W and Lewis, W. 1990. *Grounding and Shielding in Facilities*. New York, NY: John Wiley & Sons, Inc.

Everything you could possibly want to know on the subject of grounding and shielding in buildings.

Rossi, M. 1988. *Acoustics and ElectroAcoustics*. Norwood, MA: Artech House, Inc.

A very thorough mathematical treatment of acoustics and electro-acoustic devices, but break out the advanced calculus and differential equations books! (Out-of-print).

Watkinson, J. 1988. *The Art of Digital Audio*. Borough Green, Sevenoaks, Kent, England: Focal Press.

Similar to the Pohlman book but covers more applications. Newer book edition describes more current technology.

Psychoacoustics and Hearing

The books noted below cover aspects of psychoacoustics and hearing which are important for sound systems design.

Blauert, J. 1993. *Spatial Hearing The Pyschophysics of Human Sound Localization - Revised Edition.* Cambridge, MA: The MIT Press.

Covers aspects of spatial hearing for indoor and outdoor conditions, with many applications to sound systems and room acoustics. Highly recommended.

Fletcher, H. (Original Author), Allen, J. (Editor). 1995. *Speech and Hearing in Communication.* Woodbury, NY: American Institute of Physics.

One of the first books on basic and applied psychoacoustic research by one of the preeminent acoustics researchers of the 20th century. The original edition is out-of-print, but a reprint edition is available through the Acoustical Society of America.

Green, D. 1976. *An Introduction to Hearing.* Hillsdale, NJ: Lawrence Erlbaum Associates, Publishers.

A good book covering the basics of physical and psychophysical aspects of hearing.

Zwicker, E. and Fastl, H. 1990. *Pyschoacoustics Facts and Models.* Heidelberg, Germany: Springer-Verlag.

A relatively technical book covering basic and advanced psychophysical aspects of hearing, including product sound quality analysis.

Reference Data and Handbooks

The books noted below contain a variety of acoustical, architectural, and electronic data useful in the design, application, installation, and operation of sound systems.

Ballou, G. (Editor). 1987. *Handbook for Sound Engineers The New Audio Cyclopedia.* Indianapolis, IN: Howard W. Sams & Company.

Pound-for-pound one of the largest and most useful sources of information on acoustics and audio. Superceded by newer edition.

Bensen, B. (Editor). 1988. *Audio Engineering Handbook.* New York, NY: McGraw Hill, Inc.

A broad based treatment of all aspects of acoustics and audio equipment. (Out-of-print).

Borwick, J. (Editor). 1988. *Loudspeaker and Headphone Handbook.* London, England: Butterworths.

A good book summarizing most aspects of loudspeaker design and featuring one of the few sources of design information on headphones. Book sections on horn theory are weak.

Crocker, M. (Editor). 1997. *Encyclopedia of Acoustics*. New York, NY: John Wiley & Sons, Inc.

A four volume edition covering all aspects of acoustics with chapters on transducer theory, microphones, loudspeakers, and sound systems. Highly recommended.

Davis, G and Jones, R. 1989. *Sound Reinforcement Handbook - 2nd Edition*. Milwaukee, WI: Hal Leonard Publishing Corporation.

A nice easy to read book on basic theory, set-up, and operation of sound systems with emphasis towards live sound for music performance.

Eargle, J. 1996. *Loudspeaker Handbook*. New York, NY: Chapman & Hall.

A thorough introduction to drivers, loudspeakers, and enclosure design with a less mathematical treatment. Standard and unusual loudspeaker types are covered. Highly recommended. (Out-of-print).

Eargle, J. 1994. *Electroacoustical Design Data*. New York, NY: Van Nostrand Reinhold.

A good summary of relevant and hard-to-find design data, formuli, and nomograms for acoustics, audio, electronics, loudspeakers, music, and psycho-acoustics.

Eargle, J. 1990. *The Microphone Handbook*. Plainview, NY: Elar Publishing Company.

A thorough treatment of microphone design and usage geared mostly towards recording, but does have information on sound reinforcement usage. (Out of print).

Eargle, J. 1989. *Handbook of Sound System Design*. Plainview, NY: Elar Publishing Company.

An excellent summary of various aspects of sound system design presented in a clear concise manner with an approach different from other sound system design books. (Out-of-print).

Earley, M. , Sheehan, J., and Caloggero, J. (Editors). 1999. *National Electrical Code® Handbook - 8th Edition*. Quincy, MA: National Fire Protection Association, Inc.

An interpretive treatment of the National Electrical Code (NEC®) which is helpful to understand the arcane minutia contained within the NEC®. The book is routinely issued with each NEC® update.

Glerum, J. 1997. *Stage Rigging Handbook - 2nd Edition*. Carbondale, IL: Southern Illinois University Press.

An excellent book covering aspects of theatrical rigging systems with much information applicable to hanging loudspeaker systems.

Harris, C. (Editor). 1975. *Dictionary of Architecture and Construction*. New York, NY: McGraw -Hill, Inc.

A general dictionary for architectural and construction terms including everything from "Aaron's rod" to "zwinger." Contains many acoustics terms.

Mapp, P. 1989. *The Audio System Designer Technical Reference*. Cheltenham, England: The Chapman Partnership.

Similar to the Eargle design data book above, but having somewhat different useful data. Book is available through Synergetic Audio Concepts.

Morfey, C. 2001. *Dictionary of Acoustics*. San Diego, CA: Academic Press.

A thorough compilation of abbreviations, definitions, and terms used in acoustics.

Parker, S. 1987. *Acoustics Source Book*. New York, NY: McGraw Hill, Inc.

A nice collection of short articles on basic physical acoustics, psycho-acoustics, and linguistics with extensive sections on transducers, high fidelity equipment, sound reproduction, and sound reinforcement systems. Some information is dated, even at the time of its printing, such as descriptions of phonograph records and 8-track tape technology. (Out-of-print).

Sands, L. 1978. *Sound System Installers Handbook*. Indianapolis, IN: Howard W. Sams Publishing Company.

Basic information covering operation and equipment interconnection for a variety of sound system types. (Out-of-print).

Tremaine, H. 1973. *Audio Cyclopedia - 2nd Edition*. Indianapolis, IN: Howard W. Sams Publishing Company.

Precursor to the Ballou audio handbook, but topics are presented in an annoying question and answer format that only Alex Trebeck would like. (Out-of-print).

Various. 1938. *Motion Picture Sound Engineering*. New York, NY: D. Van Nostrand Company. State-of-the art information when published, the book still has lots of useful information geared towards the motion picture industry and vacuum tube audio equipment. (Out-of-print).

White, G. 1995. *The Audio Dictionary - 2nd Edition*. Seattle, WA: University of Washington Press.

A nice compendium of abbreviations, definitions, and terms used in acoustics and audio.

Acoustical and Audio Measurement Systems

The books noted below describe acoustical and audio measurement systems and equipment having application to sound systems.

Beranek, L. 1949. *Acoustic Measurements*. New York, NY: John Wiley & Sons, Inc.

The first textbook covering acoustical and electro-acoustical measurements, written by one of the foremost acousticians of this century. The testing procedures described are generally outdated. The original edition is out-of-print, but a reprint edition is available through the Acoustical Society of America.

D'Appolito, J. 1998. *Testing Loudspeakers*. Peterborough, NH: Audio Amateur Press.

The only book devoted entirely to testing and evaluating drivers and loudspeakers using a variety of current acoustical instrumentation. Highly recommended.

Ginn, K. 1978. *Architectural Acoustics - 2nd Edition*. Nærum, Denmark: Brüel & Kjær.

Basics of architectural acoustics and acoustic measurements with applications using Brüel & Kjær equipment. (Out-of-print).

Gottlieb, I. 1985. *Basic Electronic Test Procedures - 2nd Edition*. Blue Ridge Summit, PA: TAB Books, Inc.

Electronic system measurements are described in detail in a clear easy-to-understand manner. (Out-of-print).

Hassall, J and Zaveri, K. 1979. *Acoustic Noise Measurements - 4th Edition*. Nærum, Denmark: Brüel & Kjær.

Basics of acoustical instrumentation and measurements with applications using Brüel & Kjær equipment. (Out-of-print).

Meltzer, B. 1993. *Audio Measurement Handbook*. Beaverton, OR: Audio Precision, Inc.

Different types of audio measurements are described in brief with examples taken using Audio Precision instrumentation.

Peterson, A. 1980. *Handbook of Noise Measurement - 9th Edition*. Concord, MA: GenRad.

Basics of acoustical instrumentation and measurements with applications using GenRad equipment. (Out-of-print).

Prohs, J. (Editor). 1988. *Time Delay Spectrometry - An Anthology.* New York, NY: Audio Engineering Society.

A collection of papers by Richard C. Heyser describing developments and application of TDS measurement techniques. Highly recommended.

History of Acoustics and Audio Systems

The books noted below provide historical information and references on the development of acoustics, sound systems, and transducers.

Beyer, R. 1999. *Sounds of Our Times - Two Hundred Years of Acoustics.* New York, NY: Springer-Verlag New York.

An exhaustive survey covering the historical development of all aspects of acoustics with several good chapters on transducers used in electro-acoustics. Highly recommended.

Hunt, F. 1982. *Electroacoustics The Analysis of Transduction, and its Historical Background.* New York, NY: American Institute of Physics.

A book which traces the development of electro-acoustics from the 1800s to the early 1950s. The original edition is out-of-print, but a reprint edition is available through the Acoustical Society of America.

Thompson, E. 1992. *"Mysteries of the Acoustic": Architectural Acoustics in America 1800 - 1932.* Ph.D. Dissertation, Princeton University.

An excellent historical survey with copious references to other source material relating to the early development of acoustics and electro-acoustics in the United States.

Conference and Collected Papers

The books noted below are compilations of important peer-review papers covering various aspects of acoustics, sound systems, and transducers which were published in professional audio journals.

Klepper, D. (Editor). 1978. *Sound Reinforcement Volume 1.* New York, NY: Audio Engineering Society.

An excellent collection of important papers on sound reinforcement and related topics published in the Journal of the Audio Engineering Society between 1953 and 1978.

Klepper, D. (Editor). 1996. *Sound Reinforcement Volume 2*. New York, NY: Audio Engineering Society.

A continuation of the earlier AES sound reinforcement anthology but with papers published between 1967 and 1996.

Uzzle, T. (Editor). 1989. *The Proceedings of the AES 6th International Conference - Sound Reinforcement*. New York, NY: Audio Engineering Society.

A collection of papers on various sound reinforcement topics presented at a special Audio Engineering Society conference.

Manufacturer's Publications

The sources noted below provide useful information on equipment components and sound system design based on manufacturer's products, although much information is generally manufacturer-independent.

Bartlett, B. 1997. *The Crown Boundary Microphone Application Guide*. Elkhart, IN: Crown International, Inc.

Basics of boundary layer microphone applications are covered.

Bartlett, B. 1997. *The Crown Microphone Application Guide CM, LM, and GLM Series*. Elkhart, IN: Crown International, Inc.

General application of microphones for voice and musical instrument pick-up are covered.

Bartlett, B. 1997. *The Crown Microphone Application Guide for Speech Sound Reinforcement*. Elkhart, IN: Crown International, Inc.

General application of microphones for voice pick-up are covered.

Eargle, J. 1999. *JBL Sound System Design Reference Manual*. Northridge, CA: JBL, Inc.

An updated version of the 1982 edition. Highly recommended. Available for download from www.jblpro.com.

Eargle, J. 1982. *JBL Sound System Design Reference Manual*. Northridge, CA: JBL, Inc.

The precursor to Eargle's later books on sound system design. (Out-of-print).

Gutmann, J. 1998. *Planning Brochure Practical Applications in Infra-Red Technology*. Wedemark, Germany: Sennheiser Electronic.

Everything you could possibly want to know about infrared technology as applied to assistive listening systems.

Jones, B. 1993. *Wireless Microphone Systems - Operational Basics and Applications.* Rio Rancho, NM: Lectrosonics, Inc.

Thorough description of wireless radio microphone fundamentals and application.

Jones, B. 1996. *LecNet™ Audio Components - A Guide to Applications and Design.* Rio Rancho, NM: Lectrosonics, Inc.

Good description of automated audio systems using this particular manufacturer's products, but the concepts are applicable other products.

Kamlet, R. 1999. *Control Contractor Ceiling Loudspeakers Technical Application Guide.* Northridge, CA: JBL Professional.

Detailed design and application guidelines for ceiling distributed loudspeaker systems using the JBL Control series loudspeakers, with much information applicable to any ceiling distributed loudspeaker system. The publication has an excellent discussion of polar versus listening plane loudspeaker coverage. Available for download from www.jblpro.com.

Mapp, P. 1985. *Audio Systems Design and Engineering.* Kidderminster, Worcestershire, England: Klark-Teknik.

The title is a bit deceiving as the book only covers sound system equalization and signal delay, but very good information nonetheless. (Out-of-print).

Various. 1986 to present. *Application Notes.* Van Nuys, CA: Jensen Transformers.

Articles and schematics on uses of transformers for solving a variety of common audio problems.

Various. 1984 to present. *JBL Technical Notes.* Northridge, CA: JBL, Inc.

An aperiodic series of short articles on sound system components and application.

Various. 1982 to present. *Rane Notes.* Mukilteo, WA: Rane Corporation.

An excellent in-depth series of application notes on audio equipment and design written in a very humorous fashion.

Various, 2000. *Safe & Quiet.* Petaluma, CA: Furman Sound, Inc.

A design guide on requirements for AC power and application of various electrical power conditioning equipment for sound systems.

Various. 1996. *Shure Guide to Audio Systems for Meeting Facilities.* Evanston, IL: Shure Brothers, Inc.

Basics of microphone selection and application for meeting room sound systems are covered.

Various. 1979 to present. *The PA Bible.* Buchanan, MI: Electro-Voice, Inc.

An aperiodic series of short articles on various aspects of sound reinforcement system design and application, but geared mostly towards live sound and rock-and-roll performances.

Vear, T. 1994. *Microphone Selection and Application for Church Sound Systems.* Evanston, IL: Shure Brothers, Inc.

Basics of microphone design, selection, and application are covered, but the concepts are suitable for buildings other than houses of worship.

Vear, T. 1996. *Selection and Operation of Wireless Microphone Systems.* Evanston, IL: Shure Brothers, Inc.

Excellent source of information on wireless radio microphones and application.

Whitlock, B. 2001. *Causes and Cures for System "Ground Loop" Problems.* Van Nuys, CA: Jensen Transformers, Inc.

An excellent overview of electrical noise generation, interference, and grounding practices in audio systems to reduce system noise.

Magazines and Newsletters

The periodicals listed below can provide additional information on sound system equipment, applications, and installation. Some periodicals are by paid subscription and others are free but through controlled circulation.

Acoustics Bulletin. Institute of Acoustics. 77A St. Peter's Street, St. Albans, Herts, AL1 3BN, United Kingdom.

Moderately technical journal with many practical applications in architectural acoustics. Published six times per year.

Architectural Record. McGraw Hill Companies. 1221 Avenue of the America, New York, NY 10020.

General purpose architectural magazine often featuring articles on theaters and similar spaces where sound systems are used. Published monthly.

Architecture. BPI Communications, Inc. 1515 Broadway, New York, NY 10036.

General purpose architectural magazine often featuring articles on theaters and similar spaces where sound systems are used. Published monthly.

Building Acoustics. Multi-Science Publishing Company, Ltd. 5 Wates Way, Brentwood, Essex, CM14 9TB, United Kingdom.

Technical journal on architectural acoustics and its applications to different building types. Published quarterly. Highly recommended.

Journal of the Acoustical Society of America. Acoustical Society of America. Suite 1NO1, 2 Huntington Quadrangle, Melville NY 11747.

Highly technical journal on all aspects of acoustics. Published monthly.

Journal of the Audio Engineering Society. Audio Engineering Society. 60 East 42nd Street, Suite 2520, New York, NY 10165.

Technical journal covering engineering aspects of audio equipment. Published 10 timer per year.

LIVE SOUND! International. Royle Publications, Inc. 112 Market Street, Sun Prairie, WI 53590.

General purpose trade magazine with articles featuring live sound reinforcement geared towards the touring sound industry. Published eight times per year.

Sound and Video Contractor. PRIMEDIA Intertec. 9800 Metcalf, Overland Park, KS 66212.

General purpose trade magazine with articles on equipment and featured system installations. Published monthly.

S*ound and Communications*. Sound and Communications Publishing, Inc. 25 Willowdale Avenue, Port Washington, NY 11050.

General purpose trade magazine with articles on equipment and featured system installations. Published monthly.

Syn-Aud-Con Newsletter and Tech Topics. Synergetic Audio Concepts. 8780 Ruffing Road, Greenville, IN 47124.

Audio newsletter and technical topics bulletin covering different aspects of acoustics, sound system design, equipment, and measurements. One of the few sources describing the latest state-of-the-art in sound systems design. Published quarterly. Highly recommended.

Systems Contractor News. Miller Freeman PSN, Inc. 460 Park Avenue South, 9th Floor, New York, NY 10016.

General purpose trade magazine with articles on equipment and featured system installations. Published monthly.

Industry Trade Groups and Standards Organizations

The organizations noted below can provide information on relevant standards applicable to acoustics, audio equipment, electronics, and sound systems.

Acoustical Society of America (ASA)
Suite 1NO1
2 Huntington Quadrangle
Melville, NY 11747
USA
Tel: 516.576.2360
Web: www.asa.aip.org

American Institute of Architects (AIA)
1735 New York Avenue, NW
Washington, DC 20006
USA
Tel: 202.626.7300
Web: www.aia.org

American National Standards Institute (ANSI)
11 West 42nd Street
New York, NY 10036
USA
Tel: 212.642.4980
Web: www.ansi.org

American Society of Heating Ventilation and Air Conditioning Engineers (ASHRAE)
1791 Tullie Circle, NE
Atlanta, GA 30329
USA
Tel: 404.636.8400
Web: www.ashrae.org

Architectural and Transportation Barriers Compliance Board (ATBCB)
Suite 1100
1331 F Street, NW
Washington, DC 20004
USA
Tel: 202.272.5434
Web: www.access-board.gov

Audio Engineering Society (AES)
Suite 2520
60 East 42nd Street
New York, NY 10165
USA
Tel: 212.661.8528
Web: www.aes.org

British Federation of Audio
19 Charing Cross Road
London WC2H 0ES
UNITED KINGDOM
Tel: 44 (0) 171 930 3206
Web: www.british-audio.org.uk

British Standards Institution (BSI)
British Standards House
389 Chiswick High Road
London W4 4AL
UNITED KINGDOM
Tel: 44 (0) 181 996 9000
web: www.bsi.org.uk

Canadian Acoustical Association (CAA)
P.O. Box 1351 Station F
Toronto Ontario M4Y 2V9
CANADA
Tel: 613.993.0102
Web: www.uwo.ca/hhcru/caa

Canadian Standards Association (CSA)
178 Rexdale Boulevard
Etobicoke Ontario M9W 1R3
CANADA
Tel: 416.747.4000
Web: www.csa-international.org

Electronic Industries Alliance (EIA)
2500 Wilson Boulevard
Arlington, VA 22201
USA
Tel: 703.907.7500
Web: www.eia.org

Factory Mutual (FM)
Suite 300
655 Engineering Drive
Norcross, GA 30092
USA
Tel: 770.662.5700
Web: www.factorymutual.com

Federal Communications Commission (FCC)
445 12th Street, SW
Washington, DC 20554
USA
Tel: 202.418.0200
Web: www.fcc.gov

Institute of Acoustics (IOA)
77A St. Peters Street
St. Albans
Hertfordshire AL1 3BN
UNITED KINGDOM
Tel: 44 (0) 172 784 8195
Web: www.ioa.org.uk

Institute of Electrical and Electronic Engineers (IEEE)
17th Floor
3 Park Avenue
New York, NY 10016
USA
Tel: 212.419.7900
Web: www.ieee.org

Institute of Noise Control Engineering of the USA (INCE)
PO Box 220
Saddle River, NJ 07458
USA
Tel: 201.760.1101
Web: www.ince.org

Institute of Sound and Communications Engineering (ISCE)
PO Box 71
Wallasey CH45 1WA
UNITED KINGDOM
Tel: 44 (0) 151 639 5211
Web: www.isce.htm

International Communications Industries Association (ICIA)
11242 Waples Mill Road
Fairfax, VA 22030
USA
Tel: 703.273.7200
Web: www.icia.org

International Electrotechnical Commission (IEC)
3 rue de Varembé
Ch-1211 Geneva 20
SWITZERLAND
Tel: 41 22 919 02 11
Web: www.iec.ch.

International Organization for Standardization (ISO)
1 rue de Varembé
CH-1211 Geneva 20
SWITZERLAND
Tel: 41 22 749 01 11
Web: www.iso.ch

National Council of Acoustical Consultants (NCAC)
Suite 1A
66 Morris Avenue
Springfield, NJ 07081
USA
Tel: 973.564.5859
Web: www.ncac.com

National Electrical Code (NEC®)
Suite 201
7310 West McNab Road
Tamarac, FL 33321
USA
Tel: 888.632.2633
Web: www.mikeholt.com

National Electrical Manufacturers Association (NEMA)
Suite 1847
1300 North 17th Street
Arlington, VA 22209
USA
Tel: 703.841.3200
Web: www.nema.org/staging/index/html

National Fire Protection Association (NFPA)
1 Batterymarch Park
Quincey, MA 02269
USA
Tel: 617.770.3000
Web: www.nfpa.org

National Systems Contractors Association (NSCA)
Suite 420
625 First Street, SE
Cedar Rapids, IA 52401
USA
Tel: 319.366.6722
Web: www.nsca.org

Professional Lighting and Sound Association (PLASA)
38 St. Leonards Road
Eastbourne
East Sussex BN21 3UT
UNITED KINGDOM
Tel: 44 (0) 132 341 0335
Web: www.plasa.org.uk/plasa

Society of Motion Picture and Television Engineers (SMPTE)
595 West Hartsdale Avenue
White Plains, NY 10607
USA
Tel: 914.761.1100
Web: www.smpte.org

Underwriter's Laboratories (UL)
333 Pfingsten Road
Northbrook, IL 60062
USA
Tel: 847.272.8800
Web: www.ul.com

United States Institute for Theater Technology (USITT)
6443 Ridings Road
Syracuse, NY 13206
USA
Tel: 800.938.7488
Web: ffa.ucalgary.ca/usitt

Appendix C

Glossary of Technical Terms Used in the Text

Absorption — The reduction in energy level as sound strikes a surface and is converted into other forms of energy such as heat.

Acoustic Center — The frequency dependent apparent point from which sound radiates from a source, which may or may not correspond to the source's physical center.

Acoustic Power — The inherent sound energy radiated by a sound source referenced to 10-12 watt (units: acoustical watts).

Active — An audio component that requires a source of AC power to operate; also an audio device which provides signal amplification.

Ampacity — The current carrying capacity of a wire, cable, or electrical circuit (units: amperes).

Analog — An exact replica of the acoustical signal in which the audio signal voltage varies directly as the sound pressure of the original signal source.

Archival Recording — Recording of event proceedings to be used for future reference, such as court proceedings and governmental legislation.

Attack Time — The speed at which the gain of a compressor/limiter is reduced in response to an increase in the input signal level (units: ms).

Auralization — The audible simulation by acoustical modeling of a source sound field in a space which replicates the binaural listening impression of the source within the modeled space.

A-weighted (dBA) A standardized frequency equalization curve which attenuates the signal level starting at 1,000 Hz to make the measured data more closely approximate hearing loudness characteristics.

AWG An acronym for American Wire Gauge which is a scale indicating the size of electrical wire ranging from 0000 to 40 where a higher numerical value represents a smaller wire diameter.

Balanced A three-wire configuration where the signal is carried on two signal conductor wires having a common shield wire which does not carry the signal.

Baud A variable unit of digital data transmission speed equal to one bit per second.

Bi-Amplified A method of powering drivers so the outputs from an active crossover are routed to two power amplifier channels (or two separate power amplifiers), each connected to a separate driver, such as low-frequency and high-frequency sections of a loudspeaker system.

Bi-Directional Microphone A microphone type with roughly a figure-of-eight polar pattern which uses an element where the sensitivity is greatest at the front and rear, theoretically completely insensitive at the perpendicular sides, and has a maximum sound rejection angle of 90°. Bi-directional microphones commonly use a ribbon element in which the motion is proportional to the velocity of the air particles.

Bias Current An ultrasonic signal, approximately 100 kHz, which is applied to a tape recording head and mixed with the recorded audio signal to reduce distortion in the recording process.

Block Diagram A simplified drawing of an audio system showing the signal flow, component interconnection, and system operational features.

Bridged A method of connecting the two channel outputs of a stereo power amplifier to give double the power output of a single channel.

Buss A point in an electronic circuit where several connections are made; also a grouping of wires not connected which are used for parallel transmission of data.

Cardioid Microphone A microphone type with roughly a heart-shaped polar pattern which uses an element where the sensitivity is greatest at the front, 6 dB less sensitive at the perpendicular sides, theoretically completely insensitive at the rear, and has a maximum sound rejection angle of 180°.

Carrier Frequency A high-frequency signal transmitted as part of a radio or television broadcast which is either amplitude or frequency modulated by the audio signal.

Clipping	A type of distortion caused when an audio signal passes through an amplification stage and either the input is over driven or the output is overloaded resulting in generation of harmonic distortion.
Codec	An audio device used at both the send and receive locations in a teleconferencing system which serves as an interface between the network and the audio equipment and performs mixing of audio and video signals, A/D and D/A conversion, and compression.
Coherence	A mathematical function calculated in the frequency domain which represents the linearity between two signals and helps evaluate how closely the input is related to the output. A value of 1 represents high correlation and 0 represents poor correlation between input and output.
Comb Filtering	A frequency selective cancellation or boosting due to a signal combining with a time delayed version of itself resulting in a series of peaks and nulls in the frequency domain.
Condenser Microphone	A microphone type which uses an element consisting of a thin stretched diaphragm positioned closely to a metal backplate, both of which are polarized by an electrical charge. Sound impinging on the diaphragm causes the diaphragm to move resulting in a change in the capacitance and the generation of an output voltage corresponding to the acoustical signal amplitude.
Constant Percentage Bandwidth	A frequency spectrum having a constant bandwidth on a linear frequency scale yielding fine resolution at high-frequencies and coarse resolution at low-frequencies.
Continuity	The property of a signal transmission medium having a continuous electrical connection so signals can pass through.
Coverage Angle	The radiation pattern of a loudspeaker as viewed on a polar diagram and is defined by the -6 dB down points relative to the on-axis sound intensity level. Coverage angles are normally plotted for both the horizontal and vertical planes (units: degrees).
Crest Factor	The ratio of the peak-to-RMS values of a signal (units: dB).
Critical Bandwidth	The frequency band within which the loudness of a band of noise of constant level is independent of its bandwidth. In practice critical bands are approximately one-third octave wide.
Critical Distance	The distance from the sound source in a room where the direct sound level is equal to the reverberant sound level and is governed by the directional properties of the sound source and the acoustical absorption in the room (units: ft).

Crossover Frequency	A specific frequency which is used to divide an audio signal into two passbands for distribution to specific drivers.
Crosstalk	The phenomenon where one signal can "leak" over to one or more adjacent channels or cables.
Damping	The dissipation of energy with time which can be enhanced by placing an elastic material on the vibrating object.
Damping Factor	A dimensionless number which is the ratio of the load impedance of an amplifier to the output impedance of the amplifier which determines the ability of the amplifier to control the driver motion after the signal applied to the driver has stopped.
Data Window	A mathematical function used to reduce the negative effects of data reduction when a limited number of FFT points are used to transform time domain data into the frequency domain by reducing the data amplitude at the ends of the sample.
DeciBel	A dimensionless number which expresses a ratio of two numbers, typically power ratios, with a reference quantity in the denominator. Literally one-tenth (deci) of a Bel.
Digital	The application of computer-based technology to audio signal processing, transmission, and storage where the audio signal is sampled and represented as a discrete series of ones and zeros.
DIN	An acronym for the German standards organization Deutsche Institut für Normung.
Direct Sound	The portion of the sound field in a room which arrives at the receiver without reflecting off the room surfaces and is the shortest transmission path length between source and receiver.
Direct-to-Reverberant Ratio	The ratio of the direct sound level to the reverberant sound level in a space and affects the perceived reverberance and speech intelligibility (units: dB).
Directivity Factor (Q)	A dimensionless number which is a measure of the directivity of the source sound radiation.
Directivity Index (DI)	A dimensionless number which is a measure of the directional properties of a device by comparing the ratio of the on-axis sound level to the omnidirectional sound level at a specified distance for a constant sound power output.
Driver	A single loudspeaker transducer.

Durometer	The physical hardness rating of a material, with higher numerical values representing greater hardness, and often called shore hardness.
Duty Cycle	The operating condition of an electrical component describing the load or output versus time.
Dynamic	The electromagnetic principle of a coil of wire moving in a magnetic field which produces a current flow.
Dynamic Range	The ratio of the magnitude of the greatest to weakest portions of a signal (units: dB).
Early Decay Time (EDT)	The time required for the sound level in the room, after the source has stopped radiating, to decay 10 dB and is extrapolated to a full 60 dB decay. Provides a more subjective sense of room reverberance than the reverberation time (units: s).
Early Reflections	The first reflections from the room surfaces closest to the sound source or the receiver which arrive within approximately 40 ms after the direct sound in a small space and 200 ms in a large space.
Electret Microphone	A microphone type similar to a condenser microphone in which either the diaphragm or the backplate has been permanently charged during manufacturing and thus does not require an external polarizing voltage.
Electro-Magnetic Interference (EMI)	Electro-magnetic interference that can be picked-up by audio systems and result in the generation of audible noise. Some common sources include fluorescent lighting ballasts, computers, automobile ignition systems, commercial radio and television stations, and household appliances.
Electro-Magnetic Compatible (EMC)	Electro-magnetic compatible is used to designate audio equipment which is designed to be immune from EMI.
Equivalent Acoustical Distance (EAD)	The distance at which the talker and listener can communicate without the aid of sound reinforcement and is used to determine gain requirements for a sound reinforcement system (units: ft).
Fader	A variable volume attenuator which has typically a linear operating action.
far-field	The portion of the sound field where the sound level decreases at 6 dB per doubling of distance from the source.
Far-Throw	A type of loudspeaker horn which can project sound over long distances and having a narrow coverage angle, such as 40° by 20°. Sometimes called long-throw.

Feedback	The process of feeding back a portion of an electrical signal, as in negative feedback used to reduce distortion and noise in an amplifier circuit; also acoustical feedback where a portion of the loudspeaker output signal is unintentionally picked-up by the microphone and reamplified by the sound system, resulting in audible noise or reduced gain.
Feedback Stability Margin (FSM)	A correction value applied when calculating sound reinforcement system PAG to provide a margin of safety to prevent the sound system from starting to ring or go into uncontrolled feedback.
Foldback	The process of routing amplified sound back to the stage area so performers can hear themselves. Often called monitoring.
Free Field	A field in a homogenous, isotropic medium free of any reflecting boundaries.
Frequency Modulation (FM)	The instantaneous variation of a periodic carrier signal by a modulating frequency signal represented by the audio waveform.
Frequency Response	The amplitude response versus frequency characteristic of a device describing how the gain and phase of the device varies with frequency.
Fast Fourier Transform (FFT)	A mathematical computation which transforms data in the time domain (the waveform) into data in the frequency domain (the spectrum).
Full-Duplex	A communications channel that permits simultaneous sending and receiving of audio signals.
Gain	The increase in the power of an acoustical or electrical signal characterized as a ratio of the output to input levels (units: dB).
Gain-Before-Feedback	The amplification potential of a sound system before it is limited by acoustical feedback (units: dB).
Gating	The operation of a circuit which performs similar to a switch and permits a signal to pass or not pass through.
Ground Loop	The generation of circulating noise currents in audio signal circuits from the connection of several audio components to the safety ground conductor which results in audible noise.
Ground Plane	An electrical approximation of a zero potential reflective surface at the base of an antenna which is commonly used with one-quarter wavelength antennae.
Group Delay	The rate of change of phase as a function of frequency of an audio component or system (units: s).

Harmonic Distortion	A form of distortion where harmonics related to the fundamental frequency of the input signal are produced by the audio component and are superimposed on the output signal (units: percent).
Headroom	The difference in level between the highest portion of an audio signal and the maximum level the audio component can reproduce without distortion (units: dB).
Helmholtz Resonator	A type of acoustical resonator having a characteristic resonant frequency comprising an enclosed air volume connected to the surrounding atmosphere by a short neck.
High-Level	An audio signal characterized by a signal level between 0.15 V and 70 V, often called a loudspeaker level signal; also refers to amplified sound at a loud level.
Hypercardioid Microphone	A cardioid microphone type with roughly a heart-shaped polar pattern with a "tail" at the rear which uses an element where the sensitivity is greatest at the front, 12 dB less sensitive at the perpendicular sides, 6 dB less sensitive at the rear, and has a maximum sound rejection angle of 110°.
Impedance	The numerical measure of the complex reaction to current flow generated in an electrical circuit when an AC voltage is applied (units: ohms).
Induction Loop (IL)	A wire coil routed around the room connected to a voltage source in which a current circulates and induces a magnetic field picked-up by a telecoil.
Infrared (IR)	A type of electromagnetic radiation outside the visible light spectrum which is at the red spectrum end and characterized by frequencies having wavelengths between 700 nm and 1 mm.
Inrush	A momentary current surge up to 50 times the rated full power current, which occurs in the building electrical power supply when a large power amplifier is turned on.
Insertion Loss	The level reduction of an audio signal due to inherent resistances in a passive audio device (units: dB).
Intermodulation Distortion	A form of distortion where one signal is amplitude modulated by another signal resulting in upper and lower sidebands equal to the sum and difference frequencies of the two signals (units: percent).
Inverse-Square Loss	A physical property of sound or light when radiating as a point source in a free field where the intensity decreases by one-fourth (6 dB) for every distance doubling from the source.

Impulse Response A short duration impulse of sufficient energy to excite a linear system and when analyzed yields the time response properties of the device, or when convolved using the FFT process, yields the frequency response properties of the device.

Knee-Point The point in the response characteristic of a compressor, limiter, crossover, or frequency equalizer where the slope of the curve changes.

Lobing The phenomenon where an acoustical transducer exhibits a non-uniform and frequency dependant radiation or pick-up pattern resulting in pronounced "lobes" or "fingers" projecting from the central polar pattern.

Line Level An audio signal characterized by a signal level between 8 mV and 1.2 V.

Low-Level An audio signal characterized by a signal levels between 0.02 mV and 0.13 V, often called a microphone level signal; also refers to amplified sound at a quiet level.

Mass Break-Point Frequency The point in a frequency response curve of a compression driver where the high-frequency response begins to roll-off due to the equivalent shunt capacitance of the compression driver (units: Hz).

Maximum Length Sequence (MLS) An electronic test signal having a flat energy versus frequency characteristic over a wide frequency range which resembles white noise with a slow periodic repetition characteristic.

Mid-Throw A type of loudspeaker horn which can project sound moderate distances and having a medium coverage angle, such as 90° by 40°. Sometimes called medium-throw.

Mix-Minus A type of signal mixing circuit where there is one output associated with each input that includes all other inputs except the one it is associated with, resulting in an output mix minus the one input. These circuits are often used in automatic microphone mixers to help minimize feedback by the active microphone near the closest loudspeaker.

Monophonic A type of sound transmission which uses only one channel, but can use multiple loudspeakers each radiating the same signal.

near-field The sound field close to the sound source where the sound pressure and particle velocity are not in phase.

Near-Throw A type of loudspeaker horn which can project sound short distances and having a wide coverage angle, such as 120° by 60°. Sometimes called short-throw.

Needed Acoustical Gain (NAG) — A measure of the increase in sound level at the distant listener's position which must be provided by the sound reinforcement system (units: dB).

N Factor — A dimensionless number which is ratio of the number of loudspeakers providing direct sound to the listener to the total number of loudspeakers in the sound system.

Noise Criterion (NC) — A composite single number rating for assessing HVAC noise levels in the octave frequency bands between 63 and 8,000 Hz.

Nyquist Criteria — The requirement that in order to accurately sample a frequency requires it to be sampled at two times the frequency.

Nyquist Frequency — The highest frequency that can be accurately sampled and is one-half the sampling frequency (units: Hz).

Number of Open Microphones (NOM) — The number of microphones that are gated on and are transmitting an audio signal. Each doubling of the number of open microphones decreases the gain-before-feedback by 3 dB.

Ohm — The unit of electrical resistance which opposes an electrical current in a conductor.

Omnidirectional Microphone — A type of microphone which uses an element where the sensitivity is equal in all directions and has a circular polar pattern.

Order — The number of poles which make up an electrical filter or crossover network.

Overcurrent — A current that exceeds the rated current or ampacity of a conductor resulting from ground fault, overload, or short circuit.

Pad — A circuit element that has a fixed amount of attenuation, or insertion loss, for a signal which passes through.

Pan — Slang for positioning the apparent acoustical source using a pan pot to achieve a sense of source localization.

Pan Pot — A ganged set of two variable volume attenuators which are wired so that a high signal level in one results in a low signal level in the other. Taken from the term panoramic potentiometer.

Passband — The frequency range passed through a filter that is 3 dB down at the lower and upper frequency extremes relative to the maximum signal level (units: Hz).

Passive — A type of circuit that has no amplification or results in a loss of level for a signal which passes through.

Pattern Flip	The phenomenon where the directional pattern of a loudspeaker rotates approximately 90° at one frequency band relative to adjacent frequency bands resulting in an uneven coverage versus frequency characteristic.
Percent Articulation Loss of Consonants (AL_{CONS})	A measure of speech intelligibility which assesses the percentage loss of speech consonants by the sound system and the acoustical environment, developed by acoustician V. M. A. Peutz.
Period	The smallest increment of an independent variable for which a function repeats itself; also the inverse of frequency (units: s).
Phantom Power	A source of direct current carried on the microphone cable which powers a condenser microphone preamplifier and is supplied by a low voltage source, typically between 12 and 52 V.
Phase Shift	The change in phase (time) as a function of frequency due to a time delay as the signal passes through the audio component or acoustical environment (units: degrees).
Phasing Plug	An element of a compression driver having a series of slits with the purpose of transmitting in-phase sound from the diaphragm to the horn throat.
Piezoelectric	The property of certain materials, such as quartz, which either generates an electrical charge when mechanically stressed, or when the material is stressed produces an electrical charge.
Pink Noise	A type of random noise in which the spectrum density varies as the inverse of frequency and has a constant amount of energy in each octave frequency band.
Point Source	An idealized sound source which is very small compared to the wavelengths which radiate from it and exhibits an omnidirectional radiation pattern.
Polar Pattern	A graphical representation of the sensitivity of an audio transducer as a function of the angle.
Polarity	The relative phase relationship between two signals equivalent to 180° across the entire frequency domain of the signals.
Poles	The number of orders which make up an electrical filter or crossover network.
Potential Acoustical Gain (PAG)	A measure of the maximum increase in reinforced sound level relative to no reinforcement at the distant listener's position (units: dB).

Power Response A type of frequency response which is a graphical representation of the acoustic power of the device as a function of frequency; also the combined direct and reverberant sound levels from a loudspeaker as measured with a real time analyzer (units: dB).

Preamplifier A type of amplifier that increases the level of small signals and may also contain network circuits to adjust the audio signal frequency spectrum.

Primary The transformer wire windings (coil) which the signal current is input from the previously connected audio component.

Proximity Effect The effect which occurs as a directional microphone is brought closer to a sound source and results in a boost in microphone low-frequency output due to an increase in the microphone sensitivity (units: dB).

Presbycusis The decrease in high-frequency hearing sensitivity which occurs naturally with age.

Psycho-acoustical Of pertaining to the psychological and subjective aspects of the perception of sound.

Q A dimensionless number which is used to describe a variety of electrical and electro-acoustic phenomena. For a loudspeaker system it refers to the directivity of the sound radiation; for resonant systems it refers to "quality factor" and measures the sharpness of the resonant peak of the audio equipment component or system; and for capacitors it refers to the efficiency of the device and is the ratio of the capacitive reactance to the resistance.

Quantization The process of digitizing the amplitude of an analog waveform into a series of finite discrete levels as part of analog-to-digital process.

Quiescent The inactive state of an electrical component that may draw current or voltage when no signal is applied.

Raceway A full or partial enclosure designed to hold cables made of metal or an insulating material including conduits, underfloor raceways, surface metal raceways, cable trays, cable troughs, and gutters.

Rack Space Used as a unit of size in sound system equipment where one rack space is equivalent to 1¾ inches in height. Equipment racks are rated as to storage capacity by the number of rack spaces and the rack height is based on rack space multiples.

Rack Unit An industry standard unit of height measurement equal to 1¾ inches which is applicable to sound system equipment components and equipment racks.

Radio Frequency Interference (RFI) — An acronym for radio frequency interference which can be picked-up by audio equipment from radio, television, or other broadcast systems, and results in the generation of audible noise. RFI is very high-frequency EMI.

Rapid Speech Transmission Index (RASTI) — A shortened version of the STI and is a composite measure of the speech intelligibility of a room or sound system at the 500 and 2,000 Hz octave frequency bands which assesses the degradation of the amplitude modulation of speech due to reverberation and noise.

Real Time — Of pertaining to an event or signal process occurring at the exact moment it occurs in absolute time.

Release Time — The speed at which the gain of a compressor/limiter is restored to the original value after the input stimulus is removed (units: ms).

Resonant Frequency — The frequency at which resonance occurs where a system in forced oscillation requires little energy to produce its greatest motion or amplitude (units: Hz).

Reverberant Sound Level — The magnitude of reflected sound due to multiple reflections from the room surfaces (units: dB).

Reverberation Time — The time required for the sound level in the room to decay 60 dB, equivalent to one-millionth its original intensity, after the source has stopped radiating acoustical energy (units: s).

Ribbon — A thin corrugated conductive piece of metal which vibrates when exposed to sound, commonly used in some microphone and high-frequency driver designs.

Ring Modes — Specific frequencies (modes) which at the onset of feedback have a period several times longer than nearby frequencies causing a ringing in the response of a sound system.

Room Constant (R) — A measure of the amount of sound absorption in a room based on the room surface area and the average sound absorption coefficient of the room finishes (units: ft^2).

Root-Mean-Square (RMS) — The square root of the mean of the squares of a signal which is proportional to the energy content of the signal.

Sampled — The technique of representing the signal amplitude at a particular point in time.

Secondary — The transformer wire windings (coil) which provide the output signal current to the subsequently connected audio component.

Self-Noise	The inherent noise power generated by an audio device without an input signal applied and is due to Brownian motion of the electrons within the various electrical components (resistors, capacitors, and inductors).
Sensible Heat	Heat which changes the surrounding air temperature without increasing the humidity level (units: °F).
Sensitivity	The minimum input signal applied to an audio device to produce a rated output, with a higher sensitivity rating implying a lower signal input is necessary to achieve the rated output.
Signal-to-Noise Ratio (S/N)	The ratio of the power of the signal to the power of the residual noise if the signal content were removed (units: dB).
Skirts	The slope of a filter response curve outside its passband, commonly illustrated by a bandpass filter response curve.
Slewing Rate Distortion	A form of distortion where the rate at which the signal is changing is limited which reduces the high-frequency handing capacity or output of an audio device or system; also referred to as transient intermodulation distortion (units: V/s).
Sound Masking	A psycho-acoustic phenomenon where the threshold of hearing for one sound is raised by another sound and the second sound is not perceived due to masking by the first sound.
Speech Intelligibility	An objective measure or subjective perception of the percentage of speech correctly understood by a listener compared to the total speech transmitted.
Speech Transmission Index (STI)	A composite measure of the speech intelligibility of a room or sound system between the 125 through 8,000 Hz octave frequency bands which assesses the degradation of the amplitude modulation of speech due to reverberation and noise.
Spring Constant	A constant of proportionality for a spring that responds linearly to an applied load (units: lb/in).
Stereophonic	A type of sound transmission which uses two channels to provide an improved sense of directional realism over that provided by monophonic transmission.
Stopband	The frequency range of an audio signal which is not passed through a filter or other audio component (units: Hz).
Submix	A group of audio signals that are mixed together and can be conveniently controlled as a separate signal independent of the other audio signals.

Supercardioid Microphone A cardioid microphone type with roughly a heart-shaped polar pattern with a small "tail" at the rear which uses an element where the sensitivity is greatest at the front, 8 dB less sensitive at the perpendicular sides, 12 dB less sensitive at the rear, and has a maximum sound rejection angle of 126°.

Transducer A device which receives an input signal of one type and produces an output signal of a different type such that the characteristics of the input signal are preserved and appear in the output signal.

Transfer Function The quotient of the Fourier or Laplace transform of an output signal by the same transform of the input signal with all initial conditions at zero enabling changes in the frequency response of the system to be evaluated.

Transmissibility A dimensionless number which is the ratio of transmitted force output to applied force input and measures the vibration isolator efficiency.

Transondent The property of a material which will permit sound to pass through it with little or no reduction in amplitude or change in phase.

Tri-Amplified A method of powering drivers so the outputs from an active crossover are routed to three power amplifier channels (or three separate power amplifiers), each connected to a separate driver, such as low-frequency, mid-range, and high-frequency sections of a loudspeaker system.

Tuning Ratio A dimensionless number which is the ratio of the driver compliance to the enclosure compliance used in designing acoustic suspension loudspeaker systems.

Tweeter Slang for high-frequency driver in a multi-way loudspeaker system, taken from the sound of a chirping bird.

Ultra High-Frequency (UHF) An acronym for ultra high-frequency and covers the frequency range between 300 and 3,000 MHz.

Unbalanced A two-wire configuration where the signal is carried on both the signal conductor wire and the shield wire.

Very High-Frequency (VHF) An acronym for very high-frequency and covers the frequency range between 30 and 300 MHz.

Volt The unit of voltage expressed as the difference in electric potential between two points on a conductor carrying a current of 1 ampere when the power dissipated between the two points is 1 watt and is equal to 1 joule/coulomb.

Volume Unit (VU) A measure of signal strength with ballistic characteristics which approximates the human hearing response.

Volume Velocity	The rate of alternating flow of a medium through a surface due to a sound wave (units: ft^3s^{-1}).
Watt	The unit of power expressed as a rate of energy transfer and is equal to 1 joule/second.
White Noise	A type of random noise in which the spectrum density is independent of frequency and has a constant amount of energy in each frequency.
Woofer	Slang for low-frequency driver in a multi-way loudspeaker system, taken from the sound of a barking dog.