HOWARD W. SAMS & COMPANY Audio Library 22598

Microphone Manual
Design and Application

David Miles Huber

Microphone Manual
Design and Application

Sound recording is an art form. And basic microphone technique and placement are comparable to an artist's choice of medium and colors. Knowledge of acoustics, placement, and varying microphone designs therefore is critical to achieving the best overall quality on the recording engineer's or recordist's finished "canvass."

Written for the beginner and the professional alike, *Microphone Manual: Design and Application* provides the latest tips, techniques, and tools needed to keep you abreast of the ever-changing technology of microphone design, placement, and environmental acoustics.

In these pages, you'll learn about:

- The microphone pickup and its method of operation
- Cable/connector interfacing
- Single-microphone placement
- Stereo "miking" fundamentals, including spaced-pair, coincident, and near-coincident techniques
- Applied placement techniques within the music production environment, video/film media, and speech and music reinforcement
- Microphone accessories

David Miles Huber currently is working with PLAYBACK in Vancouver, British Columbia, as an engineer, author, and instructor in professional recording. Born in Connersville, Indiana, he received his degree in recording techniques (I.M.P.) from Indiana University. For a large portion of his studies, David was enrolled in the Tonmeister program at the University of Surrey in Guildford, Surrey, England.

0 81262 22598 1

$29.95/22598

ISBN: 0-672-22598-0

HOWARD W. SAMS & COMPANY
A Division of Macmillan, Inc.
4300 West 62nd Street
Indianapolis, Indiana 46268 USA

Microphone Manual

Related Titles

Audio Production Techniques for Video
David Miles Huber

Handbook for Sound Engineers: The New Audio Cyclopedia
Glen Ballou, Editor

Introduction to Professional Recording Techniques
Bruce Bartlett
(John Woram Audio Series)

John D. Lenk's Troubleshooting & Repair of Audio Equipment
John D. Lenk

Modern Recording Techniques, 2nd Edition
Robert E. Runstein and David Miles Huber

Musical Applications of Microprocessors, 2nd Edition
Hal Chamberlin

Sound System Engineering, 2nd Edition
Don and Carolyn Davis

Recording Studio Technology
John Woram (John Woram Series)
(forthcoming)

Principles of Digital Audio
Ken C. Pohlmann

Audio IC Op-Amp Applications, 3rd Edition
Walter G. Jung

How to Build Speaker Enclosures
Alexis Badmaieff and Don Davis

For the retailer nearest you, or to order directly from the publisher, call 800-428-SAMS. In Indiana, Alaska, and Hawaii call 317-298-5699.

Microphone Manual
Design and Application

David Miles Huber

HOWARD W. SAMS & COMPANY
A Division of Macmillan, Inc.
4300 West 62nd Street
Indianapolis, Indiana 46268 USA

FIRST EDITION
FIRST PRINTING—1988

International Standard Book Number: 0-672-22598-0
Library of Congress Catalog Card Number: 88-60062

Acquisitions Editor: *Greg Michael*
Manuscript Editor: *Don MacLaren*
Illustrators: *T. R. Emrick, David Cripe*
Cover Illustration: *Ron Troxell*
Indexer: *Don Herrington*
Compositor: *Photo Comp Corp.*

Printed in the United States of America

Trademark Acknowledgments

All terms mentioned in this book that are known to be trademarks or service marks are listed below. In addition, terms suspected of being trademarks or service marks have been appropriately capitalized. Howard W. Sams & Company cannot attest to the accuracy of this information. Use of a term in this book should not be regarded as affecting the validity of any trademark or service mark.

GLM is a trademark of Crown International, Inc.

Isomax is a registered trademark of Countryman Associates, Inc.

Pressure Zone Microphone, PZM, and PCC are registered trademarks of Crown International, Inc.

Swintek is a registered trademark of Swintek Telecommunications Division

Teflon and Mylar are trademarks of E. I. DuPont de Nemours and Co., Inc.

Contents

This book is dedicated to Phil and Vivian Williams, two folks who get more mileage out of life than anyone I know.

This book is also dedicated to those who seek to study and preserve the history of the audio, video, and broadcast media, so that future generations may learn from the past and build upon its technology.

Preface

In response to the great variety of sound sources and acoustical environments, a similarly great variety of microphone designs and microphone techniques have been developed. Using the proper microphone and knowing techniques for achieving the best sound possible are among the most important tools of the sound engineer and recordist.

Microphone design refers to the specific means by which a microphone is able to operate. A basic knowledge of design helps us understand how the microphone works, and it gives us an insight into the characteristics of how a microphone operates in an acoustic environment. *Microphone application* refers to the various microphone placement techniques.

Microphone design and microphone application are not cut-and-dried fields. The many varieties of microphone offer a wide range of sonic characteristics for a given application. Even a minor difference in the design of a microphone can yield a totally distinctive sound from that of an otherwise similar microphone.

Sound recording is an art form and, as such, should always be open to experimentation and change; it is this flexibility that keeps the music and audio industries fresh. Similarly, basic microphone placement is often an individual's statement borne out of experimentation. What may be considered poor choice of technique today might be the standard industry placement five years from now. As new music styles and equipment develop, new sound-recording techniques may also evolve, giving the recorded character a new sound. In light of this, it is wise to remember an old recording adage known as the first rule of recording: *There are no rules*. However, I would like to supplement this foundational rule: *There are no rules, only guidelines*.

The goal of this book is to present some of these guidelines in light of advances in the field of microphone design and application.

Writing with both the amateur and the professional audio user in mind, I wish to introduce:

- the microphone and its method of operation
- the physical and electrical characteristics of the microphone
- cable-connector interface considerations
- microphone accessories
- single-microphone placement
- fundamental stereo microphone techniques, including spaced-pair, coincident, and near-coincident techniques
- applied placement techniques in the music production environment
- applied placement techniques in video/film media, ENG and EFP placement techniques, clip microphone techniques, and wireless microphone systems
- the microphone in speech and music reinforcement

Higher-quality audio in both music and video require that the engineer and recordist place a premium on the choice of microphone design, placement, and environmental acoustics. I hope that this manual will help you reach these goals.

DAVID MILES HUBER

Acknowledgments

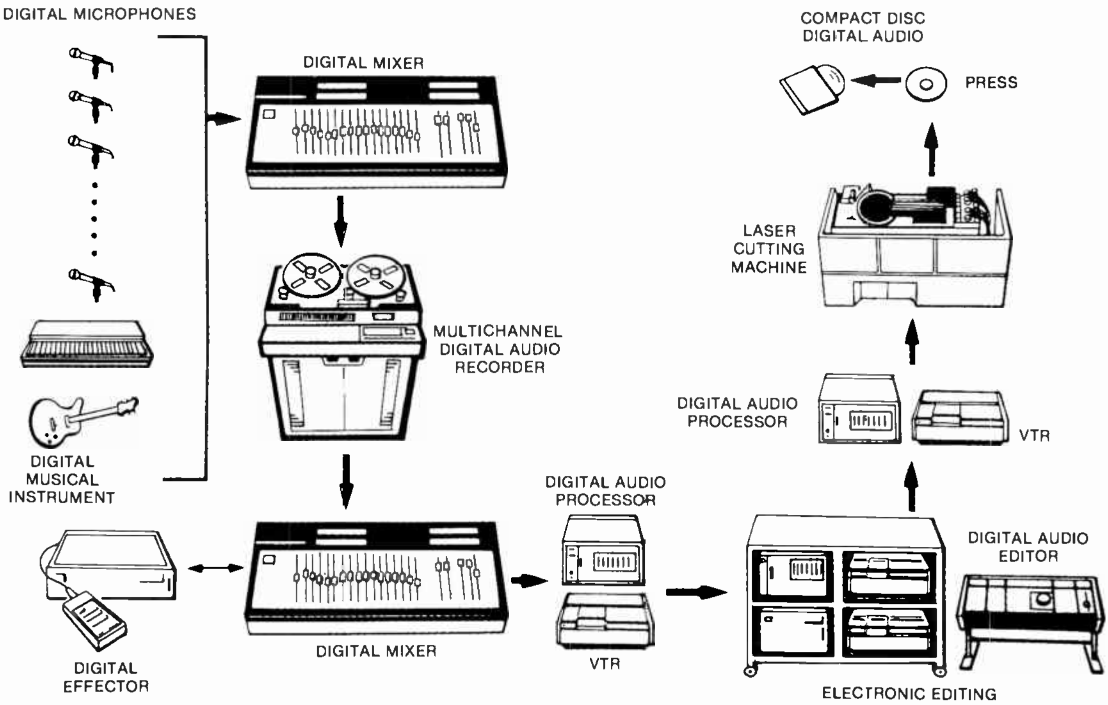
For their assistance in providing photographs, technical information, and personal insights, I want to thank Beth Marshall and Dave Talbot of AKG Acoustics, Inc.; Richard Matyskiel of Atlas Sound; Wes Dooley of Audio Engineering Associates; Robert H. Lowig of Beyer Dynamic, Inc.; Patricia M. Macdonald of Audio Engineering Society, Inc.; John R. Bareham of Bruel & Kjaer Instruments, Inc.; Nigel Branwell of Calrec by AMS; R. Wm. Wanamaker of Gotham Audio Corporation; Kathy Beardsley of Karlberg Enterprises; Ron Streicher of Pacific Audio-Visual Enterprises; Joe Hassan of Pan Communications, Inc.; Lenora Hatcher of Peavey Electronics Corporation; Terry Pennington of Rane Corporation; Anthony J. Tudisco of Sennheiser Electronic Corporation (N.Y.); Schalltechnik Dr.-Ing. Schoeps GmbH; Lee Habich of Shure Brothers, Inc.; Hiro Konno of Sony Corporation of America; Bob Speiden of Speiden & Associates; Swintek; Jeff Fossan of Toa Electronics, Inc.; Robert T. Davis of Yamaha International Corporation.

I would like to extend special thanks to Derek Pilkington of AKG Acoustics; Kathy Varley of Abbey Road Studios; Bruce Bartlett of Crown International; Keith Monks of Keith Monks Ltd.; and James G. Bonis of Whirlwind Music Dist., Inc., and Steve L. Royea for their much appreciated assistance.

1 Introduction

In the production of audio for broadcast, music, video, or live sound, the microphone ("mike") is usually the first device to be found in the chain of a system (Fig. 1-1). Its purpose is to act as an electromechanical link between the acoustical environment of music and/or speech and the electrical environment of audio production. As such, the microphone may be classified as a *transducer*. By definition, a transducer is any device that is capable of changing one form of energy into another, corresponding form of energy. For example, a violin is a transducer in that it transforms the mechanical energy of bowing into acoustical energy (sound waves). Bowing the strings sets up

Fig. 1-1. Digital recording chain with the microphone as first system link.
(Courtesy of Sony Corporation of America)



complex vibrations in the wooden body of the instrument, which, in turn, acts as an acoustic coupler to amplify the signal. These larger vibrations displace the air molecules surrounding the violin and are heard as minute changes in the surrounding atmospheric pressure. Additionally, if a microphone is placed in this sound field (Fig. 1-2), the sound waves act upon the *diaphragm* of the microphone and are converted into a corresponding electrical voltage for further processing.

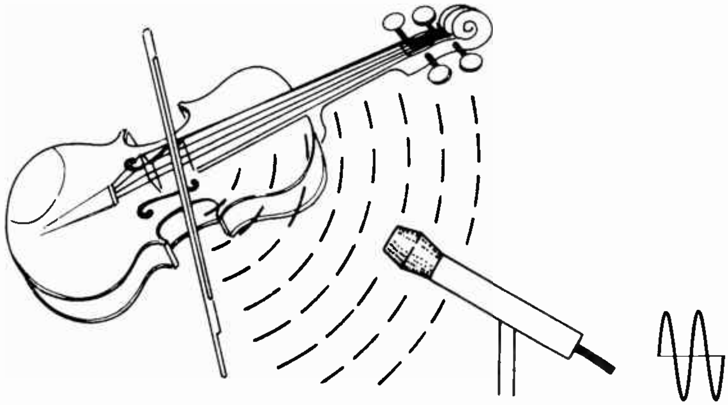


Fig. 1-2. Violin and microphone as transducers.

In this book, the microphone and its orientation will be indicated by the symbol shown in Figure 1-3. For our reference, the *axis* of a microphone can be viewed as an imaginary line drawn perpendicular to the front plane of the diaphragm.

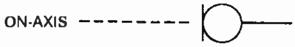


Fig. 1-3. Symbol of the microphone.

Basic Characteristics of Sound

To understand how the process of transduction works, let's look at the basic characteristics of the acoustic waveform and its movement through air.

When a vibrating mass is in direct contact with the air, this mass compresses the air molecules in direct relation to the degree and frequency of the vibrations. Figure 1-4 shows the effects that a vibrating source has on the air molecules.

When the mass is pushed against the molecules, they are squeezed together, forming a high-pressure area known as a *compression*. Conversely, as the mass is pulled away, the molecules are pulled farther apart, forming a low-pressure area known as a *rarefaction*. The higher atmospheric pressure of the compressed molecules always moves away from the sound source toward the lower pressure of the rarefacted molecules, setting an outward wave motion into play. The movement of such a waveform through a medium is known as the *propagation* of a wave.

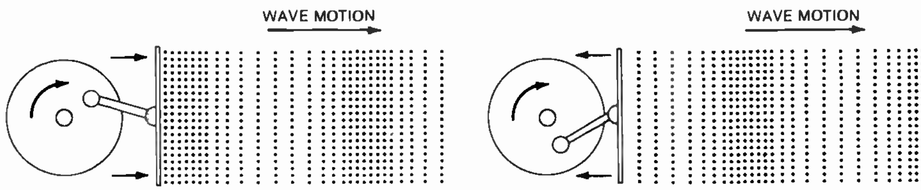


Fig. 1-4. Effects of vibrating mass upon air molecules.

(A) Compression. *(B) Rarefaction.*

Although there are several basic characteristics that allow one waveform to be distinguished from another, all waveforms display two fundamental characteristics: amplitude and frequency.

Amplitude

The distance above or below the centerline of a waveform, such as the pure sine wave shown in Figure 1-5, represents the *amplitude level* of that signal. The greater the distance, or displacement, from the centerline, the more intense the pressure variation, electrical signal, or physical displacement within the medium.

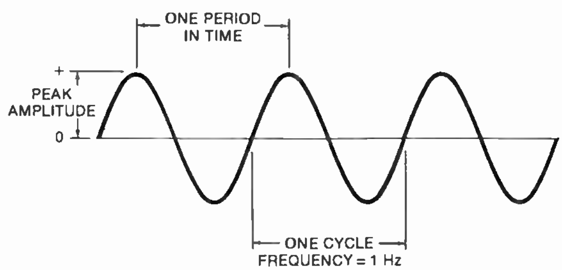


Fig. 1-5. Graph of waveform showing amplitude vs. time.

Frequency

The rate at which an acoustic generator, electrical signal, or vibrating mass repeats the cycle of positive- and negative-going amplitude is known as the *frequency* of that signal. One completed excursion of a wave is known as a *cycle*. The number of cycles that occur in one second is measured in *hertz* (Hz).

The Microphone Diaphragm

The role of a microphone is to convert acoustic energy into an analogous electrical signal by electromechanical means. Such a system comprises a diaphragm and either an integral or an attached element to generate the electrical signal.

The *diaphragm* of a microphone is the elastic mechanical link that interfaces between the acoustic environment and the means of electrical generation. In certain types, such as ribbon and condenser microphones, the diaphragm itself is used to generate an electrical output signal.

When an acoustic wave falls on the diaphragm, the compression (high-pressure area) within the wave displaces the diaphragm inwardly, proportional to the amplitude and frequency of the acoustic pressure (Fig. 1-6). As the wave propagates in a forward movement, the diaphragm is soon presented with a rarefaction (low-pressure area), which displaces the diaphragm outwardly. The repetition of this motion, in relation to a given acoustic waveform, generates an electrical output that represents the sound wave at the microphone.

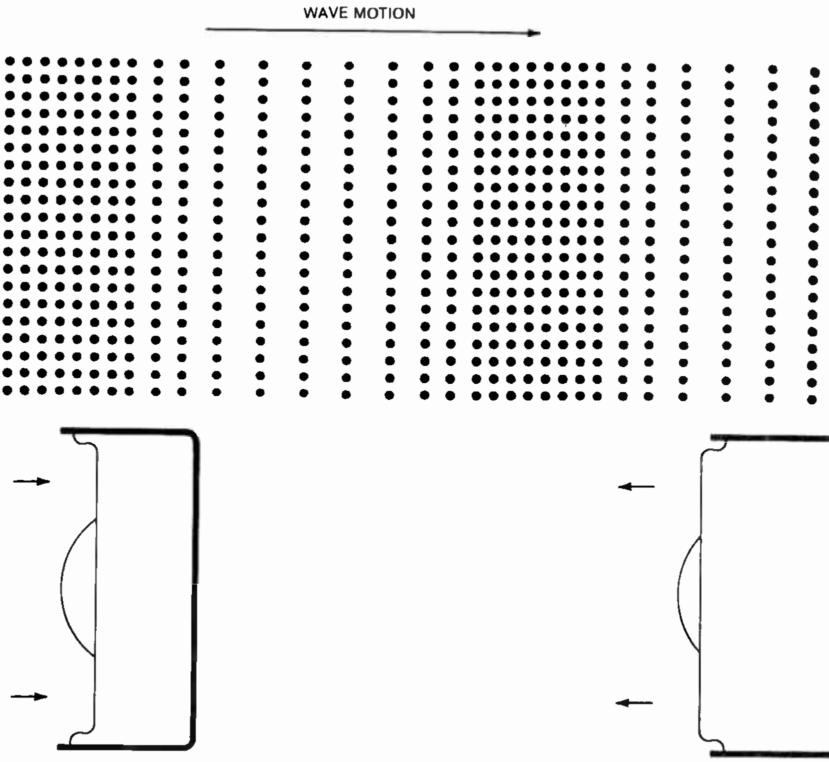


Fig. 1-6. Diaphragm displacement with sound motion.

Microphone Characteristics: An Introduction

In audio production, the microphone and the quality of its reproduction are subject to many external variables (such as the acoustic environment and the mike's placement within this environment) and internal variables (such as the electromechanical design of the microphone itself). All of these interrelated elements work closely together to affect the sound quality of a microphone. In order to satisfy the requirements of a wide range of applications and the subjectivity of human taste, an equally wide range of microphones is available to the professional user. Each microphone has its own particular sound characteristic that best suits a specific application or range of applications. In short, a microphone can be viewed as a specialized tool enabling the best sound possible to be obtained from a specific sound source.

In choosing the best microphone and placement for a specific application, two rules should be considered by the user.

Rule 1. *There are no rules, only guidelines.*

Although specific guidelines suggest a good pickup of a sound source, don't hesitate to experiment in getting the sound that best suits your own taste or the application at hand.

Rule 2. *The overall quality of a signal is no better than the quality of each individual component that contributes to making up that signal.*

Restated, the overall sound of an audio signal is no better than its weakest link. Since the microphone is a transducer to which the variables of acoustics, placement, and human subjectivity apply, its choice and placement greatly affect the final, reproduced quality of the signal.

2 The Microphone Transducer

There are various methods by which a mike is able to convert acoustical energy to corresponding electrical voltages. Each of these operating systems, known as dynamic, condenser, ceramic, and carbon transducers, presents a unique sonic and electrical characteristic.

The Dynamic Microphone

In principle, the dynamic microphone operates by electromagnetic induction to generate an output signal voltage. When an electrically conductive metal is made to cut across the flux lines of a magnetic field, a current of specific magnitude and direction is generated within that metal. This is shown in equation 2-1.

$$e = Blv \tag{2-1}$$

where,

e = induced potential (volts)
B = magnetic flux density (teslas)
l = conductor length (meters)
v = velocity of diaphragm displacement (meters/second)

The dynamic microphone may be subdivided into two types: the moving-coil and the ribbon microphone.

The Moving-Coil Microphone

The moving-coil microphone (Fig. 2-1) generally consists of a Mylar diaphragm of roughly 0.35 mil thickness. To this, a finely wrapped core of wire, called a "voice coil," is attached so that it is suspended precisely within a high-flux magnetic field.

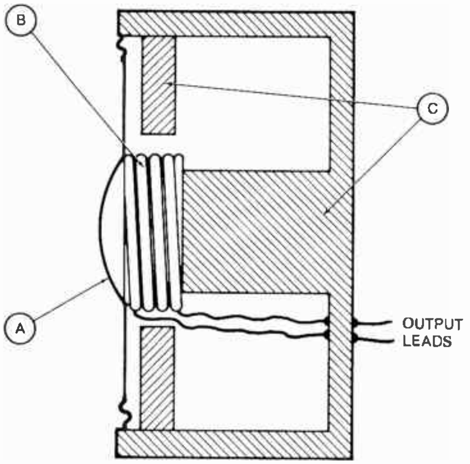
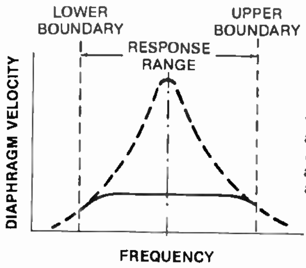


Fig. 2-1. Moving-coil microphone.

Whenever a sound-pressure wave falls on the diaphragm (A), the attached voice coil (B) is displaced in proportion to the amplitude and frequency of the wave, causing the coil to cut across the lines of magnetic flux supplied by the permanent magnet (C). As it does so, an output voltage (of a specific magnitude and direction) is generated across the voice-coil leads. The electrical output signal is analogous to the acoustical input signal, in accordance with equation 2-1.

In order for the moving-coil microphone to display a relatively uniform or flat frequency response, the velocity of the voice coil must be uniform with the frequency. To achieve this, a felt resistive damping ring is often used to control voice-coil motion at these frequencies. The motion of the voice coil may thus be said to be *resistance controlled* (Fig. 2-2). Without proper acoustical or mechanical damping in the design of a dynamic mike, the coil's velocity will be nonuniform and will peak either near 150 Hz or 1 kilohertz (kHz), depending on design.



For constant sound pressure, the particle *velocity* of air is equal at all frequencies. *Dynamic pressure microphones* are, therefore, mid-band tuned and are resistance controlled.

Fig. 2-2. Frequency responses for typical dynamic microphone.

All moving-coil microphones are designed to have their diaphragm's main resonance placed within the middle of the audio response range. This resonance is not noticed at the output because the diaphragm is forced by additional resonators in the form of slots, holes, air passages, and trapped air spaces. These correct the response-curve irregularities and extend the frequency response toward the upper and lower ends of the audio spectrum (Fig. 2-3).

The moving-coil microphone may be designed to be either a *pressure* (omnidirectional) or *pressure-gradient* (directional) pickup device.

Figure 2-1 shows an example of a basic pressure system; the microphone is equally sensitive to sound waves coming from all directions. Since these waves are only collected at the front of the diaphragm, they are combined equally in the signal output.

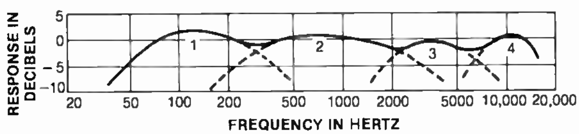


Fig. 2-3. Frequency response of a moving-coil microphone: (1) damped resonance of the vibrating system (2, 3, 4) resonance of additional air pockets.

Figure 2-4 shows the pressure-gradient pickup; its sensitivity varies with respect to the direction of the sound source. These variations in sensitivity may be plotted (with respect to the front/on-axis sensitivity) on a chart. Known as the *polar response* or *polar pattern* of a microphone, the chart can be used to evaluate the mike sensitivity with respect to direction and frequency. Polar patterns are covered in detail in Chapter 3.

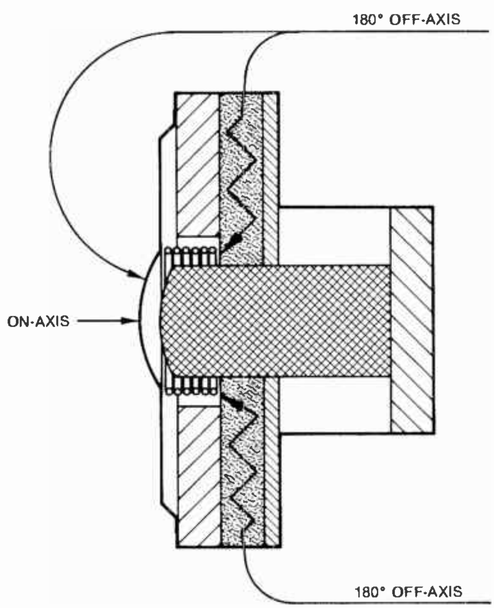


Fig. 2-4. Pressure-gradient microphone.

The Ribbon Microphone

The ribbon microphone (Fig. 2-5), like the moving-coil microphone, operates on the electromagnetic theory of induction. This transducer utilizes an extremely thin (.08" or 2 millimeter [mm]) aluminum ribbon as its diaphragm, which is corrugated transversely, or in certain cases longitudinally, along its length. The ribbon diaphragm is suspended within a strong field of magnetic flux and is exposed to the acoustic environment.

As sound pressure variations in the air displace the ribbon diaphragm in accordance with air particle velocity, the ribbon cuts across lines of flux supplied by the permanent magnet. In accordance with equation 2-1, this induces a voltage into the ribbon that is proportional in amplitude and frequency to the acoustic signal.

The ribbon mike is often called a "velocity" or pressure-gradient mike. This is because the motion of the ribbon results from the difference in pressure between its front and rear faces, which is a function of the particle velocity of sound in air. The force of this acting velocity is a function of signal wavelength and, in fact, rises with frequency at a rate of 6 decibels (dB)/octave. Because of its subsonic resonance frequency, however, the ribbon will exhibit a complementary 6 dB/octave reduction in high-frequency response (roll-off), which produces the net effect of a flat frequency response (Fig. 2-6).

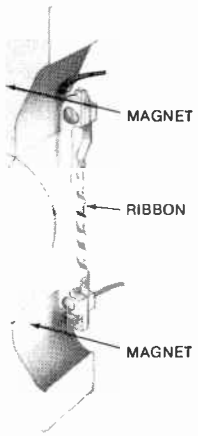


Fig. 2-5. Construction detail for a ribbon microphone. *(Courtesy of Shure Brothers, Inc.)*

The Ribbon Microphone's Most Common Polar Pattern: Bidirectional

Since the ribbon is exposed to sound waves from both the front and rear, it is equally sensitive to sounds emanating from both axial directions with sounds from the rear producing a voltage that is 180 degrees (180°) out of phase with an equivalent on-axis voltage signal (Fig. 2-7A). Thus, the polar response of a pure velocity pickup will exhibit a bidirectional or figure-eight pattern. Sound waves arriving 90° off-axis will produce an equal, but opposite, pressure at both the front and rear of the ribbon (Fig. 2-7B), giving a net result of no output signal.

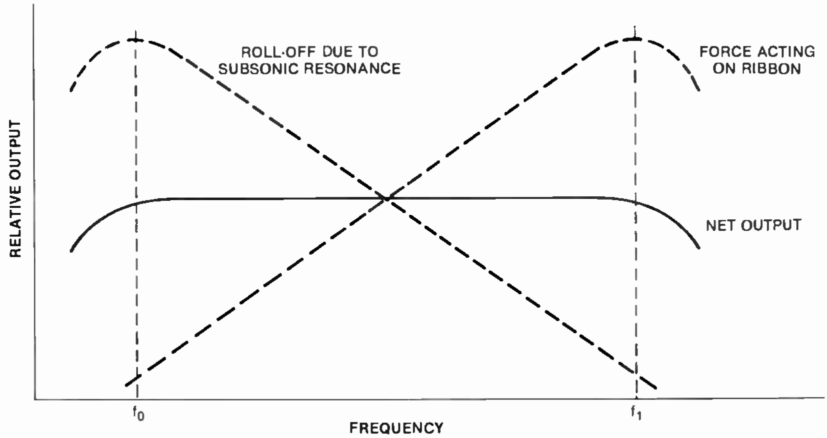
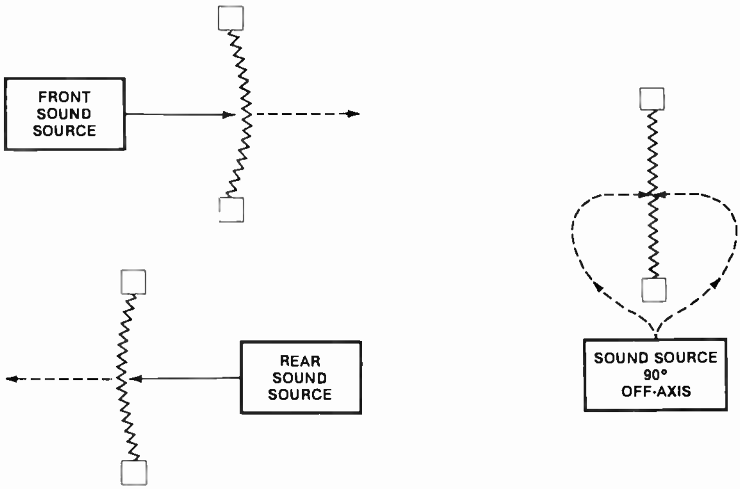


Fig. 2-6. Effect of an acoustic force on the ribbon diaphragm.



(A) Ribbon sensitive to sounds at front and rear.

(B) Sound waves from 90° off-axis.

Fig. 2-7. Sound sources on-axis and 90° off-axis at ribbon microphone.

The Printed Ribbon Microphone

One recent advance in ribbon technology has been the development of the *printed ribbon* (Fig. 2-8). In principle, the printed ribbon operates in precisely the same manner as the conventional ribbon microphone. The diaphragm, which is approximately .001″ (.025 mm) thick, is made of a polyester film upon which is printed a spiral aluminum ribbon. The magnetic structure is made of two ring magnets in front of the diaphragm and two in back. This creates a magnetic flux "wash," such that the displaced motion causes the ribbon to cut across the magnetic lines of force. This type of microphone is available in both omnidirectional and bidirectional polar patterns and exhibits a general frequency response of 70 Hz–15,000 Hz.

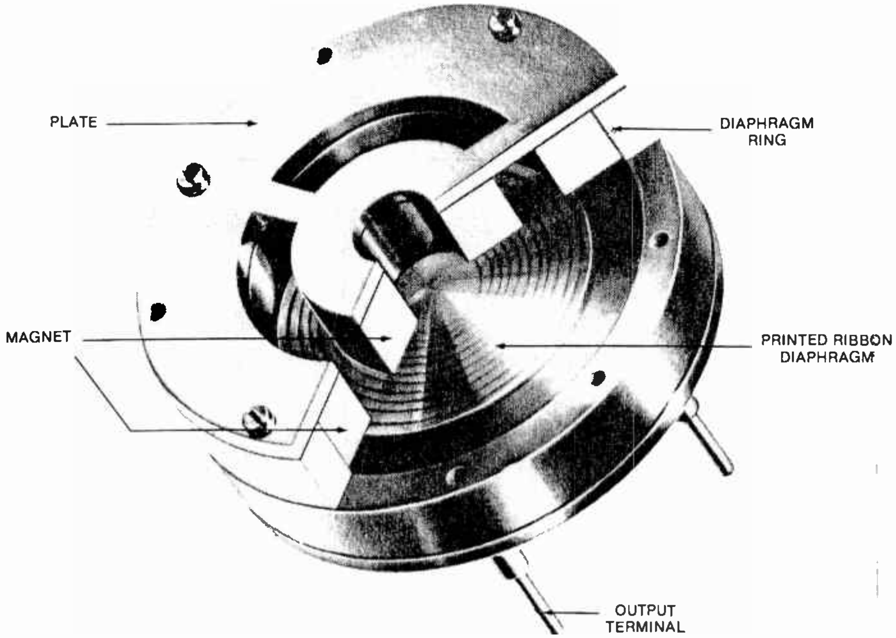


Fig. 2-8. Printed ribbon microphone. *(Courtesy of Fostex Corporation of America)*

In general, the ribbon microphone is less rugged than its dynamic counterpart. Care must be taken not to permanently deform or destroy the ribbon element. Although the newer double and printed ribbon designs offer a certain degree of ruggedness, older ribbon designs require that air currents be prevented from overloading the element. Large pressure bursts, such as blowing directly into the mike or close positioning to a high signal-level instrument, may result in permanent deformation or fracture of the element.

The Condenser Microphone

Unlike the dynamic microphone, which operates on the electromagnetic principle, the condenser mike is a purely electrical system that depends upon variations in internal capacitance for its operation (Fig. 2-9).

The condenser, or *capacitor*, microphone may be divided into two classes: externally polarized and the prepolarized electret-condenser. The

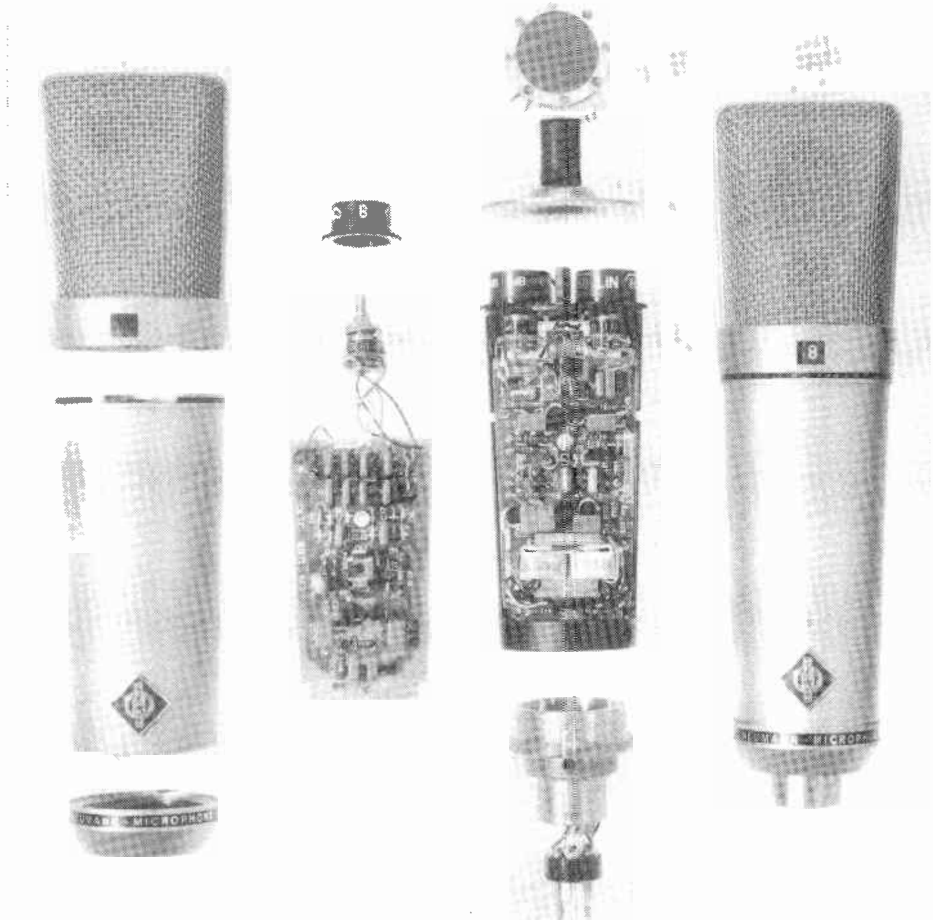


Fig. 2-9. Detail of the Neumann U-89 condenser microphone. *(Courtesy of Gotham Audio Corporation)*

function of the polarizing voltage, or its equivalent, is to translate the motion of the device's diaphragm into a linearly related audio output voltage. This voltage is amplified by a high-impedance field effect transistor (FET) or tube preamplifier, which must be located close to the microphone capsule.

The diaphragm of a condenser mike consists of a very thin metal-coated front plate, coated on one side with gold or nickel at a thickness of 50 nanometers. This diaphragm plate is spaced approximately 0.001″ (.025 mm) from a stationary backplate (Fig. 2-10). The capacitance between these two electrode plates varies as the freely suspended diaphragm is displaced with respect to the fixed back electrode by acoustic pressure variations in air.

A capacitor is, in effect, an electrical device that is capable of storing an electric charge. The amount of charge that is stored is determined by the value of capacitance and applied voltage, according to the formula in equation 2-2.

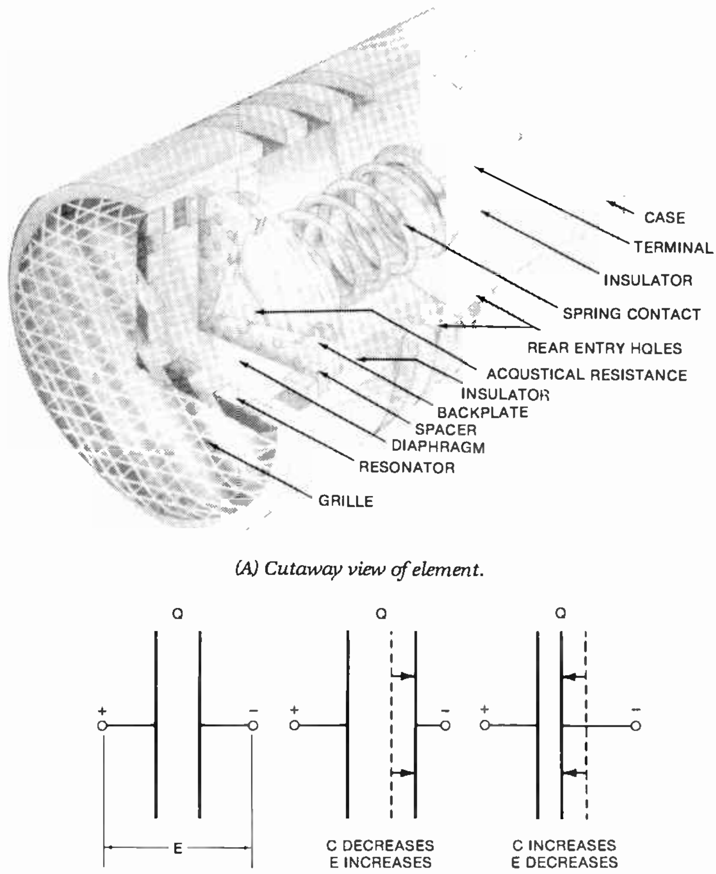
$$Q = CE \tag{2-2}$$

where,

Q is the charge, in coulombs
C is the capacitance, in farads
E is the voltage, in volts

The capacitance value of a capsule is determined by:

- the composition and surface area of the plates (which are fixed values)
- the dielectric or substance between the plates (which is air and is also fixed)
- the distance between the plates (which varies with sound pressure)

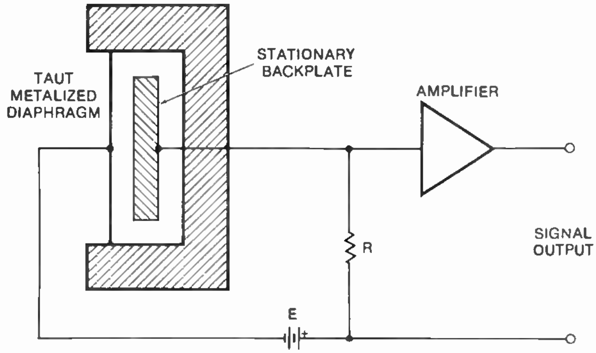


(A) Cutaway view of element.

(B) Output and potential relationships as a result of changing capacitance.

Fig. 2-10. Details of a condenser capsule. *(Courtesy of Shure Brothers, Inc.)*

Fig. 2-10 (cont.)



(C) Circuit diagram.

As the capacitance (C) changes with the impinging sound waves and as the charge (Q) is held constant, the output voltage (E) changes inversely with the varying capacitance between the diaphragm and backplate.

The proper operation and construction of a condenser microphone element is determined by several variables (equation 2-3), all of which must be accurately controlled.

$$E(oc) = \frac{EPA}{4DT} \tag{2-3}$$

where,

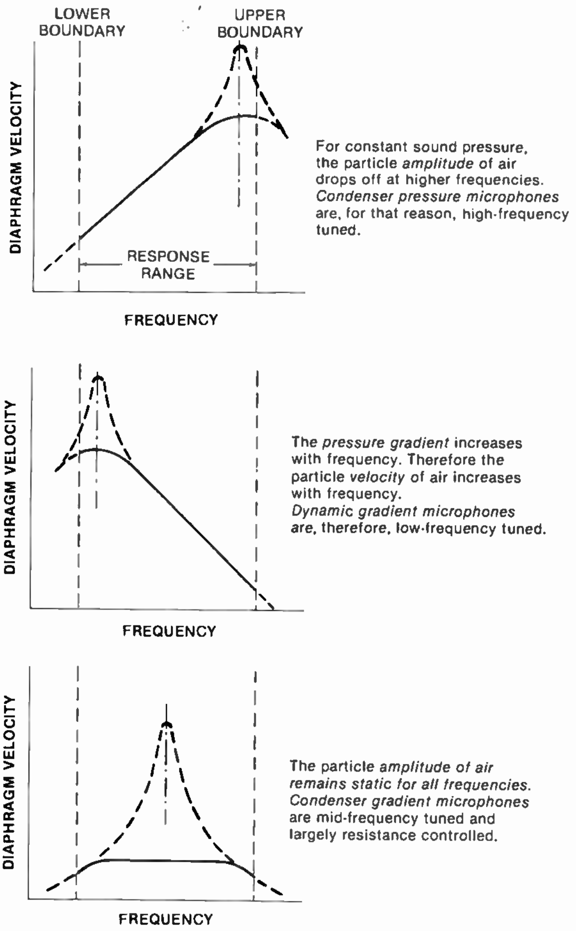
E(oc) = open circuit output voltage
E = polarizing voltage, across plates
P = applied acoustical pressure, in dyne/cm^2
A = radius of diaphragm, in cm
D = distance between charged plates, in cm
T = membrane tension, in dyne/cm^2

The most critical component in the condenser microphone is the capsule itself, which is most affected by four design considerations:

- *Tension* of the membrane (compliance), which determines the low-frequency characteristic. The greater the stiffness at which the diaphragm is held upon its support rings, the greater the drop-off at low frequencies. The low-end roll-off is measured in dB/octave and the drop-off point may be shifted by varying degrees of diaphragm tension.
- *Mass* determines the high-frequency response and transient response of a diaphragm's movement. As the mass of the diaphragm increases, the high-frequency limit decreases. The mass of the diaphragm depends on the thickness and composition of the membrane.

- *Diameter* of the head also determines the mass of the membrane, therefore affecting the overall frequency response. Thus, a small-diameter diaphragm tends to have a different response from that of a comparable large-diaphragm microphone.
- The *spacing* between the plates affects the capacitance of the circuit itself. Even more important, this spacing creates an air pocket that acts as a spring cushion to dampen the vibration of the membrane, thus decreasing the system's compliance.

The condenser element and its associated acoustic system are known as a *microphone capsule*. The diaphragm resonance of a condenser element is resistance controlled. The pressure-operated (omnidirectional) mike displays a "high-frequency tuned" resonance (Fig. 2-11A), while the pressure-gradient (directional) transducer resonance is placed at mid-to-upper-middle frequencies (Fig. 2-11B).



(A) Pressure mike.

(B) Pressure-gradient mike.

Fig. 2-11. The resonance and transmission range for the condenser microphone.

The plates of a condenser element are connected to opposite sides of a dc power supply, which provides polarizing voltage to a capsule (Fig. 2-12). Electrons are drawn from the diaphragm, which is connected to the positive side of the power supply, and are forced through a high-value resistor onto the backplate, which is connected to the negative side of the supply. This continues until the charge on the capsule (i.e., the difference in potential between the positive and negative plates) is equal to the capacitance of the capsule multiplied by the polarizing voltage. The high value of this series resistor, in conjunction with the capacitance of the plates, produces a circuit time constant that is longer than a cycle of an audio frequency. This constant represents the time needed for a capacitor to charge or discharge. Since the resistor prevents the capacitive charge from varying with the rapid changes in capacitance (caused by variations in applied sound pressure), the voltage across the capacitor changes in accordance with equation 2-2.

As the voltage across the series resistor is low, and the resistive component is extremely high (10^9 ohms [Ω]), the signal must be amplified and the impedance dropped by a preamplifier within the body of the microphone. This preamplifier must be powered by an external power source, either a phantom power supply or, in the case of the electret condenser, a battery supply to drive the FET preamp directly.

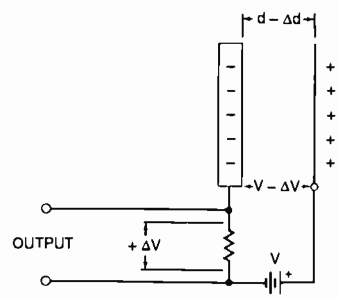


Fig. 2-12. As a sound wave decreases a condenser element's spacing by Δd, the capacitance increases by ΔC. Conversely, the voltage across the plates falls by ΔV.

The Electret-Condenser Microphone

The electret-condenser capsule operates like the externally polarized condenser, except that the required polarizing charge is stored permanently within the diaphragm or backplate. For this reason, no form of external powering is required in order to charge the diaphragm. The high output impedance of the capsule still requires an impedance-changing amplifier, thus an internal battery supply or phantom power supply is needed.

The simplest type of electret microphone is the charged-diaphragm type, using an electret foil diaphragm as a compromise between good electret and good mechanical properties.

There are several methods of making an electret; exact details are kept as trade secrets by the manufacturers. Typically, a plastic film, such as Mylar, of approximately 0.0002″ (.005 mm) is spattered with a conductive metal such as gold or nickel. The film is then heated and charged with a high dc potential, with the electret-forming electrode facing the nonconductive side of the film (Fig. 2-13).

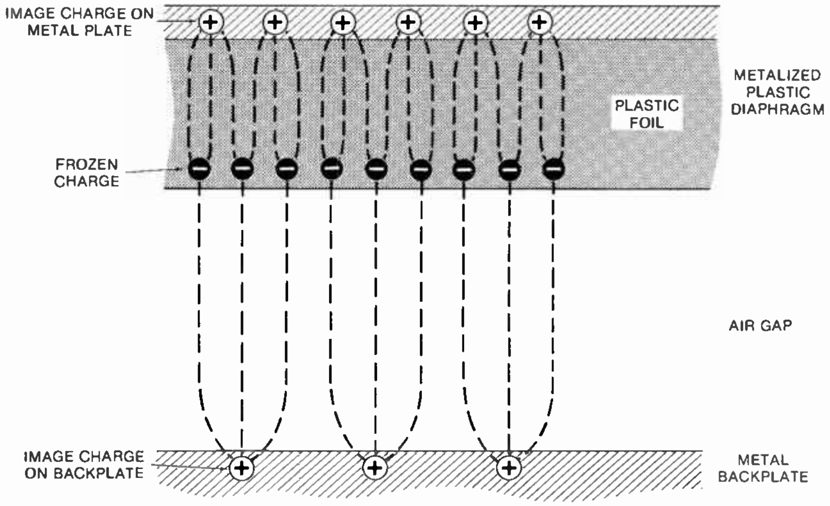


Fig. 2-13. Position of charges when electret is an integral part of the diaphragm.

A well-designed electret capsule will retain its charge for a period of ten years; it is predicted that it would take a period of from thirty to a hundred years before the capsule sensitivity would drop by 3 dB.

The charged-diaphragm electret generally cannot withstand the tension required to obtain the high resonance frequency that is commonly found in the externally polarized condenser mike. One solution to this has been to reduce the tension and support the diaphragm at many points with a grooved backplate. While this capsule system does not possess the extended high-frequency response of the externally polarized condenser, its advantage is that it can be fabricated easily and cheaply.

An improved version of the electret condenser is the *back electret* or *charged backplate electret* design (Fig. 2-14). The capsule diaphragm is coated on one or both sides with a film of gold or another metal. The charged electret material is placed upon the backplate, which also contains a conductive surface in order to form the system's "high" output terminal. The electret element, which is constructed of a fluoric film such as Teflon, is charged like the charged-diaphragm electret. However, since the diaphragm does not contain the electrostatic charge, the materials and thickness may be chosen for optimum sensitivity and stability. This capsule system is essentially identical to the externally polarized condenser microphone, except that the backplate holds a permanently charged potential.

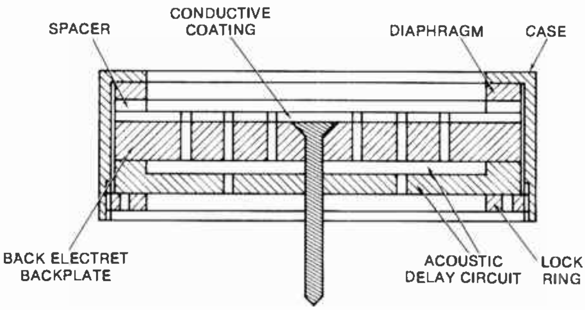


Fig. 2-14. Back-electret capsule.

The RF Condenser Microphone

Before low-noise FET preamplifiers were available, when transistor technology held desirable benefits for condenser microphones, it was necessary to utilize a radio-frequency (rf) condenser microphone circuit to obtain high stability. In such a circuit, the microphone capsule operates as an "active transducer," controlling the frequency or phase of an rf oscillator. Alternatively, the capsule may vary a tuned rf circuit in a rhythm sympathetic to the audio frequency with only the demodulated audio frequency voltage appearing at the output of the microphone. This type of microphone is no longer in popular use.

The rf oscillator is crystal controlled and operates at a fixed frequency of about 8 megahertz (MHz), which gives the capsule capacitance a relatively low impedance at radio frequencies. For example, a capsule capacitance of 50 picoFarads (pF) at 10 MHz represents only about 300 Ω.

The demodulator stage resembles that of the traditional ratio detector. When the capsule is stimulated by sound waves, it shifts the phase of the rf current flowing in the demodulator circuit according to the sound pressure variations. The detector diodes receive an unequal rf voltage and modulation is produced at the output. Noise that is introduced by oscillator frequency fluctuations is held to a minimum by the circuit's locked crystal. The rf condenser system provides about the same signal-to-noise ratio as a high-quality dc polarized condenser system.

Types of Condenser Capsule Acoustical Construction

The condenser microphone capsule falls into one of three categories based on its acoustical characteristics: pressure, pressure-gradient, or Braunmuhl and Weber.

The pressure transducer (Fig. 2-15) is a capsule system in which only one side of the diaphragm is exposed to the sound field. The diaphragm is thus sensitive to pressure variations upon its surface regardless of the sound source's direction, and the resultant pickup pattern is omnidirectional.

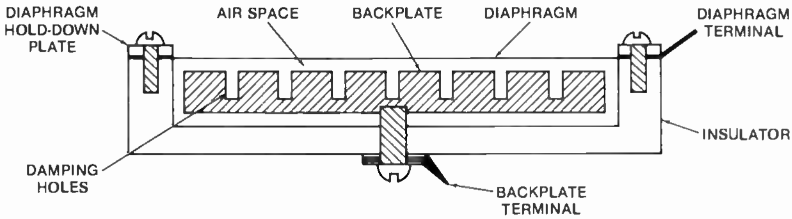


Fig. 2-15. Single-membrane omnidirectional condenser capsule.

Perforations are made within the backplate in order to damp out various mechanical resonances inherent in the diaphragm structure. To ensure acoustic isolation, the perforations do not go completely through the backplate.

The pressure-gradient capsule (Fig. 2-16) is similar to the pressure transducer, except that a difference in pressure between the front and back of the diaphragm is created by admitting acoustic energy to the rear of the diaphragm. This is accomplished by allowing a few of the damping perforations to extend entirely through the backplate's surface.

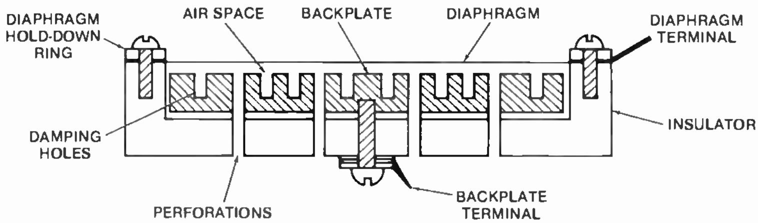


Fig. 2-16. Single-membrane cardioid condenser capsule showing damping holes and rear backplate perforations.

These extended perforations are positioned such that any sound waves originating from the rear axis impinge upon the rear of the diaphragm 180° out of phase, with respect to the front of the diaphragm, creating a cardioid directional pattern. The degree of directionality is determined by the size, length, and positioning of these perforations in the backplate.

The third capsule system, known as the *Braunmuhl and Weber configuration*, consists of dual membranes that are mounted about a central backplate (Fig. 2-17).

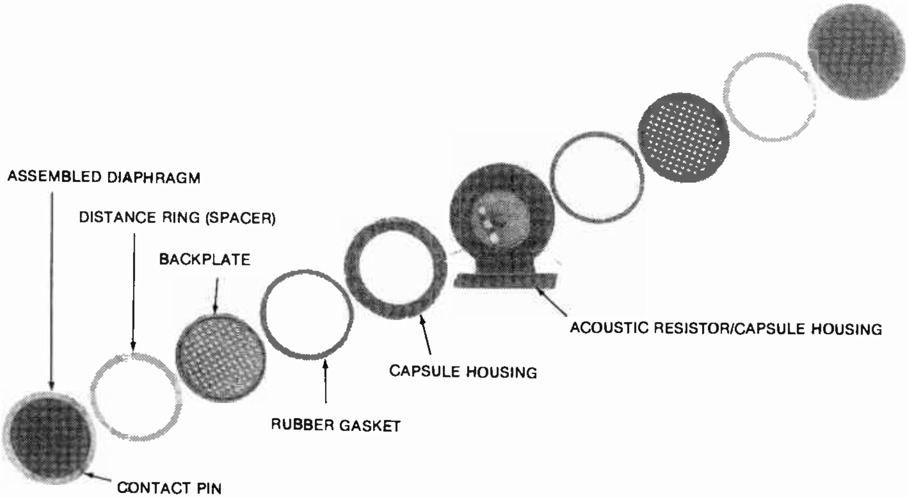
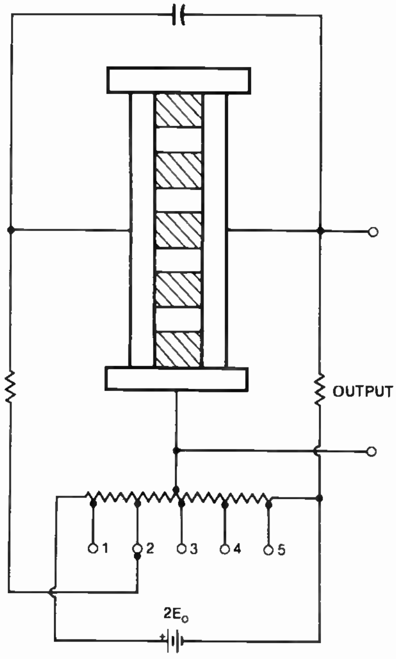
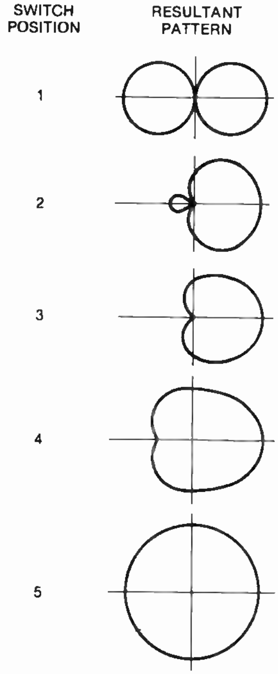


Fig. 2-17. Exploded view of a dual-diaphragm (Braunmuhl-Weber) type condenser pickup element. *(Courtesy of AKG Acoustics, Inc.)*

Basically, this system consists of the equivalent of two back-to-back cardioid capsules (Fig. 2-18A), with each diaphragm acting as part of a pressure-gradient capsule. The opposing diaphragms, in combination with the phase-shift port holes, combine to create an effective 180° phase shift relative to the on-axis angle, producing a dual-cardioid system within a single housing. This gives the user a choice of several polar patterns (Fig. 2-18B),



(A) Capsule detail.



(B) Polar patterns.

Fig. 2-18. Braunmuhl-Weber multidirectional microphone.

depending upon the internal signal ratio and polarity between the two elements. With this arrangement, the front diaphragm is given a maximum polarizing voltage (Eo) at all times and maintains a cardioid response with maximum sensitivity. The polarizing voltage on the rear diaphragm may be varied from 0 to +Eo. By mixing these respective voltage and/or polarity ratios, multiple directional responses can be obtained. We'll look further at combined pressure and pressure-gradient microphones in the Directional Sensitivity section of Chapter 3.

The Ceramic Microphone

Certain crystalline elements, when deformed along a specific axis, generate an electrical potential between the opposing sides of their structure. The ceramic microphone, also known as the *piezoelectric* microphone (Fig. 2-19), operates upon such a principle. The piezoelectric element (from the Greek *peizein*, meaning "to press") is mounted in a cantilever fashion and is actuated by the diaphragm through the use of a coupled drive pin. The generated output voltage is proportional to the displacement of the diaphragm in the frequency range below system resonance.

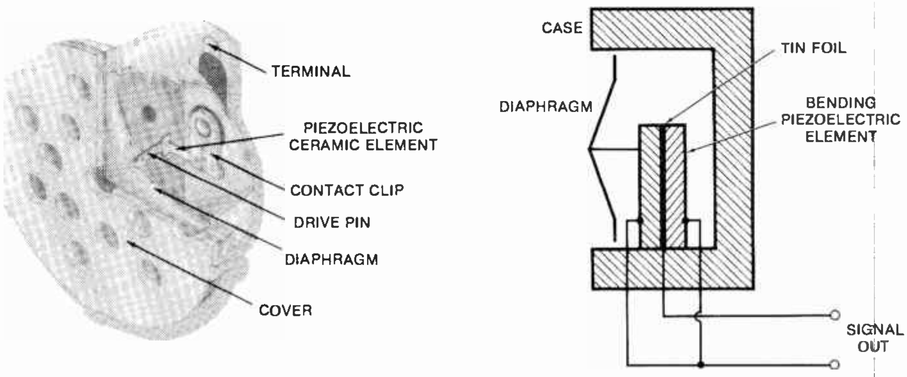


Fig. 2-19. Piezoelectric microphone. *(Courtesy of Shure Brothers, Inc.)*

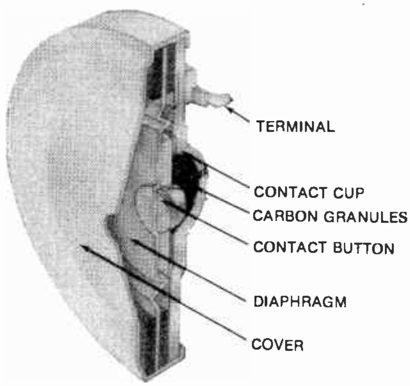
(A) Ceramic microphone. *(B) Circuit diagram.*

Prior to 1960, Rochelle salt crystals were used as the generating element, but these proved to be unstable with changes in humidity or heat. Since this time, they have been replaced by newer ceramic materials that are more resistant to environmental extremes.

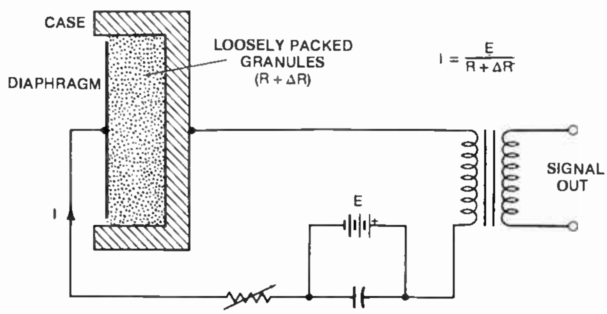
The major advantage of the piezoelectric or ceramic microphone is that the output voltage is sufficient to drive a high-impedance input stage directly. The ceramic microphone was popularly used with tube-type home tape recorders and communications equipment. Since the advent of the higher quality dynamic and electret-condenser elements, however, the piezoelectric microphone is seldom used.

The Carbon Microphone

The carbon microphone (Fig. 2-20) is a pressure device that depends upon variations in resistance between carbon contacts for its operation.



(A) Detail of diaphragm.



(B) Circuit diagram.

Fig. 2-20. Carbon microphone. *(Courtesy of Shure Brothers, Inc.)*

The carbon mike consists of a contact cup filled with carbon granules, which are generally made of anthracite coal. These granules are placed in contact with an electrically charged aluminum diaphragm by a mounted contact button. The acoustic displacement of the diaphragm causes a variation in mechanical pressure that is applied to the granules. The result is an analogous variation in electrical resistance between the diaphragm and the opposing contact cup.

This microphone is almost universally used in telephone communications, as its high sensitivity eliminates the need for audio amplification within a telephone set. Its restricted frequency range, distortion, and carbon noise limit its application in fields other than voice communications. For improved performance, as with the piezoelectric mike, the carbon telephone microphone is now being replaced with the dynamic or electret system.

3 Microphone Characteristics

Directional Sensitivity/Polar Response

The directional pattern of a microphone may be broadly classified into two types:

1. the omnidirectional polar response
2. the directional polar response

The Omnidirectional Microphone

The omnidirectional microphone (Fig. 3-1) is basically a pressure-operated device, since it responds to the acoustic sound pressure, which is nondirectional. In theory, the diaphragm reacts equally to all sound-pressure fluctuations that occur at its surface, regardless of location of source.

Because of its all-around pickup, the omnidirectional mike may be selected for use where a maximum ratio of reverberant-to-direct sound is required. Normally, its frequency-response curve is only valid for sound arriving on-axis (Fig. 3-2A) as a pressure microphone tends to become directional at high frequencies. Consequently, it needs to be aimed at the sound source.

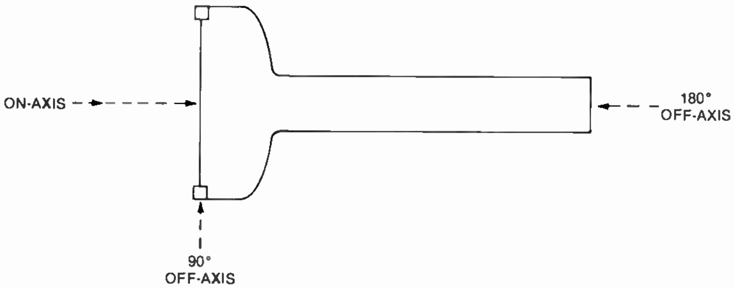
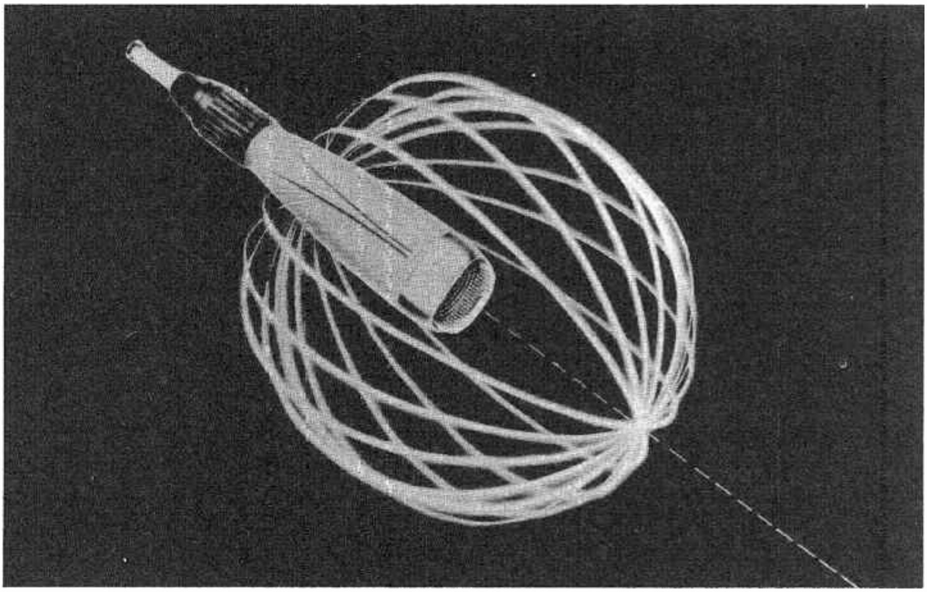
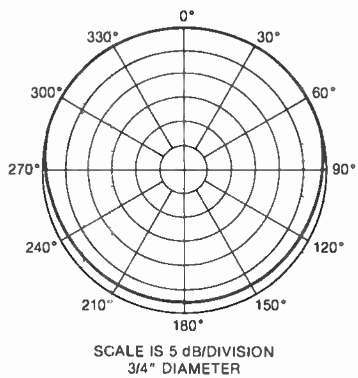


Fig. 3-1. Detail of omnidirectional capsule showing on-axis pressure build-up, given any angle of sound incidence.



(A) Field of incidence.



(B) Typical response chart.

Fig. 3-2. The omnidirectional microphone. *(Courtesy of Sennheiser Electronic Corporation [N.Y.])*

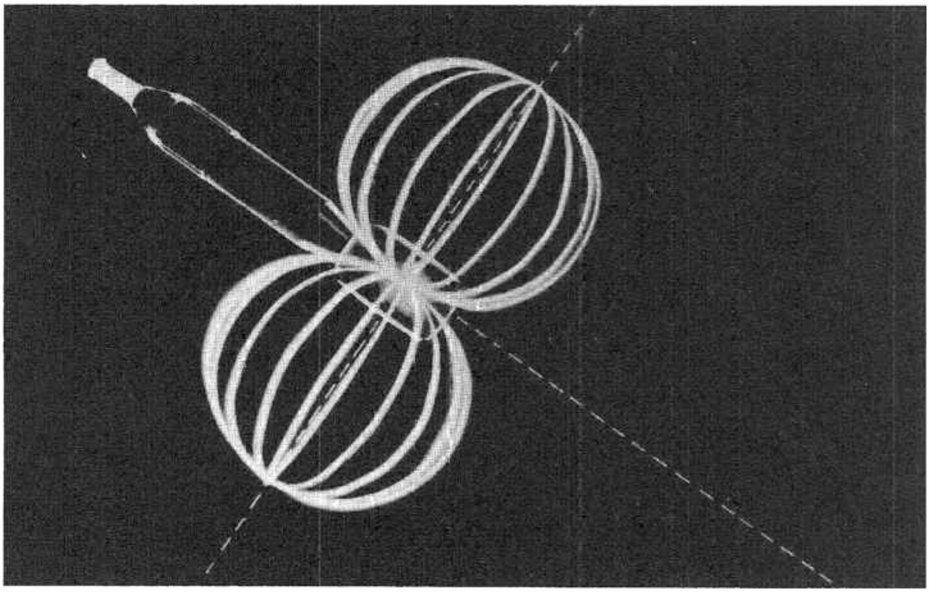
The Directional Microphone

A microphone that displays directional properties is a pressure-gradient device. This means that the system is responsive to differences in pressure between the two faces of the diaphragm. Other microphone designs may depend upon reflectors to achieve their directivity.

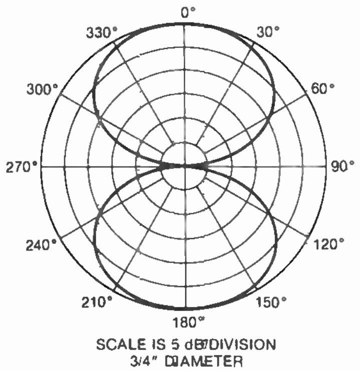
Directional microphones discriminate against excessive reverberation or background noise. They can enable a microphone to be used at a greater distance from a sound source, and they can give a sense of sound location when used with some stereophonic sound recording systems.

A purely pressure-gradient microphone will exhibit a bidirectional (cosine or figure-eight) polar pattern, as shown in Figure 3-3. The electrical response of such a system results from the velocity of air particles, which is a vector quantity and, as such, possesses both magnitude and direction.

Fig. 3-3. Bidirectional microphone. *(Courtesy of Sennheiser Electronic Corporation [N.Y.])*



(A) Field of incidence.



(B) Typical response chart.

The electrical response of the combination pressure and pressure-gradient microphone is also proportional to particle velocity and, as such, is directional in nature. Figure 3-4 illustrates graphically how the outputs of bidirectional and omnidirectional microphones may be mixed to obtain other directional patterns. In effect, there are an infinite number of directional patterns to be obtained from this combination; however, the most widely known patterns resulting from these combinations are the cardioid, supercardioid, and hypercardioid polar-response curves (Fig. 3-5).

Many single-transducer microphones of the moving-coil, ribbon, and condenser types offer a variety of patterns for best pickup of a specific sound source. The modern dynamic microphone is designed to exhibit a specific, nonadjustable polar-response pattern that is chosen to best match the task at hand. The directivity of certain outdated ribbon systems are controlled through the use of an acoustically variable delay system, allowing the pattern to be varied from bidirectional to omnidirectional or cardioid.

The directivity response may be designed into a condenser microphone in any of three ways:

1. It may be designed to exhibit only one polar response pattern, as with the dynamic mike.
2. A multiple-capsule system may be employed, in which any number of task-specific pickup capsules may be fitted to a matching preamplifier module.
3. The directivity of a dual-membrane condenser system may be variably controlled through the use of an internal electrical arrangement between the diaphragms, thus allowing a single microphone to display multiple polar patterns.

To better understand the directional properties of a microphone system, let's explore the pressure and pressure-gradient operating principles.

The Directional Dynamic Microphone

The directional characteristics of a dynamic microphone are achieved mechanically, through the use of a rear port that serves as an acoustic delay or *labyrinth*. A labyrinth consists, in one form, as a folded pipe or path that is longer than the actual acoustic path, thus forming an acoustic delay. The pipe is often lightly packed with felt or ozite in order to operate effectively over the entire frequency range.

In Figure 3-6A, a dynamic microphone of cardioid response is shown receiving a sound signal at 0° (on-axis). The diaphragm receives two signals: the incident signal and the acoustically delayed rear signal. Since there is an instantaneous difference in pressure between front and rear, an output signal is produced. Figure 3-6B shows a sound source 180° off-axis. Here, the off-axis signal, which is delayed by the rear labyrinth, is made to be acoustically 180° out of phase with the sound arriving at the diaphragm's front face. Therefore, these signals acoustically cancel, resulting in almost no signal output. The attenuation of such an off-axis signal, with respect to an equal on-axis signal, is known as the *front-to-back discrimination* of a microphone and is rated in decibels.

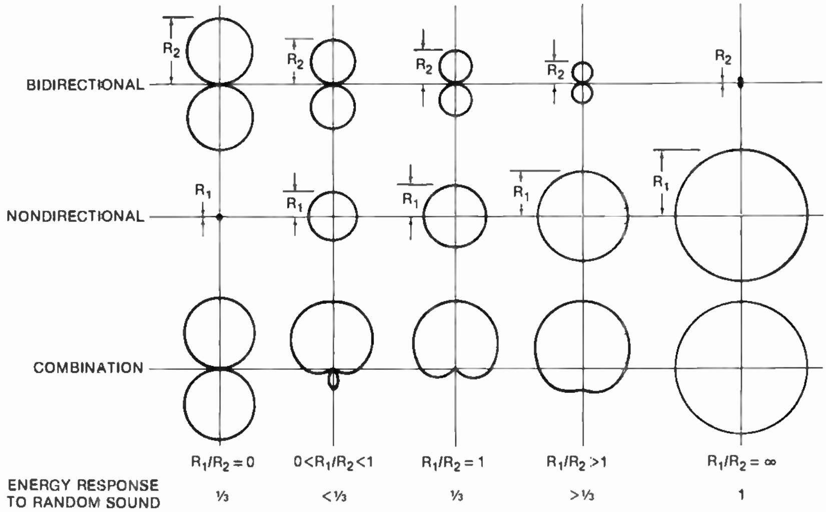
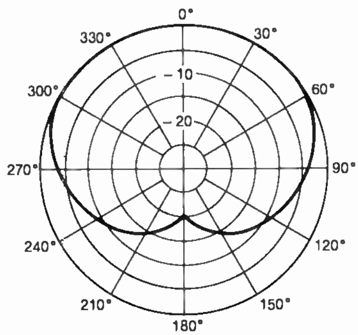
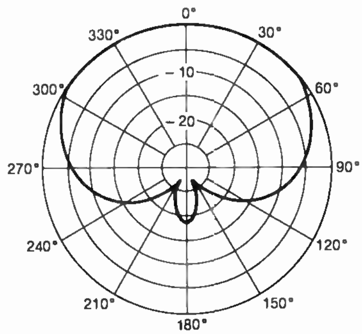


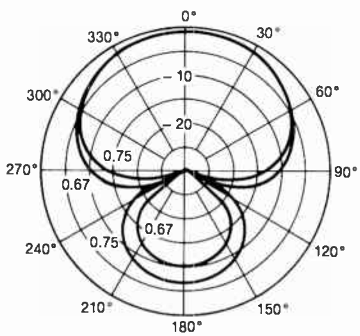
Fig. 3-4. Directional diagrams of various bidirectional and nondirectional pickup patterns and their energy response to random sounds.



(A) Cardioid.



(B) Supercardioid.



(C) Hypercardioid.

Fig. 3-5. Various cardioid-type polar-response charts. Scale is 5 dB/division.

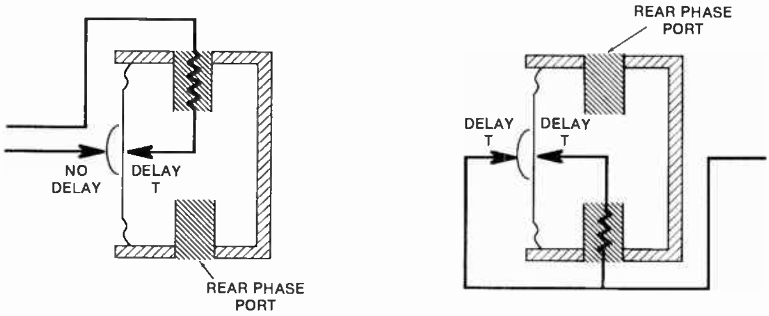
The Directional Condenser Microphone

The condenser capsule can be made directional in one of two ways:

1. by using an acoustic labyrinth at the rear port
2. by using a dual-membrane transducer, whose directional patterns can be varied electrically

In the cardioid single-diaphragm condenser capsule (Fig. 3-7A), a pressure gradient (a difference in pressure between the front and rear of the

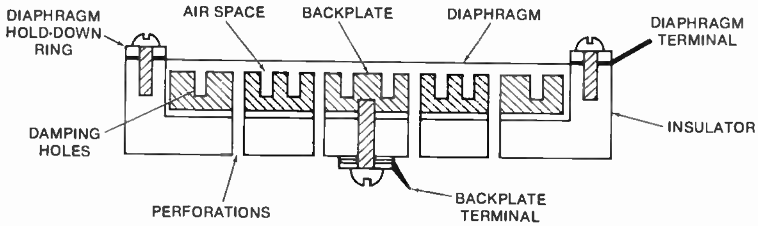
Fig. 3-6. Directional response of pressure-gradient mike.



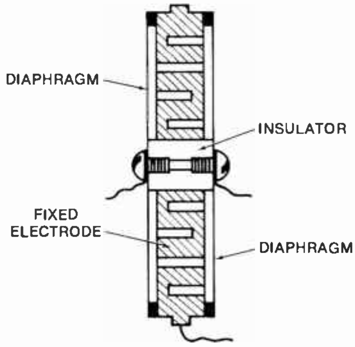
(A) Front signals arriving at rear of diaphragm are phase shifted, relative to signals at front, producing an output.

(B) Rear signals arriving at front and rear of diaphragm simultaneously, cancel each other, resulting in reduced output.

Fig. 3-7. Directional condenser microphone.



(A) Single-membrane cardioid condenser capsule.



(B) Dual-membrane multidirectional condenser capsule.

diaphragm) is created by admitting some acoustic energy to the rear of the diaphragm. This is accomplished by drilling perforations completely through the backplate beside the damping holes. These perforations are designed as an acoustic labyrinth such that any sound wave that impinges directly on the rear of the diaphragm will be 180° out of phase with respect to sound waves at the front of the diaphragm plate.

The second type of directional condenser mike (Fig. 3-7B), as described in 1935 by Braunmuhl and Weber, is a dual-membrane capsule system housed around a centrally mounted backplate. Within the Braunmuhl and Weber system, two cardioid capsules are placed back-to-back within a single housing. This arrangement gives the user a choice of several directional patterns, depending upon the electrical configuration between the two capsules (Fig. 3-8).

When the two capsules are configured electrically in phase, the resultant pattern is omnidirectional. When either capsule is used alone, the pattern is cardioid. When both capsules are connected electrically out of phase, the pattern is bidirectional. Certain condenser microphones can vary the capsule

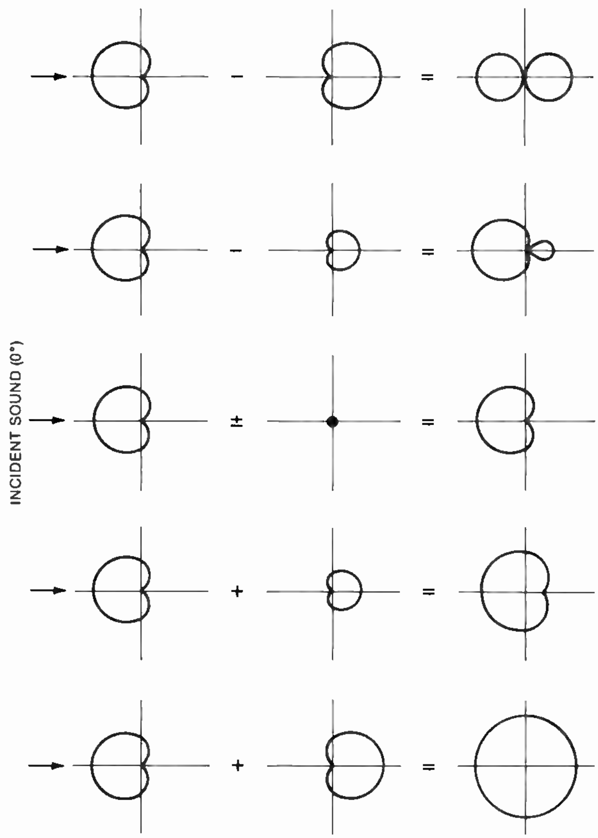


Fig. 3-8. Variety of polar patterns that may result from combining the output of two cardioids back to back.

polarizing voltages, making a continuous mixture possible between pressure and pressure-gradient polar-response characteristics.

Random Energy Efficiency

A directional microphone substantially reduces the random (off-axis) signal level relative to the desired (on-axis) signal. The measurement of such a ratio is known as the *random energy efficiency (REE)* of a microphone. This mathematical concept is useful in sound-reinforcement applications where system feedback is a potential problem. The REE indicates how well a microphone rejects reverberant sound and feedback. Figure 3-9 shows the characteristics of various polar patterns, including REE.

REE also predicts the direct-to-reverberant ratio of a microphone relative to that of an omnidirectional microphone at the same distance. As may be seen in Figure 3-10, a directional microphone may be placed further from the

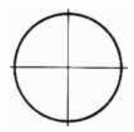
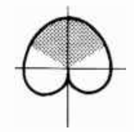
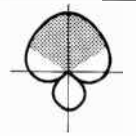
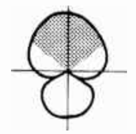
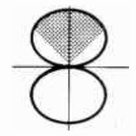
CHARACTERISTIC	OMNI DIRECTIONAL	CARDIOID	SUPERCARDIOID	HYPERCARDIOID	BIDIRECTIONAL
Polar response pattern					
Polar equation	1	$0.5 + 0.5 \cos\theta$	$0.375 + 0.625 \cos\theta$	$0.25 + 0.75 \cos\theta$	$\cos\theta$
Pickup ARC 3 dB down	—	131°	115°	105°	90°
Pickup ARC 6 dB down	—	180°	156°	141°	120°
Relative output at 90° in dB	0	−6	−8.5	−12	−∞
Relative output at 180° in dB	0	−∞	−12	−6	0
Angle at which output = 0	—	180°	127°	110°	90°
Random energy efficiency (REE)	1 0 dB	0.333 −4.8 dB	0.268 −5.7 dB	0.25 −6.0 dB	0.33 −4.8 dB
Distance factor (DF)	1	1.7	1.9	2	1.7

Fig. 3-9. Characteristics of various polar patterns.

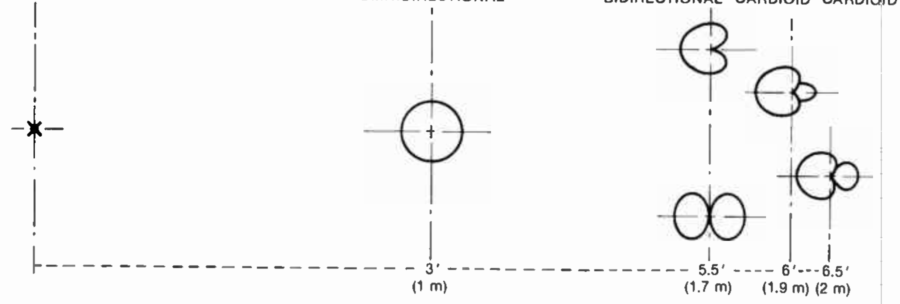


Fig. 3-10. Distance factors of omnidirectional and directional microphones.

sound source than an omnidirectional microphone while maintaining an equivalent direct-to-reverberant ratio and/or feedback threshold. For example, a cardioid microphone can be placed at a distance 1.7 times that of an equivalent omnidirectional microphone and maintain the same ratio of direct-to-reverberant pickup.

Multiple-Capsule Condenser Microphone System

The comparatively new *multiple-capsule condenser microphone system* (Figs. 3-11 and 3-12) has many applications in the fields of recording, film, video, and theater. The system utilizes a modular microphone preamplifier housing, along with a set of interchangeable microphone capsules and their associated accessories for multiple applications.

The multiple-capsule microphone system was developed for two specific reasons:

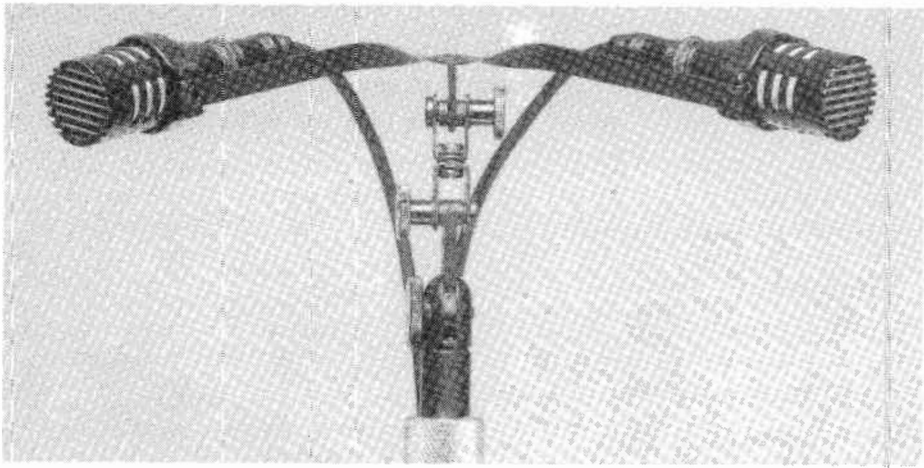
- The capsules can be interchanged, in a modulator fashion, to best suit the required polar- and frequency-response application.
- The capsules can be mounted remotely from the preamplifier housing by an angled extension tube or via a flexible cable connection. This allows the capsule to be mounted or hung off-camera or in difficult-to-reach locations.

Fig. 3-11. AKG C-460-B multiple-capsule system. *(Courtesy of AKG Acoustics, Inc.)*

(A) C-460-B "ULS" combo.

Fig. 3-11 (cont.)

(B) CK-2X and CK-1X capsules.



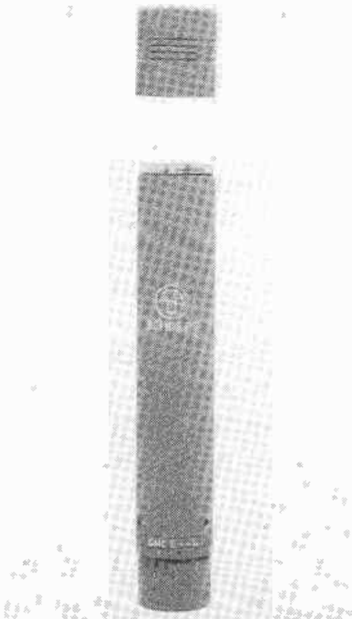
(C) H-52 stereo suspension in O.R.T.F. configuration.

The Ultradirectional Microphone

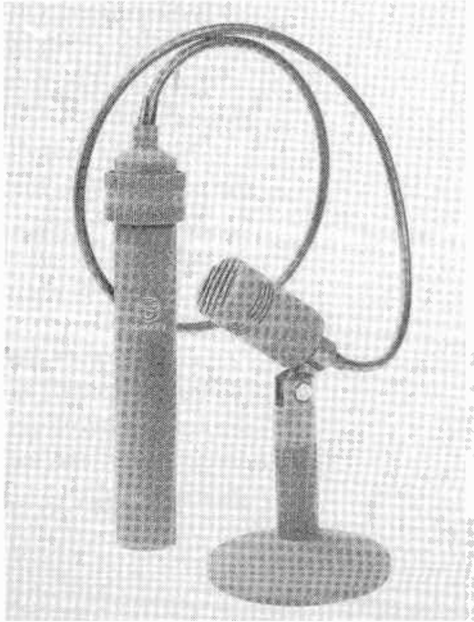
The *ultradirectional* or *shotgun* microphone is used in applications requiring a high degree of directionality. This allows the microphone to be placed at a greater working distance, while rejecting extraneous off-axis sounds that would otherwise be problematic. Additionally, the sensitivity of this pickup system is raised by 6 dB over the entire frequency range.

Located in front of the microphone diaphragm is a tube containing many slits or openings cut into its sides, perpendicular to the axis (Fig. 3-13). Sound impinging at an oblique angle of incidence changes its direction of propagation after entering the openings. The sound component at any one opening is out of phase with the sound components entering at any other openings and

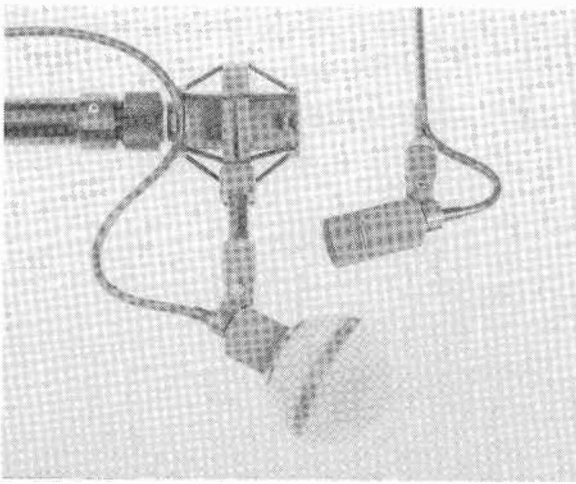
Fig. 3-12.
Schoeps "Colette" condenser microphone series.
(Courtesy of Schalltechnik Dr.-Ing Schoeps GmbH)



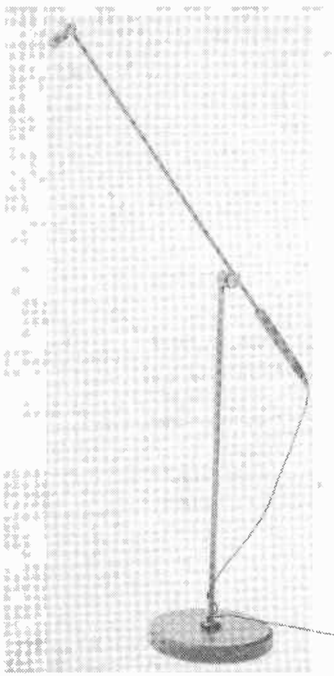
(A) CMC 5 preamplifier.



(B) With active KC 5 cable.



(C) With ACA elastic boom suspension and HC cable hanger.



(D) With RC 700 active tubes.

so is attenuated. Only those sound waves travelling parallel to the tube (arriving on-axis) will have all sound-pressure components in phase and not be attenuated. The result of this effect is a lobe-shaped directional characteristic (Fig. 3-14).

For the polar diagram to be of similar shape at all frequencies within the response range, the effective tube length must grow shorter with increasingly higher frequencies. For this reason, the side slits are covered with a layer of acoustically semitransparent material, arranged so that the acoustic impedance increases toward the end of the tube. This also provides damping of tube resonances.

At the lower frequency range (below 500 Hz), the tube length is not great when compared to the lower wavelengths, so the polar diagram becomes broader. Therefore, in order to maintain a degree of directionality at lower frequencies, the capsule is designed to operate with a hypercardioid polar characteristic.

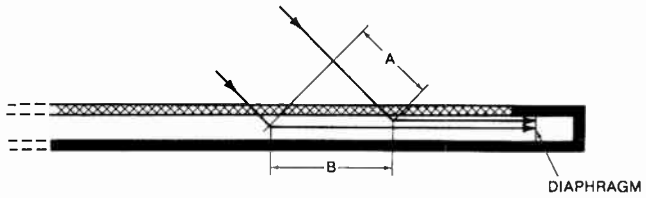


Fig. 3-13. Operating principle of single-tube shotgun directional microphone.

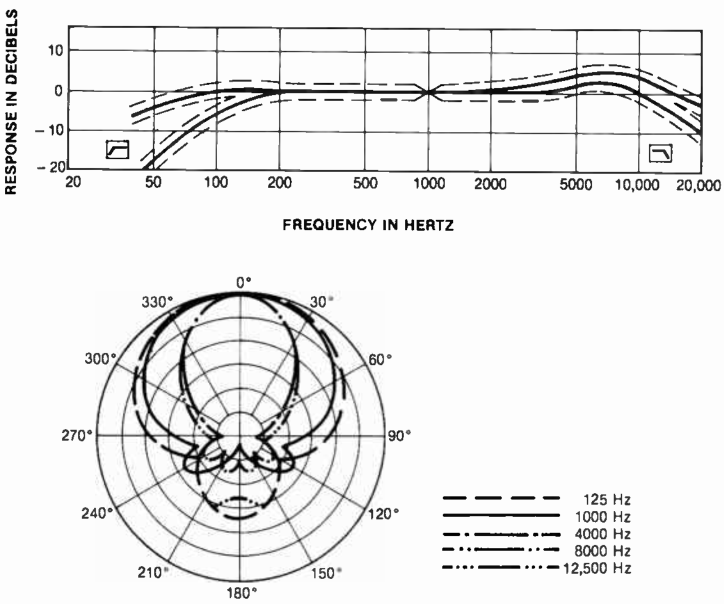


Fig. 3-14. Typical frequency and polar response of an ultradirectional microphone. *(Courtesy of Gotham Audio Corporation)*

The Parabolic Microphone

The *parabolic* or *dish* microphone functions as a highly directional unit that operates most effectively within the middle- and high-frequency range. Often used in sports, surveillance, and the recording of nature, it operates by placing an omnidirectional microphone at the focal point of a parabola (Fig. 3-15A). The parabola, which is often constructed of clear plexiglass, serves to collect on-axis audio signals at this focal point, while off-axis signals are rejected by not being focused (Fig. 3-15B). The larger the dish, the better the directionality at low frequencies.

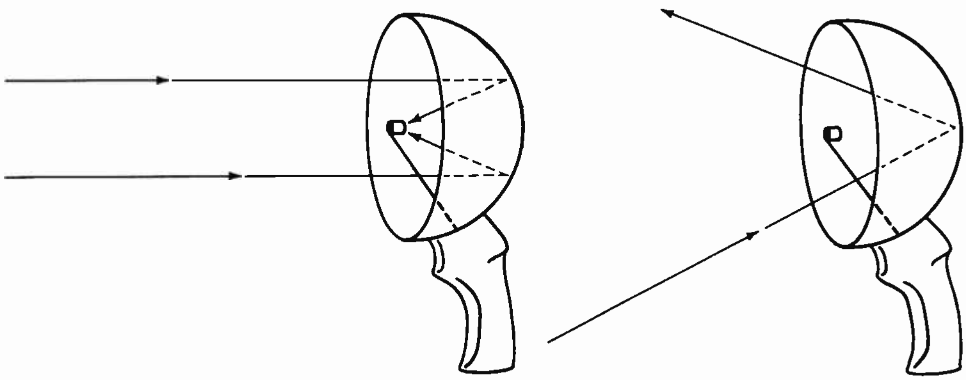


Fig. 3-15. Directional principle of the parabolic microphone.

(A) On-axis signals are focused at the microphone.

(B) Off-axis signals are directed away from the microphone.

The Noise-Cancelling Microphone

The *noise-cancelling* microphone picks up close-proximity signals (most often vocal), while effectively cancelling out high-level background noise. One example of a noise-cancelling microphone is the Electro-Voice model T-45. This single-button carbon microphone (Fig. 3-16) operates on the principle that two acoustically balanced chambers (A and B) contain a single diaphragm (C) which is exposed equally to the acoustic environment through two apertures or openings (D) and (E). Sounds originating from a distance of greater than 0.5″ (12.7 mm) pass through both apertures at an equal intensity and arrive simultaneously at the diaphragm with an equal and opposite force. The net force results in a cancellation of the signal at the diaphragm, so the output is near 0. Whenever a sound source is close to either aperture (less than half the distance between the openings), this out-of-phase balance is upset by less than 180°, thus placing the desired sound source in phase at the diaphram.

A simple, high-quality noise-cancelling device can be readily assembled by placing two matched omnidirectional microphones side by side and by

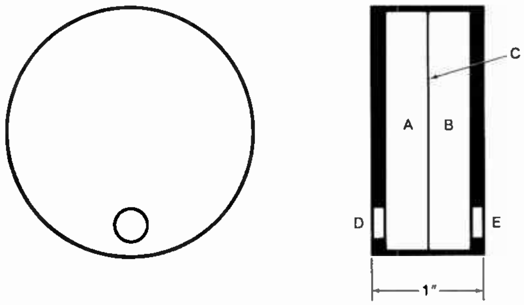


Fig. 3-16. Side and face detail of Electro-Voice T-45 noise-cancelling microphone. *(Courtesy of Electro-Voice, Inc.)*

combining their outputs out of phase (opposite polarity) with each other (Fig. 3-17). When a sound source is placed close to one of the microphones (less than half the distance between the diaphragms) the signal is louder at one microphone than the other, so a signal is produced. Distant sounds produce equal-level, opposite-polarity signals from the two microphones, so the signal is cancelled out. One application for such a device is in live performance where background noise or feedback potential is problematic.

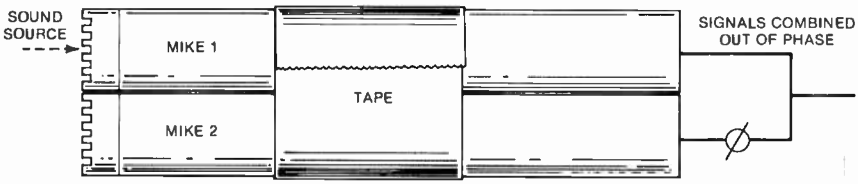


Fig. 3-17. Two omnidirectional microphones side by side wired in opposite polarity, making a quality noise-cancelling device.

Frequency Response

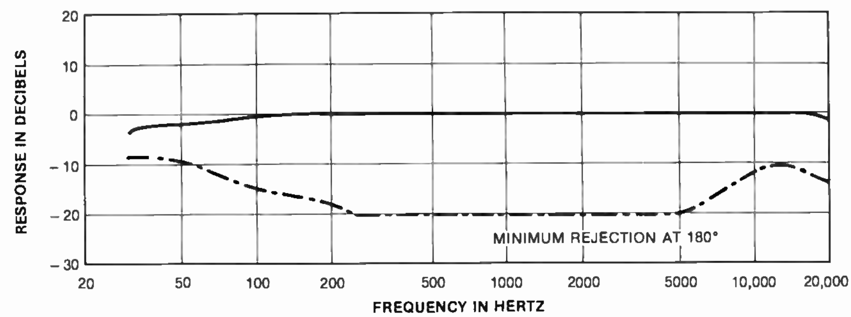
Microphones do not perform equally under all circumstances. Since it is impossible for any one design to represent the "perfect" microphone, the frequency response of each microphone is acoustically tailored to function optimally within a specific application or range of applications.

On-Axis Frequency Response

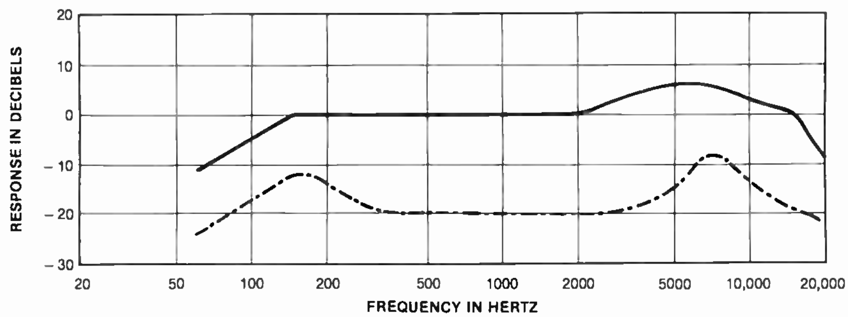
The on-axis frequency-response curve of a microphone is the measurement of its output over the audible frequency range, given a constant-level input signal on-axis to the microphone. The response curve is plotted in output level versus frequency.

A microphone can be designed to respond equally to all frequencies. Such a device is said to exhibit a flat frequency response (Fig. 3-18A). Other microphones may be designed to emphasize or de-emphasize the high-end, middle, or low-end response of the audio spectrum (Fig. 3-18B).

Fig. 3-18. Frequency-response curves. *(Courtesy of AKG Acoustics, Inc.)*



(A) Response curve of the AKG C-460B/CK61 "ULS."



(B) Response curve of the AKG D321.

The response curve indicates how a microphone affects the reproduced tonal balance of the source it is picking up. Boosted high frequencies give a bright sound, a flat response gives a natural sound, and so on.

Off-Axis Frequency Response

The frequency-response curves shown in Figure 3-18 were measured on-axis to the microphone and usually exhibit a totally acceptable response. However, certain designs, when measured off-axis, have a "peaky" or erratic curve. This signal coloration may become evident when the microphone is operating where there are off-axis sounds (in the form of leakage) arriving at the microphone. The result is often a change in the tone quality of the off-axis source. The curve, which is an effective tool in determining the sound character of this leakage, is known as the *off-axis frequency response* of the microphone and may be charted along with the on-axis curves (Fig. 3-19).

The off-axis response of a microphone is important. As we have seen, a microphone for use in music applications may degrade the overall quality of a mix through frequency coloration and/or phase distortion. But in a public-address or sound-reinforcement application a uniform off-axis frequency response (absence of severe peaks) is essential. Here's why: An on-stage car-

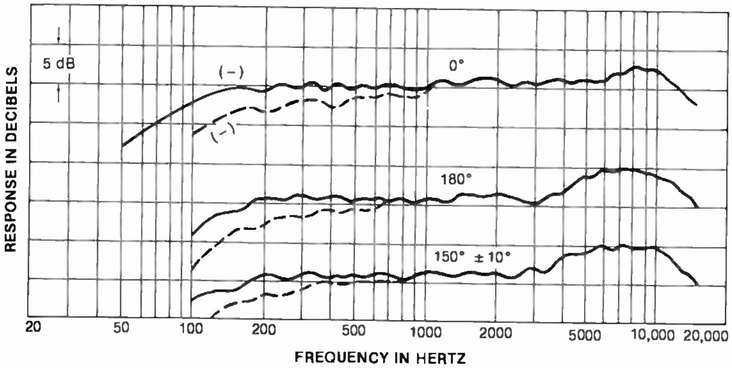


Fig. 3-19. Frequency response at various polar coordinates. *(Courtesy of Electro-Voice, Inc.)*

dioid microphone that displays a strong peak within its operating range may receive too much energy at that peak frequency from the off-axis stage monitors or audience loudspeakers. This might cause feedback at that frequency.

Low-Frequency Response Problems

Currently, "ideal" designs exist that possess a flat frequency response from 20 Hz to 20 kHz with a plus or minus variation of no more than 1 dB. From a practical viewpoint, however, such a high-bandwidth response could cause the pickup of unwanted signals. A wide-range pickup could easily become compromised by conditions over which the recordist has little control. For example:

- low-frequency noise in the recording room
- proximity effect: the bass boost of some directional microphones when used up close

At the low-frequency portion of the spectrum, rumble (high-level vibrations in the 3–25 Hz region) may be transmitted within a studio, hall, or room or along the surface of a large unsupported floor space. If such a condition exists, three possible solutions may be employed:

- Isolate the microphone from the vibrating surface and floor stand through the use of a shock mount.
- Choose a microphone displaying a restricted low-frequency response.
- Restrict the low-frequency response of a wide-range microphone through the use of a highpass (low-frequency cut-off) filter.

Proximity Effect

Many directional microphones show an increase in their bass response as the signal source is brought closer to the microphone capsule. This phenomenon, known as *proximity effect*, starts to become noticeable when the source is brought to within 1′ (30 cm) of the microphone, and it increases as the distance decreases.

As you may recall, a pressure-gradient microphone operates by responding to the acoustic force that acts upon the front face of the diaphragm, relative to the rear face. Due to the inverse-square law, which is constant over frequency, the acting force will rise greatly as working distances become small. A greater pressure buildup at lower frequencies, combined with a short path-length between the front and rear faces of the diaphragm, results in a rising bass response (Fig. 3-20). Given that this boost depends solely upon the pressure-gradient component of a microphone, the proximity effect is somewhat greater in figure-eight microphones than in cardioid microphones that have a combined pressure/pressure-gradient output.

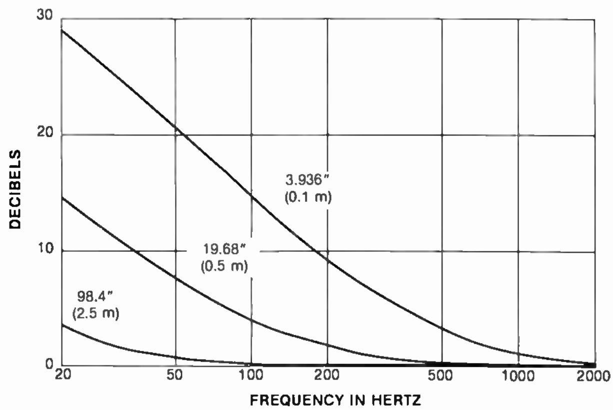


Fig. 3-20. Proximity effect low-frequency boost curves.

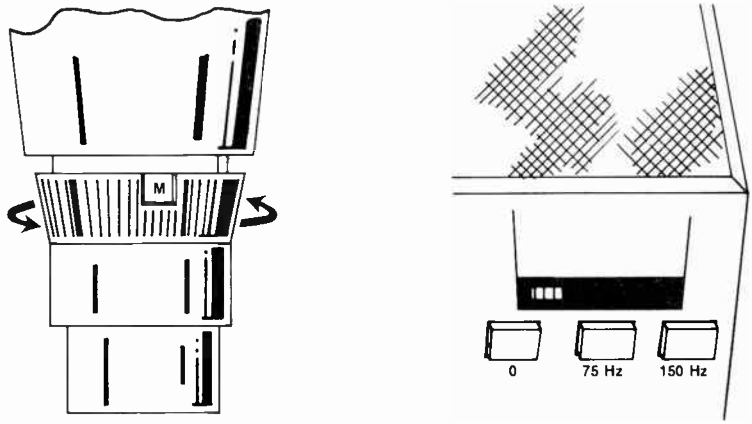
To compensate for this effect, a low-frequency roll-off filter is often provided to restore the bass response to a flat and natural sounding balance. This filter is most often a selective switch located on the body of the microphone (Fig. 3-21).

Certain microphones designed for close working conditions may be internally rolled-off in the low-frequency range, while using proximity effect to restore the response to its natural balance. One example of this is the clip microphone that employs a bass roll-off in order to compensate for the natural resonance of the human chest cavity.

Another method for eliminating proximity effect, and the associated "popping" of the letters "P" and "B," is to replace the directional microphone with an omnidirectional (pressure) microphone for close-miking applications.

On a more positive note, the increase in bass response produced by the directional mike has long been appreciated by vocalists. Its use has become a central tool in the recording and sound-reinforcement arts to give a full,

Fig. 3-21. Detail of various microphone bass roll-off switches.



(A) *Sennheiser MD-421-U.*

(B) *AKG C-414-EB.*

"larger-than-life" quality to a thin-sounding voice. In many cases, the directional mike becomes a primary component of the artist's sound.

High-Frequency Response Characteristics

At the high-frequency range, certain microphones may exhibit a brittle or "pingy" response in certain situations. For this reason, a microphone design may be chosen either to restrict the top-end response or to add presence by boosting the upper mid-range or top-end response. Again, the choice of response and bandwidth will depend upon application and personal taste.

Microphone Transient Response

A significant piece of data that has no presently accepted standard is the *transient response* of a microphone. Transient response is the measure and audible effect of how rapidly a microphone's diaphragm reacts to an incident acoustic waveform.

The rise times of speech and music are 10–100 milliseconds (ms), while the rise time of a high-quality condenser microphone may be less than 1 ms. This ability to closely follow the waveform shape is what allows such microphones to grasp the complex structure of sound.

A modification of this complex structure by the microphone will often alter the characteristic "sound" of the reproduced source in one of two ways:

1. Due to an employed acoustic design, a microphone may color the original pickup signal by adding audible components.

2. The impulse (transient) content during the rise time of a signal may be so short that a microphone's transducer would be too slow to fully transmit the sound's fine impulse structure.

Figure 3-22 shows the transient response of a studio-quality microphone which properly reproduces the audio pulse within its physical limitations and effectively produces no sound coloration.

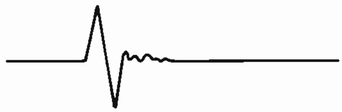


Fig. 3-22. Transient response of a studio-quality microphone.

By contrast, some microphones have oscillation-prone components built into their design. A microphone may, for example, incorporate a small acoustic chamber that is used to boost a certain frequency range to improve high-end response or to increase an insufficient frequency bandwidth. Figure 3-23 shows the transient response of a microphone that boosts frequencies in the 8-kHz range by 2 dB. Within this range, this oscillation-prone system is exited into resonance (ringing) and thus adds a coloration of its own to the sound output.

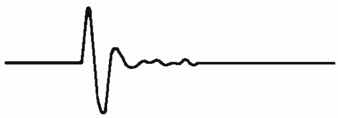


Fig. 3-23. Transient response of a microphone displaying a high-frequency boost.

Figure 3-24 shows the transient response of a condenser microphone designed for a flat frequency response. Although the frequency response is flat, the transient response of this particular microphone is only fair. A flat frequency response does not necessarily guarantee a microphone will sound good. Conversely, not all microphones with a flat response have a poor transient response.

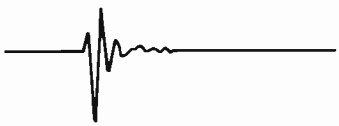


Fig. 3-24. Transient response of a microphone with flat frequency response.

Both condenser and dynamic microphones generally have a diaphragm self-resonance within the audible range. This resonance may be damped by a friction system. A dynamic mike usually has a large diaphragm membrane of generally ¾″–2½″ (19–63 mm), with a correspondingly large mass and compliance. As a result, most dynamic mike's (Fig. 3-25) display a much longer decay (transient after-ring) and are unable to reproduce sharp transient sounds accurately. Such mikes lack in upper brilliance compared to the condenser microphone, with its smaller diaphragm and less-compliant suspension.

Fig. 3-25. Transient response of a typical dynamic microphone.

A ribbon microphone (Fig. 3-26) derives its good transient response from the small mass of the diaphragm itself. As does the condenser diaphragm, the ribbon diaphragm responds quickly to transients, although a sonic "ribbon" character of its own is sometimes added.

Fig. 3-26. Transient response of a double-ribbon microphone pickup.

Electrical Characteristics

The electrical characteristics of a microphone refer to a specific model's measured output response, such as its sensitivity, equivalent noise, overload characteristics, and impedance.

Microphone Sensitivity

The *sensitivity* of a microphone is the output level (in volts [V]) that a microphone will produce, given a specific standardized input signal (rated in decibels of sound pressure level [dB SPL]). This specification implies the degree of amplification that is required to raise the microphone signal to an operating line level (−10 dBV or +4 dBm). This value also allows the recordist to easily judge the output-level differences between two mikes. A microphone that exhibits higher sensitivity will produce a stronger output-signal voltage than that of a microphone with lower sensitivity when both are driven by the same sound pressure level.

Microphone sensitivity is often confusing as it can be specified relative to a number of standard references:

1. dBm per 10 dynes/cm^2
2. dBm per 10 microbars
3. dBV per dyne/cm^2

4. dBV per microbar
5. dBV per 10 dynes/cm^2
6. dBV per pascal
7. mV per pascal

To understand these standard reference ratings, note that:

1. 10 dynes/cm^2 = 10 microbars = 1 pascal = 94 dB SPL
2. 1 dyne/cm^2 = 1 microbar = 74 dB SPL

Typical sensitivity ratings of a professional microphone are:

condenser mike: −45 dBm/10 microbars
dynamic mike: −55 dBm/10 microbars

or expressed to another standard reference:

condenser mike: −65 dBv/dyne cm
dynamic mike: −75 dBv/dyne cm^2

We cannot directly compare the sensitivities of two microphones that are specified to differing references. They must be converted to the same reference, using these simple formulas:

1. dBV/10 microbars = dBV/pascal = dBV/microbar + 20 dB
2. dBm/10 microbars = dBm/10 dynes/cm^2 = dBV/microbar + 20 dBm/10 microbars = dBV/microbar + 22.2 dB (if microphone impedance is 150 Ω)
3. $\text{dBV/microbar} = 20 \log \dfrac{\text{mV/pascal}}{1000} - 20$

As an example, let's compare the sensitivities of two mikes that are rated in different ways.

Mike 1: −70 dBV/microbar (impedance = 250 Ω)
Mike 2: −55 dBm/10 microbars

In converting from dBV to dBm, for an impedance of 250 Ω, you add 20 dB to the dBV figure. This yields an equivalent sensitivity figure for Mike 1 of −50 dBm. Thus, the sensitivity of Mike 1 is −50 dBm and Mike 2 is −55 dBm. Microphone 1 has 5 dB more sensitivity than Microphone 2.

Equivalent Noise Rating

The *equivalent noise rating* of a microphone may be viewed as the device's electrical self-noise. It expresses the equivalent dB SPL that would produce a voltage equal to its noise voltage.

As a general rule, the microphone itself does not contribute a great deal of noise to a system, compared to the amplification stages and tape used in the analog recording chain. However, with recent advances in digital technology at both the professional and consumer levels, these noise ratings are of increased interest.

In the moving-coil or ribbon microphone, this noise is generated by electrons moving within the coil or ribbon itself. In a condenser device, most of the noise is generated by the built-in preamplifier. Certain microphone designs display a greater degree of self-noise than other designs, and thus, for critical applications, care must be taken in the choice of microphone.

According to Eargle (see Appendix F), when discounting the effects of outside electrostatic and electromagnetic interference (in the form of hum and radio-frequency pickup), the minimum limit of a dynamic microphone's internal noise is an output level of −125 dBm (given a source resistance of 50 Ω at a temperature of 300° Kelvin and a bandwidth of 20 kHz).

The equivalent noise rating (in SPL) of this 50-Ω dynamic microphone may be calculated using the following equation:

$$e(enr) = e(t) + e(r) - e(s) \tag{3-1}$$

where,

e(enr) = equivalent noise rating (SPL)
e(t) = theoretical noise rating (dBm)
e(r) = sensitivity reference level (SPL)
e(s) = microphone sensitivity (dBm)

As an example, let's assume that a 50-Ω dynamic microphone displays an output sensitivity of −55 dBm (10 dynes/cm^2 = 94 dB SPL). By plugging the appropriate values into equation 3-1, we find that this pickup will display an equivalent noise rating of 24 dB SPL.

$$-125\ \text{dBm} + 94\ \text{SPL} - (-55\ \text{dBm}) = 24\ \text{dB SPL}$$

Such a microphone may be quite adequate for critical recording within a low-noise environment.

The self-noise level of a microphone may be measured by one of two means, known as the *weighted* and *unweighted* measurement response curves. The unweighted curve is a direct measurement of all frequency components constituting the overall noise content. With the weighted curve, the signal is measured using a weighted equalization network that boosts the mid- and high-frequency components which are louder to the ear and de-emphasizes the low-frequency components to which the ear is less sensitive.

Often this measurement is made using an averaged response through the weighted A-filter of an internationally standardized sound-level meter (IEC 179). This measurement yields a substantially lower value than its unweighted counterpart and is given in technical specifications as dBA. This difference in measurement must be taken into account when technical specifications of microphones are compared.

Microphone Overload

As the use of a microphone is limited at low levels by the microphone's self-noise, its use is also limited at high levels by the SPL it can accept without causing *overload distortion.*

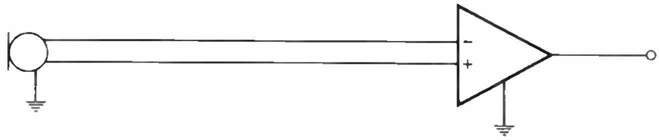
Due to its construction and damping arrangement, the dynamic microphone is an extremely low-distortion device, often capable of accurate transduction over an overall dynamic range of 140 dB.

The diaphragm of most condenser microphones will not distort, except under the most severe sound pressure levels. However, the condenser system differs from the dynamic system in that, at high acoustic levels, the capsule's output signal may be high enough to overload the impedance-convertor circuit contained within the microphone. To prevent this condition, many condenser microphones contain a built-in attenuation pad that immediately follows the transducer output. It reduces the signal level to prevent overloading of the internal impedance convertor. When inserting such an attenuation pad into the microphone circuit, keep in mind that the signal-to-noise ratio of the device is degraded by the amount of attenuation. Thus, when using the microphone in normal SPL's, it is wise to remove the inserted pad. Additional information regarding the microphone attenuation pad may be found in Chapter 4.

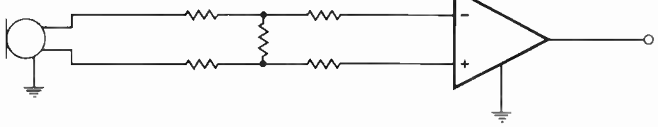
Microphone Preamplifier Overload

Sometimes distortion is heard that is not due to the microphone, but due to the microphone signal overloading the microphone preamp in the mixer or recorder. In many recording or sound-reinforcement devices (e.g., VCRs, cassette recorders, and small mixers) control over the input gain is located after the first stage of amplification (Fig. 3-27A). If the microphone output exceeds the signal-handling capacity of the microphone preamplifier, the entire signal chain becomes distorted and no amount of post-attenuation can solve the problem.

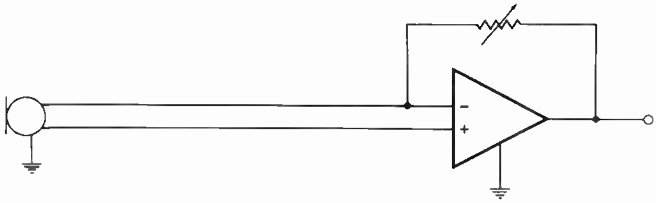
To prevent such an overload of the amplified input signal, it is necessary to place an attenuation device before this first preamplification stage (Fig. 3-27B). Such an attenuator is known as a *microphone pad*. It can be an "in-line" device plugged into the microphone cable (Fig. 3-27B) or a variable switch built into the mixer. Alternatively, many professional audio production and recording consoles provide a continuously variable-gain input stage (Fig. 3-27C) which allows the preamplifier to accept a wide range of input levels without overloading.



(A) *Input stage with level control following amplification.*



(B) *Pad preceding the input stage.*



(C) *Variable-gain input stage.*

Fig. 3-27. Various microphone-preamplifier configurations.

Microphone Impedance

The element of *impedance* within a circuit is the opposition to the flow of current and, in a microphone, includes a resistive component. Impedance is measured in ohms and is denoted by the symbol Z.

The most commonly used microphone output impedances are 50 Ω, 150 Ω, 250 Ω (low-Z) and 20kΩ to 50kΩ (high-Z). The output impedance of a microphone may vary from one design type to another and may be internally selectable. The dynamic microphone is by nature a low-impedance device; sometimes requiring a step-up transformer to achieve a higher impedance rating.

Each impedance range has its own advantages and disadvantages. One major disadvantage of the high-Z microphone is its high susceptibility to the pickup of surrounding electrostatic noise, such as that caused by fluorescent lights and motors. As the length of the microphone cable increases, its capacitance also increases. At about 10′ (3 m), the cable capacitance begins to short out much of the high-frequency information picked up by the microphone. Therefore, a high-impedance mike is generally limited to use with a cable length of 10′ (3 m) or less for good high-frequency response. For this reason, high-impedance microphones are seldom used any more.

Low-impedance microphones have very little high-frequency loss, even when used with cable lengths of up to several hundred feet. Low-impedance mikes (under 300 Ω) cause their cables to be fairly insensitive to electrostatic pickup. Induced hum pickup from electromagnetic fields can be eliminated through the use of a twisted-pair cable. As a result, the low-impedance microphone used with a shielded, twisted-pair cable offers the best attainable signal-to-noise figure in conjunction with a transformer/transformerless balanced input circuit.

The impedance rating of a microphone's signal line may be changed through the use of a microphone-matching transformer (Fig. 3-28), in order to best adapt to the immediate needs of the cable/equipment requirements.

A microphone that is not connected to a device still exhibits an output voltage. This potential is known as the *unloaded* or *open-circuit* voltage of the microphone. Often, the input stage of a professional audio console will be designed to have an input impedance at least 7 to 10 times the microphone impedance so that the microphone is effectively unloaded and so produces maximum voltage. A microphone that is connected in an unloaded fashion operates according to its designed open-circuit sensitivity and frequency-response specifications.

The classifications of microphones commonly encountered in audio production are delineated in Figure 3-29. Table 3-1 suggests the choice of microphone for various applications.

In this chapter we saw that the microphone can be designed with a wide range of directional, frequency, transient, and electrical-response characteristics. Each of these parameters contributes to the sound of a microphone, influencing its use for achieving the best possible sound quality.

Fig. 3-28. The LITTLE IMP impedance transformer. *(Courtesy of Whirlwind Music Dist., Inc.)*

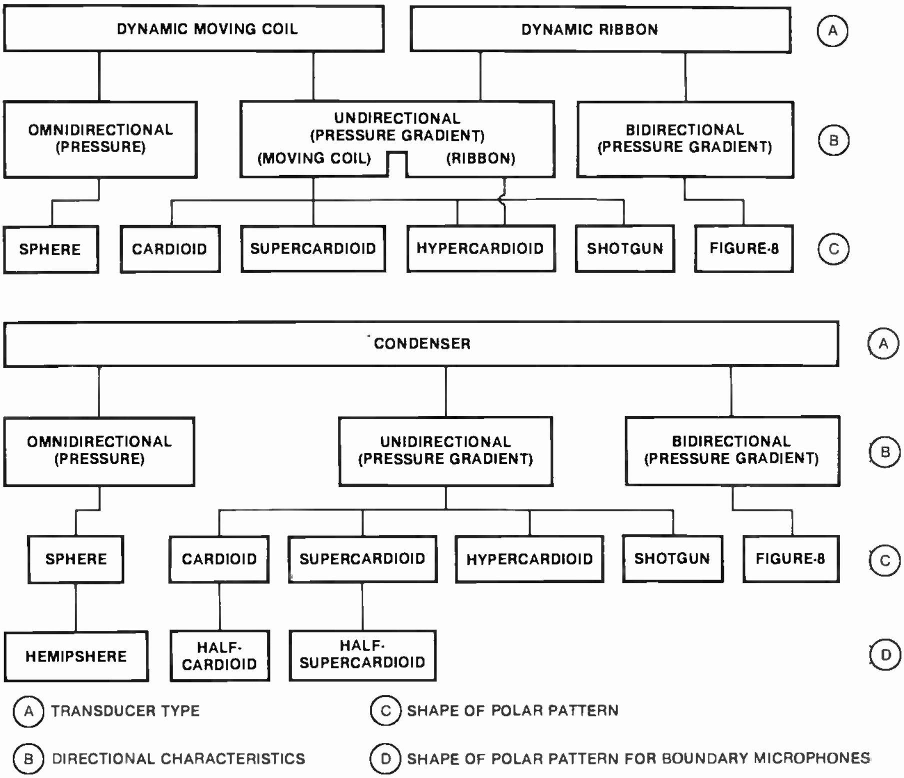


Fig. 3-29. Microphone classification chart. *(Courtesy of Bruce Bartlett)*

Table 3-1. Microphone selection chart.

APPLICATION	MICROPHONE CHARACTERISTIC/CHOICE
Natural, smooth tone quality	Flat frequency response
Bright, present tone quality	Rising frequency response
Extended lows	Omni dynamic or condenser with extended low-frequency response
Extended highs (detailed sound)	Condenser
Reduced "edge" or detail	Dynamic
Boosted bass up-close	Directional microphone
Flat bass response up-close	Omnidirectional, Multiple-D directional
Reduced pickup of leakage, feedback, and room acoustics	Directional microphone, or omnidirectional microphone at close working distances
Enhanced pickup of room acoustics	Omnidirectional microphone, or directional microphone at greater working distances
(1) Miking close to a surface	Boundary or clip microphone
(2) Even coverage of moving sources or large sources	
(3) Inconspicuous mike	
Extra ruggedness	Moving-coil microphone
Reduced handling noise	Omnidirectional, or directional microphone with shock mount
Reduced breath popping	Omnidirectional, or directional microphone with pop filter
Distortion-free pickup of very loud sounds	Condenser with high maximum-SPL spec, or dynamic
Noise-free pickup of quiet sounds	Low self-noise, high sensitivity

4 Electrical Interface: The Cable and Connector

In Chapter 3, we discussed the various characteristics inherent in the microphone itself. These are, however, only part of the overall picture. The electrical interface—the cable and its means of interconnection—play an important role in the electrical characteristics and operation of the microphone system.

In this chapter, then, we'll focus on: the effect of cable length upon electrical performance, unbalanced and balanced signal lines, polarity considerations in the microphone/cable, microphone connectors, the microphone pad, and the use of phantom powering for condenser microphone systems.

Signal Losses in a Microphone Cable

The major disadvantages in the use of a high-impedance microphone (20–50 kΩ) are its great susceptibility to electrostatic pickup and to cable capacitance, which limits the cable length for flat high-frequency response to about 10′ (3 m). The high-frequency component of a signal is reduced as cable length increases. Because of this limitation, the low-impedance microphone (150–250 Ω) is used almost exclusively for professional audio work.

The cable employed in carrying this low-impedance signal is typically made of two conductors, one for the audio signal plus one for a grounded shield. The shield is a braided copper wire for movable cables or a thin aluminum foil for permanent cables.

The average length of microphone cable in the production/recording studio (25–30′ [8–9 m]) is not long enough to create appreciable losses due to capacitance between the audio leads. However, at lengths approaching 200′ (60 m) or more, however, the capacitive reactance may be sufficient to alternate high frequencies. This attenuation depends on four factors: cable capacitance, cable length, signal source impedance, and frequency. The attenuation increases as any of these variables increases (Fig. 4-1). For example, a 200′ (60-m) cable of 22-pF capacitance per foot reduces the signal by 1.2 dB at 20 kHz relative to 1 kHz for a sound source having an impedance of 200 Ω.

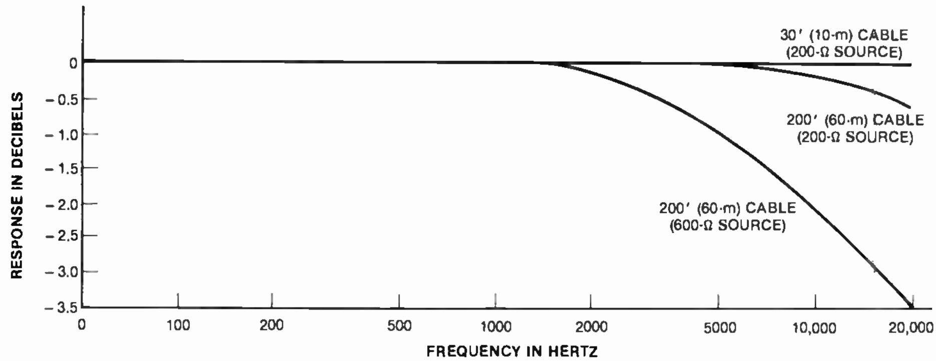


Fig. 4-1. Effect of various cable lengths upon frequency response (cable capacitance: 22 pF/foot).

The impedance of professional microphones generally ranges between 150 and 250 Ω. Cables running with such a source impedance will display insignificant signal losses at cable runs of up to 200′ (60 m), while those operating with a 50-Ω microphone may be used with lengths up to 2000′ (600 m) without serious losses at the upper frequency range. If you doubt whether you can use a long cable run, try this simple and accurate procedure for measuring the capacitive losses shown in Figure 4-2.

1. Plug an oscillator in series with a resistor (R_1) that is equal to the impedance of the microphone.
2. Connect the cable leads across the resistor.
3. Measure the output voltage vs. frequency at the oscillator output (E_i).
4. Measure the voltage at the console input position (E_0).

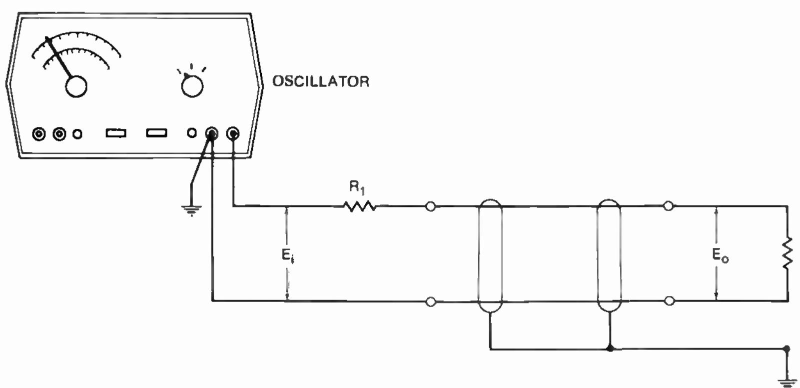


Fig. 4-2. Method of measuring microphone cable losses with frequency.

The dB loss at any frequency is 20 log $(\frac{E_0}{E_i})$.

With many microphone designs, the user can select the correct impedance of the microphone's output transformer by using an internal switch or by soldering the output terminals to the correct impedance tap. If this option is not available, an external matching transformer can be used.

Balanced/Unbalanced Signal Lines

The cable connecting a microphone's output to an input preamplifier uses either an unbalanced line or a balanced line.

An unbalanced line (Fig. 4-3) is most often used with an inexpensive, high-impedance microphone. A single conductor carries the positive (in-phase) signal, while a shield carries the negative (return) signal of the circuit. This shield also reduces the pickup of electrostatic noises that might be induced across the audio leads.

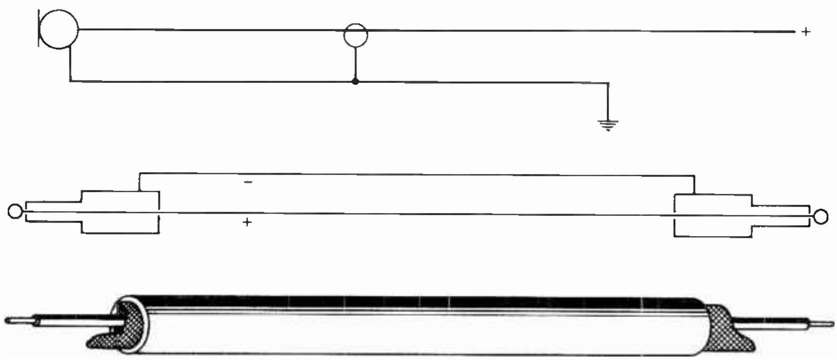


Fig. 4-3. Unbalanced microphone circuit.

In practice, the microphone level may be quite low compared to an unfavorable, high-level electrostatic pickup that might emanate from light or power-line fields. In an unbalanced circuit, this field is injected across both the signal "hot" conductor and shield. Due to the shield's properties, however, the stray noise is reduced across the signal "hot" conductor. As the leads pick up different amounts of noise, the stray noise is amplified to become audible with the audio signal.

As low-impedance microphones are often very low in output level and may use long cable runs, such external electrostatic noises could easily be greater in level than the audio signal itself. To correct this problem, a balanced microphone circuit (Fig. 4-4) is almost always used.

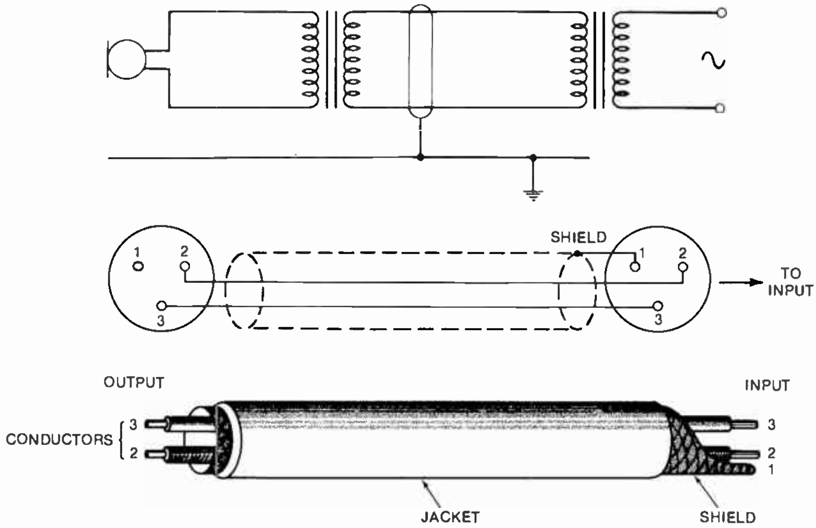


Fig. 4-4. The balanced microphone circuit.

In a balanced transmission line, the output signal of a microphone is carried by two conductor leads. Neither of these leads is directly tied to the circuit ground of the audio system; the ground circuit conductor is carried by an electrically separate outer shield.

A balanced line operates on the following principle: Audio is carried in two leads. An external electrostatic or electromagnetic field is presented to both leads equally. This field induces a voltage with equal phase at both leads simultaneously. The induced voltage appears at the input of a transformer or a balancing amplifier with equal amplitude and polarity at both leads. Since the microphone preamp amplifies the *difference* in voltage between the leads, the equal interference signals are cancelled at the input, while the low-level audio signal is allowed to pass through unaffected.

When connecting a balanced line to an unbalanced circuit or vice versa, install a 1:1 isolation transformer (Fig. 4-5) into the audio line. This ensures relative isolation from unwanted ground loops or induced hum and noise into the input circuit.

Cable/Connector Specifications

In professional applications, a microphone cable has one of two forms: the movable cable/connector run or the permanent installation run.

A good movable microphone cable is typically made of two conductors, 18 to 22 gauge, placed within a braided copper or similar conductive shield. Each end of a cable is soldered to a male connector on one end and a female connector on the other end. Nearly all professional audio connectors are the three-contact XLR-type (IEC standard 268-14B) shown in Figure 4-6A. However, a five-contact XLR may be used for stereo microphone applications. Newer, low-cost home recording systems often employ the two-conductor ¼″ phone plug/jack in a balanced configuration (Fig. 4-6B). Certain European recording and broadcast organizations have standardized on the three-, five-, or twelve-contact DIN (Fig. 4-6C) or the Tuchel-type connector.

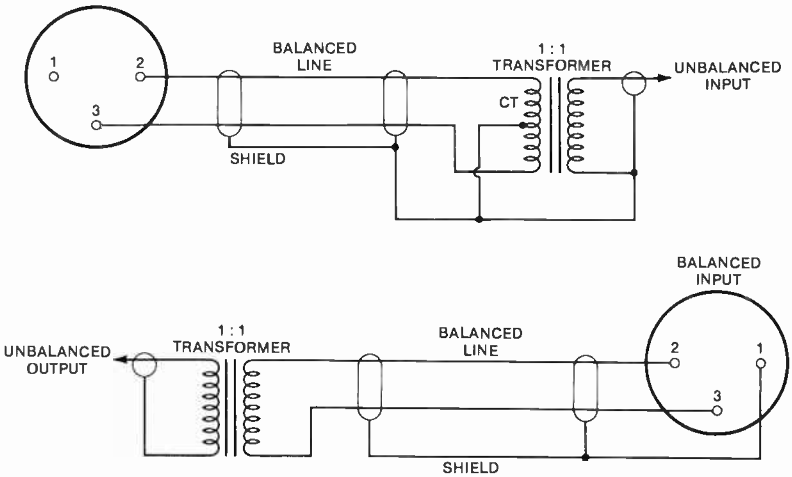
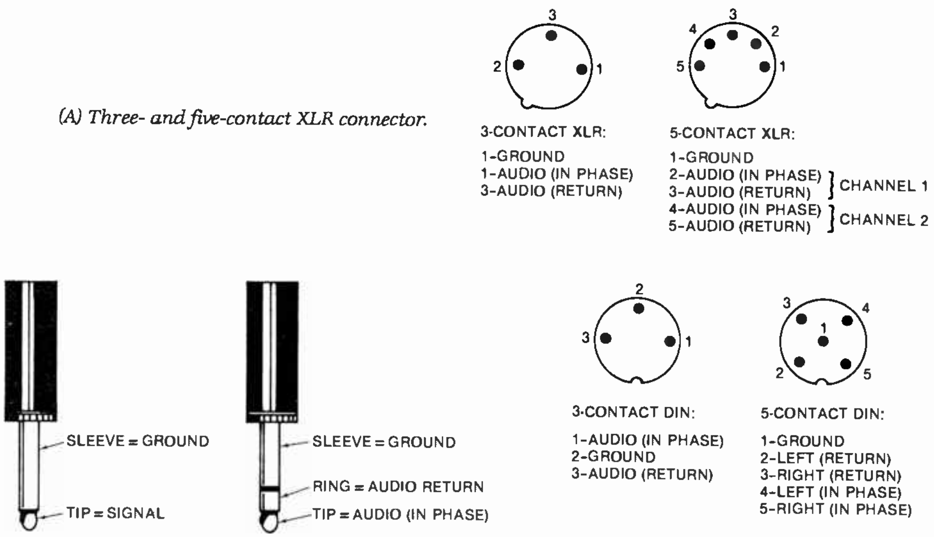


Fig. 4-5. Transformer connections between balanced and unbalanced equipment.



(A) Three- and five-contact XLR connector.

(B) Unbalanced/balanced ¼″ phone plug.

(C) Three- and five-contact DIN connector.

Fig. 4-6. Microphone connector standards.

Cables that are used within a permanent installation often differ from those of the studio cable run. Permanent installations use multiple runs of two-conductor, balanced mike lines that may be either a multicable "snake," or permanent cable lines run through the walls and/or floor of a production structure.

In live reinforcement applications, the multiple-cable lines that are used to carry audio are placed within a single plastic sheath to form a single run known as a *microphone snake* (Fig. 4-7). A snake is a reliable and flexible system that may be reused in a wide variety of reinforcement situations. Such runs are often constructed in cable lengths of 500′ (150 m) or more. For ease of storage, a microphone cable drum may be used (Fig. 4-8).

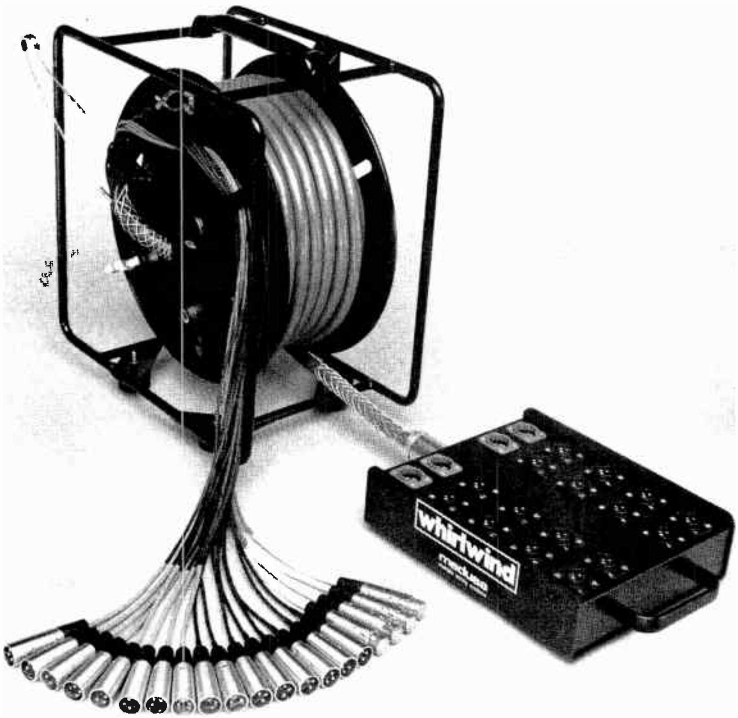
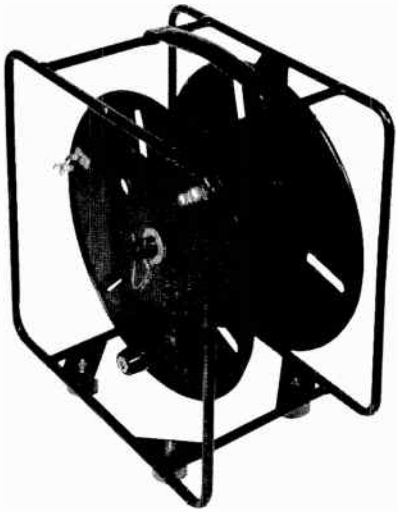
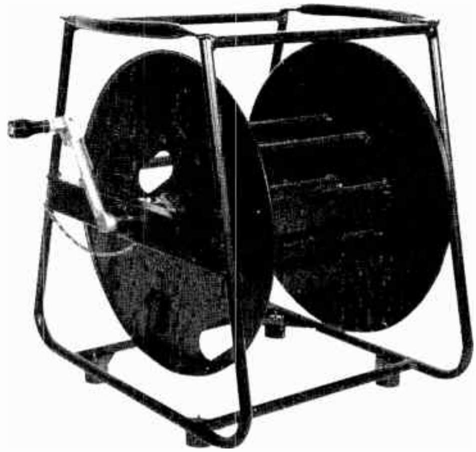


Fig. 4-7. Medusa Multiwiring snake system. *(Courtesy of Whirlwind Music Dist., Inc.)*

In an audio or video production studio, adaptability is not as important as reliability. Special-purpose microphone cables are often run via metal shielding conduit from a microphone box through the shortest path to the console inputs. All female wiring connections are made at a centralized microphone input panel (Fig. 4-9), which houses all mike input and cue-return (headphone) connector points. Wiring at the console position is by either XLR connectors, multiple-pin connectors, or punch block.



(A) Keith Monks CD1.



(B) Keith Monks CD2.

Fig. 4-8. Microphone cable drum. *(Courtesy of Keith Monks Ltd.)*

Microphone/Cable Polarity Considerations

When more than one microphone is present in a system, the polarity of all microphones and cables must be standardized. *Polarity* is the phase relationship between the motion of a mike's diaphragm and its output voltage. When two or more microphones are placed near a sound source, each microphone receives a portion of the sound (possibly in the form of leakage). If the audio leads of any of these microphones are reversed, it is likely that signal phase cancellations could result, causing degradations in frequency response or complete electrical cancellation when combined into a mono signal. To eliminate this possibility, all microphone cables must be properly phased at the connector ends. The polarity of the connector leads can be readily checked with a volt-ohmmeter or a microphone cable tester (Fig. 4-10).

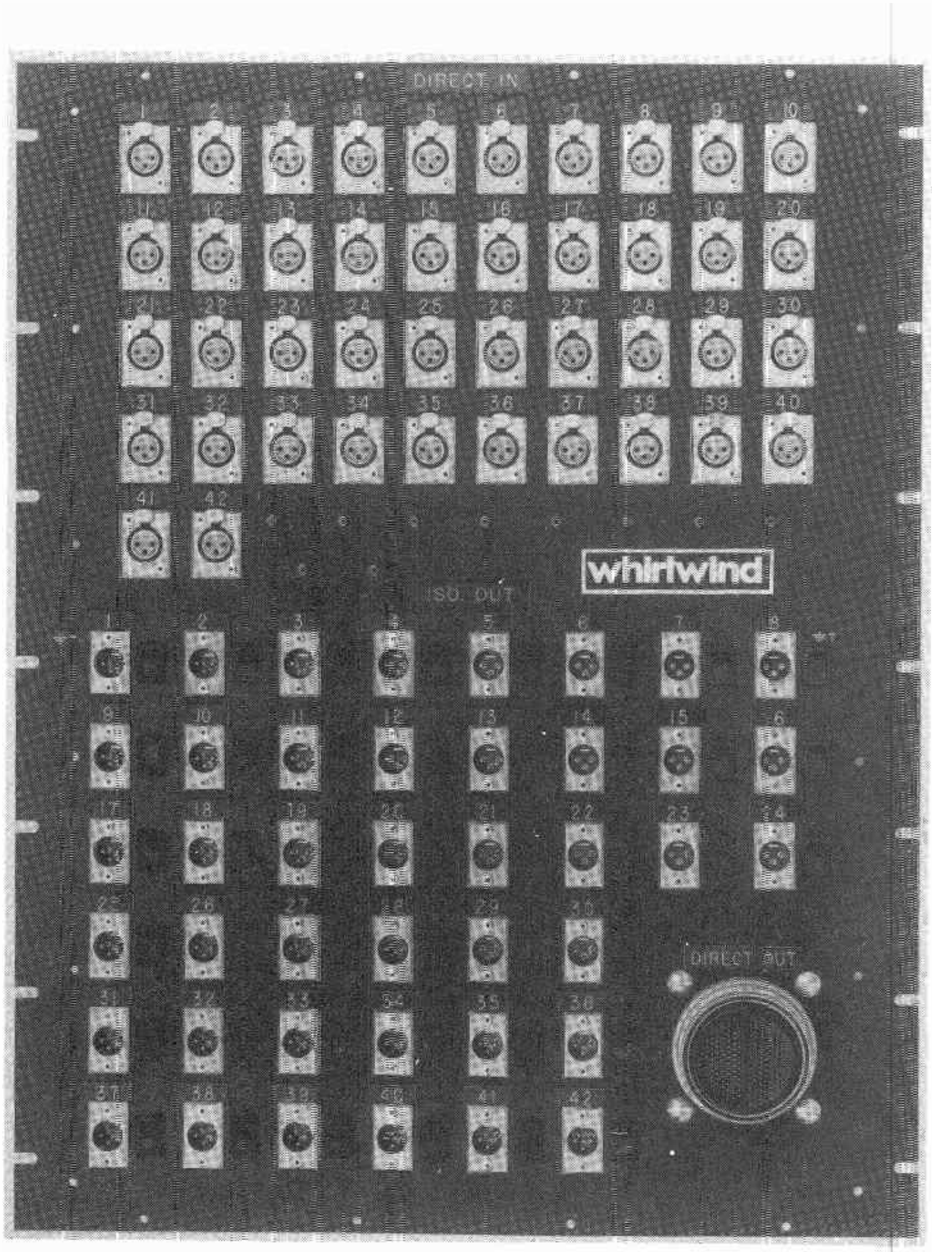


Fig. 4-9. Microphone input panel. *(Courtesy of Whirlwind Music Dist., Inc.)*

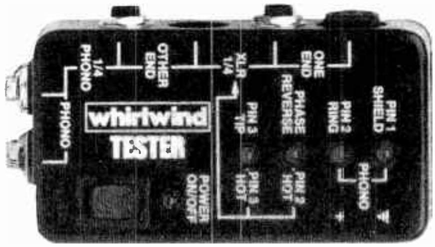
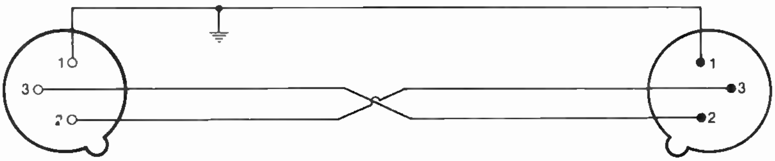


Fig. 4-10. Microphone cable tester. *(Courtesy of Whirlwind Music Dist., Inc.)*

If microphones of different polarities cause tonal changes when combined onto one track or mixed to mono, try these remedies:

- If it is available, press the phase "Ø" button located on the input strip of a production console to reverse the polarity of the problematic input.
- Insert a phase-reversal adapter (Fig. 4-11) into the problem microphone cable.
- Replace the cable with a cable of proper polarity.
- Rewire the problem cable in correct polarity.



(A) Schematic.

(B) Adapter.

Fig. 4-11. Phase-reversal adapter. *(Courtesy of Whirlwind Music Dist., Inc.)*

Note that the XLR connector may be phased to two differing standards. The American standard has pin 2 in phase (+), while in Europe, pin 3 is in phase. To be sure of the standard in use, consult the manufacturer's data sheet for the microphone.

Cable Grounding Considerations

As we have seen, the balanced microphone-cable system uses two signal paths: two conductors for the transmission of audio signal. The shield conducts induced electrostatic charges to system ground. The unbalanced cable uses a center conductor for the audio in-phase signal and uses the shield both for the audio-return signal and shielding.

With both the balanced and unbalanced audio circuits, watch out for external noises caused by a ground loop (Fig. 4-12). In short, the ground loop is a buzz or 60-Hz hum in the audio signal, caused by a difference in relative voltage potential between two or more circuits with respect to system or absolute ground. This occurs when two interconnected devices are connected to ground via two or more paths: local chassis grounding through the power cable and grounding via the cable shield.

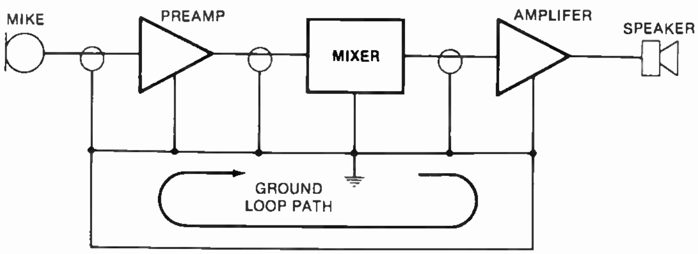


Fig. 4-12. Existence of ground loop between two or more devices.

Ground loops are most often encountered in an unbalanced signal circuit, where the system ground is part of the signal path itself. Given this condition, it is important that all equipment be referenced with respect to ground at only one point; usually the "ground bus" of the production console or mixer. Such conditions may also be present in a balanced system if proper care is not taken in the wiring of the two-conductor cable-connector combination. Within a balanced microphone/line cable, it is often advisable to tie the cable shield directly to the outer shell of the XLR (or equivalent) connector at the *male end/chassis ground end* only (Fig. 4-13). This provides an improved shielding both against radio interference and other undesirable pickup and against ground loops, which may induce hum into the signal, if the outer shields are grounded at more than one circuit end point.

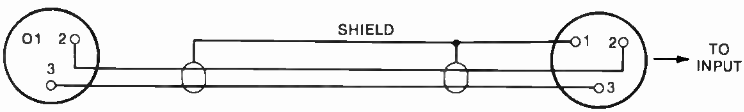
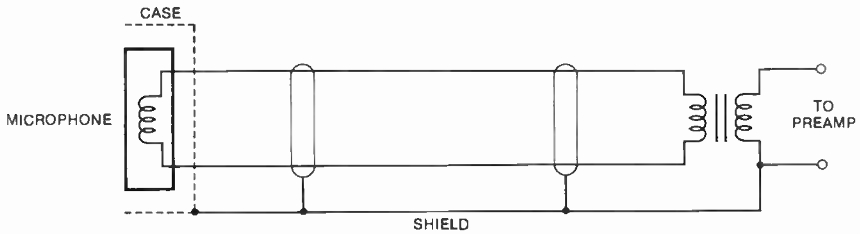
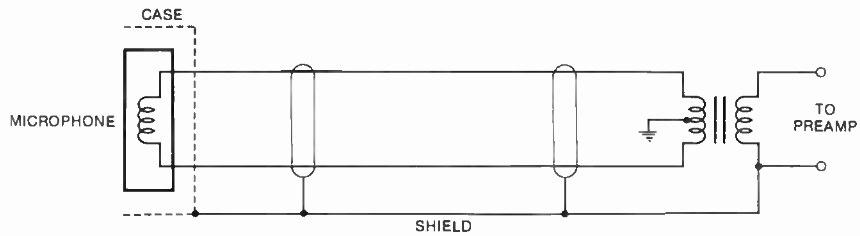


Fig. 4-13. Diagram of grounded shell at male connector.

The audio leads that are connected to a transformer or to a balanced transformerless input stage of a microphone preamplifier may be designed to *float* with respect to circuit ground (not be tied to circuit ground), shown in Figure 4-14A, or they may be tied to ground via a transformer center-tap configuration (Fig. 4-14B).



(A) Floating line.



(B) Grounded line.

Fig. 4-14. Balanced microphone line.

Remote Powering of the Condenser Microphone

Independent Phantom Powering for Each Microphone

We learned in Chapter 2 that all condenser microphone systems require a dc potential or electrostatic potential across their capacitive plates in order to provide an audio output signal. Power is also needed for the transducer's impedance-convertor circuit. The methods of providing this potential vary with microphone design and time of manufacture. Early condenser microphones required a dedicated, external power supply for each microphone (Fig. 4-15). Both the audio signals and voltage for the capacitive element were handled by a single multiconductor cable, a system called *phantom powering*. A male XLR-type connector in the power supply provides signal output to the audio chain. In this system, a positive voltage is supplied to both microphone audio leads (XLR pins 2 and 3) through a set of matched value resistors (6.8 kΩ at 48 V, 1%, ¼ W/680 Ω at 12 V, 1%, ¼ W) or transformer center tap. The negative side of the supply is returned to the power supply ground via the cable shield.

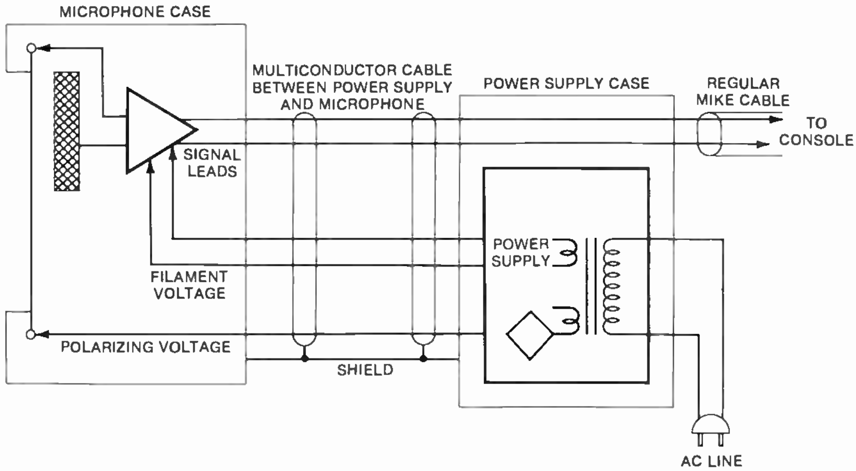
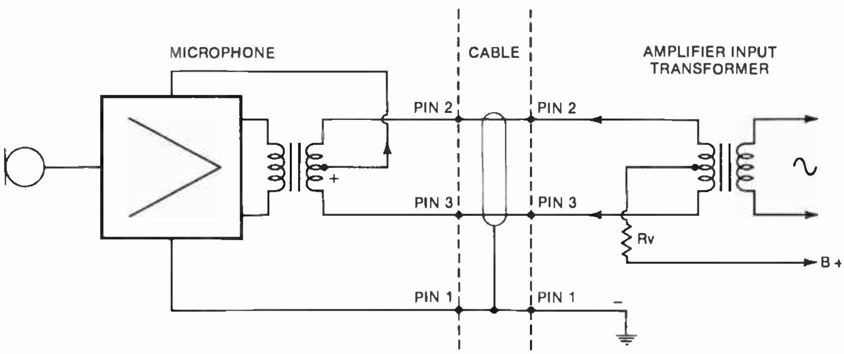


Fig. 4-15. Simplified schematic of microphone requiring external power supply.

Console Phantom Powering for all Microphones

The dedicated power supply is now seen as cumbersome and unnecessary. A new method has been designed for providing power in which all externally powered condenser microphones obtain their required dc polarizing voltage and circuit powering from one, centrally located and standardized power source in the console or mixer.

Console phantom powering (Fig. 4-16) provides power to all mike inputs through the standardized XLR-type connector/cable arrangement. This eliminates the need for in-line power supplies. Condenser microphones using centralized phantom power need no internal batteries, external battery packs, or individual ac-operated power supplies for operation. Typical phantom-power specs are 48 V ±4 V, max. 2 milliamps [mA] or 12 V ±1 V, max. 10 mA.



(A) Artificial-tap (resistive) schematic.

Fig. 4-16. Phantom powering system.

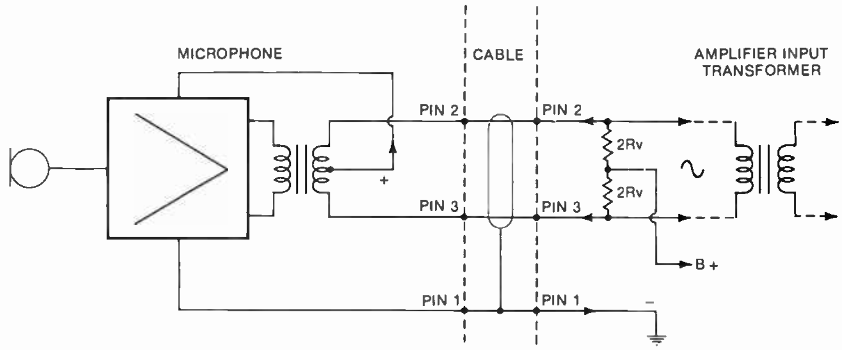
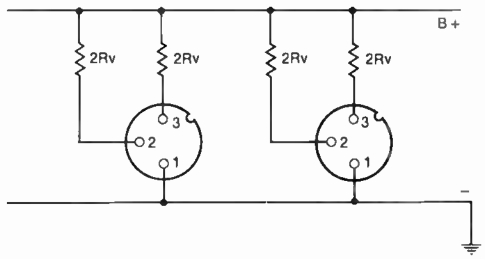


Fig. 4-16 (cont.)

(B) Center-tap schematic.



(C) Multiple-microphone common tap.

This powering method does not interfere with the normal operation of a moving-coil or electret-condenser microphone, as the positive voltage is applied to both signal leads equally. Recent professional consoles and mixers are equipped with an individual phantom power on-off switch at each input strip. This option is provided because phantom powering might destroy the element of a ribbon microphone. For this reason, it is wise to switch the phantom power supply out of circuit for each channel using a ribbon microphone.

For convenience, the phantom supply is most often available from the console's microphone inputs, automatically providing studio-wide power. However, where no such power is available (such as in a simple, remote situation), a portable battery or ac phantom supply (Fig. 4-17) may be called into use.

A condenser microphone can be operated from a dc battery source, if so designed. A few condenser microphone systems, such as the Neumann U-87, can work on internal batteries or phantom powering. Such a microphone can be incorporated into any system without the need for an external power supply. External battery packs can be purchased or designed for powering one or more microphones, as long as the microphone's voltage requirements are met. Most externally powered condenser microphones are designed to oper-

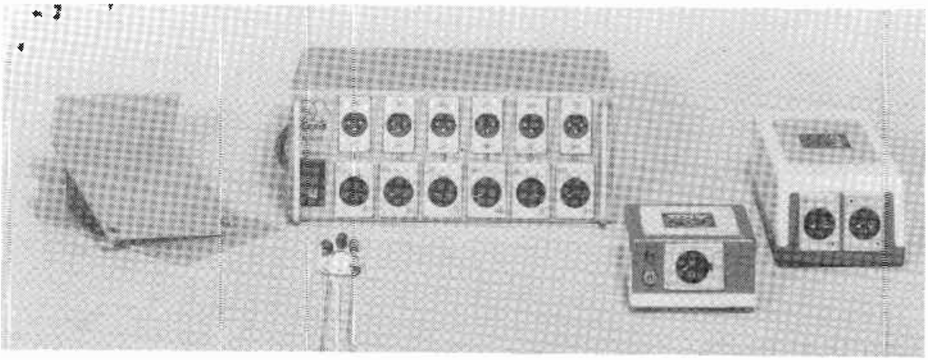


Fig. 4-17. Various battery and ac phantom power supplies. *(Courtesy of AKG Acoustics, Inc.)*

ate at the nominal 48-V range, ± 4 V. Should a lower potential (such as 12 V) be applied, a degradation in distortion, noise, and frequency-response figures may occur.

Radio Frequency (RF) Interference

The function of the grounded shield within a signal cable is to reduce the effect of electrostatic pickup such as *radio frequency interference* (rfi). However, in an improperly grounded audio circuit, a microphone or signal line may serve as an excellent antenna for the pickup of radio waves. These low-level modulated radio signals may then be detected and amplified by an unbalanced input stage as an audible signal (commonly your least-favorite radio station or citizen's-band transmission). Electrostatic pickup also may be caused by electronically controlled lighting systems, such as those used at a rock concert.

Radio frequency interference can be controlled by one of several means, the best of which is to employ sets of properly grounded, balanced transmission lines throughout a particular audio production setup. Whenever possible, low-level transmission lines, such as microphone cables, should be physically isolated from higher-level power cords and grids. For example, within a permanent installation where high-power fields exist, it is wise to consider a common ground system where all mike cables are electrostatically isolated within runs of metal conduit.

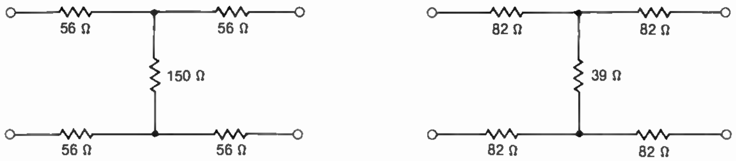
Microphone Pad Specifications

In most present-day microphone designs, an extremely high sound pressure is necessary in order to overload or distort the transducer element itself. It is more likely, however, that the output signal of the microphone preamplifier following this element will be distorted by a microphone signal that is too high in level. As we saw in Chapter 3, this signal may be reduced by a number of means:

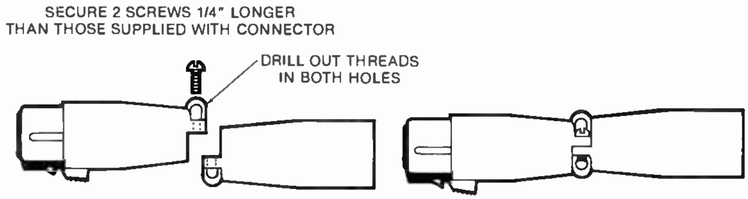
1. attenuation within the microphone itself
2. attenuation built into the console/mixer
3. attenuation (padding) plugged into the cable between the microphone and the microphone input (Fig. 4-18)

Attenuation is used *only* when the sound source is so loud that distortion occurs. For distortion in the microphone itself, use an internal pad, if available. For distortion in the console mike preamp, adjust console attenuation or console mike preamp gain.

Fig. 4-18. Microphone pad to prevent the overloading of an input amplifier. *(Courtesy of Whirlwind Music Dist., Inc.)*



(A) Construction values for a −10 dB and −20 dB H-type pad.



(B) Simple construction details.

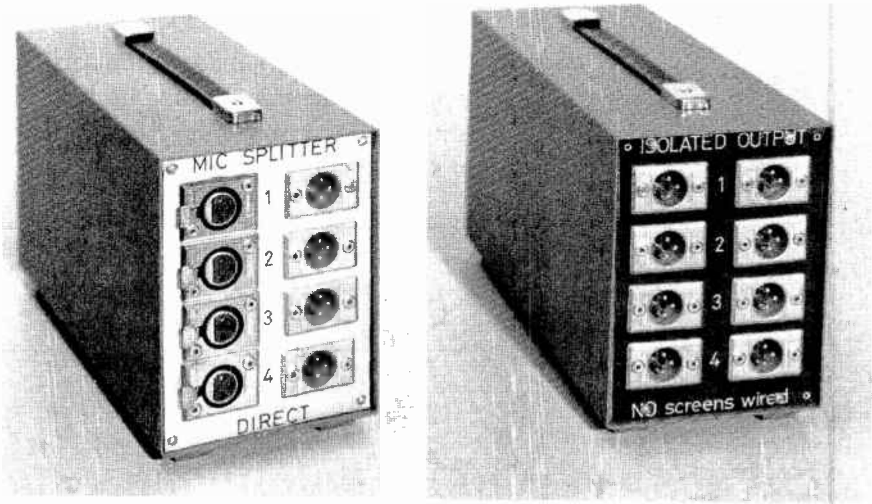
(C) Commercially available microphone line attenuator.

The Microphone Splitter Box

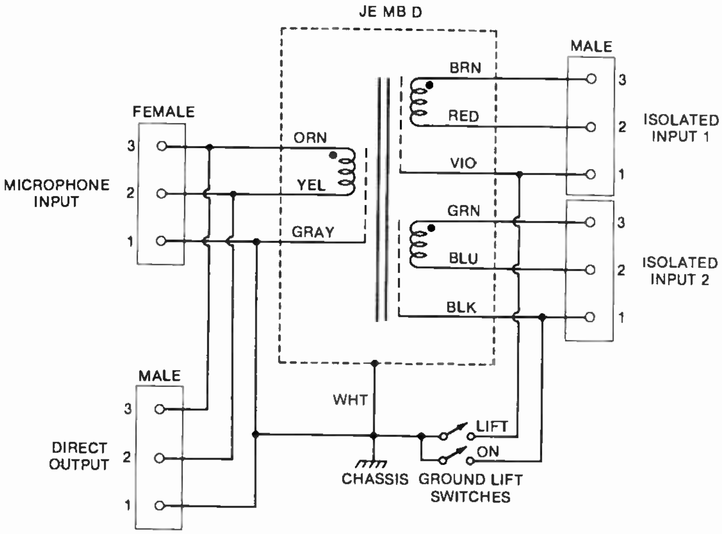
A microphone splitter divides the output of one or more microphones to feed two or more sound systems, allowing simultaneous use for public address, broadcast, or recording applications. Thus, a minimum number of on-stage microphones allows the performers to have a greater degree of movement, along with a cleaner stage appearance.

A single microphone can drive two audio systems simultaneously by splitting the microphone output in a "Y" fashion. Unfortunately, some forms of electrical interaction may occur: degraded frequency response, increased distortion, and rf interference. In order to prevent such a degraded interaction, a transformer-isolated *microphone splitter* (Fig. 4-19) can be used.

Fig. 4-19. Microphone splitter.



(A) Microphone splitter box. (Courtesy of Keith Monks Ltd.)



(B) Signal path diagram. (Courtesy of Jensen Transformers)

When using a splitter box in conjunction with phantom-powered condenser microphones, you must provide powering from the audio system connected via the direct signal path. This is because the isolated output path serves to isolate the phantom power signal from the microphone, thus preventing the required voltage from reaching it.

In summary, the method and type of wiring used in an audio installation affect the quality of the recording, broadcast, or reinforcement system. Care and attention to wiring details help eliminate trouble spots and improve the quality of the final audio product.

5 Microphone Accessories

In modern-day audio production, microphones are used for a wide range of applications. For the most effective mike operation in these applications, an equally wide range of accessories is often required. Such accessories include:

- microphone mounts (desk stands, floor stands, booms, fishpoles, etc.)
- microphone stand mounts (clamps, shock mounts, hangers, etc.)
- stand adapters and accessories
- wind screens
- multiple-capsule condenser microphone systems

Microphone Mounts

The means of mounting a microphone may be broken down into one of four categories: flush mount, desktop mount, floor stand mount, and boom mount.

The Flush Mount

The *flush mount* places a microphone near a surface (such as a stage floor). The mount may be a sculptured windscreen, which is specifically designed to house a microphone (Fig. 5-1), or simply a soft sponge (placed underneath the microphone's body), which places the microphone ¼″ (6 mm) to ⅛″ (3 mm), above the support surface (Fig. 5-2). The sponge-like material also isolates the microphone from structure-borne noise and resonances.

This method operates like a boundary-type microphone, giving a smooth frequency response within its operating range. However, because the diaphragm is about an inch above the surface (depending upon the microphone size), cancellations occur within the upper range of the audible frequency band, possibly causing a dull sound. Recently, boundary

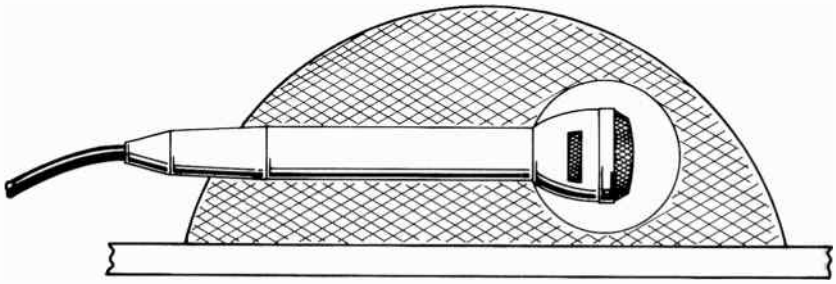


Fig. 5-1. Prefabricated foam flush mount.

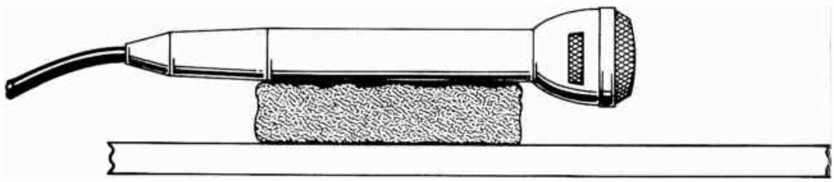
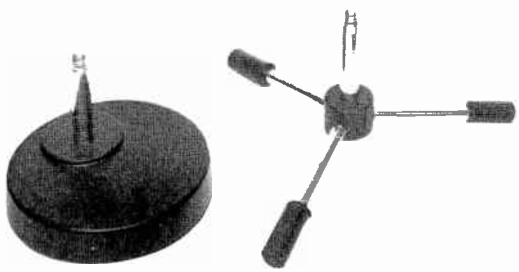


Fig. 5-2. Homemade flush mount using a simple sponge.

microphones with wide-range response on a surface have made the flush mount relatively obsolete.

The Desktop Mount

The *desktop mount* (Fig. 5-3) enables tabletop or low floor stand mounting of a microphone. Such stands are either fixed or adjustable in height, with



(A) Fixed height.

(B) Adjustable height.

Fig. 5-3. Desktop stand mount. *(Courtesy of Keith Monks Ltd.)*

threading provided for the microphone clamp. Depending on the standard adopted, these threads may be sized either at ⅝″—27 threads per inch (USA) or ⅜″ (9.5 mm)—Whitworth (UK).

Desktop communications microphones (Fig. 5-4) are often useful in paging applications for areas having high background noise and/or where the operator is able to spare only one hand for speaking over the paging system.

Fig. 5-4. Shure model 522 paging base station microphone. *(Courtesy of Shure Brothers, Inc.)*

The Floor Stand Mount

The *floor stand mount* is generally designed to telescope vertically between an adjustable height of 37—66″ (1—1.6 m). The base may be either solid cast, using its weight to stabilize the microphone (Fig. 5-5), or may be a tripod tubing arrangement (Fig. 5-6). Tripod models are often designed to fold down to save storage space.

The Boom-Mount Attachment

The *boom mount* is a counterbalanced arm assembly that attaches to the standard floor stand mount. This arm allows more flexible microphone placement because the arm can be adjusted up or down, swiveled horizontally, or telescoped in or out. The boom mount is available is varying sizes and lengths in order to fit the application and the desired extension or "reach."

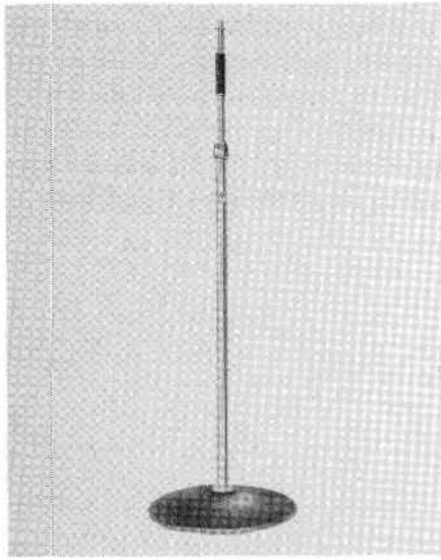


Fig. 5-5. Atlas Sound MS-12S automatic locking microphone floor stand. *(Courtesy of Atlas Sound)*

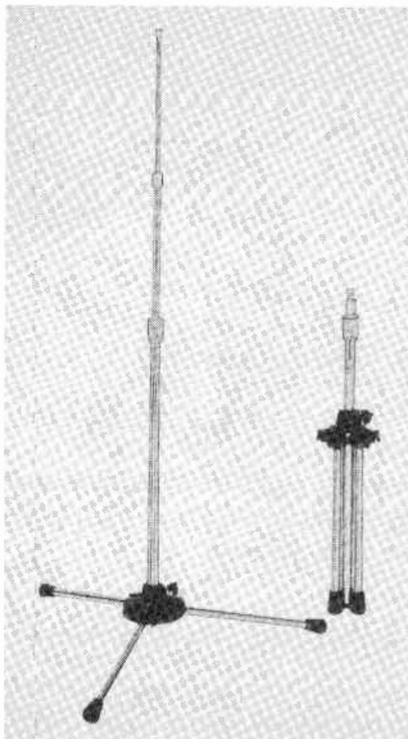


Fig. 5-6. Atlas Sound PS-C3 Porta-Series stand. *(Courtesy of Atlas Sound)*

The small boom (Fig. 5-7) is found in many studio and professional audio applications. With it, a microphone can be easily placed in a "hard-to-reach" position. The small boom mount permits drum-set miking in "tight fitting" positions between a myriad of stands and drum shells.

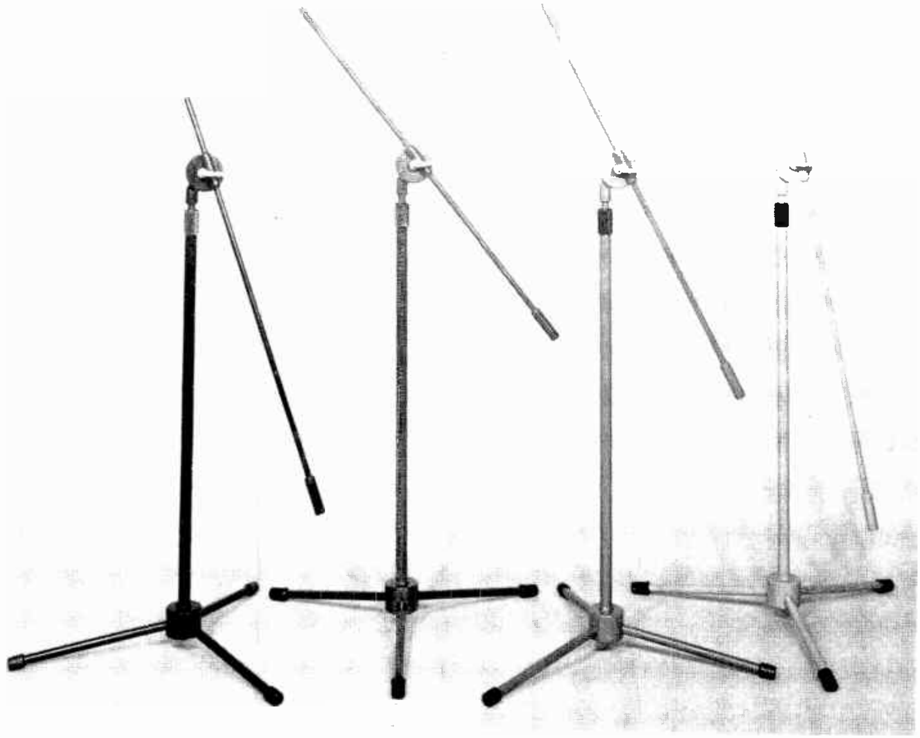


Fig. 5-7. Keith Monks MS/M with BA/M boom arm. *(Courtesy of Keith Monks Ltd.)*

The drum boom (Fig. 5-8) can suspend a microphone from a wall or ceiling when space is at a premium. The Keith Monks DB/1 and DB/2 drum-boom system allows 360° of coverage along a centrally supported stem.

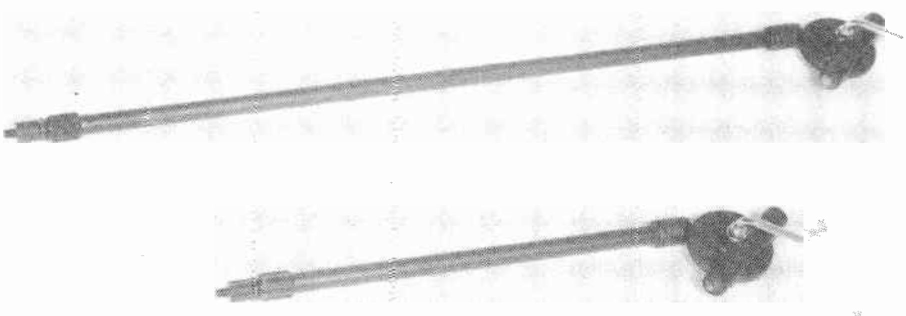


Fig. 5-8. Keith Monks DB/1 and DB/2 drum boom. *(Courtesy of Keith Monks Ltd.)*

The overhead boom stand and the "steerable" soundstage boom are larger microphone booms. The overhead boom stand (Fig. 5-9) is required for extended reach or height and is most commonly used in the large studio, audio-for-visual sound stage, and on-location for orchestral pickup. The "steerable" microphone boom can be used in the set of a sound stage, where on-camera dialogue must be followed by one or more overhead microphones. The boom lets the user steer the microphone toward the talent or the sound source. Larger sound-stage booms can be mounted on a driven, motorized dolly. Such booms are often fitted with automatic telescoping sections that can be operated remotely.

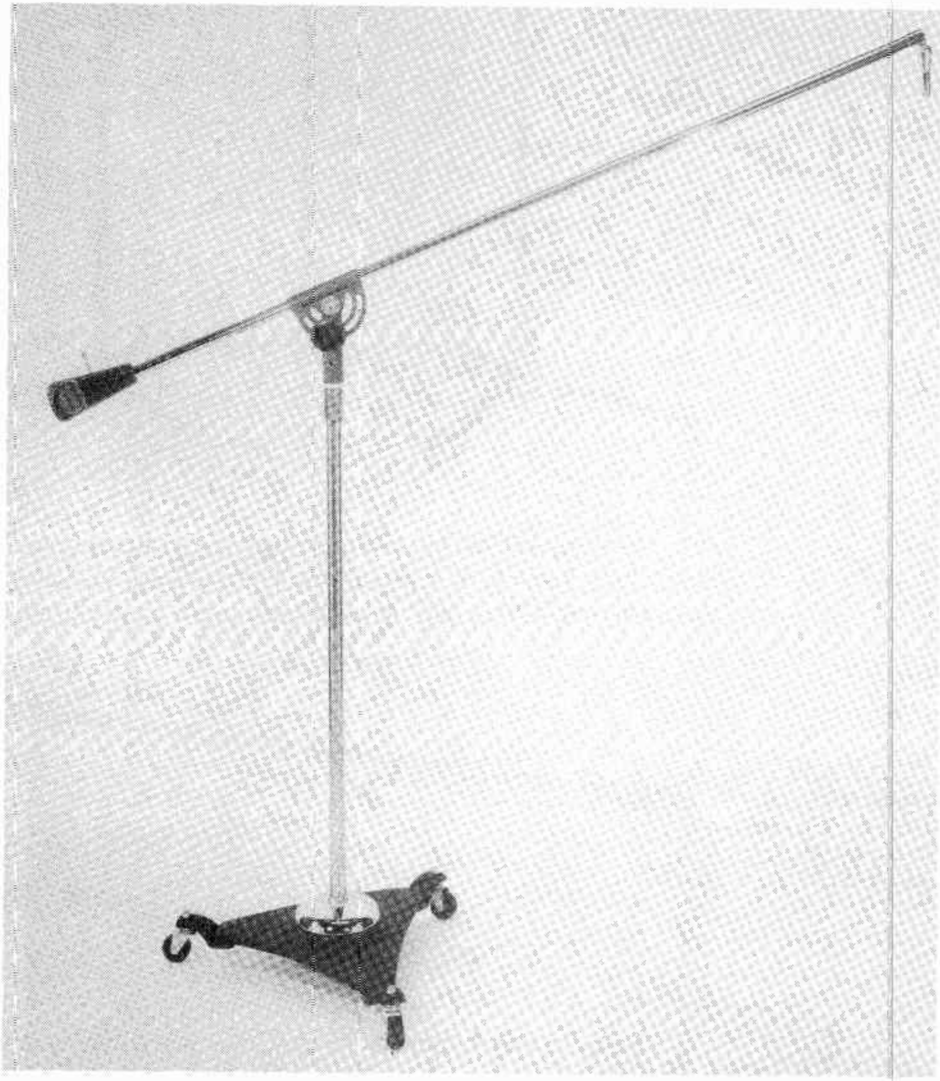


Fig. 5-9. Atlas Sound SB-100W overhead boom. *(Courtesy of Atlas Sound)*

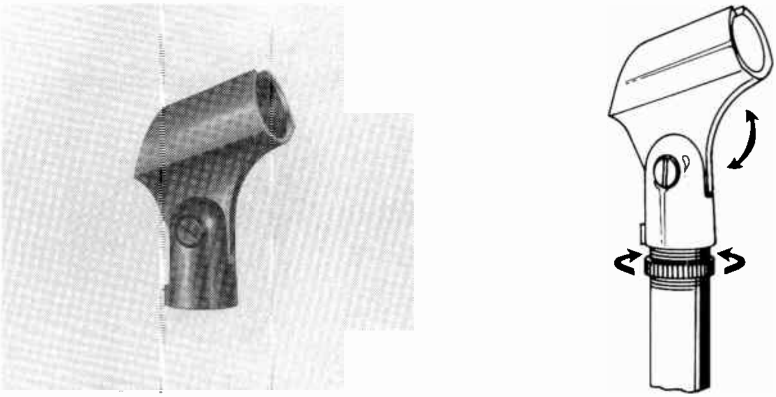
Microphone Stand Mounts

The microphone stand mount takes such forms as a microphone clamp or a shock mount.

Microphone Clamp

The *microphone clamp* (Fig. 5-10) attaches to the end threads of a microphone stand or boom, enabling quick and accessible mounting of the microphone. Most clamps allow the microphone to be tilted over a 90° angle independent of the stand or boom's angle, while the rotation or swivel of the clamp can be locked through the use of a lock-ring, which is attached to the stand threads (Fig. 5-11).

Fig. 5-10. Microphone clamp.



(A) Adapter. *(B) Detail drawing.*

Fig. 5-11. Tilt-and-lock assembly used by the average microphone clamp. *(Courtesy of Shure Brothers, Inc.)*

The popular adjustable microphone clamp (Fig. 5-12) is made of two spring-loaded armatures that adjust to fit most microphones. This contrasts with most clamps, which are designed for only one microphone diameter.

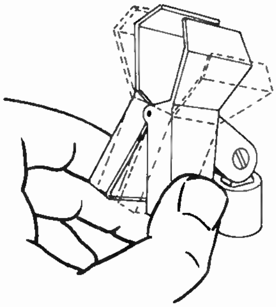


Fig. 5-12. Adjustable microphone clamp. *(Courtesy of Beyer Dynamic, Inc.)*

The dual microphone mount (Fig. 5-13) can be used to place two microphones on one stand or mount to ensure reliable coverage of a speech or multimedia event. Such mounts are also available for use with the lavaliere-type tie-clip microphone.

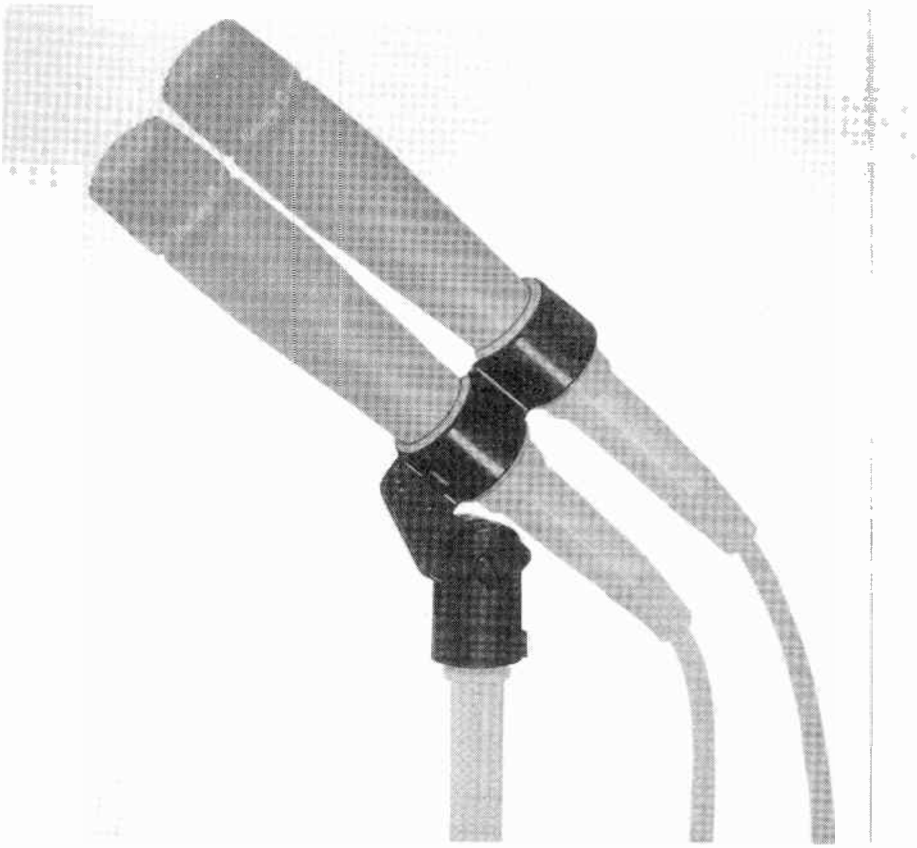


Fig. 5-13. Dual microphone mount. *(Courtesy of Shure Brothers, Inc.)*

Snap-On and Lock-On Stand Adapters

The *snap-on accessory* (Fig. 5-14) allows quick fastening or disconnection of the microphone holder or boom attachment. The lock-on accessory prevents unnecessary clamp and microphone detachment.

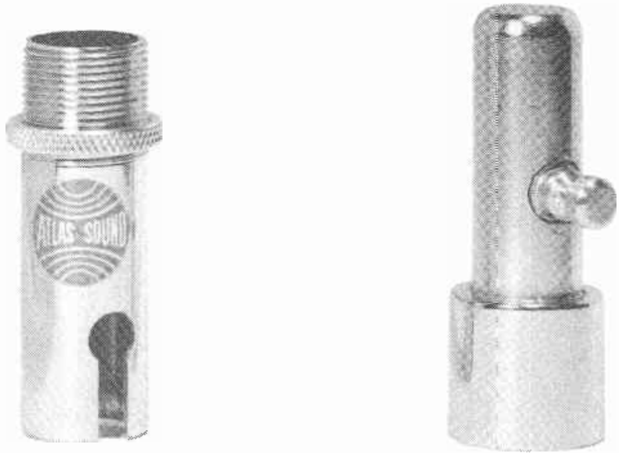


Fig. 5-14. Atlas LO-2B lock-on accessory. *(Courtesy of Atlas Sound)*

Side Clamp

The microphone-stand *side clamp* (Fig. 5-15) allows an additional microphone to be placed at any point along the stand length. For instance, such a clamp may be attached to a vocal stand, allowing a guitar microphone to be placed at the proper height, without the need for an additional microphone stand.

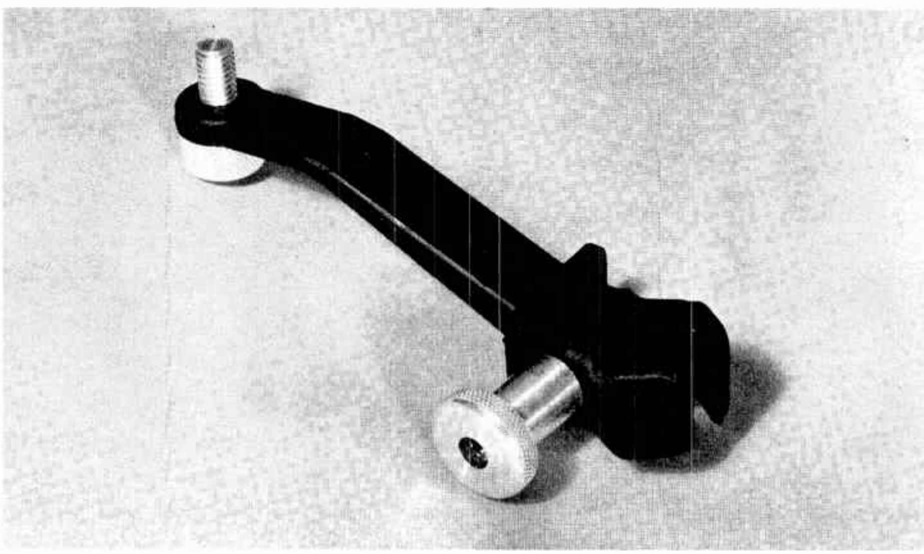
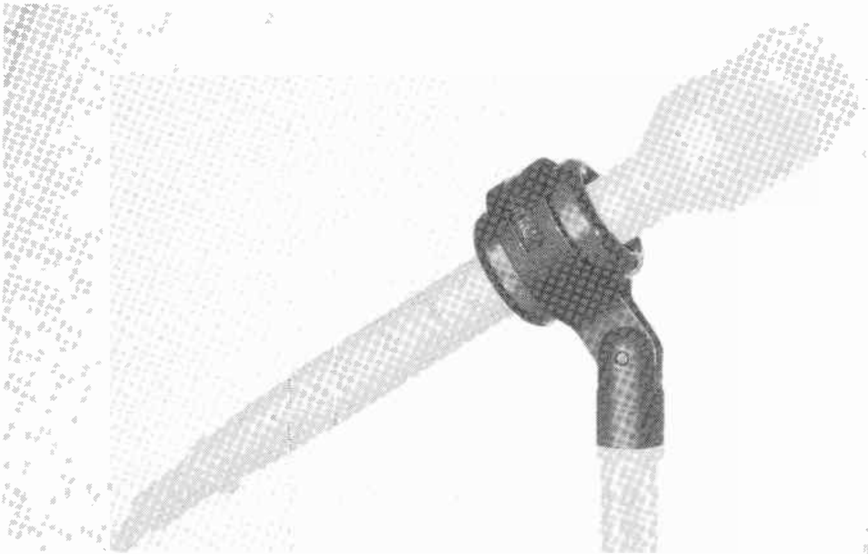


Fig. 5-15. Keith Monks SC1 ($\frac{3}{8}$" Whitworth) side clamp. *(Courtesy of Keith Monks Ltd.)*

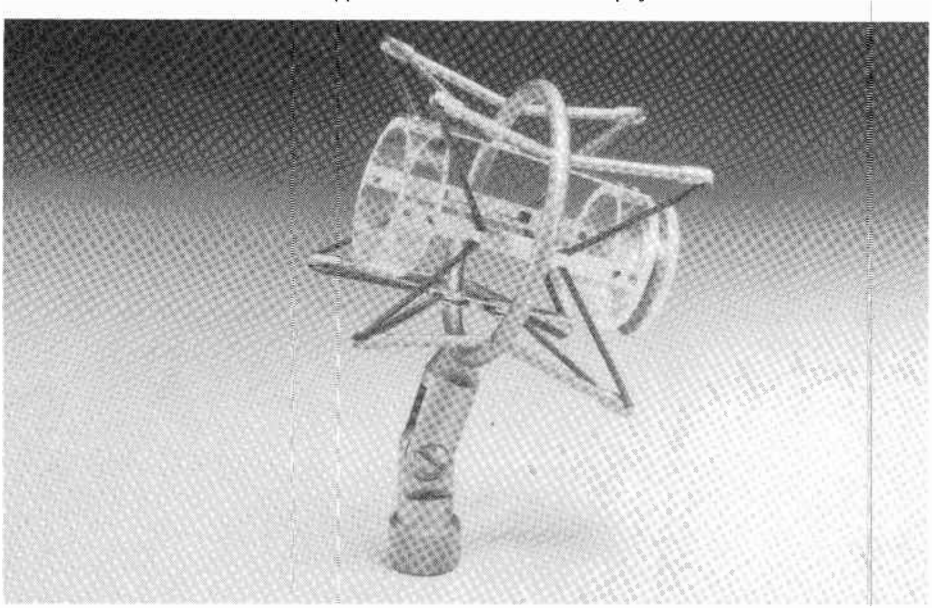
Shock Mount

The microphone *shock mount* (Fig. 5-16) is a microphone clamp that elastically isolates the microphone from the microphone stand. The purpose of the shock mount is to attenuate low-frequency rumble and vibrations which are otherwise transmitted from the base of the microphone stand directly to

Fig. 5-16. Microphone shock mount.



(A) Shure A55M "Shock-Stopper" isolation mount. (Courtesy of Shure Brothers, Inc.)



(B) Sanken S-41 Elastic suspension mount. (Courtesy of Sanken/Pan Communications, Inc.)

the microphone housing and capsule. Such rumble often occurs when the microphone-stand combination is placed upon a pliable surface (such as a wooden stage) where footsteps or dancesteps cause excessive vibration. This is particularly true of microphones with an extended low-frequency response. In Figure 5-17 the output of a direct-mounted condenser microphone (solid line) is compared to that of a shock-mounted condenser microphone (dashed line). The microphone directly mounted on the stand shows a high degree of low-frequency pickup, while the shock-mounted shows a transmission reduction of up to 35 dB.

When shock mounting a microphone, be sure to isolate the microphone cable as much as possible from the microphone boom, stand, or vibrating surface. This is necessary because the mike cable can transmit vibrations. When possible, create a strain relief (Fig. 5-18), such as a loop of cable, by clipping or taping the cable to the stand or boom. Where a high degree of motion vibration is encountered (as with the steerable boom), further isolate the microphone from the boom by using a short length of thin coiled microphone cable between the microphone and the boom tubing.

Microphone Stand Accessories

The Stereo Bar

The *stereo bar* (Fig. 5-19) allows you to mount two microphones on any microphone stand for stereo miking, without the need for an additional stand mount.

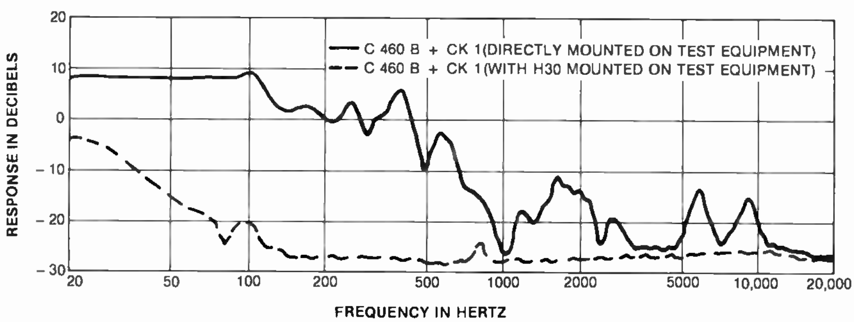


Fig. 5-17. Damping response of the AKG H-30 elastic suspension shock mount. *(Courtesy of AKG Acoustics, Inc.)*

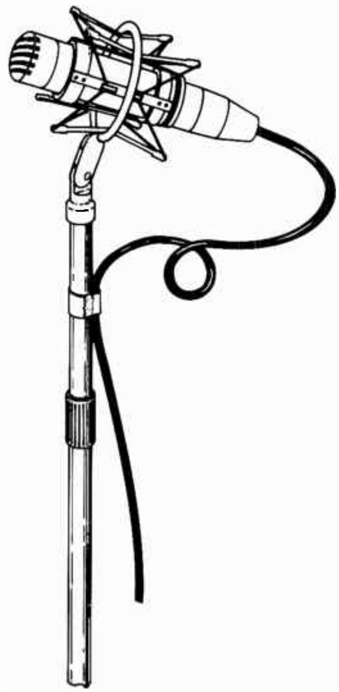
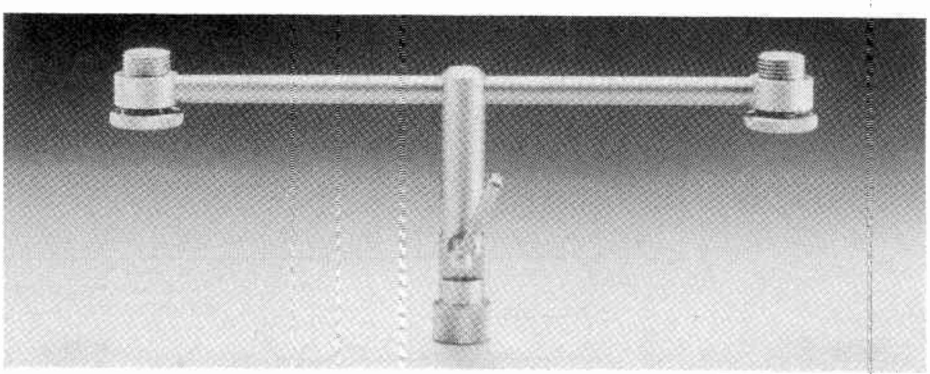


Fig. 5-18. Shock mount cable relief.

Fig. 5-19. Stereo bar.



(A) Sanken X-Y adjustable stereo bar. (Courtesy of Sanken/Pan Communications, Inc.)

(B) Atlas TM-1 two/three microphone bar. (Courtesy of Atlas Sound)

The Gooseneck Adapter

The *flexible gooseneck* (Fig. 5-20) adapter allows a high degree of mobility in microphone placement, as the spring-wound metal adapter can be easily flexed into almost any convenient position.

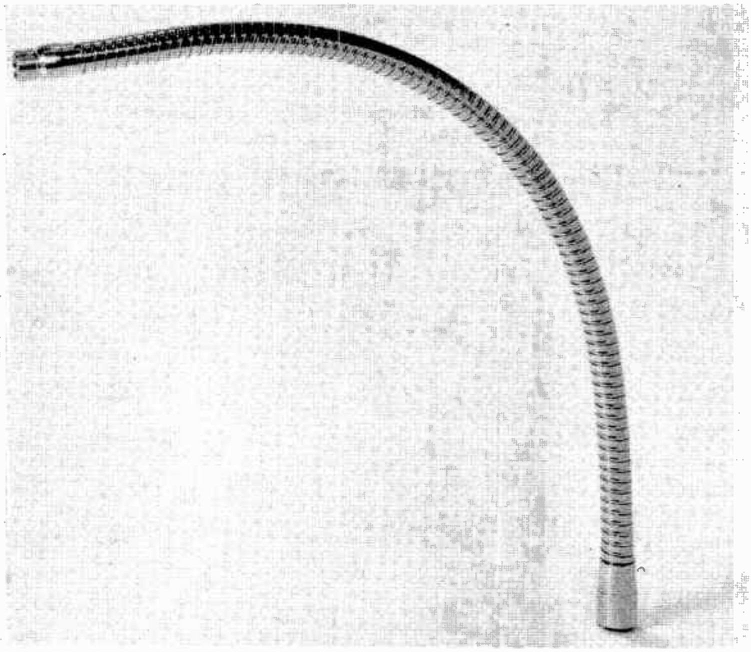


Fig. 5-20. Gooseneck adapter. *(Courtesy of Atlas Sound)*

Windscreens

Most microphones are susceptible to wind noise. This is especially true of directional microphones, in which large differences in pressure build between the front and rear of the diaphragm. When wind noise is excessive, the most common solution is a *windscreen*, a foam or fabric screen that resists bursts of wind and its noises, while keeping the attenuation of audible frequencies to a minimum. The most popular windscreens are made of open-cellular foam (Fig. 5-21) that has been dipped in an acid solution to etch away any honeycomb cells that would impede the flow of high frequencies.

A style of windscreen that is returning to style is the "nylon stocking" windscreen (Fig. 5-22). These may be bought or they can be made from a stocking and hoop, such as a crochet hoop or clothes hanger. They are mounted on another mike stand and positioned between the sound source and the microphone.

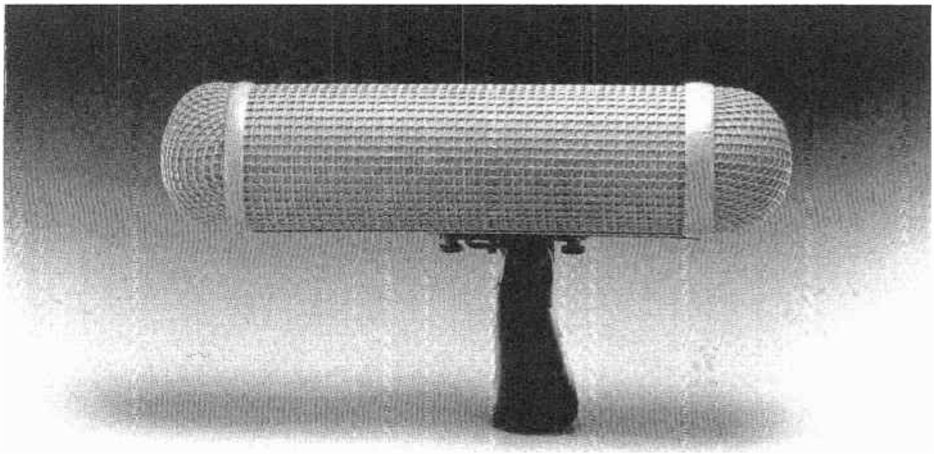
Fig. 5-21.
Open-cell foam windscreen.

(A) WS-30 foam windscreen. (Courtesy of Sanken/Pan Communications, Inc.)

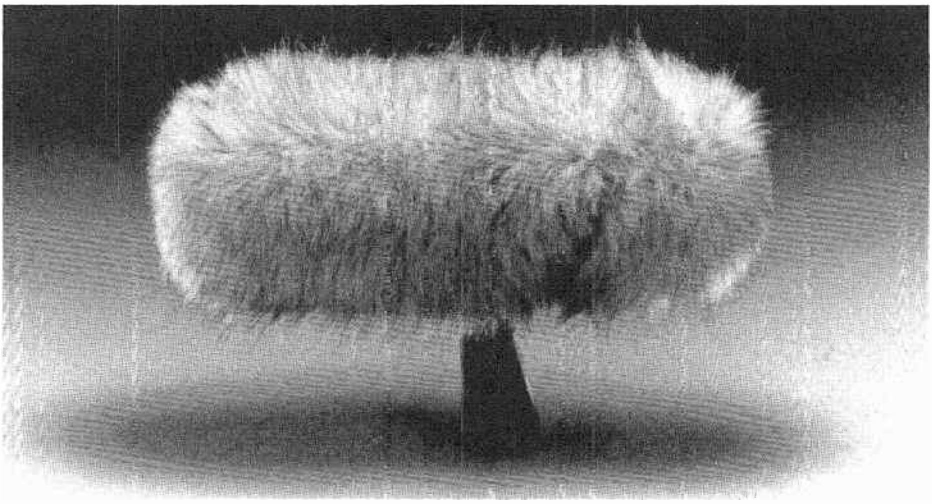
(B) A53WS foam windscreen. (Courtesy of Shure Brothers, Inc.)

Fig. 5-22.
"Nylon stocking" windscreen.
(Courtesy of The Stocking Screen)

The basket-type windscreen (Fig. 5-23) is often used to protect the overall surface of an ultradirectional or shotgun microphone. These highly effective shields can be easily adapted for use with a pistol grip handle, fishpole, or boom mount.



(A) Sanken WS-7 Basket-type windscreen.



(B) WS-7 "Windjammer" windscreen.

Fig. 5-23. Basket-type windscreen. *(Courtesy of Sanken/Pan Communications, Inc.)*

6 Fundamentals of Single-Microphone Techniques

Microphone Placement

The choice of microphone and its placement are of prime importance as working tools for the professional sound engineer or recordist. There is a great variety among mike designs, operating principles, and applications, with each microphone configuration yielding a different sonic character.

Microphone placement is a subjective art form. Although guidelines have their place in the professional media, what may be presently considered as bad technique could easily be the best solution to a particular application in the future. As new music styles and equipment develop, sound recording techniques will evolve (or come back into vogue), giving the recorded sound a new and interesting sonic character.

This chapter covers basic guidelines of microphone placement and describes the many choices of single-microphone technique that are used today. By the single-microphone technique, we refer to the pickup of one sound source by a singular, monaural ("mono") device. We'll examine additional techniques, in the form of stereo placement, in Chapter 7.

Sound Characteristics as a Function of Working Distance

In modern studio and sound-stage recording, four basic styles of microphone placement are directly related to the distance of a microphone from its sound source: distant miking, close miking, accent miking, and ambient miking.

Distant Microphone Placement

Distant miking refers to the positioning of one or more microphones about three or more feet (1 m) from the sound source. This technique serves two functions:

- To place the mike at such a distance that an entire musical instrument or ensemble is picked up, thus preserving the overall tonal balance of that instrument or ensemble (Fig. 6-1). A natural tonal balance may often be achieved by placing the microphone at a distance roughly equal to the size of the sound-radiating portion of the instrument or sound source.
- To place the mike such that the acoustic environment is included within its pickup and thus is combined with the direct signal of the sound source (Fig. 6-2).

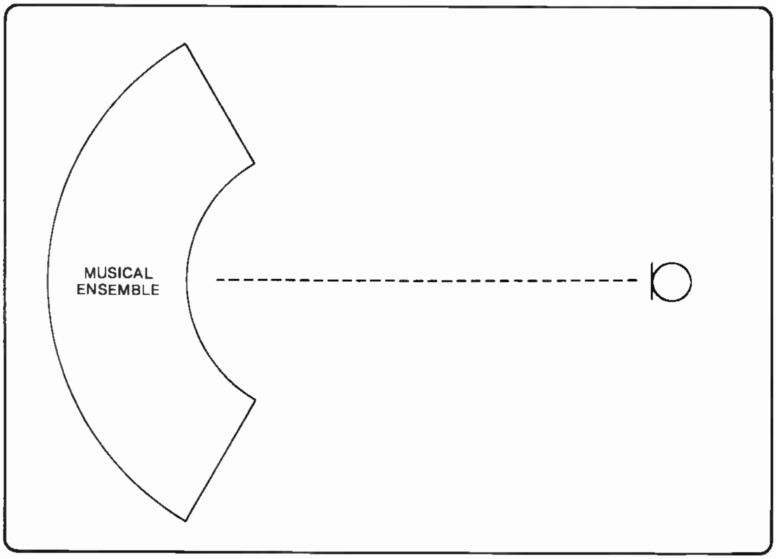


Fig. 6-1. Example of an overall distant pickup.

Distant miking is often used in the pickup of a large instrumental ensemble (such as a symphony orchestra or choral ensemble) and thus relies heavily upon the quality of the acoustic environment. In such a situation, the microphone is placed at a distance that strikes an overall pickup balance between the ensemble and environmental acoustics. This balance may be determined by a number of factors, including the size of the instrument or sound source and the reverberant characteristic of the room.

Distant miking techniques tend to add a live, airy, or open feeling to a recorded program because distant microphones cover a larger incident angle

(Fig. 6-3), allowing for fewer microphones to be required in the successful pickup of a sound field.

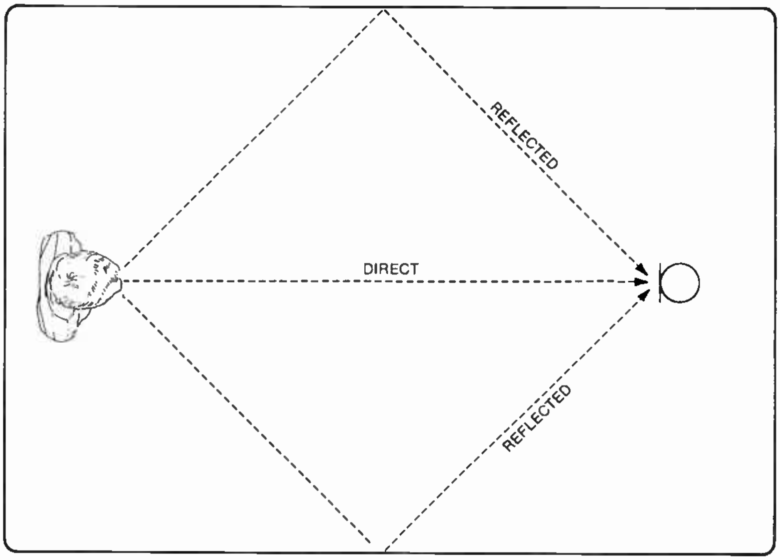


Fig. 6-2. Distant microphone receives both direct and reflected sounds.

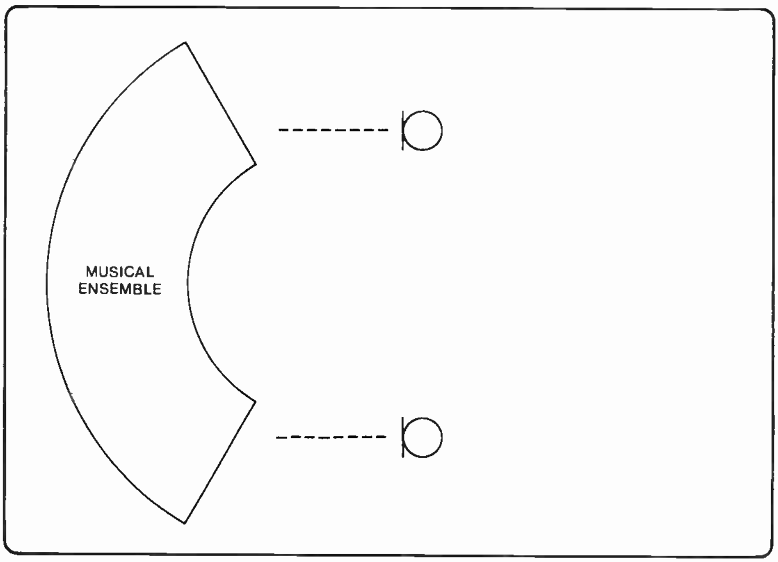


Fig. 6-3. Distant pickup using two microphones.

A disadvantage of the distant microphone technique is the potential pickup of poor hall or studio acoustics. Improper or bad room reflections are picked up, often creating a muddy or poorly defined recording. To avoid this liability, the recordist can:

- temporarily correct for reflections by using absorptive or offset reflective panels
- place the microphone closer to its source and add artificial reverberation

The Effects of Boundary Interference on Distant Miking

Another disadvantage of distant miking is phase cancellations due to boundary reflections. These cancellations result in a hollow sound and are caused by sound travelling to the microphone via two paths: directly from the sound source and reflected off a nearby boundary surface.

In Figure 6-4, the sound that is reflected off the floor travels further than the sound that reaches the mike directly. Specific frequencies within the audio range for which this extra path length is one-half of a wave length (or an odd-integer multiple thereof) arrive 180° out of phase with respect to the direct sound. This phase interference may produce severe dips of up to 15 dB in the frequency response of the microphone.

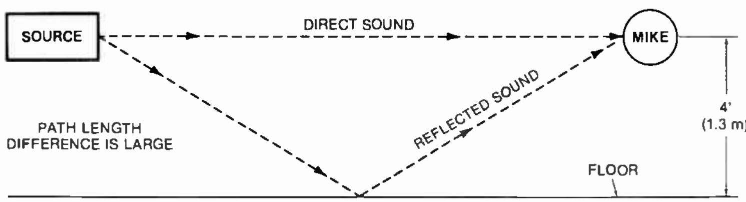


Fig. 6-4. Cancellations occur as a result of differing direct and reflected acoustic path lengths.

These cancellations may be reduced or eliminated by placing the microphone very near the boundary, thus reducing the difference in relative path lengths (Fig. 6-5). A mike height of $\frac{1}{16}-\frac{1}{8}''$ (1.5 − 3 mm) keeps the lowest cancellation frequency above 10 kHz. Boundary microphones operate on this principle by placing the microphone diaphragm well within these height restrictions.

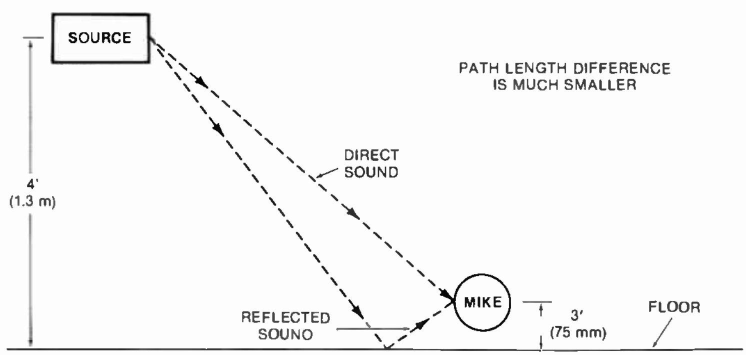


Fig. 6-5. Placing microphones close to boundary raises lowest frequency of cancellation.

The placement of a mike close to a reflecting surface has the added advantage of increasing its output by 6 dB. Experiments have shown that cancellations are a problem whenever the mike-to-source distance is greater than one or two times the distance of the source to the reflecting surface. The effects of these cancellations also depend on the ratio of direct-to-reflective pickup and the reflective qualities of the nearby boundary surface.

Close Microphone Placement

Close microphone techniques refer to placing the mike from about 1″ to 3′ (25 mm to 1 m) away from the sound source. This technique is often used in multitrack music production and in audio production for video. Close miking serves two major functions:

- to create a tight, present sound quality
- to effectively exclude the acoustic environment from being picked up

Because sound diminishes with the square of its distance, a sound originating 6′ (2 m) from a mike will be insignificant in level when compared to that of the same sound originating 3″ (7.5 cm) from the microphone (Fig. 6-6). As a result, only the desired on-axis sound will be recorded on tape: extraneous sound will not be picked up by the microphone, for all practical purposes.

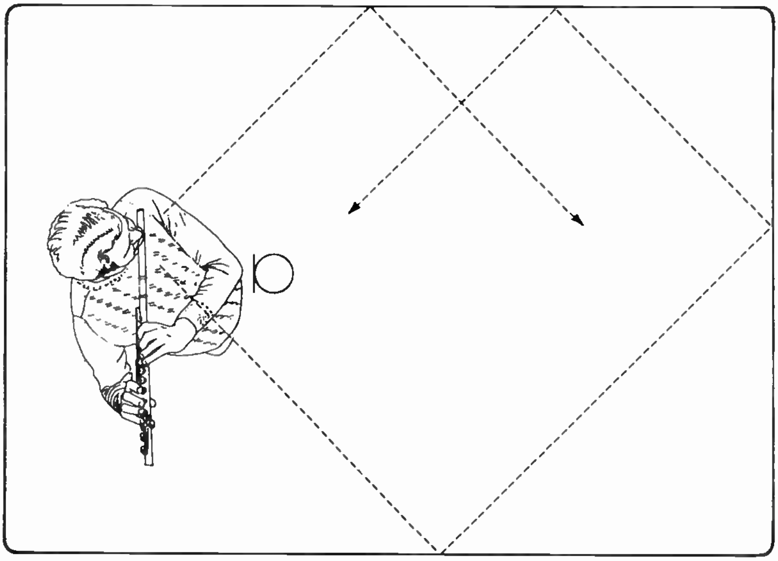


Fig. 6-6. Close miking reduces the effects of the acoustic environment.

Although close miking offers several advantages, a microphone should be placed only as close to the source as is necessary, not as close as is possible.

Miking too close may color the recorded tone quality of the source. Since such techniques commonly involve distances of 1″ to 6″ (2.5 to 15 cm), the *entire* tonal balance (timbre) of the sound source may not be picked up. Rather, the mike may be placed so close to the source that only a small portion of the surface may actually be "heard," giving an area-specific pickup balance. At these close distances, moving a mike only a few inches may change the overall tonal balance of the pickup. Three remedies are available (Fig. 6-7):

- equalizing the signal until a desired balance is achieved
- moving the microphone along the surface of the sound source until a desired balance is achieved
- placing the pickup a greater distance from the sound source to allow for a wider pickup angle and thus an improved blend of the entire sound source

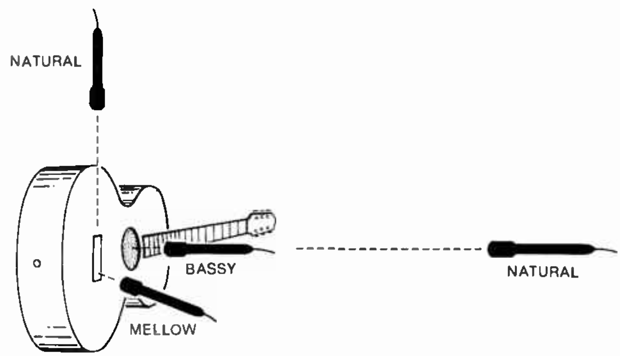


Fig. 6-7. At close working distances, moving a mike by only a few inches may affect the overall tonal balance of a pickup.

The Effects of Boundary Interference at Close Distances

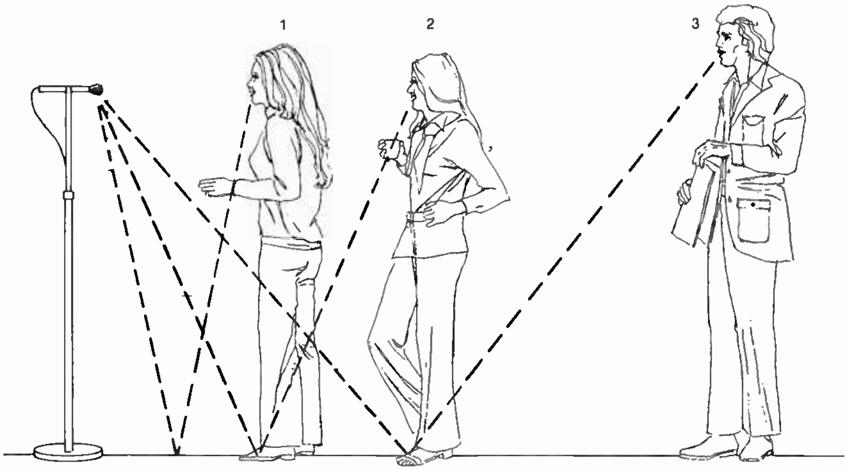
As with distant microphone techniques, the effects of phase cancellations as a result of reflections from nearby boundaries may affect the quality of a signal pickup at close working distances.

In typical applications, the use of a microphone stand places a microphone from 3′ to 6′ (1 to 2 m) off the floor (Fig. 6-8). At a greater distance, the effects of frequency cancellation, as a result of the differing direct and reflected path lengths, may affect the resultant frequency response. However, as this source-to-microphone distance is being reduced, the reflected sound path will have an increasingly negligible effect upon sound quality.

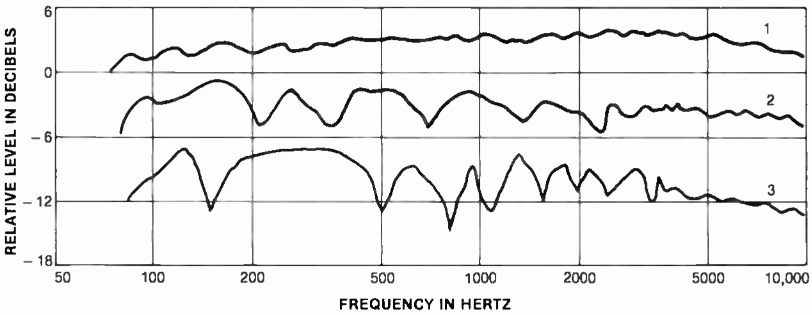
Commonly, a podium pickup requires that a microphone be mounted on a stand above the podium or on a similar desktop surface (Fig. 6-9A). In this setting, the effects of phase cancellation are great indeed. To avoid such cancellations, the microphone needs to be lowered nearly flush with the podium in order to place the cancelled frequencies above the audible range (Fig.

6-9B). The directional boundary microphone may be the ideal choice to solve such close-range boundary problems.

Fig. 6-8. Boundary interference problems associated with typical floor stand heights.



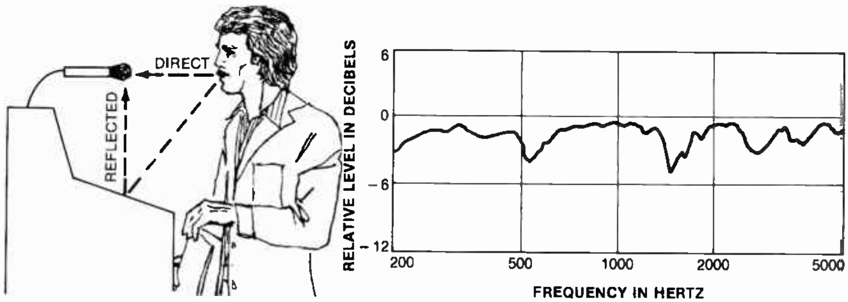
(A) Cancellation due to varying microphone—source path lengths.



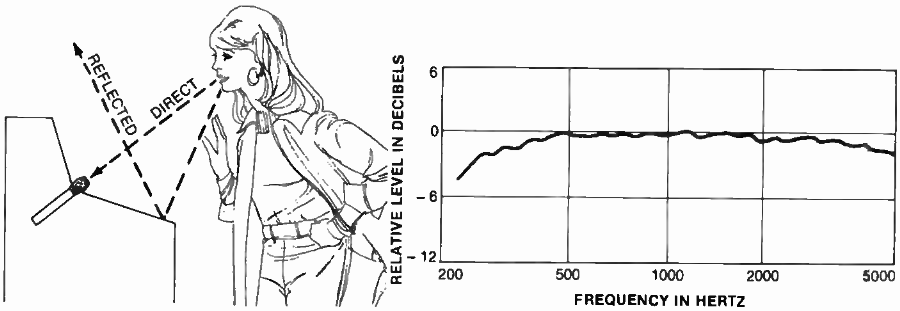
(B) Resultant response curve.

The Effects of Multiple-Microphone Interference

The premise of the multitrack recording process is to get control over each sound source by maintaining a moderate-to-high degree of isolation between sources. Should an extraneous instrument or sound be picked up by a mike that is recording an adjacent source, a condition known as *leakage* occurs (Fig. 6-10). Since the pickup contains a mixture of direct and indirect signal, getting control over this signal track in the mixdown phase would be difficult to achieve without also affecting the level and sound character of the indirect source. Because excessive leakage tends to make a soundtrack more live and less intelligible, unwanted leakage is a condition to be avoided. Whenever this effect is wanted (for a live recorded sound), exercise caution because the recorded tracks can easily take on the negative aspects of leakage.



(A) Cancellation effects of mounted pickup above surface of podium.



(B) Effects of flush-mounted pickup.

Fig. 6-9. Phase cancellations involving podium or similar desktop.

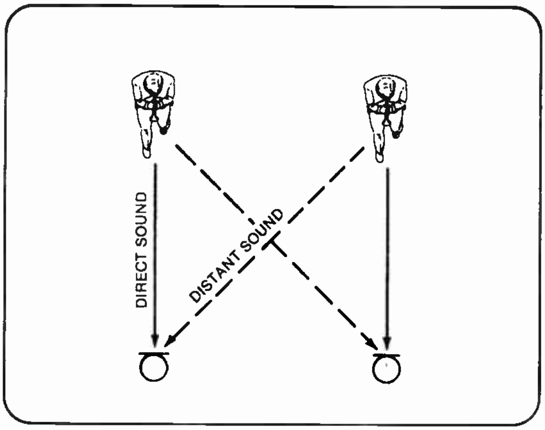
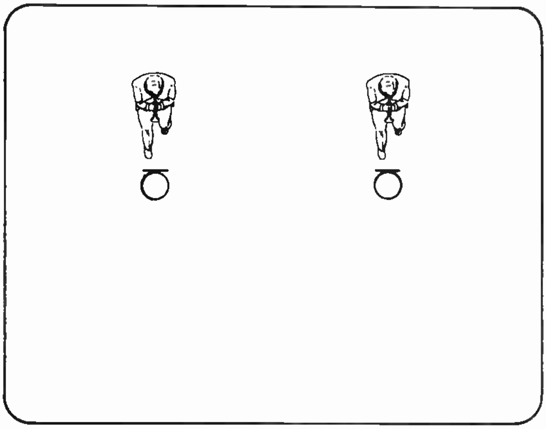


Fig. 6-10. Leakage due to direct and indirect signal pickup.

Several corrective options exist for the leakage problem:

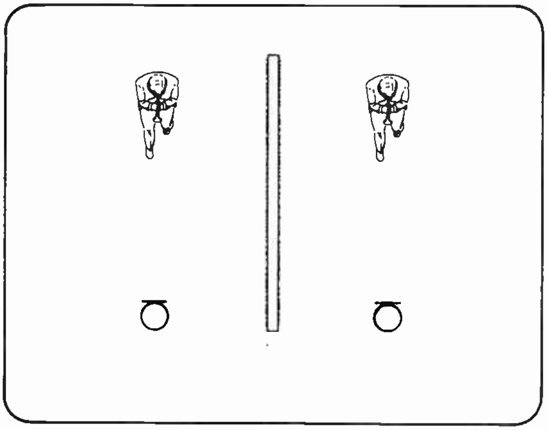
- In moderation, physically separate the involved sound sources. This works because sound (and thus leakage) drops 6 dB in level as miking distance is doubled.

- Place the involved microphones closer to their respective sound sources (Fig. 6-11A). In this manner, the desired pickup signal increases relative to the unwanted leakage. Take care to maintain the proper tonal balance of the sound source as this distance is decreased.
- Use directional microphones. A directional microphone has maximum sensitivity when placed on-axis to the desired sound source, while discriminating against unwanted off-axis leakage. In this manner, the desired source-to-leakage ratio is increased.
- Isolate the sound sources by placing an acoustic barrier known as a *flat* or *gobo* between them (Fig. 6-11B). Placing the objectionable



(A) Place mikes closer to their sources.

Fig. 6-11. Methods of reducing leakage.



(B) Use acoustical barrier.

sound source in another room, often known as an isolation ("iso") booth, achieves the same result.

In addition to these options, the multitrack audio engineer may decide to eliminate the obtrusive acoustic source by (1) recording its signal directly onto tape and bypassing the use of a microphone (if the source provides an electric signal) or (2) by later overdubbing the louder instrument onto a separate track of the tape recorder. Both options reduce leakage, as the softer instrument can be recorded without detrimental acoustic interference.

More than one microphone is often used to pick up a single sound source (as with the multimike setups associated with the drum set). This technique often produces a larger-than-life effect and improves the stereo spread.

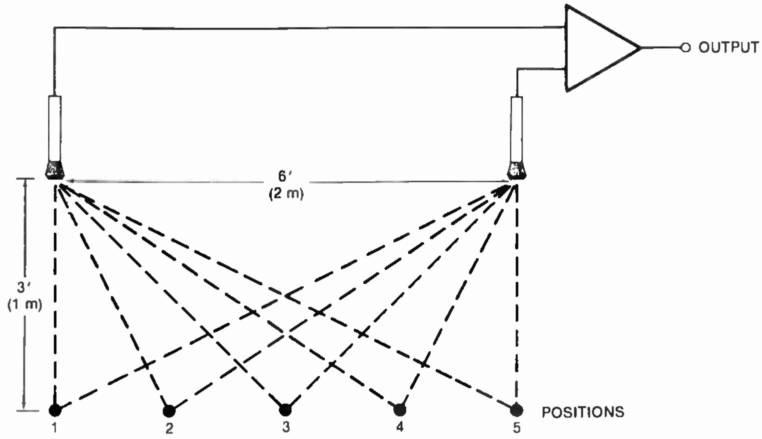
When using this technique, take care that the microphones are electrically in phase with each other. If this is not the case, signal cancellations at various frequencies may occur, changing the volume and/or the tonal balance of the pickup. In the case of a microphone pair that is wired out of phase (opposite polarity) or one that receives out of phase information, a listener may hear the recorded left and right signals in stereo. But a listener monitoring in mono (over a car radio, for example) will hear very little of the recorded signal because two out-of-phase signals tend to cancel. A simple check for an improper polarity condition is to assign the output from both (or all) of the mikes to one loudspeaker and listen for changes in the tonal character of the sound.

If one or more microphones are 180° out of phase with each other, the problem may be that a cable is miswired. This may be solved by:

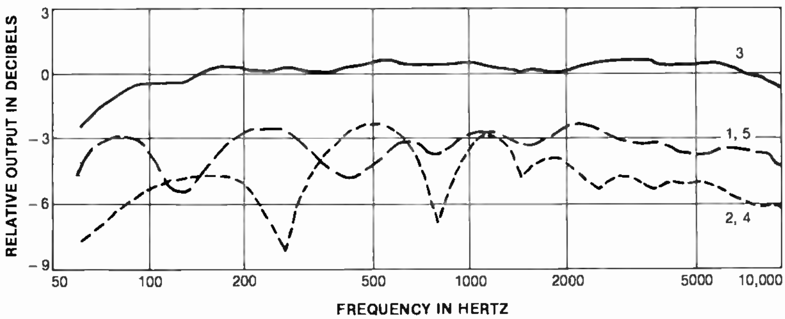
- replacing the cable with one that is properly phased
- repairing the cable to its proper polarity of pin 2 (+) and 3 (−) on the XLR-type connector by inserting a phase-reversal adapter
- reversing the connections to pins 2 and 3 in one XLR-type connector
- reversing the polarity on the console's input strip if a polarity switch is available

Acoustical phase-cancellation problems may also be produced by using too many microphones too far from the sound source, a condition that allows both excess leakage and a degraded pickup response. Should a single sound source be picked up by two nearby microphones at roughly an equal intensity, variations in a phase could result from the differing path lengths from the source to each microphone. Should these signals be combined into a single channel at any point, as often occurs in the mixing stage, cancellations may occur, producing severe frequency-response dips in the pickup of the sound source (Fig. 6-12).

Fig. 6-12.
Variations in pickup response due to phase cancellation.



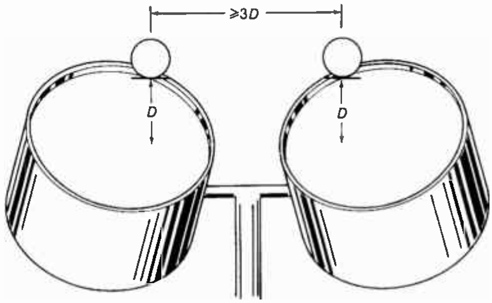
(A) Variations with microphone/source path lengths.



(B) Resultant response curve.

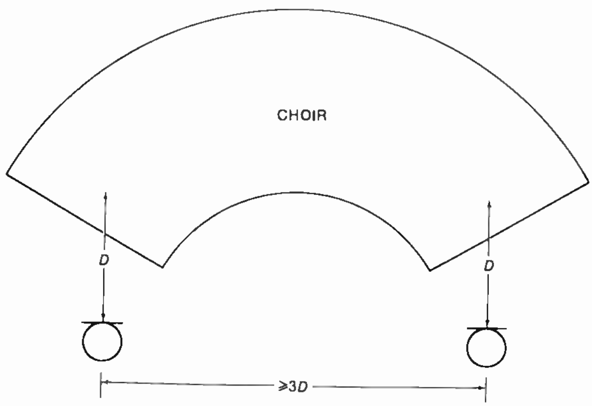
In order to avoid such acoustical phase cancellations under the majority of miking conditions, the *three-to-one principle* may be employed. This principle states that to maintain phase integrity between two or more microphones, for every unit of distance between each mike and its source, the distance between mikes should be at least three times that mike-to-source distance (Fig. 6-13).

Quite often, it's unnecessary to use more than one microphone in a pickup area where a single microphone might provide adequate, if not improved, coverage. In Figure 6-14A, a speaker's podium is shown, covered by two widely spaced microphones. This condition degrades the frequency response due to the movement of the lecturer and the resultant changing path lengths to each microphone. Figure 6-14B shows a single directional microphone providing an equal amount of coverage without phase-related problems.



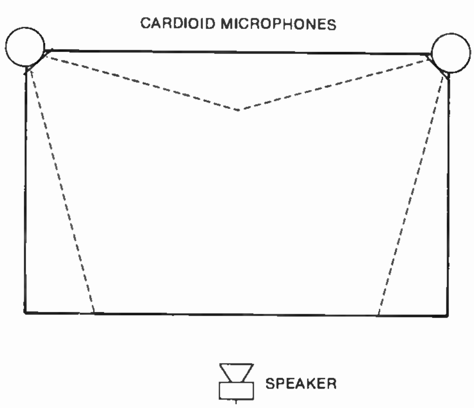
(A) Pickup of two tom-tom drums.

Fig. 6-13. 3-to-1 principle.

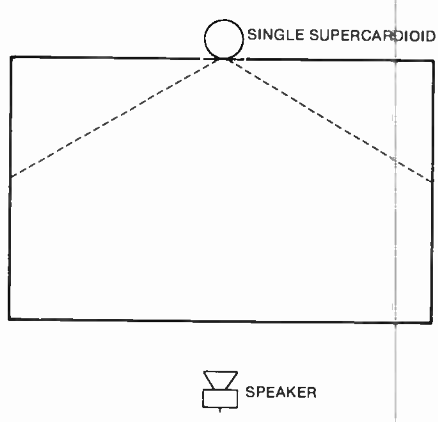


(B) Pickup of choir.

Fig. 6-14. Microphone coverage of podium.



(A) Improper coverage using spaced microphone pair.



(B) Proper coverage using single microphone.

Accent Microphone Placement

As we saw earlier, distant and close miking yield entirely different pickup and tonal qualities. Under certain circumstances, it is difficult to obtain a natural recorded balance when mixing these two techniques. For example, the classical symphonic styles of recording rely upon distant mike techniques for providing a natural balance between direct and reverberant (ambient) sound pickup. Should a solo instrumental passage occur within the score, an additional microphone may be required for improved coverage of that instrument, providing both volume and presence. Should this solo instrument be miked too closely, though, the pickup of the solo would sound too present and "out of context" with the overall distant pickup (Fig. 6-15A). To avoid this pitfall, a compromise in distance (and thus pickup balance) must be struck between close and distant placement. A microphone that is placed within this compromised range is known as an *accent microphone* (Fig. 6-15B).

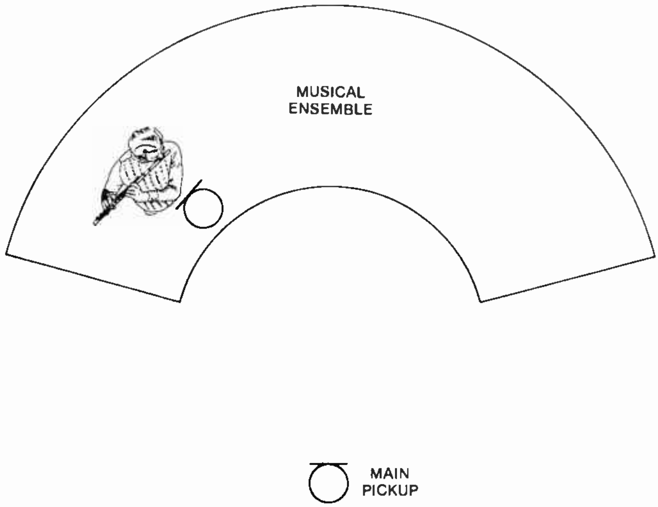
When employing an accent microphone, choose and place the microphone carefully. The amount of accent-mike signal introduced into the mix is also important, as this should not discolor or change the balance of soloist to surrounding instruments within the stereo perspective. A good accent-microphone technique only adds presence to the sound of a solo passage; the accent mike is not perceptible as a separate pickup. The proper *panning* of an accent microphone into the overall stereo image helps eliminate any wandering images that may occur with changes in solo intensity.

An accent microphone is placed significantly closer to the sound source than the overall distant microphone. This introduces a time discrepancy, such that the overall recorded sound plays the accent (solo) information before the overall signal is heard. This makes the accent mike sound closer to the source than it should be. The most recent means of compensating for this time discrepancy is to delay the signal from the accent microphone with a digital delay device. This ensures that the delayed signal is placed into the mix at the same time or slightly later (i.e., 10–16 ms) than the overall pickup signal.

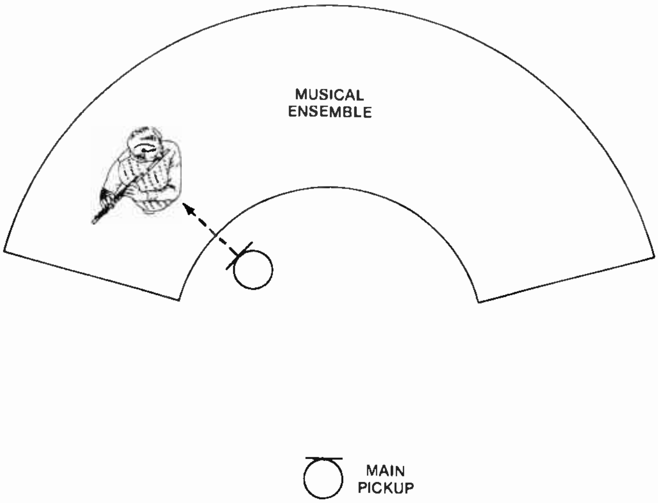
Ambient Microphone Placement

When a microphone is placed at a distance, such that the reverberant or room sound is predominant to the direct signal, that microphone is said to be an *ambient* microphone. The ambient microphone is most often omnidirectional (Fig. 6-16A); however, a cardioid polar pattern may be chosen to face away from a sound source in order to keep the direct signal to a minimum (Fig. 6-16B). In this manner, the reverberant or audience pickup may concentrate solely upon the ambient signal. Often the ambient pickup is a stereo pair, and it is usually mixed in with close-placed microphones, adding a spacious and more reverberant atmosphere to the sound source being recorded.

Fig. 6-15. Placement of accent microphone.



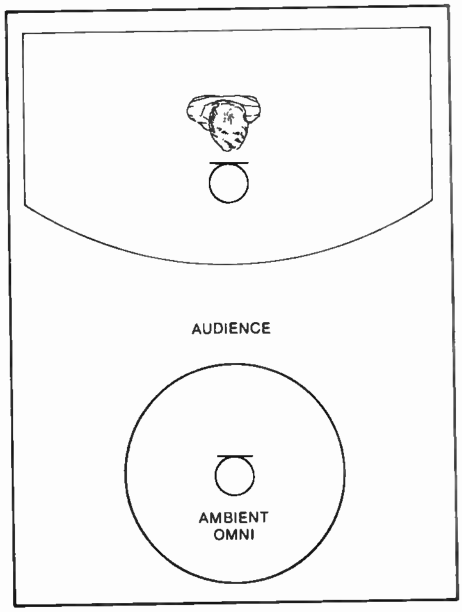
(A) *Microphone placed too close.*



(B) *Microphone placed at proper, compromise distance.*

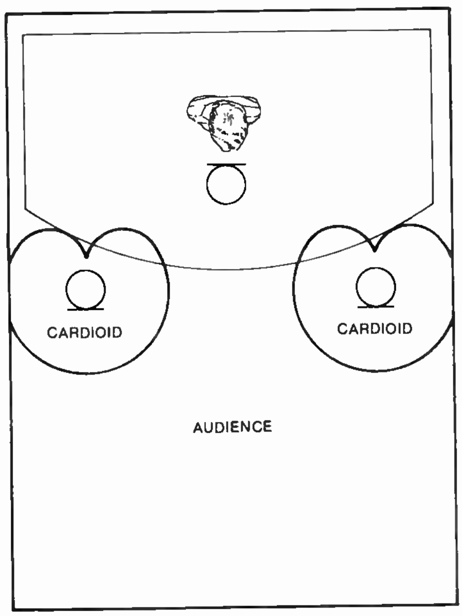
An ambient microphone can be used in three situations to enhance a recorded pickup:

- In a live concert recording, ambient microphones may be placed in the hall to restore the natural reverberation that might be lost with the use of close miking techniques.
- In a live concert recording, ambient microphones may be placed over the audience as a pickup for audience reaction and applause.



(A) Omnidirectional pickup of room reverberation.

Fig. 6-16. Ambient microphone.



(B) Cardioid pickup of audience response.

- In a studio recording, ambient microphones may be employed to add studio acoustics to a recorded sound. Ambient miking, when used as an effect, can add depth to a sound. For example, if a recorded sound is played loudly over a set of studio/stage monitors and is then rerecorded by a distant pair of mikes, the effect may yield a fatter, deeper sound on its own or when mixed with the original signal source.

Although we've dealt separately with distant, close, accent, and ambient microphone placement techniques, any of these techniques may be mixed and matched, as long as the production style fits the type of music or other source being recorded. For example, distant microphone techniques are not, by any means, limited to use with classical music, as is sometimes believed; they may be used in many circumstances, ranging from overall pickup of crowd voices to the larger-than-life, driving sounds of a distant drum in a multitrack recording studio. Additionally, a jazz flute effect may sound interesting whether it is recorded in the studio at a distance of 1′ or within the Taj Mahal at 50′. Through experimentation, the mix of these techniques can create a unique effect, given proper forethought as to production style.

The Boundary Microphone

As we have observed, when a microphone is placed near a reflective surface, the sound travels to the microphone by two paths: directly from the sound source to the microphone, and reflected off a large, nearby surface. Note that the reflected sound will travel a longer path than does the direct sound, such that the reflected sound is delayed relative to the direct sound. Sounds traveling these two paths are then combined at the microphone diaphragm.

All frequencies in the reflected sound path will be delayed by an equal length of time. Having a constant ratio of time delay effectively creates a different degree of phase cancellation at each frequency since various frequencies have differing wavelengths. For example, a time delay of 1 ms will cause a 360° phase shift for a 1000-Hz wave, but only a 180° phase shift for a 500-Hz wave (Fig. 6-17). At frequencies where the direct and delayed sounds are in phase (coherent), the signals add together, doubling the pressure and boosting the amplitude by 6 dB. At frequencies where the direct and delayed signals are out of phase, the signals cancel each other, creating a dip or notch in the response. The result is a series of peaks and dips in the net frequency response known as a *comb-filter effect* (Fig. 6-18). This bumpy frequency response has the effect of coloring the tonal reproduction, giving an unnatural sound. To solve this problem, the delay of the reflected sound needs to be shortened so that it arrives at the microphone at the same time as the direct sound.

Should a microphone be placed very close to and parallel with the nearest large, reflective surface (primary boundary), the direct and reflected waves will arrive at the diaphragm at the same time. Such a technique is known as the *boundary* or *pressure zone recording process* (Fig. 6-19). The pressure zone is defined to be the 0.04″–0.08″ (1–2 mm) region within which the mi-

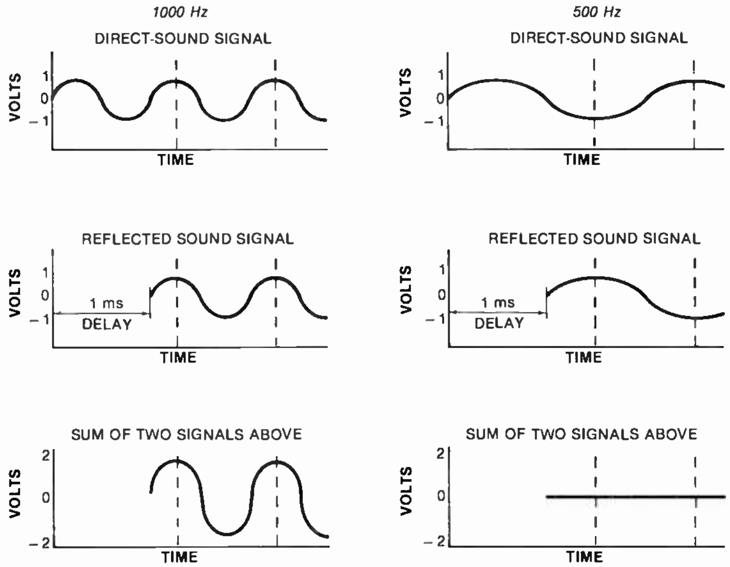


Fig. 6-17. Example of wave addition and cancellation at 1000 Hz and 500 Hz.

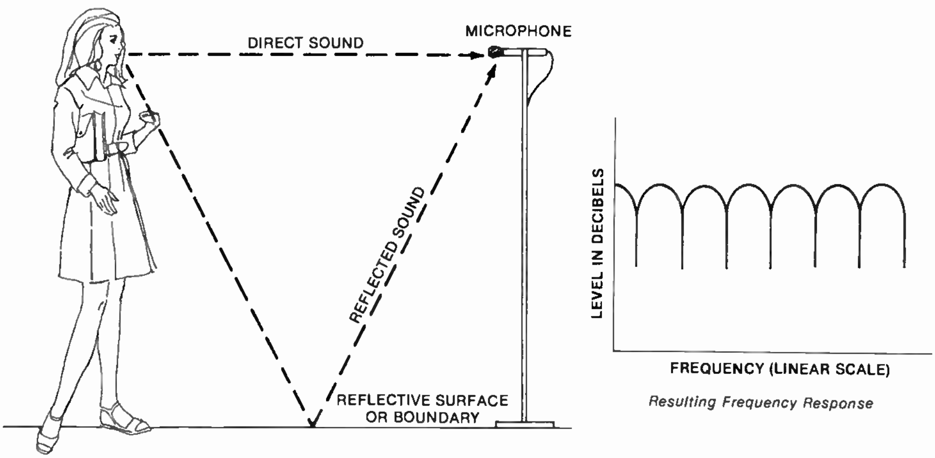


Fig. 6-18. Resulting frequency response from microphone receiving direct and delayed sound from one source.

crophone diaphragm must be placed in order to achieve a desired high-frequency response without adding adverse effects. Within this zone, may phase cancellations will be moved out of the audible frequency band, resulting in a smooth frequency response that is free of the comb-filter effect. A popular boundary microphone is shown in Figure 6-20.

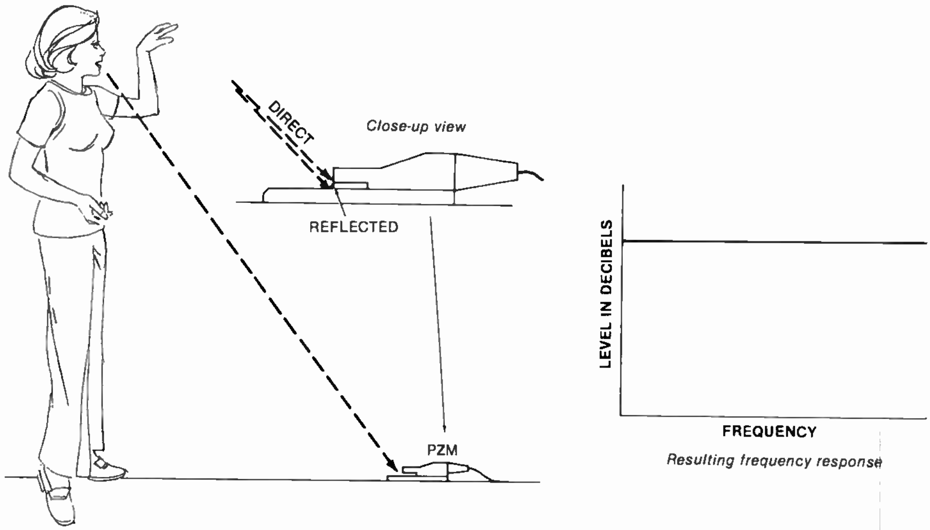


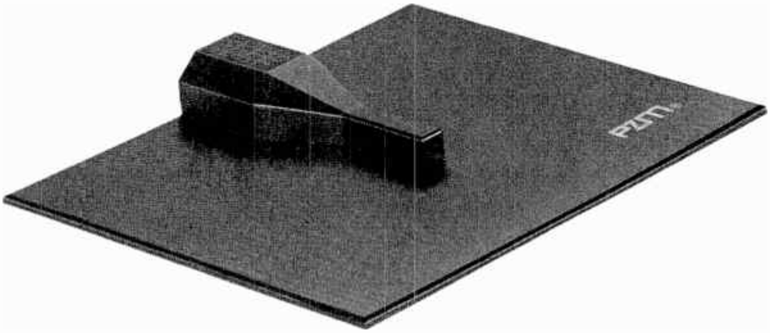
Fig. 6-19. Pressure Zone Microphone or PZM system. *(Courtesy of Crown International, Inc.)*

As the major requirement of a boundary microphone is proximity to the nearest incident surface, it is a simple matter to devise such a mike using a clip-type microphone (Fig. 6-21). Additionally, a conventional microphone may be placed near or on a reflective boundary (Fig. 6-22). A short delay may possibly exist in the reflected sound because the center of the microphone diaphragm is placed slightly above the surface. Consequently, the uppermost frequencies may be partially cancelled, providing a lightly dull sound quality.

There are several phenomena that occur when a boundary microphone is suspended in mid-air on a small boundary and away from the nearest large reflective surface (primary boundary). These are:

- diffraction
- low-frequency shelving
- increased directivity

Diffraction is the disturbance of a sound field by an obstacle, such as a boundary or hard, reflective panel. At the boundary surface, the pressure is boosted at frequencies having a wavelength on the order of the boundary dimensions. For example, a 1′-(.3 m-)diameter disk boosts the level of frequen-



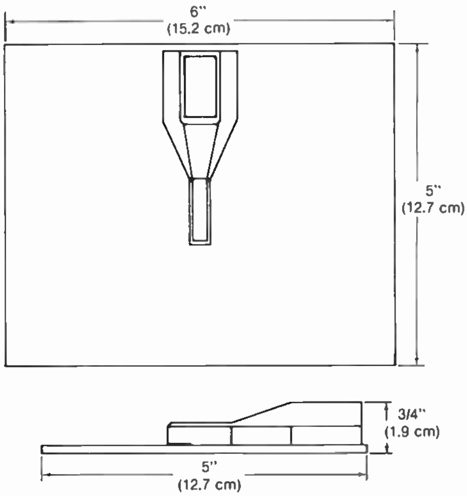
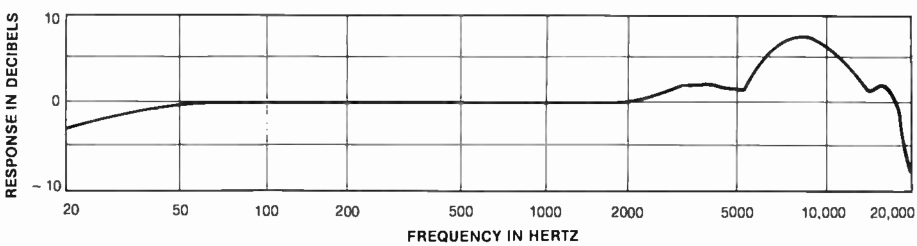


Fig. 6-20. Crown PZM-30RB Pressure Zone Microphone. *(Courtesy of Crown International, Inc.)*



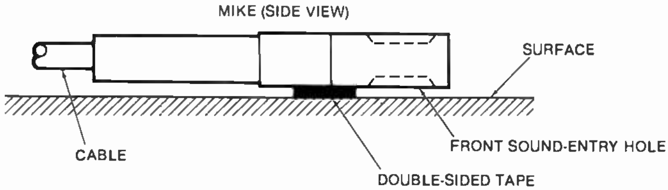


Fig. 6-21. Using clip microphone to devise boundary-type pickup. *(Courtesy of Crown International, Inc.)*

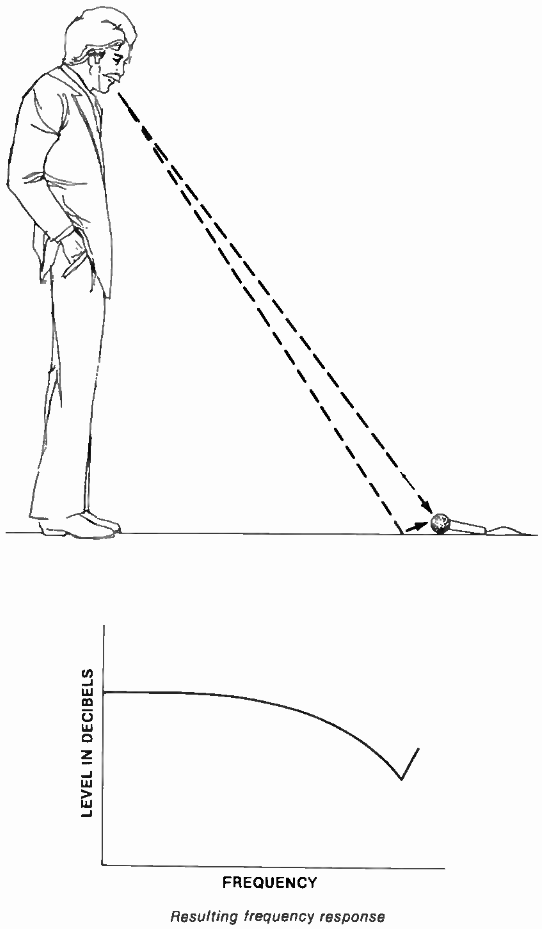


Fig. 6-22. Conventional microphone placed to receive direct sound and slightly delayed reflected boundary signal. *(Courtesy of Crown International, Inc.)*

cies around 1 kHz, creating a broad peak in the frequency response at 1 kHz. This pressure boost is greatest for sounds arriving at normal incidence, while sounds arriving at a grazing incidence are not boosted. For a square boundary, the peak occurs at frequency f = 994/D, where D is the boundary dimension in feet.

If the boundary itself is sufficiently small in size, pressure doubling occurs only at mid-to-high frequencies. In effect, a low-frequency shelf is created. At frequencies whose wavelength is greater than six times the boundary dimensions, the frequency response will be down about 6 dB. The boundary thus acts as a low-frequency shelving filter, with its response down 6 dB at and below f = 188D, where D = the boundary dimension in feet. The larger the boundary to which the pickup is attached, the more extended is the low-frequency response. Big boundary = extended bass; small boundary = bass shelf.

Sounds approaching the front face of the boundary are picked up stronger than sounds approaching the side or rear of the boundary, thus increasing the directivity of the microphone. Sounds from the rear are rejected more at high frequencies. The larger the boundary, the lower the frequencies that are rejected from the rear angle. For example, a 2′ by 2′ panel will reject sound from the rear by 3 dB at 125 Hz, by 10 dB at 500 Hz, and by greater than 20 dB at 10 kHz. Given a square panel, the rear rejection is 10 dB at f = .88C/D, where C = the speed of sound (approximately 1130 ft/s [344 m/s]) and D = the panel dimension in feet (or meters). Rejection is 3 dB at f = .22C/D and greater than 20 dB at high frequencies.

Summary

Single microphone placement technique may be broken down into five classifications: distant miking, close miking, accent miking, ambient miking, and boundary miking.

- Distant miking: a positioning technique in which the mike is placed at such a distance that an entire musical instrument or ensemble is picked up, thus preserving the overall tonal balance of that instrument or ensemble. A natural tonal balance may be achieved by placing the microphone at a distance roughly equal to, or greater than, the size of the sound source. Often, the acoustic environment will be included in the microphone pickup and may be combined with the direct signal to create an overall distant sound quality.
- Close miking: the miking of a sound source at a distance less than the size of the instrument. The miking of a sound source so closely serves

two major functions: (1) to create a tight, present sound quality and (2) to effectively exclude the acoustic environment from being picked up and/or recorded on tape.

- Accent miking: when a solo instrument that is miked at too-close a working distance is mixed together with an overall distant pickup, the latter would be detracted from by an unnatural pickup sound that is too present and "out of context" with the overall sound. In order to avoid this pitfall, a compromise is needed in distance between a close and distant pickup technique to avoid picking up a sound from a closely miked source that is too present and "out of context" with a distant pickup.
- Ambient miking techniques: when a microphone is placed at a distance, such that the reverberant or room sound is predominant to the direct signal.
- Boundary miking: places a microphone very close to a boundary of 2′ (.6 m) square or greater. Consequently, any high-frequency phase cancellations are moved out of the audible frequency band, resulting in a smooth frequency response free of adverse effects. (We look in more detail at the boundary method in Chapter 8 and Appendix C.)

7 Fundamentals of Stereophonic Microphone Techniques

Perception of Direction by the Ear

Before investigating basic stereophonic microphone technique, it helps to understand the methods by which the ear localizes or perceives the originating direction of sound.

One ear is not able to discern the direction from which a sound originates; two ears can, however. The ability of two ears to localize a sound source is called binaural localization. This faculty results from using three cues that are received by the ears:

1. interaural intensity differences
2. interaural arrival-time differences
3. the effects of the pinnae (outer ears)

Middle to higher frequency sounds originating from the right side reach the right ear at a higher intensity level than the left ear, causing an *interaural intensity difference*. This is because the head casts an acoustic shadow to direct sound waves originating from the right, allowing only sound reflected from surrounding surfaces to reach the left ear (Fig. 7-1). Since the reflected sound travels further and loses energy at each reflection, the intensity of sound perceived by the left ear is reduced, with the resulting signal being perceived as originating from the right.

This effect is relatively insignificant at lower frequencies, whose wavelengths are large compared to the diameter of the head and are able to bend easily around the head. Thus, at lower frequencies a different method of localization, known as *interaural arrival-time differences*, is employed. These time differences occur because the acoustic path length to the left ear is slightly longer than that to the right ear. Thus, the sound pressure will be sensed by the left ear at a later time than by the right ear (Fig. 7-2). This method of localization, in combination with interaural intensity differences, gives us lateral localization cues for the entire frequency spectrum.

These intensity and delay cues allow us to perceive the angle from which a sound originates, but not whether the sound originates from in front, behind, or below. The pinna (Fig. 7-3), however, has two ridges that reflect the incident sound into the ear. These ridges introduce delays between the direct sound, which reaches the entrance of the ear canal, and the sound reflected from the ridges, which varies according to the source location.

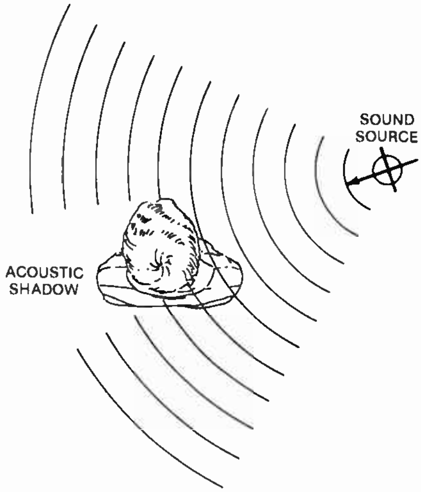


Fig. 7-1. Acoustic shadow thrown by head causes interaural intensity differences at middle to upper frequencies.

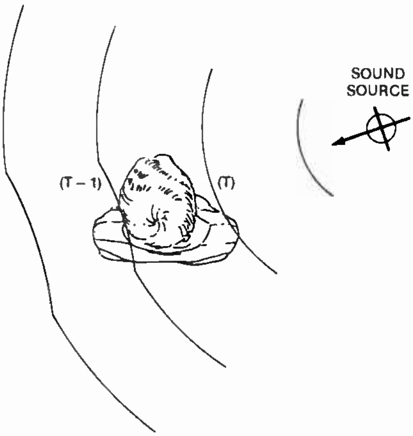


Fig. 7-2. Interaural arrival-time differences occurring at lower frequencies.

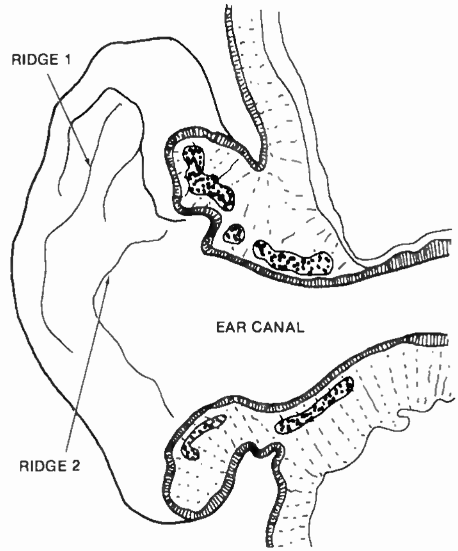


Fig. 7-3. Pinna and its reflective ridges for determining vertical location information.

It is interesting to note that from beyond about 130° from the front axis there can be no reflections from the outer ridge (1) because it is blocked by the pinna. Unreflected sounds are therefore delayed (from between 0 and 80 ms) and are perceived as originating from the rear. The inner ridge (2) produces delays between 100 and 330 ms, corresponding to a source located within the vertical plane. The delayed reflections from both ridges combine with the direct sound to produce the characteristic frequency-response colorations due to constructive and destructive interference at differing frequencies. The brain is able to compare these colorations at each ear and use this information to determine source location. Small movements of the head provide additional position information, due to the changing source perspective. This last cue is minor, however, when compared with that of the other localization cues.

If there are no differences between what the left and right ears hear, the brain assumes the source is the same distance from each ear. It is this phenomenon that allows the audio production engineer to position sound not only in the left and right loudspeakers, but also within a monophonic central position between the loudspeakers. If the same signal is fed to both loudspeakers, the brain perceives the sound identically in both ears and deduces that the source must originate from directly in front of the listener. By changing the proportions fed to the two loudspeakers, the engineer changes the interchannel intensity differences and thus creates the illusion that the sound source is positioned at any desirable point between these two loudspeakers. The source position may even be made to move between the loudspeakers. This placement technique is known as panning (Fig. 7-4) and, although it is the most widely used method, it is not the most effective positioning technique because only those listeners who are equidistant from the left and right loudspeakers will perceive the desired effect. A listener located close to the left loudspeaker will tend to locate the source as coming from that side even though the signal may be panned toward the right. The other tools of localization may be used by the engineer to assign the source a location point between two loudspeakers, such as digital delay lines (DDL), phase shifters, filters, or stereophonic microphone techniques.

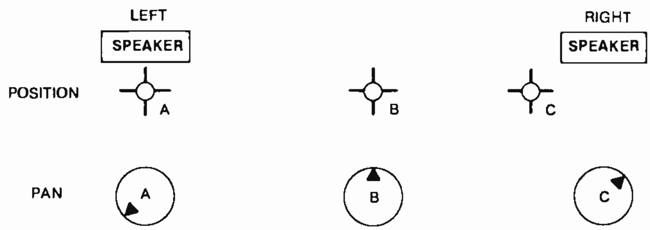


Fig. 7-4. Panning settings vs. spatial positioning.

Stereophonic Microphone Techniques

The concept of using stereophonic microphone techniques to convey a sense of direction is not new. In 1931, the British scientist Alan Dower Blumlein filed for his historic British Patent No. 394,325, listing seventy claims over twenty-two pages, on the subject of two-channel stereo for disk recording and motion pictures. In his patent application, Blumlein stated:

> The fundamental object of the invention is to provide a sound recording, reproduction and/or transmission system whereby there is conveyed to the listener a realistic impression that the intelligence is being communicated to him over two acoustic paths in the same manner as he experiences in listening to everyday acoustic intercourse and this object embraces also the idea of conveying to the listener a true directional impression and thus, in the case in which the sound is associated with picture effects improving the illusion that the sound is coming, and is only coming, from the artist or other sound source presented to the eye.

Some of Blumlein's contributions to audio in this patent are the development of the 45/45 modulated groove which is used today for stereophonic disk reproduction, as well as improvements in stereo miking techniques. His crossed figure-eight stereo arrangement was employed by Decca London in the 1950s to produce recordings with outstanding results, even by today's standards. Blumlein was also instrumental in the practical development of the moving-coil microphone, as well as in studies that led to many of the microphone polar patterns used today.

When speaking of stereo miking techniques, we are generally referring to the use of a two-microphone arrangement to attain a stereophonic image. This is different from the single-microphone pickup, in which stereo is created in the audio production console through the use of panning and effects techniques.

The dual-microphone technique may be used, with equal effectiveness, for either close-mike techniques (where a single instrument is recorded at close range within a studio or on-location environment) or for distant, overall miking of an ensemble. The only limitations in the application of techniques are those placed by the imagination.

There are four basic ways to create a stereophonic image with two microphones: the spaced technique, the coincident technique, the near-coincident technique, and the binaural technique. We'll look, too, at a fifth element, the soundfield microphone system, which makes use of four coincident pickups and is a special embodiment of the coincident technique.

The Spaced-Microphone Technique

Spaced-microphone methods were among the first popular techniques known to relay a stereo image. Generally, spaced techniques employ two (or more) matched microphones that are set symmetrically along a centerline perpendicular to the front plane of the sound source. The polar pattern of this stereo pair, their spacing, and their distance from the sound source are all variable. Stereo information in these configurations is created by differences in both amplitude and time of arrival of the sound wave. Positional information changes radically as the distance to the sound source varies.

When using spaced microphone configurations, pay special attention to these potential problems:

- low-frequency comb-filter effects on sound sources to the extreme left and right of the sound stage
- vague center imaging (instruments near the center may appear from the left or right speaker during playback)
- erratic mono compatibility

For the phase integrity of the spaced microphone technique, we again refer to the 3-to-1 principle, which states that for every unit of distance measured between the microphone and its source, the distance between the spaced pair should be at least three times that unit of measure. Since this condition is not always met within this system, phase interference in mono reproduction may result.

Spaced Omnidirectional Microphones

This style typically utilizes two (or three) spaced omnidirectional microphones. Common spacings are from 2′ to 10′ (.6–3 m) on either side of the centerline. The spacing is determined by the width of the sound source and the distance of the mike pair from that source. A general rule is that each mike should be placed one-third to one-half the distance from the centerline to the outer edge of the sound stage (Fig. 7-5). Stereo imaging is diffuse, but imaging improves if a center mike is mixed with the outer pair (a method we discuss later in this chapter).

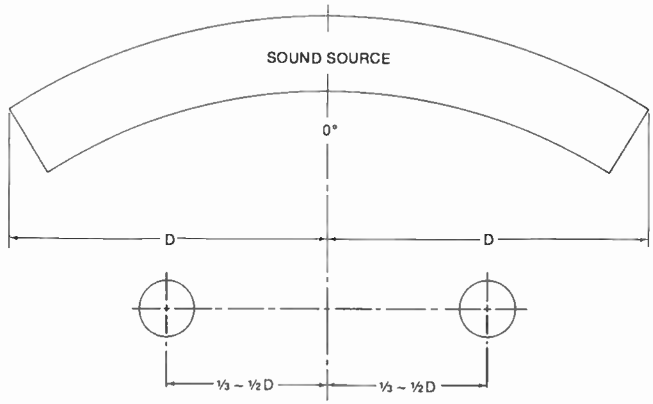


Fig. 7-5. Spaced omnidirectional pair.

Spaced Cardioid Microphones

The spaced cardioid method is similar to spaced omnidirectionals. Since these microphones are directional, they tend to favor that segment of the sound source that is most on-axis and may exhibit the effects of coloration for reverberation, audience response, and other off-axis sound sources. For this reason, orientation and placement will sometimes be more critical than with omnidirectional mikes.

Spaced Hypercardioid Microphones

The spaced hypercardioid system uses a polar pattern that is midway between the spaced cardioid and bidirectional techniques. The front lobe of the pickup is narrower than that of the cardioid, while the small rear lobe has the reverse-polarity qualities of the bidirectional. The null area, in which the rear lobe is found, is generally a cone whose edges lie near the angle of 110° off-axis. The exact null-cone angle, the amount of rear lobe pickup, and the coloration of sounds arriving from off-axis will depend on the design of the microphone being used.

Spaced Bidirectional Microphones
The rear lobe of the bidirectional mike, which provides the reverberation and audience-response component, will have the same sonic characteristics as the front lobe (i.e., there will be little off-axis coloration to these sounds).

Spaced Boundary Microphones
The spaced bidirectional mike pair tends to have more "reach" to the front than the spaced cardioid pair has, but it has an equal pickup lobe to the rear. Therefore, the bidirectional pair must be placed further from the sound source than either an omni or cardioid pair to achieve the same degree of coverage. Experiments conducted with boundary or surface-mounted mikes have given rise to several new microphone "types"; the best known of these being the *Pressure Zone Microphone*.

In operation, the spaced boundary-microphone pair is similar to the spaced omnidirectional pair, except that the boundary microphone's polar pattern is hemispherical about the boundary surface.

Three- or Four-Spaced – Microphone Techniques
Generally, these techniques are an extension of the two-microphone configurations discussed above. Most of the same qualities, advantages, and disadvantages therefore apply.

Three Spaced Microphones

The three-spaced–microphone technique was developed by the researchers at Bell Labs during their experiments with stereo in the 1930s. It employs a center microphone in addition to the two-microphone array discussed earlier. The center mike tends to fill the "hole-in-the-middle" which results from the wide spacing of the two outer mikes, and it may also be used to "tighten" the center imaging of the configuration.

A problem caused by using this additional microphone is that it compounds the effects of phase anomalies between the microphones within the array. This tends to increase the comb-filtering and raise the frequencies affected into the more noticeable mid and upper ranges of the spectrum.

Two-Spaced – Microphone Technique with Accent Mikes

An accent microphone is often added to the basic two- or three-microphone technique to emphasize a soloist within the overall sound-stage image. More than one accent mike may be used. However, in production, it is important to keep in mind the main technique that is being employed. For example, is it a stereo pair that is providing the basic overall coverage, or has the use of more than one accent mike led the sound engineer into the realm of "multi-microphone" techniques? (We noted precautions for the use of accent mikes in Chapter 6.)

The Coincident-Microphone (X/Y) Technique

Coincident (X/Y), or *intensity stereo*, technique utilizes a pair of directional microphones, vertically aligned on a common axis and set at a specific angle with respect to each other along the horizontal plane. Given that the stereo coincident pickup is very closely spaced, there is no time (phase) difference between the two capsules for sound sources within the horizontal plane. This method relies on intensity differences between the two signals for directional cues. The choice of microphone polar pattern may be varied between subcardioid and bidirectional, depending on the technique being implemented. The angles formed by the microphone pair are typically symmetrical about the centerline of the sound source.

One advantage of intensity stereo is that the angular accuracy of the stereo image is unaffected by the distance of the microphone pair to the sound source. However, a disadvantage may be that, without the interchannel delay common to some other miking techniques, the stereo image may seem to lack a "sense of space."

In addition to the use of two matched microphones operating as a stereo pair, coincident-microphone systems that are specifically designed for this purpose are currently available. These utilize two coincident-mike capsules mounted in one housing (Figs. 7-6 through 7-8). Such a stereo microphone may allow the upper element to rotate, relative to the lower element by 180°, accommodating all possible angle offsets. Certain designs enable the sound engineer to control the polar pattern of each pickup.

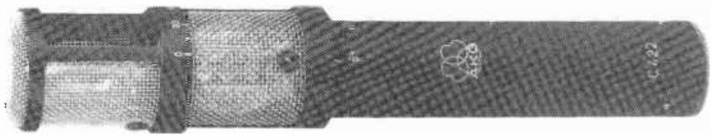


Fig. 7-6. AKG C-422 stereo microphone. *(Courtesy of AKG Acoustics, Inc.)*

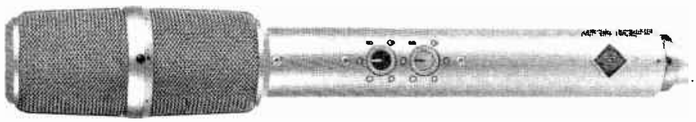


Fig. 7-7. Neumann SM-69 stereo microphone. *(Courtesy of Gotham Audio Corporation)*

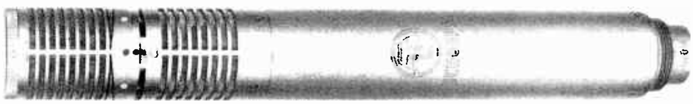


Fig. 7-8. Schoeps CMTS 301 stereo microphone. *(Courtesy of Schalltechnik Dr.-Ing. Schoeps GmbH)*

A critical factor to consider when using coincident techniques is the polar response of the microphones involved. As the individual mikes are oriented at an angle to a majority of the sound source, considerable off-axis coloration is possible. Furthermore, the pair should be closely matched for frequency response; any difference would result in a wandering image with changes in pitch.

The use of cardioid microphones is common when coincident techniques are employed, typically placed at an included angle of from 90° to 135° and fairly close to the sound source (Fig. 7-9). Often, the axes of the microphones are aimed near the extremes of the sound source. Because the direct-to-reverberant sound ratio is often high with this approach, a degree of unwanted sound rejection may occur at the rear of the pair. At times, a distant pickup with a large reverberant component is desirable. In such circumstances, included angles as large as 180° may be used. The wider the angle, the greater the stereo spread.

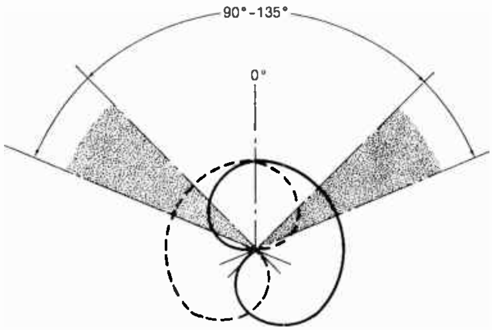


Fig. 7-9. Cardioid X/Y pair.

The hypercardioid X/Y pair is similar to the cardioid pair, except the included angle is typically narrower in order to preserve a solid center image (Fig. 7-10). The increased reach of the hypercardioid allows a more distant placement for a given direct-to-reverberant sound ratio. With their small reverse-polarity lobes, the hypercardioid pair is often a good compromise between the cardioid and the Blumlein X/Y technique.

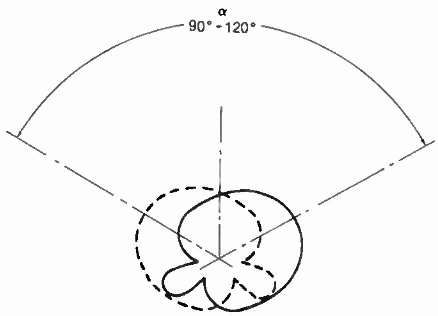


Fig. 7-10. Hypercardioid X/Y pair.

The Blumlein X/Y Technique

The Blumlein or "crossed pair of figure-eights" is the earliest of the X/Y techniques. It uses two bidirectional microphones oriented at an included angle of 90° (Figs. 7-11 and 7-12).

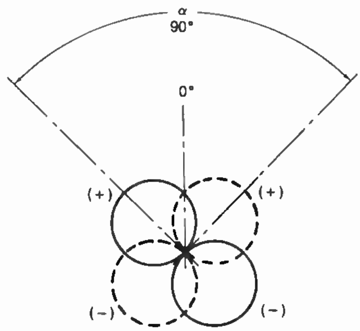
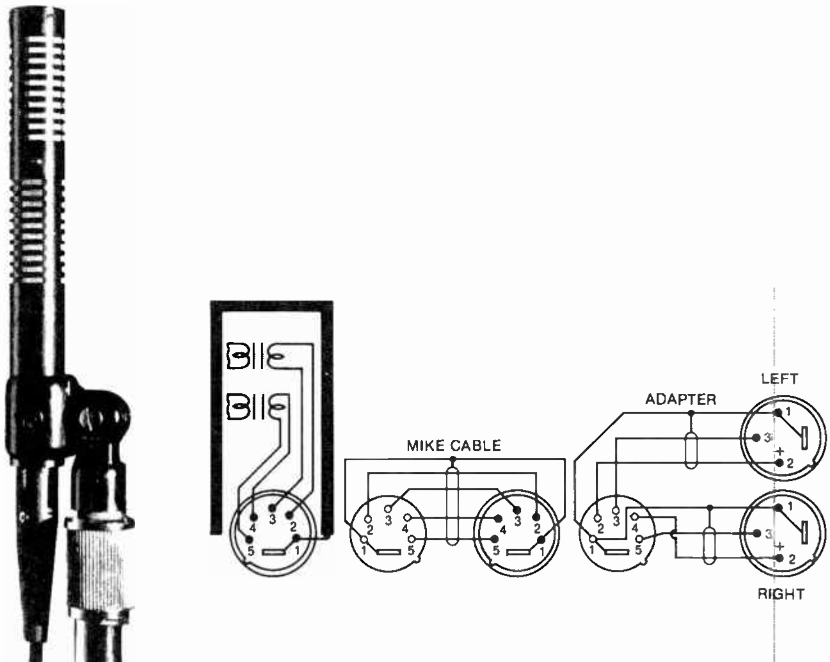
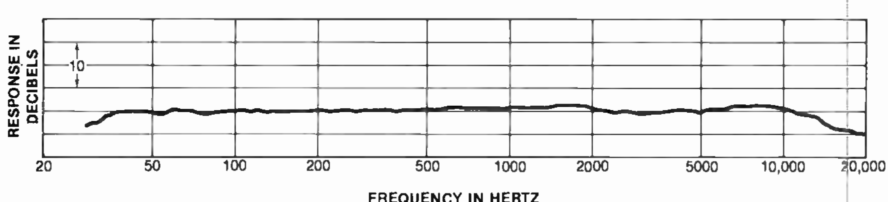


Fig. 7-11. Blumlein X/Y crossed pair (figure-eight).



(A) *Actual mike.*

(B) *Wiring diagram.*



(C) *Typical frequency response.*

Fig. 7-12. Speiden SF-12 coincident bidirectional stereo ribbon microphone. *(Courtesy of Speiden and Associates)*

One attribute of this technique is that the rear lobes of the bidirectional microphones record the rear 90°-quadrant in phase between channels but out of polarity with the front lobes. The rear quadrant combines with the front quadrant in the stereo (cross-channeled) image. Signals arriving from the two side quadrants are picked up out of phase between channels. Placement is, therefore, critical in order to maintain a proper direct-to-reverberant sound ratio and to avoid a strong out-of-phase side component. This technique works very well when used in a wide room or one with minimal side wall reflections. This configuration seems to produce a very natural sound.

The technique of incorporating the omnidirectional microphone into a coincident X/Y technique was developed by Ron Streicher for use as a close and semiclose soloist pickup. Since most omnidirectional mikes tend to exhibit a degree of directionality at higher frequencies, configuring an omni pair at an included angle of 60° to 90° will provide a stable, coherent center image combined with a sense of stereo space. Further, there will be little sense of "image shift" as the soloist moves, as is the case with directional mikes. The use of pressure capsules eliminates the proximity effect and breath blasting problems associated with the pressure-gradient microphone pair.

The M/S Technique

The *M/S* or *mid-side stereo* miking technique is similar to the X/Y method in that it operates two microphone capsules in close proximity. This may take the form of two coincident microphones or a stereo microphone system operating with the appropriate polar patterns and orientation.

In the classic M/S configuration, one of the microphone capsules is designated to be the "mid"-position microphone and is generally selected to have a cardioid pickup pattern, oriented toward the sound source (Fig. 7-13). The remaining capsule, designated to be the "side" pickup, is selected to have a figure-eight pattern and is oriented 90° laterally to the sides. Direct information is thus picked up by the mid (M) capsule, while ambient and reverberant information is received by the side (S) capsule. These outputs are then processed by a sum-and-difference matrix network, which resolves them into a conventional X/Y stereo signal: (M + S) and (M − S).

One major production advantage of this system is its absolute monophonic predictability. When the left and right signals are combined, the sum is solely the output from the mid pickup component ([M + S] + [M − S] = 2M) which contains the direct information. As it is generally more desirable that there be less reverberation in a mono signal than in a stereo signal, there is a built-in advantage to M/S stereo microphone techniques.

M/S stereo production offers the engineer a great advantage in stereo sound control. Within this system, it is possible to adjust the ratio of mid-to-side information delivered to the sum-and-difference matrix at the audio

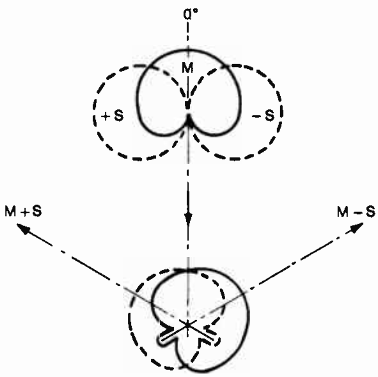


Fig. 7-13. Standard M/S configuration and its standard conversion to X/Y.

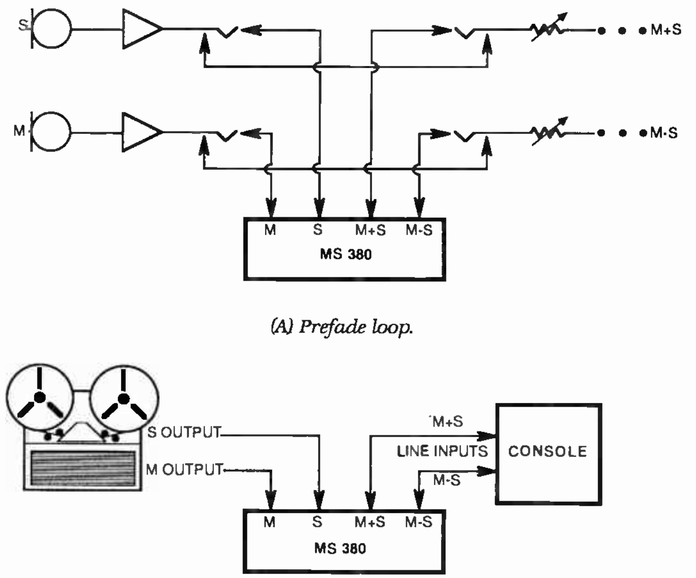
production mixer or console. This allows remote control of the direct-vs.-ambient and stereo-width information. It is equally possible, given two phase-aligned audio recording tracks, to record the mid information on one track of an audio tape recorder and the side information on another. This allows the data to be remixed into an X/Y-compatible signal in the mixdown phase. This feature also lets the producer make important decisions regarding stereo width and depth at a later, more controlled date.

One commercially available sum-and-difference matrix that utilizes an active combining circuit is the Audio Engineering Associates, MS-380 Active Matrix (Fig. 7-14). This mike/line level, active device can adjust the mid-to-side ratio within the matrix via a single knob control. It follows the microphone preamplifiers from the M and S mikes, either in a real-time recording situation (Fig. 7-15A), or during tape playback from a tape recorder during a postproduction mixdown (Fig. 7-15B).

Fig. 7-14. Audio Engineering Associates MS-380 Active Matrix. *(Courtesy of Audio Engineering Associates)*

M/S decoding may also be accomplished by a dual-transformer matrix arrangement (Fig. 7-16). This method may be readily incorporated into an existing system through the design of such a matrix into a patch bay or in a separate, portable housing.

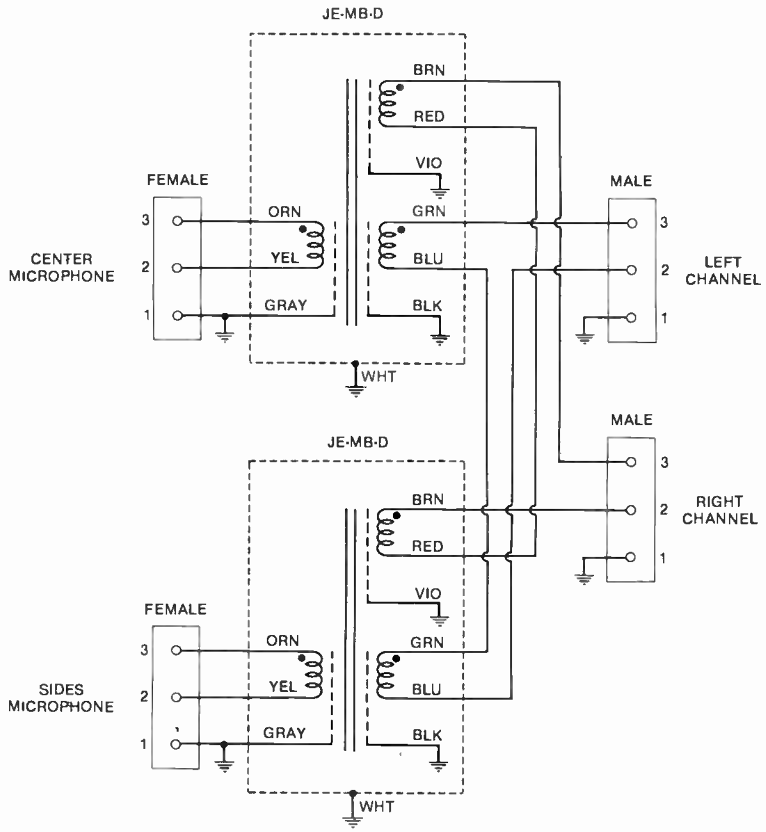
Fig. 7-15. MS-38 active M/S Matrix patch loop.



(A) Prefade loop.

(B) Postproduction mode.

Fig. 7-16. M/S transformer matrix.



The Near-Coincident Technique

The term *near-coincident* is used to describe a class of techniques in which a microphone pair is placed close enough together to be substantially coincident at low frequencies, yet far enough apart to have appreciable delay between channels for sound source localization at higher frequencies.

The value of near-coincident techniques is that they exhibit good localization combined with a sense of depth. Close miking of soloists in an ensemble is not always recommended for use with these techniques, since small movements of the sound source may produce large image shifts. Sounds arriving from the far left or far right may create problems for mono summation due to interchannel delay.

O.R.T.F.

Named for the French National Broadcasting Organization's "Office de Radiodiffusion-Television Française," the O.R.T.F. configuration, shown in Figure 7-17, consists of two cardioid microphones, oriented outward from the centerline of the sound source with an included angle of 110° and a capsule spacing of 7″ (17 cm).

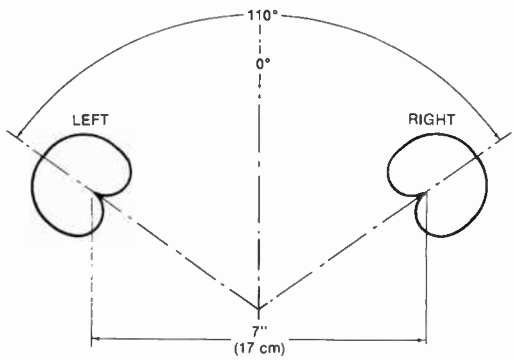


Fig. 7-17. O.R.T.F. configuration.

N.O.S.

Adopted by the Dutch Broadcasting Organization ("Nederlandache Omroep Sticting"), the N.O.S. standard consists of two cardioid microphones oriented outward from the centerline, with an included angle of 90° and a capsule spacing of 12″ (30 cm), as shown in Figure 7-18.

Faulkner

Developed by British engineer Tony Faulkner, the Faulkner configuration uses two bidirectional microphones facing directly forward, toward the sound source, and spaced 8″ (20 cm) apart (Fig. 7-19). This technique can combine the coherence of the Blumlein technique with the "openness" afforded by the

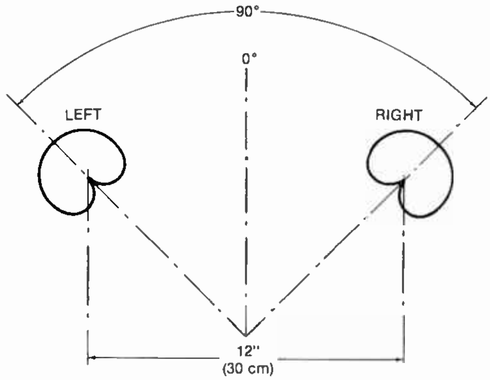


Fig. 7-18. N.O.S. configuration.

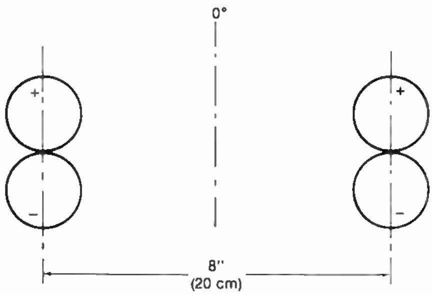


Fig. 7-19. Faulkner configuration.

time (phase) differences resulting from the spacing between the microphones. Additionally, Faulkner recommends that the microphone pair be placed further back from the sound source than is common with other coincident techniques; this provides a more natural balance between the direct and reverberant sounds.

Many people and organizations have adopted these techniques for their own needs, altering choice of mike, pattern, spacing, or included angle.

The Binaural Technique

The *binaural technique* (Fig. 7-20) is intended specifically for playback via headphones. It is often configured using two omnidirectional microphones placed into the ears of a "dummy head" to simulate the sound received at the listener's position. This technique can be very realistic, providing a good illusion in both the horizontal and vertical planes. Unfortunately, these characteristic illusions cannot reproduce during playback over loudspeakers.

Similar techniques have been configured with two omnidirectional (or even bidirectional) microphones placed approximately 3–4″ (7.5–10 cm) on either side of a sound-absorbing baffle (Fig. 7-21). This method results in a pickup similar to the "dummy head" system and can also provide sufficient isolation between channels to allow realistic stereo reproduction over loudspeakers.

Fig. 7-20. Neumann KU 81 binaural dummy head. *(Courtesy of Gotham Audio Corporation)*

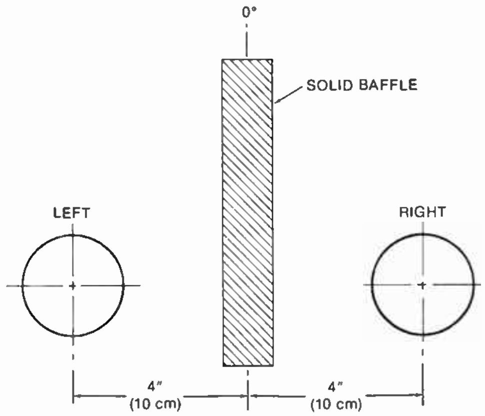
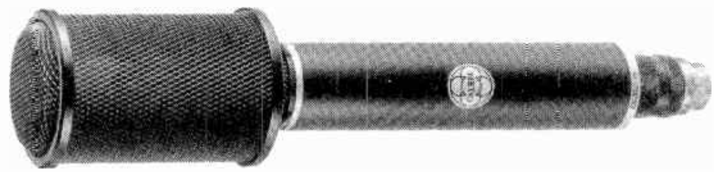


Fig. 7-21. Simple quasibinaural setup configuration.

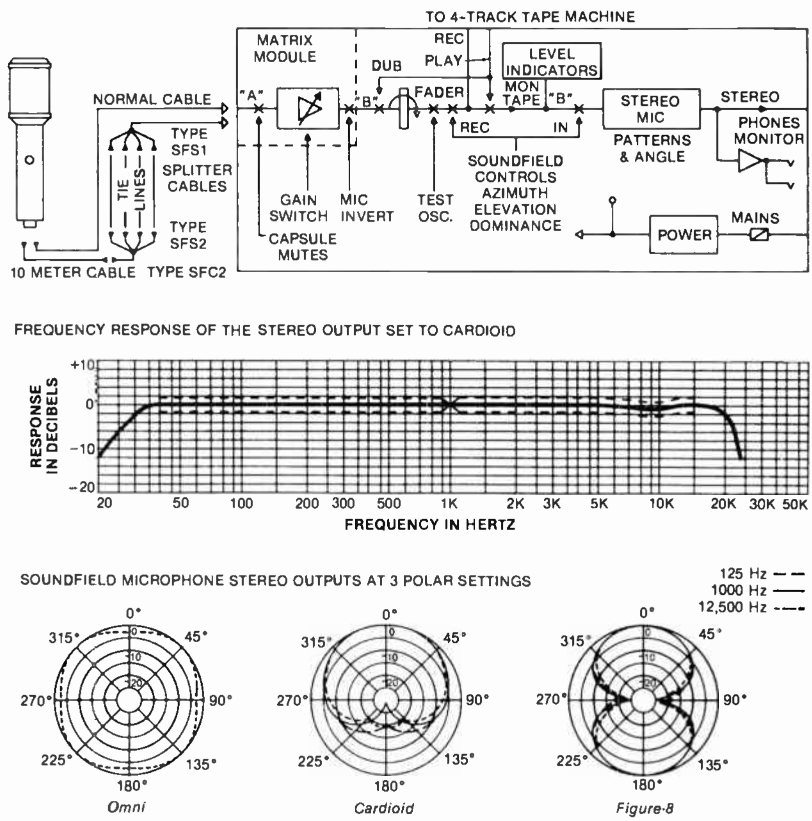
The Soundfield Microphone

The Soundfield microphone system was developed by Calrec Audio Ltd., following a theoretical development by Michael Gerzon, under the direction of the National Research Development Council in England. A multiple-microphone system, it employs four mike elements fitted into a single housing (Fig. 7-22A and B). Through electronic manipulation (Fig. 7-22C) the four signals



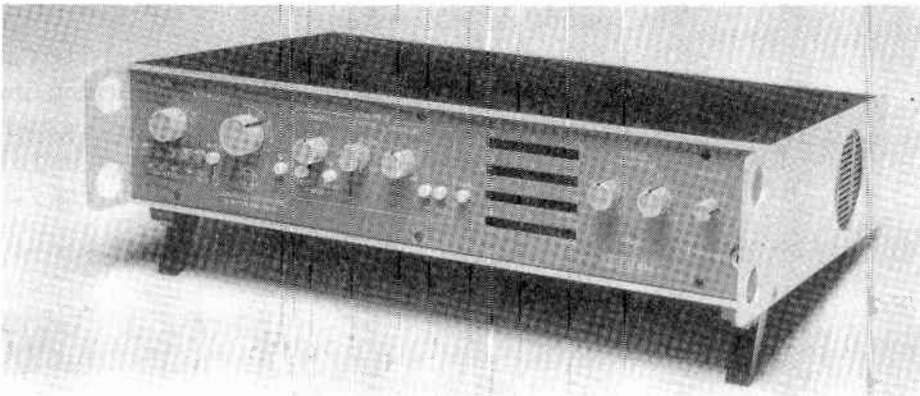
(A) Calrec Soundfield microphone.

Fig. 7-22. Calrec Soundfield microphone system and ambisonic controller. *(Courtesy of Audio Design Calrec, Inc.)*



(B) Diagram, frequency and polar response.

Fig. 7-22 (cont.)



(C) Calrec Soundfield/ambisonic controller.

may be converted into *ambisonic surround-sound signals*, in addition to conventional left and right stereo signals.

In their original form, the four signals are derived from the four mike elements arrayed in a near-coincident tetrahedron (Fig. 7-23). Their outputs are electronically matrixed to produce:

1. an omnidirectional component relating the pressure of the sound wave at the microphone (W in Fig. 7-23)
2. a pressure-gradient component, relating the vertical (up-down) information of the sound wave (Z)
3. a pressure-gradient component relating the lateral (left-right) information (Y)
4. a pressure-gradient component relating the lateral (fore-aft) information (X)

These virtually coincident signals are encoded into a B-format signal, which contains the overall horizontal, vertical, and pressure information.

The B-format signal may be stored on four-channel tape for discrete postproduction, or it may be immediately processed for resolution into quadraphonic, ambisonic, or conventional stereo signals. Additionally, the electronic control facility of the Soundfield system enables the mixing engineer to steer, pan, tilt, and vary the included angle; alter the directional pattern; and otherwise change the overall stereo perspective and sound imaging. This can be accomplished without touching the microphone placement and may be done after the recording session, direct from tape in the postproduction phase.

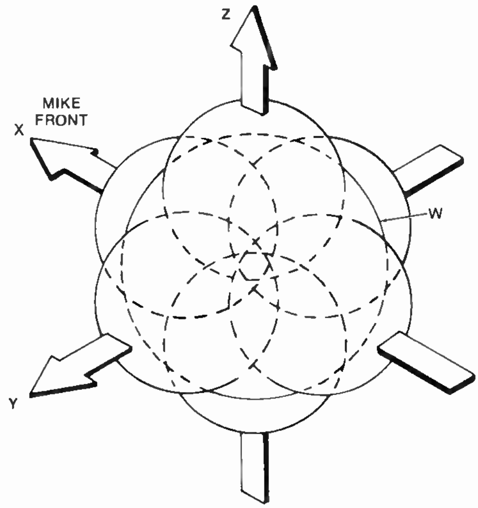


Fig. 7-23. Soundfield microphone capsules arranged in tetrahedron, representing the three dimensions of live sound. *(Courtesy of Advanced Music Systems)*

Summary

In this chapter we saw that the class of stereo microphone techniques—be they spaced, coincident, or near-coincident—are valuable tools available to the sound engineer/recordist. These techniques can be used on their own to create effects ranging from a wide stereo spread, to the natural-sounding blend of an ensemble, or to the larger-than-life sound of distant studio miking techniques found on recent music releases. In addition, through experimentation, any of these configurations can be employed or mixed to form a new effect for any of the music or sound production media.

8 Applied Microphone Techniques in Music Production

The art of capturing music on a recorded medium involves balancing a number of audio considerations, including the personal style of the musician(s), the acoustic character of the sound source, the design and acoustics of the location, the choice and placement of microphone(s), and the recording chain (recording medium, signal routing, equalization, effects, and so on).

When we speak of "applied microphone techniques," we refer to insight (scientific *and* artistic) into, and judicious control over, the sonic quality of a microphone.

In addition to talent considerations and the acoustic nature of the sound source, microphone techniques involve two primary factors:

- the effects of microphone placement on sound quality
- the correct choice of a microphone for a given application

Number of Microphones

The number of microphones to be used depends on the pickup conditions. The ambient demands of classical music may require a distant miking approach, while multiple-mike close techniques may work best with popular-music forms. The degree of control over the sound sources and the amount of leakage encountered also help to determine the number of microphones that might be used in a particular situation. Figure 8-1 shows an overall acoustic pickup of a musical ensemble. The size of group and the style of stereo pickup may vary; this arrangement, however, provides an overall natural balance that is under the control of the ensemble and is affected by room ambience.

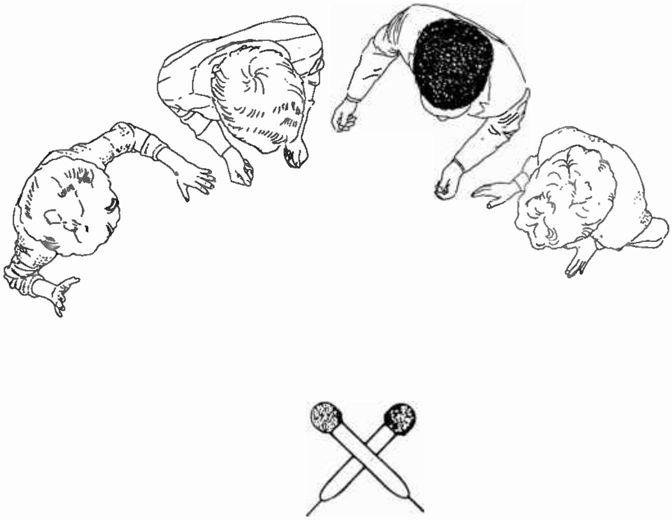


Fig. 8-1. Overall miking of musical ensemble using two distant microphones.

For greater control, the ensemble can be broken down into smaller groups (Fig. 8-2). This reduces the miking distance, diminishing leakage and improving separation during a multitrack recording session.

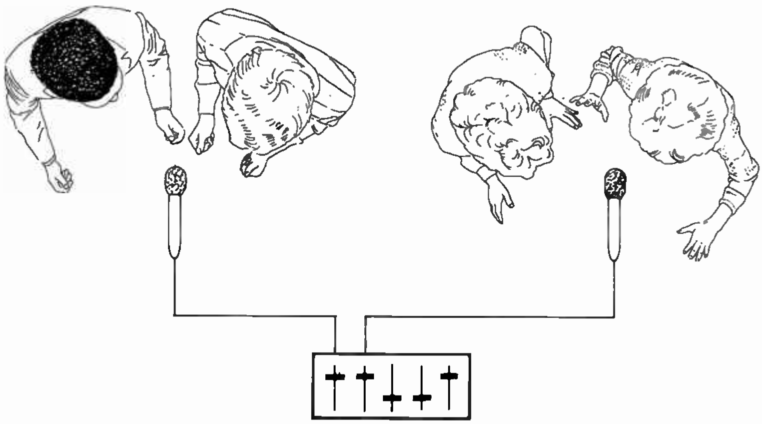


Fig. 8-2. Multiple miking for smaller ensemble groupings.

When individual control is necessary or if feedback is a problem, each individual sound source may be given its own microphone (Fig. 8-3). If the ensemble is out of balance at the time of recording, this approach enables a proper balance to be restored in the mixdown phase.

Placement Techniques: Sound Sources

Here we want to demonstrate many of the currently accepted placement techniques. These techniques are simply guidelines. The actual placement and choice of microphone type, pattern, and frequency response may vary from one instrument and environment to the next at the discretion of the user. Once these basics are mastered (and this section cannot be considered all-encompassing), remember that the "rules" of recording are meant to be broken, in order to move into new grounds of sound production.

As we examine the range of musical instruments, we will generally consider the instrument class, the tonal and frequency range of the instruments, and standard miking techniques and/or placement. Unless otherwise stated, all microphone polar characteristics may be assumed to be cardioid.

Although the majority of placements will be given as a single-mike pickup, certain results might be improved by the judicious use of a coincident pair in the same position. The two signals are panned left and right in order to provide an image spread as broad or narrow as is wanted.

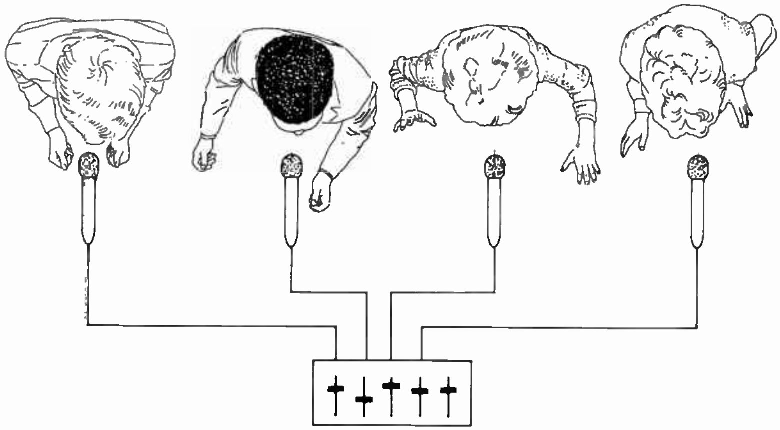


Fig. 8-3. Individual miking of each source within ensemble.

Brass Instruments

The Trumpet

The trumpet contains overtones that stretch up to 15 kHz. Its fundamental frequencies lie between E3 and D6 (165 Hz to 1.175 kHz). Below 500 Hz, the sound emanating from the trumpet projects uniformly in all directions; at 1.5 kHz and up, the projected sound becomes highly directional; above 5 kHz, the dispersion angle is a tight 30° on-axis to the trumpet bell.

The trumpet's *formants* (the relative harmonic levels that give an instrument its specific character) lie around 1 to 1.5 kHz and at 2 to 3 kHz. The tone of a trumpet can be radically changed through the use of a mute, which fits over or inside the bell. A cup-shaped mute (fitting over the bell) dampens frequencies above 2.5 kHz. A conical mute (fitting inside the bell) cuts back on the region below 1.5 kHz and emphasizes the spectrum above 4 kHz.

Because of the high sound pressure levels that may be encountered within a trumpet passage (up to 155 dB SPL), it is best to place a microphone slightly off-center to the bell, at a distance of 1′ (.3 m) or more (Fig. 8-4). When a situation necessitates a closer placement, a −10 to −20 dB pad inside the microphone may be required to prevent microphone overload. Under such close working conditions, a wind screen helps to protect the diaphragm from excessive exposure, especially for older ribbon designs.

Fig. 8-4. Typical mike positioning for single trumpet.

Due to its strong harmonic structure, the sound of a trumpet may be "shaped" in numerous ways through the choice of microphone:

- The condenser microphone might be chosen for its accurate transient response.
- The dynamic microphone might be chosen for its smoother treatment of the transients, as well as its rugged design.
- The ribbon pickup (particularly of the older variety) might be chosen for close pickup because of its "mellow" high-end response and proximity effect, which adds a fuller tone to the existing sound.

As the microphone signal of a trumpet often contains high-level transient peaks, it is necessary to hold the recorded signal level to 0 volume units (VU) or slightly lower, to reduce the possibility of tape saturation.

The Trombone

The trombone is found in an assortment of sizes. The tenor is most common and its fundamental range is between E2 and C5 (82 and 520 Hz), with an output containing overtones that provide a rich tone. Medium-loud playing gives an upper limit of 5 kHz, while harder blowing could press the upper limit to 10 kHz. Typical formants are at 480 to 600 Hz and around 1.2 kHz.

The "bone's" polar pattern is as nearly symmetrical as that of the trumpet. Below 400 Hz, frequencies are distributed evenly, while at 2 kHz and above the dispersion angle is down to 45° from on-axis to the bell. With extremely loud passages, the frequency components at and above 7 kHz may have a dispersion angle as narrow as 20°.

The trombone is used most often in jazz and classical music. The Mass in C Minor by Mozart has parts for soprano, alto, tenor, and bass trombones. These need the spacious blending that results from long miking distances. Jazz trombone requires a closer miking distance, but, like the trumpet, the danger of overload is present. From 2″ to 12″ (5–30 cm), the trombonist should play slightly to the side of the microphone. In miking a trombone section, a single microphone may be placed between two players, and the total pickup of those players may be blended on one or more recorded tracks.

The French Horn

The French horn is a difficult instrument to play with a clean intonation and sound. Its fundamentals range from B1 to B5 (65 to 700 Hz). An "oo" sound or formant gives it its round, broad quality and may be found at about 340 Hz with others between 750 Hz and 2 kHz, as well as around 3.5 kHz. French horn players often place one hand inside the bell, which mutes the sound and promotes the formant at about 3 kHz.

Frequencies between 62 and 100 Hz radiate uniformly from the instrument. With rising frequencies, however, this angle trims down to within 15° from off-axis. For this reason, a microphone is often placed behind the player, facing into the bell at a slightly off-axis angle.

Traditionally, the French-horn player or section is placed at the rear of an ensemble, just in front of a rear, reflective stage wall. Because of its curvature, the bell faces the wall, which reflects the sound back toward the listener's position, creating a fuller, more defined French-horn sound. An effective pickup may be accomplished in the following ways:

- A bidirectional pickup may be used on-axis to the horn in order to receive both the direct and reflected sound.
- The pickup may be placed in front of the horn section, on-axis to the reflective surface so only the reflected sound is picked up.

Under typical studio conditions, the latter method promotes leakage and is only advised as an accent pickup when leakage is not a major concern.

The Tuba

The bass and double-bass tubas are the lowest pitched of the brass and woodwind instruments. The range of the bass tuba is actually a fifth higher than that of the double-bass, however it is still possible to obtain a low fundamental of B (29 Hz). The overtone structure is limited, with the top response range being 1.5 to 2 kHz. The lower frequencies of the tuba (around 75 Hz) are spread evenly; as frequencies rise, though, the off-axis distribution angle reduces drastically. Normally, this class of instruments is not miked at close distances; a working range of 2′ (.6 m) or more slightly off-axis to the bell yields the best results.

Woodwind Instruments

The flute, clarinet, oboe, saxophone, and bassoon compose the woodwinds. Not all modern woodwinds are made of wood, nor do they all share the same means of producing sound. The sound of the flute is generated by blowing across the hole in a tube, while the rest of this family produces its sound by causing a reed to vibrate a column of air. Historically, the woodwind's pitch was controlled by opening or covering finger holes along the sides of the instrument, changing the length of the tube and the length of the vibrating air column. As instrumentation sophistication grew, the Boehm system of pads and levers further developed this class of instrument.

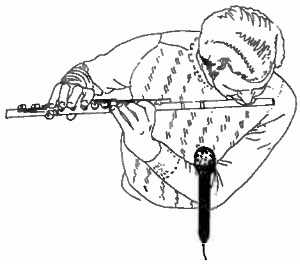
A common misunderstanding concerning the woodwind is the belief that the instrument's natural sound radiates entirely from its bell or mouthpiece. In actuality, most of the sound radiates from the mouthpiece and from the first open finger hole. In many cases, the overall sound output is produced over the entire length of the instrument.

The Flute

The fundamental range of the flute extends from about B3 to C7 (247 Hz to 2.1 kHz). For medium-loud tones, the upper overtone limit lies between 3 and 6 kHz. At frequencies up to 3 kHz, the instrument's sound is radiated along the flautist's line-of-sight, above this point the direction of radiation swings to about 90° to the right of the player.

Microphone placement depends upon the type of music and room acoustics. For classical and solo playing, the microphone should be placed on-axis and slightly above the player at a distance of 3′ to 8′ (1–2.5 m). When dealing with contemporary pop music, this distance should range from 6″ to 2′ (15–60 cm). In both circumstances, the microphone should be positioned midway between the mouthpiece and bell. In this manner, both the characteristic breath sound and tone quality will be picked up with equal intensity (Fig. 8-5). Placing the mike directly in front of the mouthpiece may be desirable in a high-feedback/leakage environment; however, lacking the full-sounding pickup of the overall body sound, breath noise is greatly accentuated. If the acoustic environment is so extreme that it would require a miking distance of ¾″ to 2″ (2–5 cm) in front of the mouthpiece, a preferable practice is to blow below the microphone as the air stream is directed downwards due to the position of the lips.

Fig. 8-5. Typical mike positioning for flute.



If instrument mobility is important, a clip microphone can be secured near the mouthpiece or a specially designed contact pickup can be integrated into the instrument's headpiece for amplifying or processing the signal.

The Clarinet

The clarinet comes in two pitches—the B clarinet with a lower limit of D3 (147 Hz) and the A clarinet, which has a lower limit of C3 (139 Hz). The highest fundamental lies around G6 or 1.57 kHz, while notes in the octave above middle C contain frequencies up to 1.5 kHz when played softly. When played loudly, the clarinet's spectrum can range up to 12 kHz.

The frequency components of the clarinet radiate exclusively from the finger holes at frequencies between 800 Hz and 3 kHz, but as the pitch rises, more of the sound emanates from the bell. A reflective floor surface reflects the upper frequencies back to the listener, producing a more brilliant sound. If the player holds the instrument correctly (Fig. 8-6), the best mike placement would be to aim at the lower finger holes from a distance of 6″ to 1′ (15–30 cm). In this manner, the sounds originating from the finger holes and the bell are picked up with equal intensity.

The Saxophone

The metal-bodied saxophone varies greatly in size and shape. The most popular models for rock and jazz are the "S" curved B-flat tenor sax, whose fundamentals span from B2 to F5 (177 to 725 Hz) and the E-flat alto, with a span from C3 to G5 (140 Hz to 784 Hz). Also in this family are the straight-tubed soprano and sopranino and the "S" shaped baritone and bass saxophone.

Fig. 8-6. Typical mike positioning for clarinet.

Playing technique affects the harmonic content of the instrument, which generally runs up to 8 kHz and is extended by breath noises that take the range to a peak between 12 to 13 kHz.

Unlike the rest of the woodwinds, the saxophone is a "closed system," radiating all of its frequencies within the 3-dB band from the bell. The mike should thus be within the dispersion angle and pointed roughly toward the middle of the instrument (Fig. 8-7). Keypad noises are considered a part of the instrument's sound, but even those may be eliminated by aiming the microphone at the outer rim of the bell.

The Harmonica

While harmonicas ("harps") come in many shapes, sizes, and keys, they are divided into two basic types: diatonic and chromatic.

The pitch is determined by the vibrating reed (its length, width, and thickness); the harp player's habit of forming his or her hands around the instrument is a method of molding the tone by forming a resonant cavity.

As part of their technique, harp players like to get close to the microphone. They may achieve a deepened tone or get a special "washing" effect by opening and closing one hand. Often, they hold the microphone in the cavity formed by their palms (Fig. 8-8). For this reason, harmonica players often carry a preferred microphone with them (Fig. 8-9), rather than be rooted in front of an unfamiliar microphone and stand.

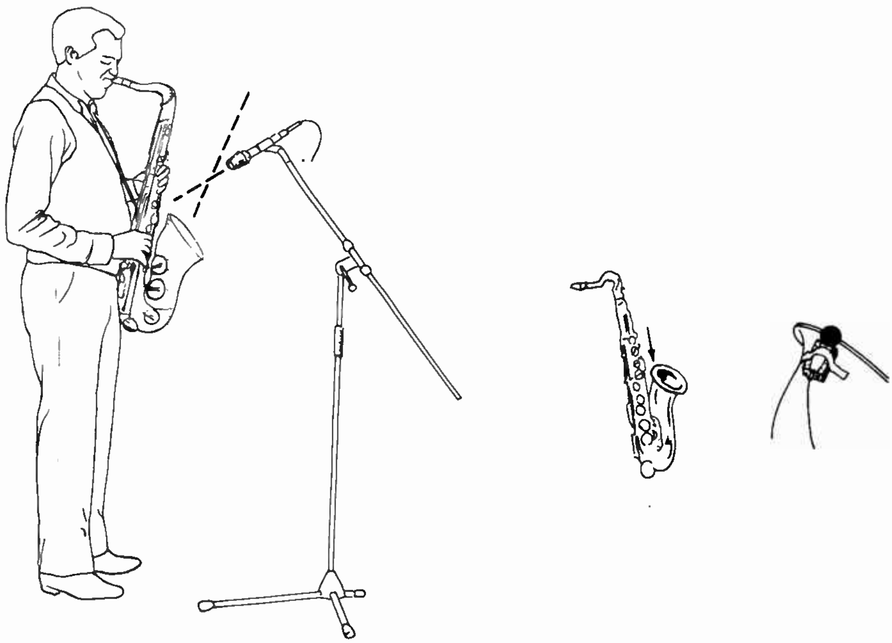


Fig. 8-7. Typical mike positioning for saxophone.

Fig. 8-8. Typical positioning for harmonica microphone. *(Courtesy of Shure Brothers, Inc.)*

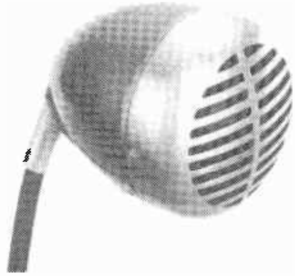


Fig. 8-9.
Shure 520D "Green Bullet" microphone.
(Courtesy of Shure Brothers, Inc.)

Keyboard Instruments

The Grand Piano

The grand piano is the classic example of the principle that "sound comes from many parts of an instrument." There are about as many approaches to the proper pickup of the piano as there are professional engineers. This is because the grand piano is not only very large, but it is also acoustically complex.

The piano's sound includes the "traditional style," but it has also evolved to include a contemporary style consisting of a compromise between a realistic sound and acoustic separation in the studio. Thus the piano may be picked up in one of a number of styles, depending on what is "right" for the given job.

Figure 8-10 shows some of the microphone positions currently accepted for the pickup of the grand piano. Keep in mind that these are guidelines and that a proper balance can be achieved through microphone choice and experimentation in microphone placement.

- In position 1, the microphone is attached to the partially or completely opened lid. For this, a boundary mike would be most appropriate. This method makes use of the lid as a collective reflector, giving an excellent pickup under restrictive conditions (stage, live video, etc.)
- Position 2 shows two microphones placed in a spaced-stereo configuration at a working distance of 6″ to 1′ (15–30 cm), with one microphone centrally positioned over the low strings and the other placed over the high strings.
- Position 3 shows a single microphone coincident-stereo pair placed just inside the piano between the soundboard and its fully or partially open lid.
- Position 4 shows a single microphone or coincident-stereo pair outside the piano facing into the open lid. This method is most appropriate for solo or accent miking of the instrument.

- In position 5, a spaced-stereo pair is placed outside the lid, facing into the instrument.
- Position 6 shows a single microphone or coincident-stereo pair just over the piano hammers at a distance of 4″ to 8″ (10–20 cm), providing a driving popular or rock sound.

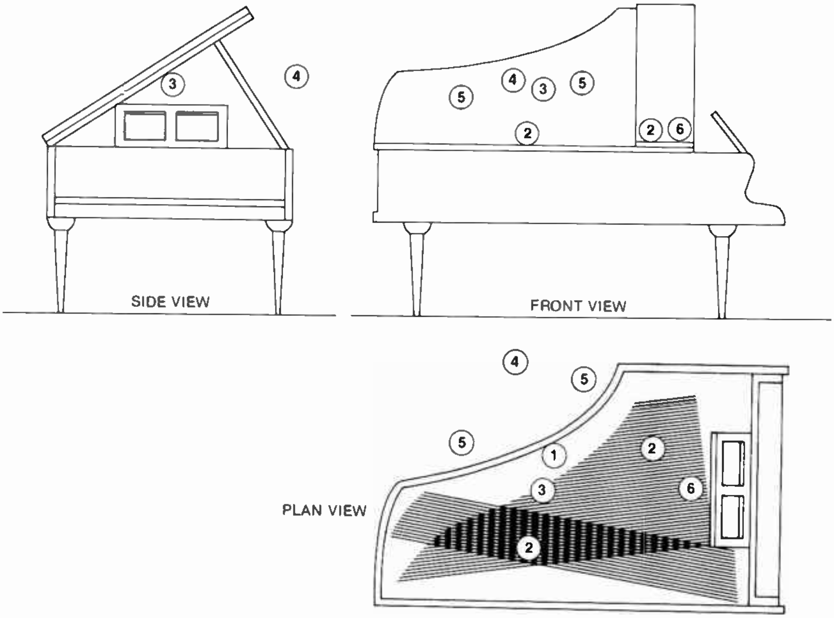


Fig. 8-10. Possible miking combinations of the grand piano.

Condenser or extended-range dynamic microphones are usually chosen to mike an acoustic grand piano, because they accurately represent the transient and complex nature of the instrument. When close miking or reduction of leakage is required, a cardioid or tighter polar pattern may be used. Where leakage is not a problem the omnidirectional pickup may be preferred to capture an overall natural sound.

Separation

Achieving *separation* is often a problem when the grand piano is placed next to raucous musical neighbors. Any of these means can be used to get separation:

- Place the piano inside an isolation booth.
- Place a flat between the piano and its loudest neighbor.
- Place the mikes inside the piano and lower the lid onto its "short stick." A heavy moving blanket or a regular blanket can be placed over the lid to further reduce leakage. This technique works best with closer microphone techniques.

- Overdub the instrument at a later time. In this instance, the lid may be propped up by the "long stick" and the mikes placed at a more natural-sounding distance.

The Upright Piano

You would expect the techniques for this seemingly harmless type of piano to be similar to its larger brother. This is generally true, but, since this piano was designed for home enjoyment and not performance, the techniques employed are slightly different and it is often more difficult to achieve a respectable tone quality.

In close miking the upright piano, two factors dominate:

- Because of the piano's design, access to its strings may be restricted, limiting microphone placement.
- Distant miking—and often close miking—tends to sound muddy when the upright is recorded in a small room.

Miking over the Top

Place two mikes in a 3-to-1 fashion just over the piano's open top: one over the bass strings and the other over the treble strings (Fig. 8-11). If isolation is not a factor, remove or open the front face, which covers the strings, in order to reduce reflections and thus the instrument's "boxy" quality.

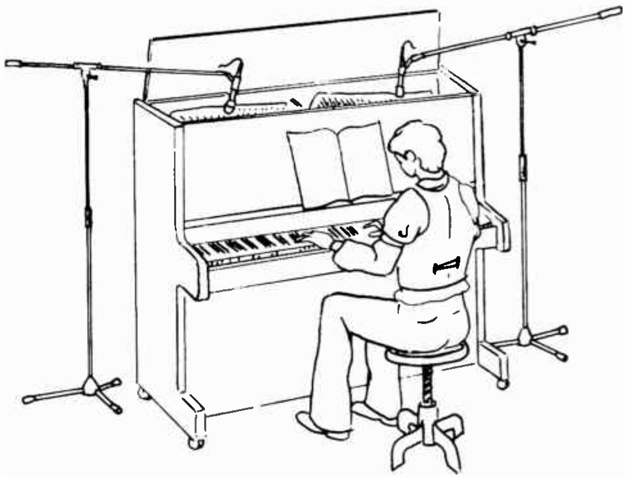


Fig. 8-11. "Over-the-top" mike placement for upright piano.

Miking the Kickboard Area

For a natural sound, remove the kickboard (at the lower front part of the piano) in order to expose the strings. Place a stereo-spaced pair over the strings, one each about 8″ (20 cm) over the bass and treble strings. If only one mike can be spared, place the mike over the treble end.

Miking the Soundboard

To reduce excessive hammer attack, place a microphone pair about 8″ (20 cm) from the soundboard, covering the bass and treble strings. To reduce muddiness, the soundboard should be facing into the room and not into a wall.

Boundary Miking

The boundary microphone often works well in the pickup of the upright piano, although placement will vary with environment and instrument. To begin, the mike(s) may be affixed to the nearest large boundary or to the top lid (which may be closed for isolation).

The Harpsichord

The similarities between the harpsichord and the piano create similar difficulties in pickup; however, the harpsichord's quietness and the rumbling or clonking noises of its action create additional difficulties.

One method is to treat the harpsichord exactly like the piano, but a microphone placed too close to the strings then will give an unacceptable plucking emphasis. The best method in a solo or accent pickup is to employ a distant coincident-stereo pair, a technique that requires a very quiet environment.

Electronic Keyboard Instruments

Most electronic musical instruments, such as synthesizers, samplers, and drum machines, are picked up directly from the device's output for maximum clarity. In the studio the direct-injection (DI) box may be employed to reduce the line level and impedance to a level acceptable for the console's input stage.

When played in the home or studio control room, many of the line-level outputs of these devices can be plugged into the line input of the audio production console or audio tape recorder.

An electronic organ may be quite different in application. A good Hammond or an older organ can sound wonderfully dirty through its own loudspeakers, a sound that is more appropriate when miked. Many of these organs may be played through a *Leslie*-type cabinet, which adds a spacial "swirl," a characteristic pitch rise and drop of the Doppler effect. Inside the cabinet is a set of rotating speakers which spin on a horizontal axis. This produces a pitch-based vibrato as the speaker is accelerated toward and away from the listener or microphone (Fig. 8-12).

The upper high-frequency loudspeaker may be miked by either one or two microphones (each panned left and right), with the low-frequency driver picked up by another microphone. The whirring of motors and baffles may produce quite a lot of wind and noise, perhaps calling for a wind screen or experimentation in placement.

The electric piano and many of the newer electronic devices often provide

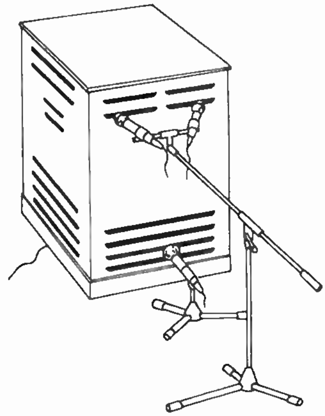


Fig. 8-12. Miking the Leslie organ.

a stereo output signal. Whenever possible, these signals should be placed on separate tracks for later panning to provide a wider stereo effect.

Percussion Instruments

The Drum Set

The standard drum set (Fig. 8-13) is usually the backbone of modern recorded music, in that it provides the "heart beat" of the basic rhythm track.

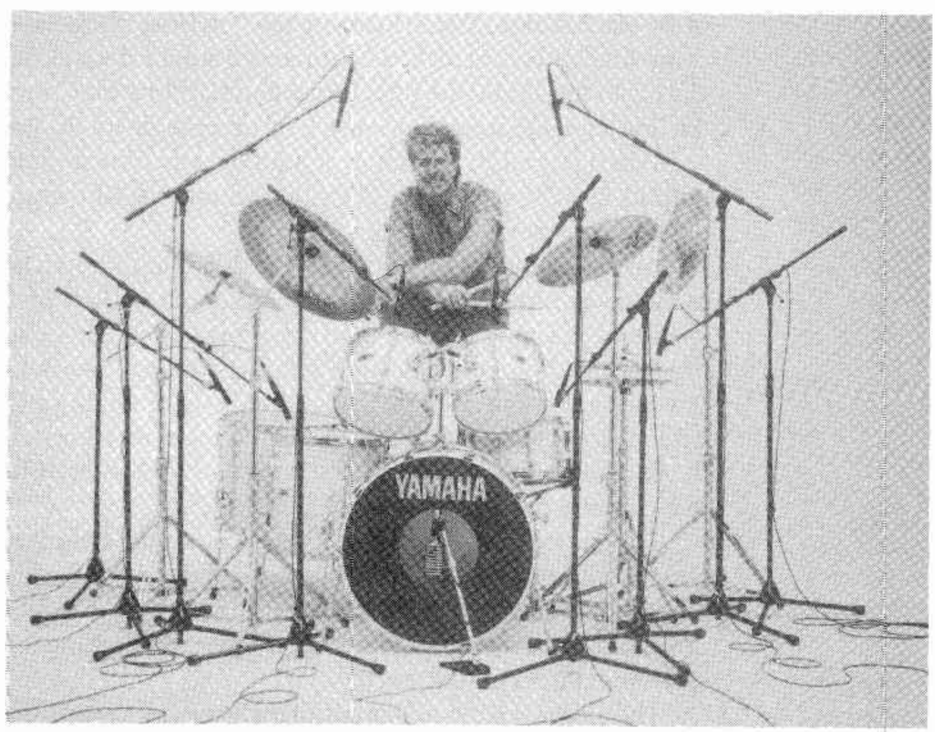


Fig. 8-13. J. R. Robinson, photographed on his studio drum set. *(Courtesy of Beyer Dynamic, Inc.)*

Because of this, a proper drum "sound" is of primary importance to the final impact of a recording.

Today's drum set generally comprises a bass (kick) drum, a snare drum, high-tom (one or more), low-tom (one or more), hi-hat, and a variety of cymbals. Since a full drum set is a series of interrelated and closely spaced percussion instruments, it is often a difficult instrument from which to obtain a proper spacial and tonal balance. The larger-than-life driving sound of the acoustic drum set is the result of expert balance between professional playing technique, proper tuning, and proper mike placement. As a general rule: a poorly played or tuned drum will sound just as bad through a good mike as it will through a bad one. Thus, it is important that the drum sound good to the ears, before microphone placement is attempted.

Tuning the Drum Heads

One secret of creating a good drum sound lies in careful tuning. Getting a good sound on tape will be much easier if the sound within the studio is right, as in the following description given by Bruce Bartlett:

> . . . we should first introduce the effect which the drum head will have upon the instrument's sound. Plain heads have a maximum ring or sustain, while hydraulic heads or heads with sound dots dampen the ring. Thin heads provide a sharp attack, good sustain and a weak projection. Thick heads have a duller attack, rapid decay and strong projection. Old used heads tend to become dull and muffled, while newer heads sound crisp.
>
> **Tom-Toms**
>
> The following is one suggested tuning procedure for the tom. First, take off the heads and remove the damping mechanism, which is a possible source of rattles. Place only the top head back onto the tom and hand-tighten the lugs. Then, using a drum key, tighten opposite pairs of lugs one at a time, one full turn each. After all of the lugs have been tightened in this manner, repeat the process, tightening each a half turn. Then apply heavy pressure to the head in order to stretch it. Continue to tighten, one-half turn at a time, until the desired pitch has been reached. The most pleasing tone is reached when the heads are tuned within the resonant range of the shell reinforcement.
>
> The bottom head may be removed from the tom for the best projection and the broadest range of tuning. In this case, pack the bottom lugs with felt to prevent rattles. Should extra control over the head be required, the bottom head may be added to the tom. In this case, projection is best if the bottom head is tighter than the top head (say, tuned a fourth above the top head), resulting in a muted attack, an "open" tone and some note bending. If the

bottom head is tuned looser than the top, the tone will be more "closed," with a good attack.

Bass (Kick) Drum

. . . a loose head will give a lot of slap attack and almost no tone. The opposite is true for a tight head. Tune the head to complement the type of music. Additionally, a hard beater adds attack.

Snare Drum

Tune the snare drum with the snares off. A loose batter head or top head gives a deep, fat sound. A tight batter head sounds bright and crisp. With the snare head or bottom head loose, the tone is deep with little snare buzz, while a tight snare yields a crisp snare response. Set the snare tension just to the point where the snare wires begin to "choke" the sound, then back off a little in tension.

Sometimes a snare drum will buzz in sympathetic vibration with other instruments. This buzz may be controlled by wedging a thick cotton wad between the snares and the drum stand.

Damping and Noise Prevention

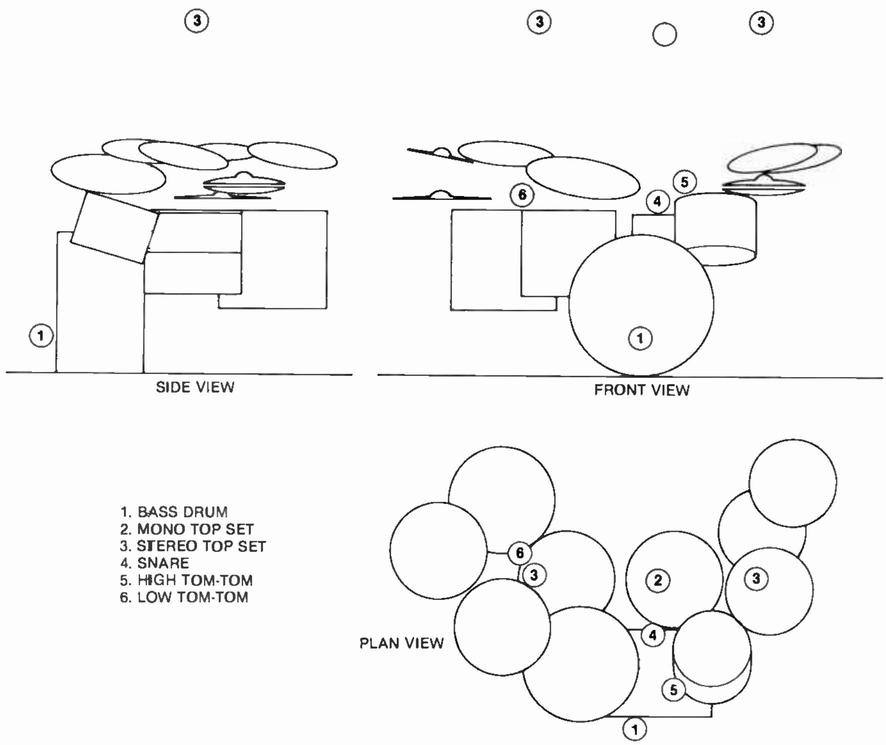
If the toms or snare rings excessively, gauze pads or folded handkerchiefs may be taped to the edge of the heads. Place the tape on three sides of the pad so that the untaped edge is free to vibrate and dampen the head motion. Do not overdo the damping or the drum set will sound too "dull" and "thuddy."

To reduce excessive cymbal ringing, apply drafting tape in radial strips from the bell to the rim. Also, the kick-drum pedal may be oiled in order to prevent squeaks. Rattling hardware may be tapped into place with drafting tape.

Miking the Drum Set

Once the drum set has been tuned and adjusted for the best sound, the microphones can be positioned (Fig. 8-14). As each part of the drum set is so different in sound and function, each should be considered an individual instrument and miked accordingly. The characteristics that assist in this match-up include frequency response, polar response, proximity effect, and transient response.

The dynamic range of a microphone is another important performance characteristic to consider when miking a drum set (Table 8-1). The set is ca-



(A) *Side view.* (B) *Front view.* (C) *Top view.*

Fig. 8-14. Typical microphone placements for a drum set.

pable of generating extremes of volume and power as well as soft, subtle sounds which add color to the music. A mike must be able to withstand the peaks without distorting and to capture the nuances without adding noise of its own.

Table 8-1. Typical sound pressure levels for drum set elements.

INSTRUMENT	MIKE DISTANCE FROM INSTRUMENT	PEAK SPL
Snare Drum	1" (2.5 cm)	152 dB
	10" (25 cm)	145 dB
	40" (1.2 m)	126 dB
Low Tom-Tom	1" (2.5 cm)	149 dB
	10" (25 cm)	142 dB
	40" (1.2 m)	134 dB
Cymbal	1" (2.5 cm)	154 dB
	10" (25 cm)	142 dB
	40" (1.2 m)	128 dB
Bass Drum	1" (2.5 cm)	154 dB
	10" (25 cm)	146 dB
	40" (1.2 m)	136 dB

As the drum set is usually one of the loudest sound sources in the sound studio, the drum set is often placed on a 1′ or 1½′ (30–45 cm) riser in order to reduce bass leakage. In preventing drum leakage to other mikes, 4′ flats are often placed around the drum set. For greater isolation, the entire set may be placed in an isolation booth—generally an acoustically dead-to-semilive room with windows for visual communication.

The Snare Drum—A compromise is needed between the ideal microphone placement and the movement patterns of the drummer. Therefore, one of the first considerations is simply that of finding a safe place for the microphone where it won't be bashed by a passing stick.

For rock drumming, the microphone is generally aimed just inside the top rim of the snare drum, at a distance of about 1″ (2.5 cm) (Fig. 8-15). The microphone should be aimed for the best separation from other drums and

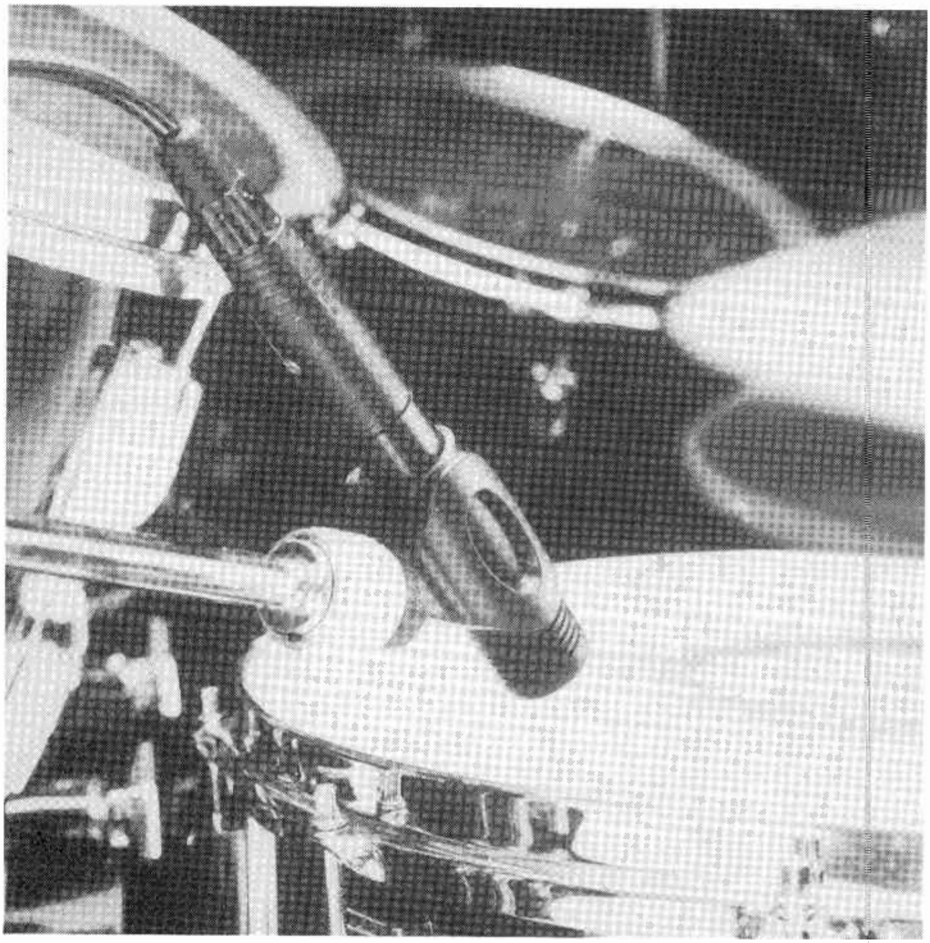


Fig. 8-15. Microphone positioning for snare drum. *(Courtesy of Beyer Dynamic, Inc.)*

cymbals. Such a microphone will most often be a cardioid, with a hypercardioid response offering the tightest on-axis angle. Its rejection angle is aimed at either the hi-hat or high-toms, depending on leakage difficulties.

In certain musical forms, such as jazz, a crisp sound may be desired. This can be gotten by facing a mike at the bottom head, in addition to the mike for the top head. As the bottom head is 180° out of phase with the top, it is wise to reverse the phase polarity of this mike.

The Hi-Hat—The sound of the hi-hat falls within the same overall frequency range as the snare drum, but the hi-hat produces much more high-frequency energy, while the snare's sound is concentrated in the mid-range.

Moving the hi-hat microphone does not change the overall sound as much as it does on the snare; however, four points should be kept in mind:

- The sound is brightest over the edge of the top cymbal.
- Placing the mike above the top cymbal (Fig. 8-16) reproduces all the nuances of stick attack.
- The opening and closing motion of the hi-hat produces rushes of air; therefore, the mike should not be placed directly off the cymbal edges.
- If only one microphone is available or desired, it can be used for both the snare and hi-hat by placing it equidistant between the two and facing it away from the high-toms.

Fig. 8-16. Microphone positioning for hi-hat. *(Courtesy of Beyer Dynamic, Inc.)*

The High-(Rack-)Toms—The mounted high-toms may be miked either for each tom individually or with a single, overall microphone placed a short distance away. When miked individually (Fig. 8-17A), the microphone may be placed close to the top head (about 1″ (2.5 cm) over and 2″ (5 cm) in from the rim). This will give a tight, "dead" sound with less shell resonance and more head attack. This distance may be increased to about 3″ (7.5 cm) in order to give a "more-live" sound. A single microphone may be used on two high-toms (Fig. 8-17B) by placing it between and slightly above the drums on the side away from the drummer. To reduce overall leakage or feedback potential, a hyper-cardioid response pattern may be chosen.

One other method for reducing leakage, and for placing the pickup out of the drummer's way, is to remove the bottom heads from the toms and mike them inside, a few inches away from the head. The sound picked up from inside the tom will generally yield less of an attack and a fuller tone.

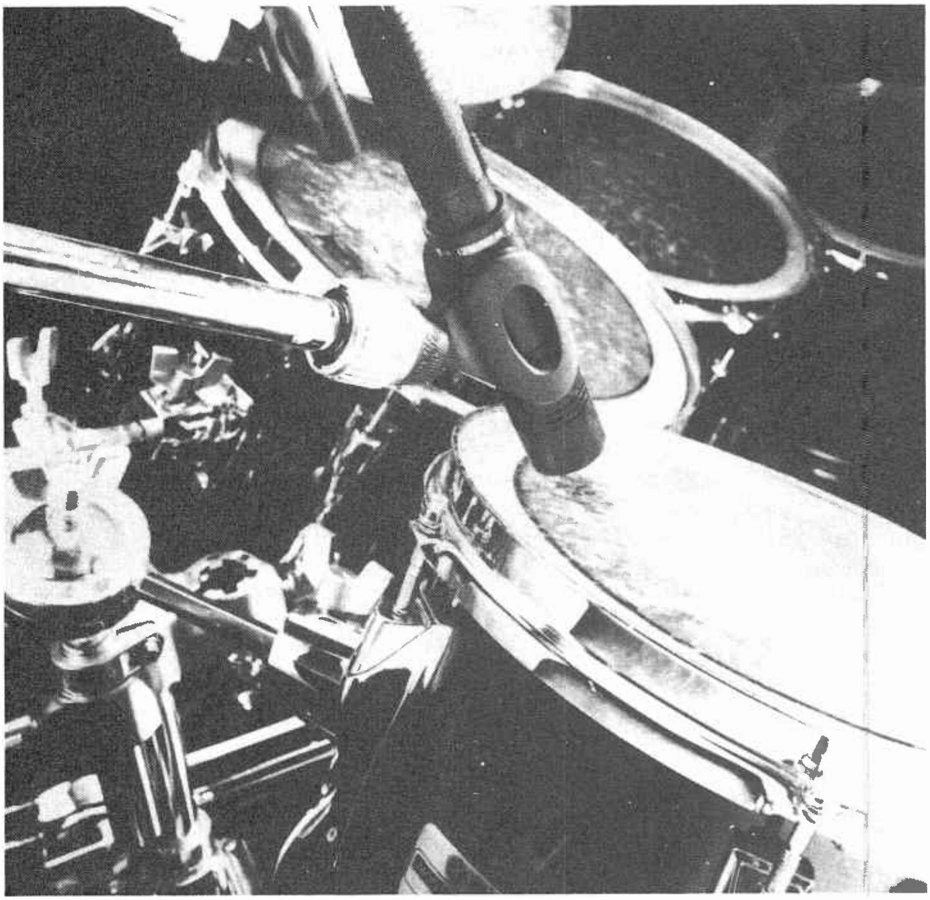


Fig. 8-17. Typical microphone placement for high-tom. *(Courtesy of Beyer Dynamic, Inc.)*

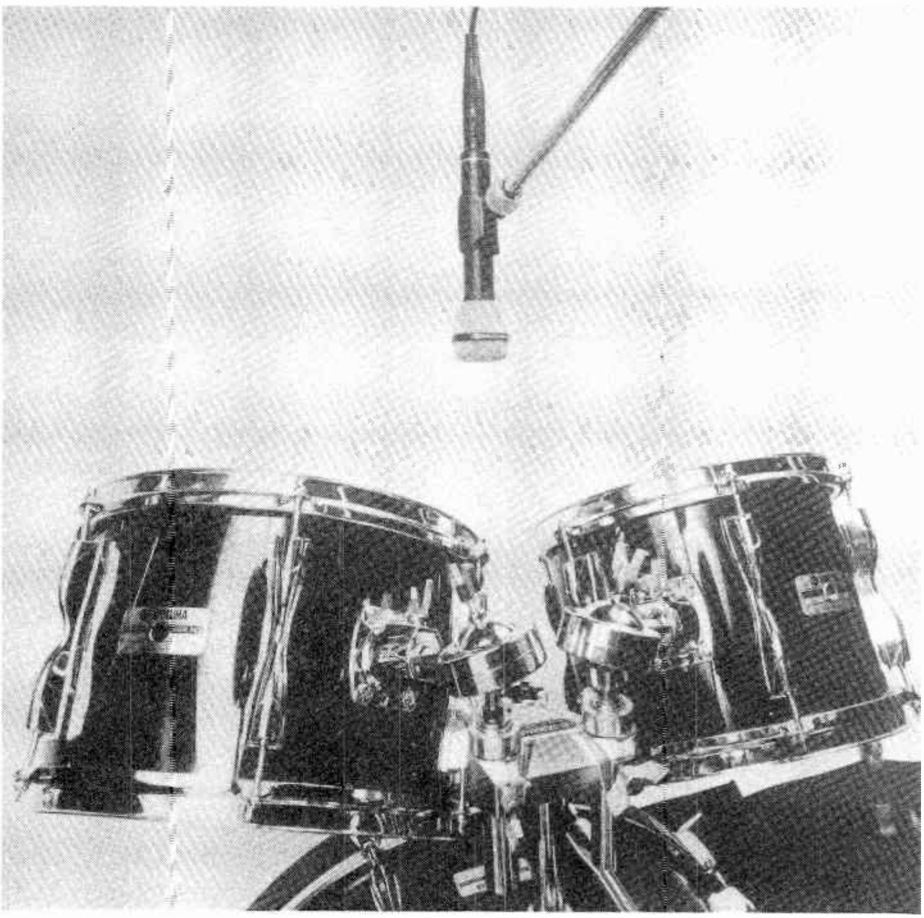
(A) Individually miked.

The Low-(Floor-)Tom—The recommended placement for the low-tom is similar to that for the high-tom (about 1–3″ (2.5–7.5 cm) above the top head). Again, one microphone may be used between two low-toms, but one mike per drum gives greater control over panning and tone color.

The Bass (Kick) Drum—Low-frequency reproduction at high sound pressure levels is essential to the quality pickup of the bass drum (also known as the kick drum). It is necessary, therefore, to choose a microphone that is designed to reproduce low-frequency signals at high working levels. This often means choosing a large diaphragm, dynamic microphone, such as the AKG D-12E, Electro-Voice RE-20, or the Beyer M-380.

Because of the extreme proximity effect encountered at close working distances when using directional microphones, even a minor change in placement may have a profound effect on the pickup. Moving the microphone

Fig. 8-17 (cont.)



(B) Single-mike placement for two high-toms.

closer to the head adds warmth and fullness, moving it further back often emphasizes the high-frequency "click" (Fig. 8-18). A mike placement close to the beater picks up a "hard" beater sound, while an off-center placement picks up more of a skin tone.

Fig. 8-18. Mike placement just outside the bass-drum head brings out the low-end and natural fullness. *(Courtesy of Beyer Dynamic, Inc.)*

Placing a blanket or other damping material inside the drum shell firmly against the beater head tightens a dull and loose kick sound to a sharper, more defined transient sound. Cutting the kick's equalization at 300 to 600 Hz, reduces the dull, "cardboard" sound, while boosting from 2.5 to 5 kHz adds a sharper attack, giving a "click" or "snap."

Overhead Miking

One or more overhead microphones may be needed to reproduce the high-frequency transients of cymbals with crisp, accurate detail, while providing an overall blend of the entire drum set. Because of these transients, a condenser microphone is often chosen for its accurate high-end response.

While the choice of overhead mike positioning is quite subjective, two popular methods dominate: 3-to-1 positioning and coincident positioning.

When using an overhead pair in a 3-to-1 spacing position, the widest stereo image is obtained by facing the mikes away from each other in an outward fashion (Fig. 8-19A). Phasing problems may be caused as a result of sound waves arriving at each pickup point at different times. If the 3-to-1 rule does not fix this problem to your satisfaction, try positioning the mikes in a closely spaced X/Y coincident array (Fig. 8-19B). In this way, a stereo image is maintained. As there is little distance between microphones, however, all sounds arrive in phase at both microphones, preventing phase cancellations if the recording is heard in mono.

With certain drum-miking setups, the need for separate overhead microphones may be reduced or entirely eliminated. This is due to the leakage from the cymbals into the snare and tom mikes. Additionally, in a live-reinforcement situation, the loud cymbals may be easily heard above the overall pickup; thus separate overhead mikes are not always necessary.

Although two overhead microphones are the industry standard, using a single mike when either the budget or mixer channels is limited is quite common. Using one microphone prevents a stereo image, but it does provide adequate coverage while reducing phase problems.

Minimal Miking

The minimal approach of drum-set miking (Fig. 8-20) is sometimes dictated by financial considerations, but there are also aesthetic considerations involved in this approach. Many people feel that the recorded or amplified sound of a drum set is simply more realistic (or more realistically simple) when just a few mikes are placed and balanced carefully. This is often true in recording jazz drumming.

For a minimal setup that combines individual control over key elements with a balanced pickup of the whole set, place a quality dynamic or condenser microphone between the snare and hi-hat, an extended bottom-range mike at the bass drum, and a quality condenser/electret-condenser overhead. If stereo imaging is important, use a pair of quality overheads.

Distant Miking of the Drum Set

When "new music" came on the scene, distant miking techniques in the recording studio became an additional tool to get a fuller, more ambient sound.

Such a distant miking technique, which may be accomplished in addition to or in place of traditional close miking techniques, uses any of these approaches:

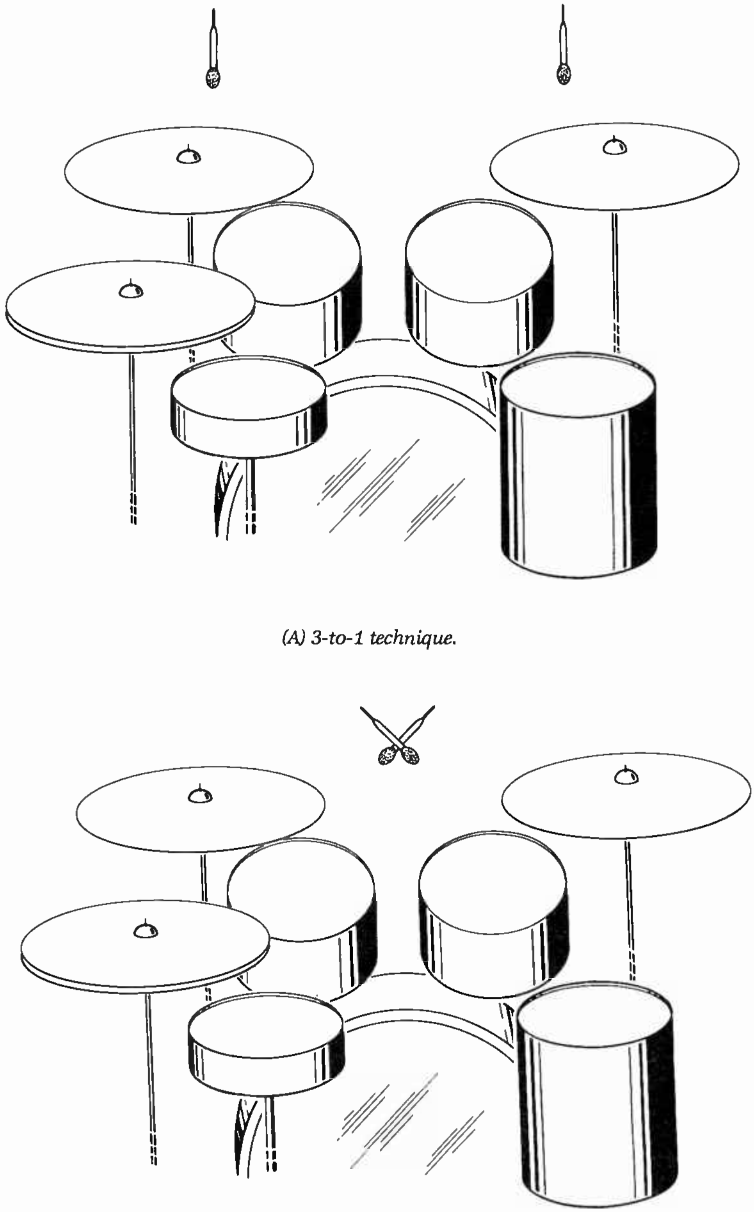


Fig. 8-19. Typical stereo overhead mike positioning.

(A) 3-to-1 technique.

(B) X/Y coincident technique.

- Place a spaced microphone pair 10′ to 20′ (3–6 m) from the set, facing towards the source.
- Place a coincident (X/Y or Blumlein) microphone pair 10′ to 20′ (3–6 m) from the set, facing toward the source.

- Place boundary microphones on two major boundaries in the room (taped to hard-surfaced flats surrounding the drummer or on major surfaces within the room).

This technique relies heavily upon the acoustics of a studio or room, which affect the recorded sound. For best results when only one sound source is being recorded, use an omnidirectional or boundary microphone.

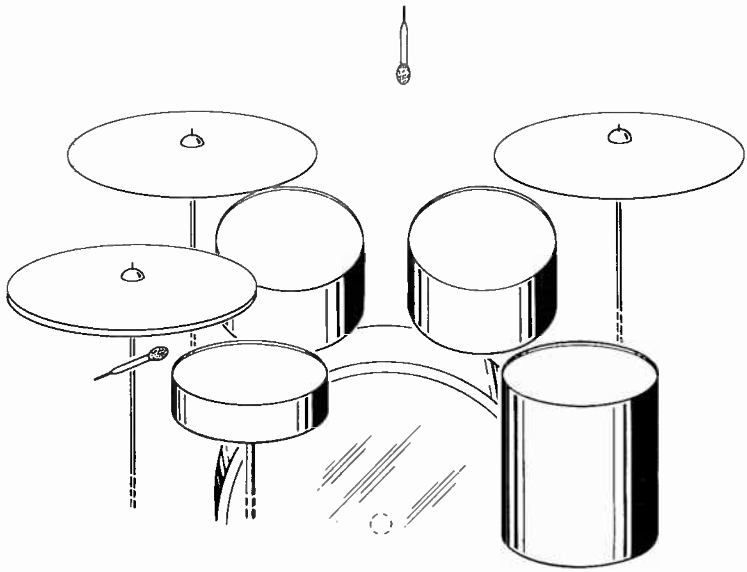


Fig. 8-20. Minimal miking technique.

Tuned Percussion Instruments

Xylophone and Vibraphone

The common technique used for the tuned percussion instruments is to place two good condenser or extended-range dynamic mikes about 1½′ (46 cm) above the playing bars, spaced about 3′ (1 m) apart. While coincident-stereo techniques eliminate phase errors, this arrangement provides good coverage and an excellent stereo image. Miking from underneath often gives a less-defined sound and picks up more room leakage.

Timpani

Capable of low, sustained rumblings as well as loud, transient bangs, the timpani are among the most commonly used drums in classical music.

As the stretched membrane works together with a kettle shell to produce the overall sound, these instruments sound best when miked at a distance of 1½′ (46 cm) or more (Fig. 8-21). These instruments are often played in groups of two or four, and a pair may share a microphone in order to reduce phase errors.

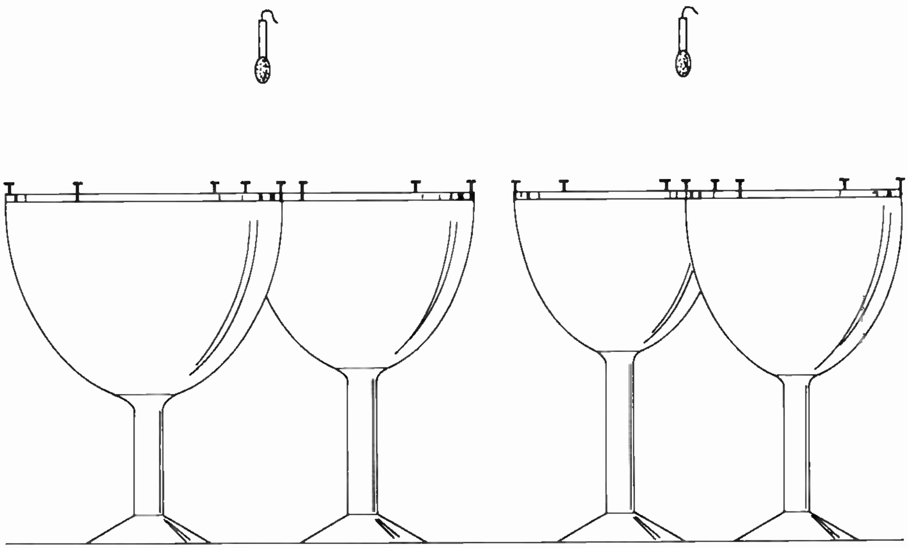


Fig. 8-21. Typical "close miked" placement for timpani.

The Steel Drum

Similar to the timpani, the steel drum doesn't take well to being close miked. It is preferable to mike the steel drum at a distance of 1′ (.3 m) or more above the head. A good deal of the "steel" sound radiates downward from the head. To get this sound, place a microphone facing up, about 1′ (.3 m) below the instrument.

Non-Tuned Percussion

The Conga and Tumba

The conga and tumba are low-pitched, single-headed drums, physically differing only in diameter. If only one microphone is available, it may be aimed between the drums, placed between 5–10″ (12–24 cm) away. Better control over the sound may, however, be obtained when the drums are miked individually. The microphones may be placed close to the top head (about 1–3″ (2.5–7.5 cm) above and 2″ (5 cm) in from the rim), giving a tight, "dead" sound with less shell resonance and more head attack. The pickup distance above the head may be increased to about 10″ (25 cm) to give a "more-live" sound. Face the heads of two directional microphones at a 45° angle for best separation and improved stereo spread (Fig. 8-22).

The Bongos

Two microphones may be employed to mike bongos, but usually a single microphone is placed between the two heads, from 5–10″ (12–25 cm) away.

Fig. 8-22. Microphone pickup of conga drums.

Percussion Toys

The toy chest of a percussionist may contain from triangle, tambourine, bird whistles, and maracas all the way to little rubber frogs that squeak. Such instruments are generally transient in quality, requiring a high-quality condenser or extended-range dynamic mike. Given adequate isolation from other instruments, a working distance of 1–4′ (.3–1.2 m) is preferred in order to pick up some of the room's natural acoustics and prevent microphone overload.

Stringed Instruments

Of all the instrumental families, the string family is perhaps the most varied. Eastern countries still use single-stringed instruments that are capable of producing subtle and rich tones. Western classical listeners have grown used to the violin, viola, cello, and double bass. Guitars abound as four-, six-, and twelve-stringed instruments. Various construction details, sound posts, and structual reinforcement schemes create asymmetry in interior design—often quite deliberately in order to enhance or cut back certain harmonic frequencies. This may also alter the sound radiation. The guitar, for instance, emits bass notes uniformly, but the radiation becomes more directional as frequency rises.

The Violin and Viola

The frequency range of the violin runs from 200 Hz to 10 kHz and a good microphone that displays as flat a frequency response as possible should be used. The violin's fundamental range is from G3 to E6 (200 Hz to 1.3 kHz). It is particularly important that the microphone be flat within the formant frequencies of 300 Hz, 1 kHz, and 1.2 kHz. The fundamental range of the viola is tuned a fifth higher and contains fewer harmonic overtones.

Up to 500 Hz, the sound radiation of the violin is uniform. Above this point, the sound projection is concentrated in a direction perpendicular to the soundboard. As the overall frequency dispersion is approximately 15° to the instrument's front face, the microphone is generally placed directly in line with this angle and pointed at the sound holes.

Under most conditions the microphone for the violin or viola should aim at the instrument's front face. The distance depends upon the musical application of room acoustics. As miking distance decreases, a scratchy, nasal quality is picked up. For a solo instrument, the microphone should be placed between 3′ and 8′ (1−3 m) away and positioned slightly above and in front of the player (Fig. 8-23). Under studio conditions, the microphone may be placed closer (between 2′ and 3′ [.6−1 m]). For a fiddle or jazz/rock violin, the microphone may be placed as close as 6″ (15 cm) away. Under these conditions, the best sound can be found by experimentation.

Fig. 8-23. Typical pickup placement of violin.

In public address (PA) applications, distant miking is likely to produce feedback. In such an instance, a clip-type microphone may be attached to the tail-piece of the instrument (Fig. 8-24). At such close working distances, less amplification than usual is required, helping reduce the possibility of feedback. The sound quality of such a microphone may be changed by aiming it differently. Facing the microphone towards the strings gives a bright sound; facing the microphone away or parallel to the strings produces a slightly mellower sound. Fortunately, the pressure design and elastic suspension of most clip mikes make them insensitive to unwanted noise transmitted through the body of the instrument.

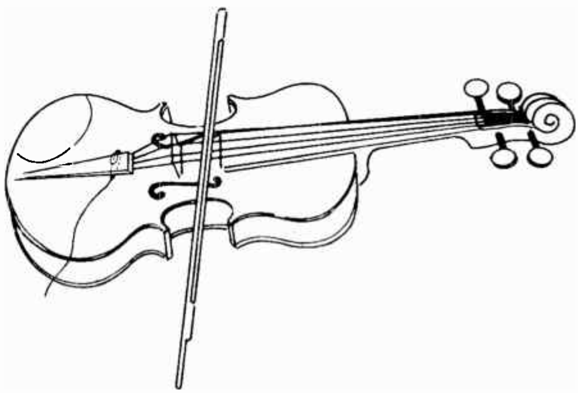


Fig. 8-24. Clip microphone placement on violin.

The Cello

The fundamental range of the cello lies from C2 to C5, corresponding to a frequency range of 56 to 520 Hz, with an overtone spectrum that rises to 8 kHz. Most important, it is the "o" formants, which lie between 350 Hz and 600 Hz, that combine to create the sonorous character of the cello.

If we assume the player's line of sight is 0°, the preferred direction of sound radiation lies between 10° and 45° to the right. A microphone is often placed level with the instrument and directed toward the sound holes. The microphone (usually a condenser or extended-range dynamic) should have a flat response and be placed between 6″ (15 cm) and 3′ (1 m) away (Fig. 8-25). In PA applications, a clip microphone may be fastened directly to the bridge of the instrument, as in Figure 8-26.

The Double Bass

The double bass is one of the orchestra's lowest-pitched instruments. The four-string type reaches down to E1 (41 Hz) and the five-string to C1 (33 Hz), while the upper limit of the fundamental range is around middle C (260 Hz). At frequencies below 70 Hz, the double bass has a very weak sound projection. This is fortunate, though, as it is the stronger frequencies that lie

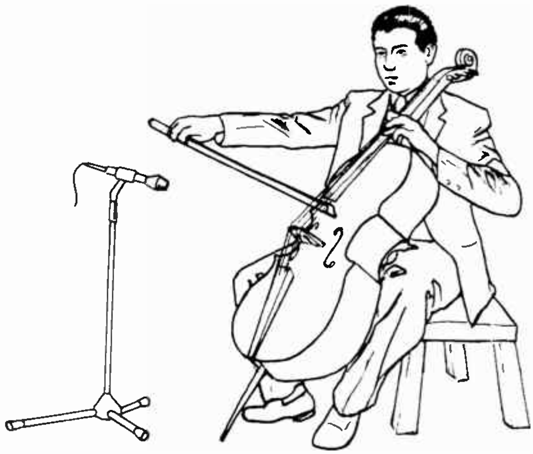


Fig. 8-25. Typical microphone placement for cello.

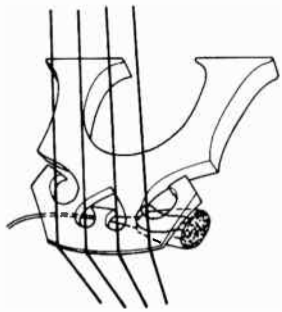


Fig. 8-26. Pickup of cello using clip microphone.

between 70 Hz and 250 Hz, giving the instrument its rich, dark quality. The overtone spectrum generally reaches up to 7 kHz; however, only a narrow band of frequencies (around 100 Hz) is radiated evenly in all directions. The angle of dispersion for all frequencies is roughly $\pm 15°$ from the player's line of sight. Once again, the microphone may be placed within this angle and aimed at the sound holes (Fig. 8-27) from a distance that depends on the type of music and acoustic environment. In the classical playing style, the double bass is usually bowed, which sets up a strong buzzing sound, often containing frequency components up to 10 kHz. In order to capture these subtle qualities, a microphone should be placed at a shorter working distance (1½ to 3′ [.45–1 m]) than that used for jazz or rock. In jazz the bass is usually plucked and its overtones are weaker and fewer in number. To reduce feedback and increase separation, the microphone may need to be placed closer, between 2–8″ (5–20 cm) away.

Some jazz players prefer to place a microphone inside the bridge (Fig. 8-28) or clip a microphone to the bridge, as this reduces the danger

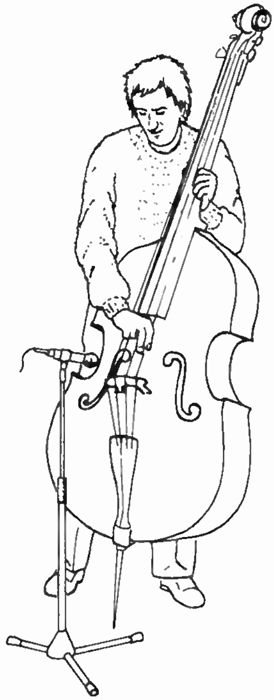


Fig. 8-27. Typical microphone placement for double bass.

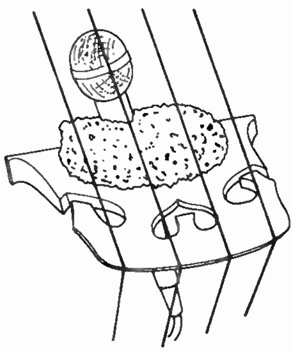


Fig. 8-28. Pickup of double bass using a mounted microphone.

of feedback. This method often produces a brittle or mid-range-heavy tone that may improve with experimentation with microphone placement and equalization.

Plucked String Instruments

The Acoustic Guitar

Nylon- and steel-string guitars are subtle instruments, requiring care in microphone choice and technique for best results. The instrument's body shape and type of strings and the player's style affect the tone, and all of these influences are best captured with proper placement.

The small-bodied classical guitar is normally strung with nylon or gut and is played with the fingertips, giving it a warm, mellow sound having few of the higher overtones produced by the steel-string acoustic. These upper harmonics can be picked up by placing the microphone near the center of the bridge, at a distance of between 8″–1½′ (20–45 cm) (Fig. 8-29). At this point the overtones are at their loudest and are reinforced by the guitar's top face. When miking a classical instrument under solo conditions, a greater working distance may be desired. In Chapter 7 we gave further details on the stereo techniques that apply to distant miking of the solo guitar.

Fig. 8-29. Typical pickup placement for classical guitar.

The steel-string guitar is a larger instrument which carries a brighter, richer set of overtones, especially when played with a pick. Microphone placement often varies from instrument to instrument, requiring that the recordist experiment. A natural tonal balance may be obtained by placing the mike at a point slightly off-axis, above or below the sound hole, between 6″–12″ (15–30 cm). A condenser microphone often is used for the acoustic guitar, as its smooth, extended frequency and transient response give a clear, detailed quality. Should this detail distract from the program material, a dynamic microphone may be used to yield a bit less detail.

Miking Near the Sound Hole

The sound hole on the front of a guitar serves as a bass port which resonates at lower frequencies (around 80 Hz to 100 Hz). A microphone placed too close to the front of this port will sound boomy and unnatural. This is, however, a popular miking position on stage or around high acoustic levels because the guitar's output is highest at this position. To achieve a more natural pickup under these conditions, the microphone's output can be rolled off at the lower frequencies (5 to 10 dB at 100 Hz).

Alternative Miking Styles

On stage, the microphone should be placed as near the guitar as is possible—not so close, however, that the player might hit the microphone, nor so close to the sound hole that the bass is emphasized. A distance of 8″ (20 cm) often works well. If mobility is required, a number of options are available to the guitarist:

- A clip microphone can be attached to the body of the instrument.
- A contact mike can be attached directly to the body of the instrument for total acoustic isolation.
- An electric mike can be incorporated into the instrument.
- Any of these systems can be connected to a wireless transmitter (this allows complete mobility).

The clip microphone, which can be fastened directly to the body of the guitar, can give a high-quality pickup with reduced leakage. A popular arrangement is to tape an omnidirectional clip mike between the sound hole and the bridge near the low E string (Fig. 8-30), or underneath the strings. The clip microphone can also be attached directly to the sound hole, often requiring that the bass be rolled off at the mixer to restore natural tone.

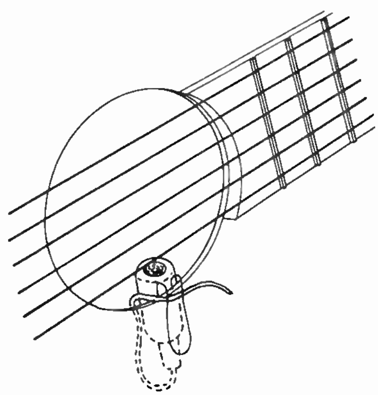


Fig. 8-30. Pickup of guitar using clip microphone.

Often the contact mike is a crystal type that directly picks up the body vibrations of the guitar. Best isolation is achieved with this method; the fidelity of the mike, however, is directly related to its position on the body of the guitar. This critical placement often varies from one instrument to the next. For starters, follow the manufacturer's recommendations. If this fails, place the mike near the bridge of the instrument and experiment.

Some acoustic guitars have an electric mike built into the bridge. These produce little or no feedback but are weak in picking up the upper and lower frequencies. Equalizing out the middle frequencies in a broadband fashion compensates for this.

The Electric Guitar

In the late 1930s, acoustic guitarists tired of being drowned out by the other musicians in a band and of not having the volume to play a solo part. This changed with the advent of the amplifier.

The fundamentals of the average 22-fret guitar extend from E2 to D6 (82 Hz to 1.17 kHz), with overtones pushing this upper range much higher. Not all of these frequencies are necessarily amplified, as the electric-guitar cord tends to attenuate frequencies above 5 kHz, unless the guitar has either a low-impedance converter or a low-impedance mike built in. The frequency limitations of the average guitar loudspeaker often add to this effect, as its upper limit is 5 or 6 kHz.

The electric guitar may be recorded by a number of means:

- a microphone placed in front of the amp's loudspeaker
- a direct box plugged into the guitar or external-speaker jack on the amplifier
- both miked and direct

Miking the Guitar Amp

The most popular guitar amplifiers for recording purposes are the small practice-type amp-loudspeaker. Such low-noise systems are designed to help out the suffering high end by incorporating a sharp rise in the response range at 4–5 kHz, thus promoting a clean, open sound.

The most popular type of microphone used for the electric guitar amp is the cardioid dynamic microphone, because it adds a full-bodied character to the sound without picking up extraneous noises from the amplifier. Often the microphone chosen has a presence peak in the upper-frequency range, giving an added clarity to the pickup. The cardioid pattern is used in order to reduce leakage. A condenser or ribbon microphone may be used in order to capture a quieter, more sensitive guitar part.

For a good pickup with a high degree of separation, a microphone may be placed at a distance of 2″ to 1′ (5–30 cm). When miking at a distance of less than 4″ (10 cm), microphone and speaker placement becomes slightly more critical (Fig. 8-31). For a brighter sound, the microphone should face directly into the center of the speaker's cone. Placing the microphone off-center to the cone produces a more mellow sound while reducing amplifier noise. When miking at a distance of greater than 6″ (15 cm), phase effects due

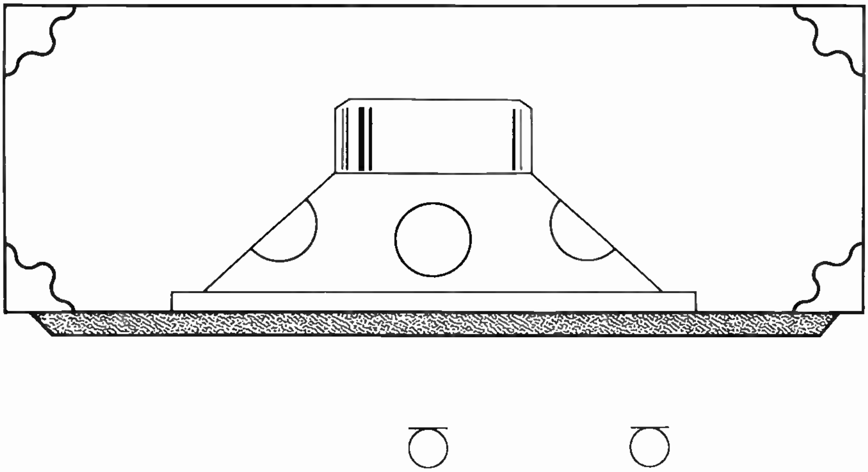


Fig. 8-31. Miking electric guitar cabinet.

to boundary interference can be reduced by placing the cabinet on top of a chair or by laying a small amount of carpet on the floor.

Recording Direct

The guitar's electrical signal can be recorded directly on tape for a cleaner, more present sound, bypassing the distorted character of the amp and eliminating leakage into the guitar-amp microphone. If the player unplugs the speaker and listens through headphones, the guitar amp does not leak into other microphones.

In the sound studio, a *direct injection (DI) box* (Fig. 8-32) is used for direct recording. It interfaces an electric instrument to the audio console (Fig. 8-33) by the following means:

- reducing the instrument's line-level output to mike level for insertion into the console's input
- changing the instrument's high-source impedance (unbalanced line) to a low-source impedance (balanced line) at the cable/console
- electrically isolating audio signal paths (thereby reducing the potential for a ground loop and thus hum)

This signal feed may be taken directly from the output of the guitar, after the effects device, or, in certain cases, from the external speaker jack of the amplifier, as in Figure 8-33.

Many such electrical instruments may be both miked and picked up with a DI and mixed at the console. This allows for a subtle combination of both the rugged mike sound and crisp DI sound. These may be combined in one

Fig. 8-32. A DI box. *(Courtesy of Whirlwind Music Dist., Inc.)*

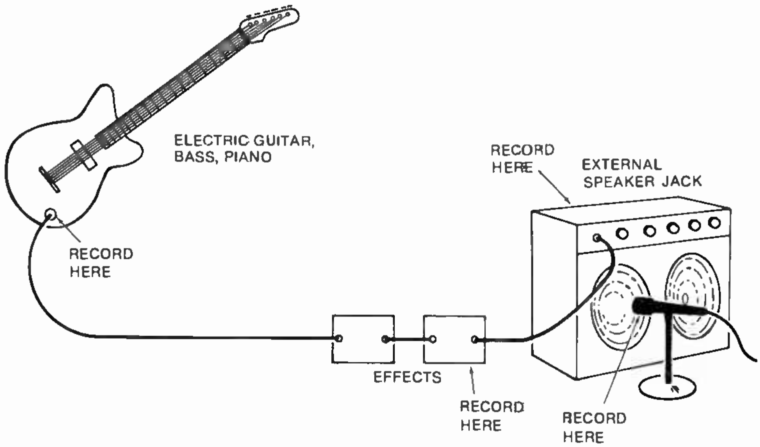


Fig. 8-33. Direct recording of electric instrument.

channel at the console or, where open tracks exist, put on separate tracks in the multitrack recorder for later decision-making.

The Electric Bass Guitar

The electric bass guitar operates in the range of E1 to F4 (41.2 Hz to 343.2 Hz), though if the player plays loudly or with a pick the added harmonics may range up to 4 kHz. Additionally, the playing style and choice of pickup greatly affect the bass sound. Playing in the "slap" style or with a pick gives a brighter, harder attack. As with the double bass, the finger style mellows the tone, which may be further deepened by choice of mike. As with the electric guitar, the electric bass may be either miked at the amplifier or picked up by direct injection (Fig. 8-34). The latter gives a cleaner, sharper attack but either style may be chosen.

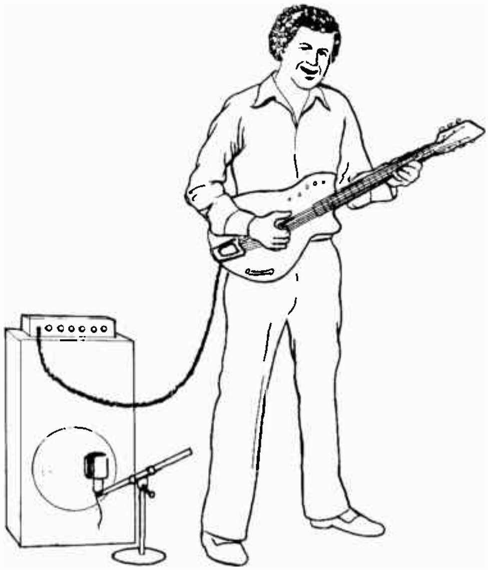


Fig. 8-34. Typical microphone placement for bass guitar.

For the cleanest possible sound, the bass guitar is almost always recorded directly. However, many players still prefer to mike the amp. To do so, the microphone should be placed 2–6″ (5–15 cm) away. Dynamic mikes are usually chosen because of their deep, rugged tone. Some large-diaphragm dynamic designs subdue the high-frequency transients and, when combined with a boosted response around 100 Hz, give a warm, mellow tone that is powerful in the lower register. Equalization of the bass signal may increase the clarity of a guitar, with the fundamental being effected from 125 Hz to 400 Hz and the harmonic punch being effected at 1.5 to 2 kHz. Care should be taken that an excessive amount of low-frequency energy is not recorded, as this may cause problems when placed on record and not be heard on most home systems.

One other tool for the electric and acoustic bass is the compressor. The signal output from one bass note may be weaker than the next note, causing frequency "holes" in the bass line. A compressor set at an input/output compression ratio of 4:1 with a fast attack (8–20 ms) and slower release time (¼–½ second) often smoothes out the levels, providing a strong, present bass line.

The Banjo

The banjo has a head stretched across its body, much like a drum, that amplifies the string's vibrations. While containing strong fundamentals in the lower mid-frequency range, the complex harmonic structure of the banjo can easily continue up to 10 kHz. This, combined with a sharp, transient quality, gives the banjo a clear, present sound.

To pick up a natural instrument blend, place either a condenser or dynamic microphone 6–12″ (15–30 cm) away. The condenser provides a sharp, cutting quality, while the dynamic provides a more rugged, fundamental sound.

A natural sound can be obtained by miking slightly off-center to the front face of the head. Positioning a microphone near the center of the head produces a harsh, thumpy sound, which thins out when the microphone is moved toward the outer edge. The closer working distances are useful for maximum isolation and stage mobility. A clip microphone may be attached to the tailpiece and aimed towards the bridge. A contact mike may be wedged behind the tailpiece, but it must rest flat against the banjo head.

The Zither

This class of instruments includes the autoharp, mountain dulcimer, other small multiple stretched-string instruments and ethnic hammered and bowed instruments.

These instruments are often quiet and harmonically complex. The conditions under which these instruments are recorded often require a degree of control and forethought. Because their signal output is very low, it is a temptation to close mike these instruments; however, an overall blend is essential to their sound, requiring a semidistant technique. This combination often leads to problems with leakage from other instruments onto the softer zither tracks. This may be solved in the studio by isolating the instrument or by later overdubbing when the instrument can be played on its own.

The condenser microphone often is the best choice because of the harmonic complexity of these instruments, which are capable of giving both a subtle and rich tone. The miking distance may be anywhere between 6″ to 3′ (15 cm–1 m), depending on instrument and conditions. Often a stereo 3-to-1 coincident mike best represents the subtle "colors" of these instruments. On stage, a closer distance or a different type of microphone may be chosen.

The Voice

From shout to whisper, the human voice displays dynamics and timbre that few other instruments can approach. The male bass voice may extend from E2 to D4 (82 to 293 Hz) with harmonics extending to 7 kHz. Sibilant "s" and "th" sounds extend to 12 kHz. The upper soprano voice may range up to 1.05 kHz with harmonics climbing to 9 kHz. The vowel sounds (a, e, i, o, and u) are formed when the mouth takes a particular shape, creating formants (strongly emphasized resonance bands), which occur even if the pitch varies.

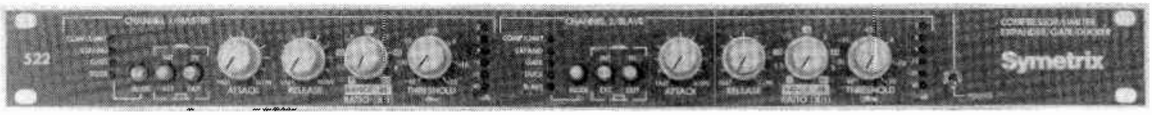
The engineer/recordist should be aware of four traps that may be encountered when recording the human voice:

- excessive dynamic range for the recording chain
- sibilants (f, s, and sh), which are overly accentuated high-frequency sounds
- plosive "p" sounds, which often create unwanted popping at the mike diaphragm
- excessive bass boost due to proximity effect

Dynamic Range

One significant problem in the miking and recording of a voice is a signal that is too wide in dynamic range for the recorded medium. For example, assume that an average vocal level is recorded on tape at approximately 0 VU (+4 dBm), while on occasion the vocalist will belt out a level that is 12 dB higher. This would cause the system to attempt the recording of a +16-dBm signal on tape. Since the maximum level that can be recorded is about +8 dBm, this signal would result in severe distortion. It is therefore necessary to reduce the level of these loud passages to a more manageable level through the use of a dynamic-range changer, such as the *compressor* (Fig. 8-35).

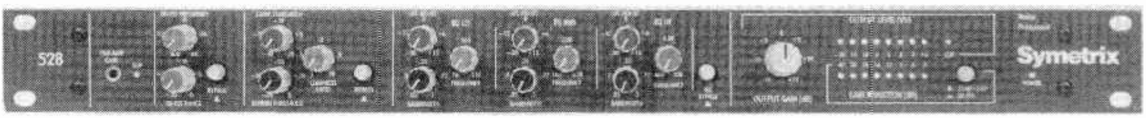
Fig. 8-35. The Symetrix 522 dynamic range finder. *(Courtesy of Symetrix.)*



Sibilance

The sibilant (s, sh, and ch) sounds may become distorted when recorded at too high a level or at too slow a tape speed. Certain microphones may contain a peak at these upper frequencies, exaggerating the sibilant sounds. This can be corrected by a frequency-selective signal-limiting device known as a *de-esser* (Fig. 8-36). Such a device is able to sense and reduce excessive sibilance at its output.

Fig. 8-36. The Symetrix 528 de-esser/voice processor. *(Courtesy of Symetrix.)*



Popping

When placed at close working distances, most directional microphones display an increased sensitivity to low-frequency signals. The most obtrusive of these signals are the plosive (p, t, k, b, d, and g) sounds, which may create a loud "pop" caused by the sudden rush of air at the front of the mike diaphragm.

This sound may be reduced by one of three means:

- Place a foam pop filter (windscreen) over the front face of the microphone grille. This serves as a barrier to the high-pressure "popping" energy, while allowing the sound energy to pass through unimpeded.
- Substitute an omnidirectional microphone.
- Place the microphone well above, or to the side of, the mouth.

The omnidirectional mike is less effected by sudden bursts in pressure and may be the most desirable polar-response choice if excessive leakage is not a problem.

Proximity Effect

At close distances, most directional microphones boost the bass. This effect can be either negative or positive, depending upon the desired audible result. For decades, the bass boost due to proximity effect has been a distinctive part of the radio announcer's voice. Vocalists often integrate this effect as part of their own "sound."

If proximity effect is not desired, an integral bass roll-off switch may be used to attenuate the lower frequencies on certain microphones. Or a roll-off may be added at the console by an equalizer. Also, using an omnidirectional microphone eliminates the bass boost.

Vocal Close-Miking Technique

In the close miking of a voice, working distances of between 1″ (25 mm) and 1′ (.3 m) may be commonly used, while the solo or accent microphone may be placed 2–8′ (.6–2.4 m) away. The type of microphone chosen depends on the sound desired. A dynamic microphone often provides a rugged sound, which is assisted by a close proximity bass boost. The ribbon microphone can provide a mellow, slightly "croony" sound when used at a distance of 4–6″ (10–15 cm), while the condenser microphone generally gives an "open," accurate sound.

At close working distances, the best effects from a cardioid microphone are often obtained when the microphone is placed slightly off-axis to the mouth (Fig. 8-37). This reduces the effects of both popping and sibilance and provides a natural tone.

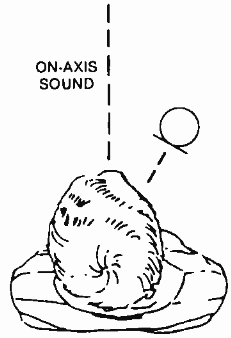


Fig. 8-37. Placement of cardioid mike at close working distances.

Placement Techniques: Microphone Choices

The Clip Microphone

This small, high-quality, multipurpose electret-condenser microphone is often used where mike visibility is to be minimized (Figs. 8-38, -39, and -40).

The clip microphone is composed of two sections: the capsule and the interface. The capsule comprises the diaphragm, the preamplifier (often a single-stage FET-transistor), and the housing. The interface comprises the final preamplification stage (optional), voltage powering interface (phantom or battery), output transformer, and in some models, battery housing and a battery on-off switch.

Their mobility, flexibility, and nominal size make clip mikes useful for amplifying and recording a number of instruments. The mikes may be mounted in a variety of devices, including drum, cymbal, horn, and boundary mounts and affixed with double-sided tape.

Acoustic Guitar (Recording)

Place the clip mike in a boundary mount and use double-sided tape to locate the mike between the sound hole and the bridge near the low E string (as in Fig. 8-41) or under the strings. The front of the mike should be aimed toward the guitar body.

To avoid putting tape on the guitar, attach a universal mount to the sound hole. Commercial adjustable universal mounts are available, but a nonadjustable mount can easily be made by placing a continuous piece of rubber or thick felt between the clip attachment and the instrument/pickup body (Fig. 8-42). Aim the clip mike at the sound hole and position it a few inches away. Roll-off the bass at the mixer until a natural tone quality is achieved.

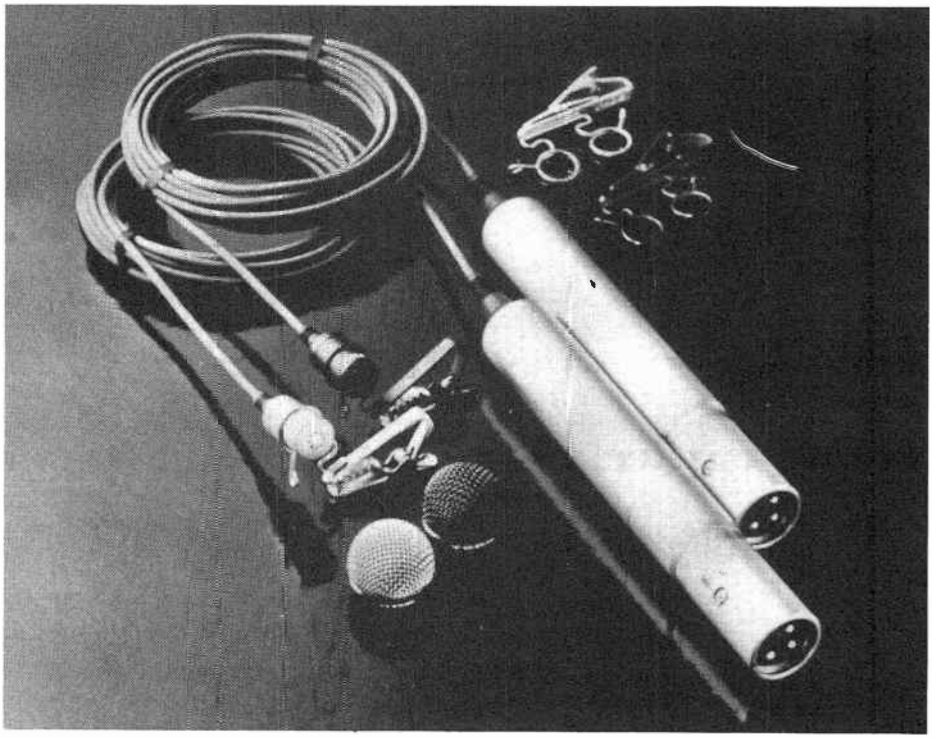


Fig. 8-38. Sony ECM-77 B/S clip mike. *(Courtesy of Sony Corporation of America, Inc.)*

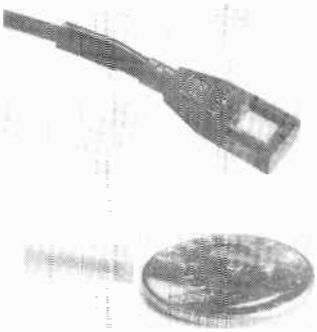


Fig. 8-39. Crown GLM-100 clip mike. *(Courtesy of Crown International, Inc.)*

Acoustic Guitar (Sound Reinforcement)

The primary need in sound reinforcement is to control monitor feedback while maintaining a balanced guitar sound. The best approach is to use a directional clip microphone with a universal mount. The mike is held about 1⁄16″ (1.5 mm) above the top of the guitar with its diaphragm pointed toward the sound hole.

Another good approach is to place an omnidirectional clip microphone inside the guitar itself. It is important to use an omnidirectional microphone and not a directional microphone *inside* an instrument or small enclosure because, when enclosed, directional mikes lose their directional properties and their frequency response changes radically. Additionally, inside an instrument a directional mike will require much more equalization to sound acceptable.

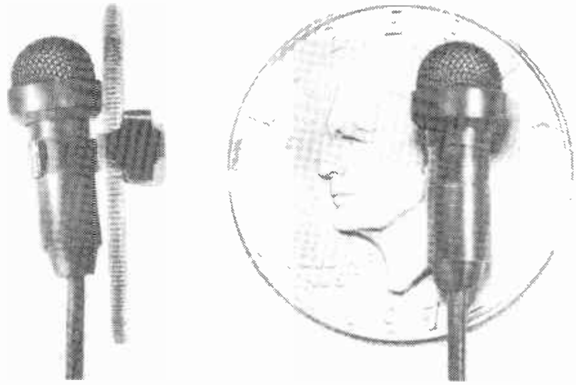


Fig. 8-40. Beyer MCE-5 clip mike. *(Courtesy of Beyer Dynamic, Inc.)*

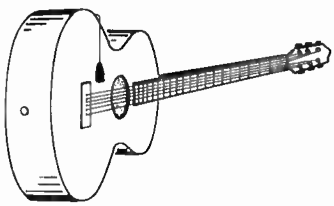


Fig. 8-41. Miking acoustic guitar with clip microphone. *(Courtesy of Crown International, Inc.)*

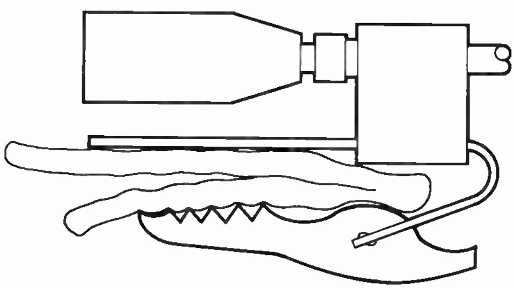


Fig. 8-42. Homemade universal mount for clip microphone.

Amplifier/Speaker for Electric Guitar, Piano, or Bass

Tape the clip microphone's cable to the amplifier grille cloth; the mike becomes invisible in use, uncluttering the stage. Make certain not to cover the microphone sound entries with tape. An omnidirectional microphone usually provides maximum SPL capability, while a directional microphone can be used for extra isolation.

Amp/speakers best radiate high frequencies near the center of the speaker cone. Consequently, mike placement near the center yields a bright tonal balance, while placement near the edge of the cone picks up a duller, more mellow tonal balance.

Drum Set

For each drum in a drum set, tape the cable to the drum rim so that the mike can "see" the drum head, or clip a universal mount to the rim in order to hold the microphone. Be sure the mikes are out of the drummer's way. For permanent, inconspicuous mounting, Crown offers a drum mount for the GLM microphone series (Fig. 8-43). With these mounts alone, the cymbals will be picked up from underneath by the tom mikes.

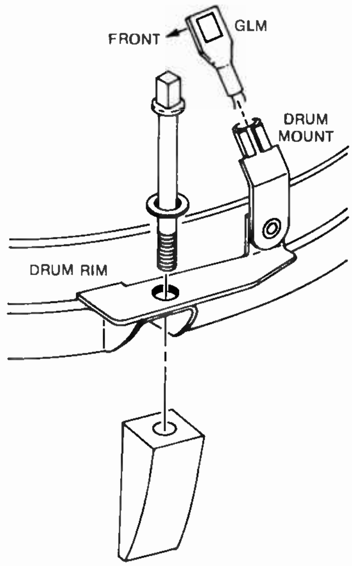


Fig. 8-43. GLM-DM drum mount for Crown GLM clip microphone. *(Courtesy of Crown International, Inc.)*

The bass drum can be damped with a pillow or blanket and a clip mike taped inside, attached to the upper surface of the shell, a few inches from the side opposite the beater head. For more beater attack, tape the cable to the upper surface of the shell inside and let the microphone hang a few inches in front of the beater, or feed the clip mike through the vent hole in the shell and tape the cable to the outer shell surface.

For added isolation, tape each tom mike inside the shell near the side opposite the head. This placement greatly reduces cymbal leakage, though, so the cymbals may need to be miked separately. If so, using microphone stands, position one or more clip mikes about 1′ (.3 m) above the edges of the cymbals. Crown offers the GLM-CM cymbal mount (Fig. 8-44) for the hi-hat and the GLM-OHM Overhead Mount for inconspicuous boom miking.

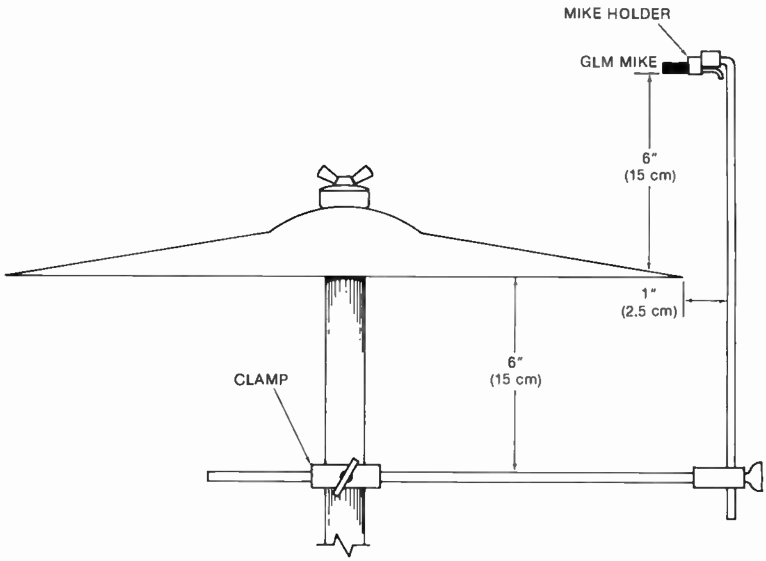


Fig. 8-44. GLM-CM Cymbal Mount for Crown GLM clip mike. *(Courtesy of Crown International, Inc.)*

Drum Set (Three Mikes)

Tape or clip a clip mike near the left high-tom and the snare drum (Fig. 8-45) and aim the front of the mike toward the hi-hat for best high-frequency reproduction. This mike will pick up the hi-hat, snare, left high-tom, and cymbals. Tape or clip another clip mike near the right high-tom and the low-toms. This will pick up these and cymbals. Experiment with placement to achieve a good balance; the bass response may need a slight boost. Place a clip or a regular microphone in the bass drum.

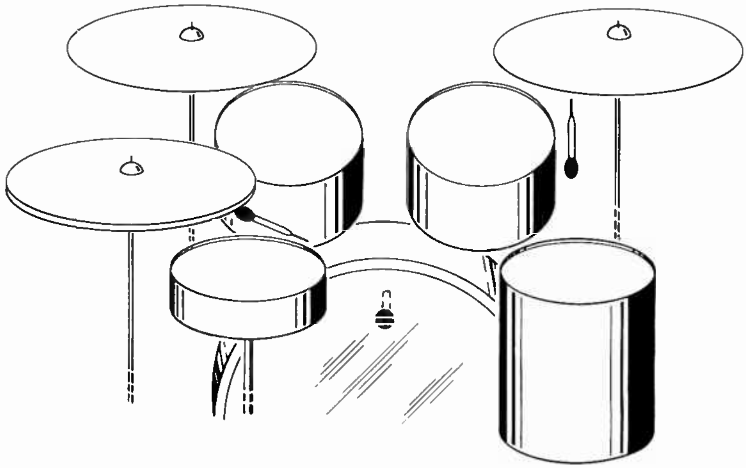


Fig. 8-45. Drums miked with three microphones. *(Courtesy of Crown International, Inc.)*

Drum Set (Two Mikes)

Attach a clip mike to the snare-drum rim and position the mike at the center of the set (Fig. 8-46). With a little bass and treble boost, the sound will be surprisingly good for such a simple arrangement. Place a clip or a regular microphone in the bass drum.

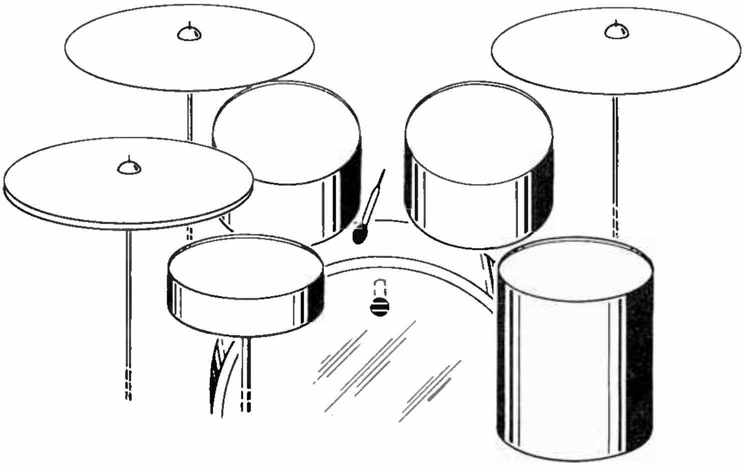


Fig. 8-46. Drums miked with two microphones. *(Courtesy of Crown International, Inc.)*

Woodwinds

Attach a universal mount to the bell and position the clip mike to pick up both bell and tone holes. This technique may be used for miking woodwinds, in general, and the flute and recorder, specifically, to achieve a natural and balanced tone (refer to Fig. 8-47).

1. Wrap two rubber bands around the instrument on either side of the topmost finger hole.
2. Trap a flat, flexible piece of wire or plastic (about ¼″ [6 mm] wide) underneath the rubber bands.
3. Attach the mike to the opposite end of this piece.
4. Bend this piece 90° so that the mike is above the center of the finger holes.
5. Attach the microphone's cable to the body under the existing or additional rubber bands.

Flute

A clip-mike alternative positioning for the flute is to use tape or a rubber band to attach the cable of the clip mike 4″ (10 cm) to the player's right of the lip plate, 1½″ (4 cm) above the flute.

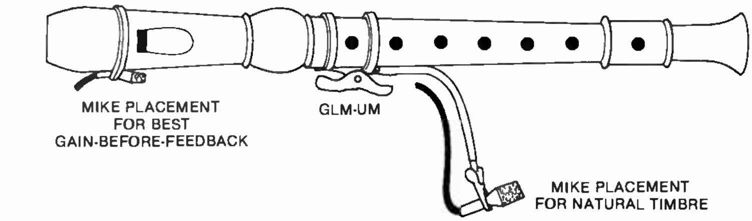


Fig. 8-47. Clip mike placement for woodwinds. *(Courtesy of Crown International, Inc.)*

Brass and Saxophones

For a bright tone quality, attach a universal clip mount to the bell, and position the pickup a few inches in front of the center of the bell. For a mellow tone quality, tape the mike a few inches inside of the bell.

The sax may sound best when miked near the tone holes. To achieve this, attach a universal mount to the outside of the bell (or to the sax hardware) and place the mike over the bell near the tone holes.

Strings

Violin or Banjo

Attach a universal mount to the instrument's tailpiece. Place the clip mike a few inches from the banjo head (Fig. 8-48) or violin sound hole.

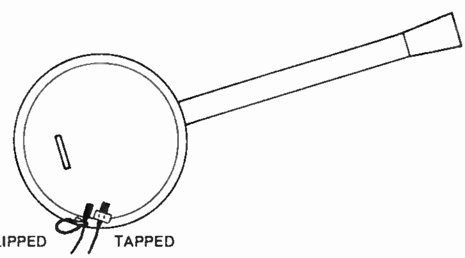


Fig. 8-48. Miking banjo with clip microphone. *(Courtesy of Crown International, Inc.)*

Experiment with miking distance in order to attain a good compromise between tone quality and isolation. Close placement gives better isolation; more distant placement sounds more natural.

If isolation and gain-before-feedback are adequate, it may be best to use an omnidirectional microphone. Otherwise use a directional microphone.

String Bass

For a full, deep tone, tape the clip mike near a sound hole. For a more defined sound, tape the mike cable to the bridge or tailpiece.

Harp

Using a boundary mount, attach an omnidirectional clip mike to the soundboard. Experiment with placement for best results.

The Piano

Grand Piano

Three methods of clip-mike placement give good results:

1. Raise the lid and use the boundary mount to attach the mike inside the audience side of the piano so the mike can see the strings. Use two mikes for stereo.
2. Using the boundary mount, attach a mike to the middle on the underside of the raised lid. Aim the front either toward the piano lid (as in the case of the Crown GLM, or (for cylindrical mikes) facing the piano hammers. This gives a brighter sound.
3. Raise the lid and stretch some masking tape across the ribs of the soundboard, about 8″ (20 cm) from the hammers. Tape one mike's cable near the treble strings, and tape another near the bass strings. (Do not tape the mikes themselves.) Hang the mikes so that their diaphragms point down toward the strings.

Upright Piano

Two methods are recommended:

1. Remove the kickboard in front of the piano to expose the strings. Tape the mike cable underneath the keyboard so that the mike will be about 8″–1′ (20–30 cm) away from the strings. Use one mike near the bass and one near the treble strings.
2. Use the boundary mount to attach the mike to the soundboard. Experiment with position for the best results.

Studio Vocals

Hang the mike (with windscreen) from a mike stand boom about 8″ (20 cm) from the mouth at nose height. To prevent phase interference from sound reflections off the sheet music, angle the sheet so that the reflections travel away from the microphone.

Orchestra, Band, Choir, or Organ

For recording, hang two or three mikes overhead, about 5–10′ (1.5–3 m) apart, about 14′ (4 m) above the floor, 5–10′ (1.5–3 m) in front of the front row of musicians (or organ). You may want to roll-off the high frequencies slightly for a more natural sound.

For sound reinforcement, mike each section a few feet away. Keep in mind the 3-to-1 rule to prevent phase interference: the distance between microphones should be at least 3 times the distance from each mike to its sound source.

The Boundary Microphone

The boundary microphone (Figs. 8-49, -50, -51) has recently become accepted in all communications and entertainment media. Boundary-microphone theory was given in Chapter 6. We'll concentrate here on boundary microphone placement in various applications (courtesy of Crown International, Inc.).

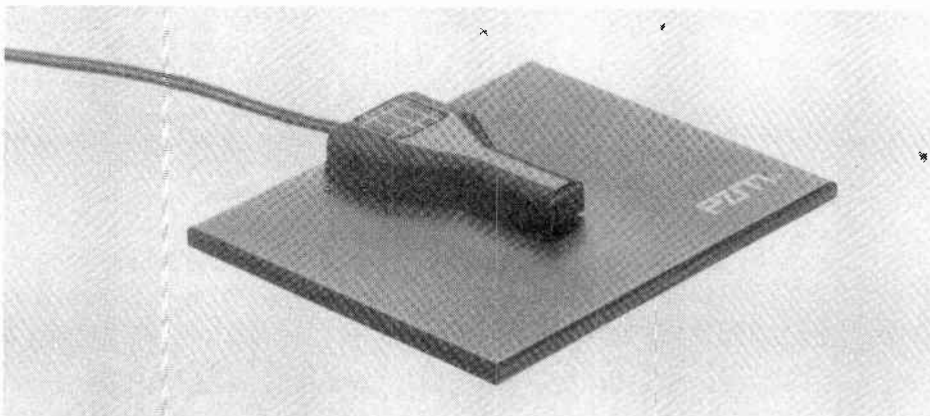


Fig. 8-49. Crown PZM-6RB. *(Courtesy of Crown International, Inc.)*

Controlling Low-Frequency Response

The low-frequency response of a typical boundary microphone depends on the size of the surface on which it is mounted: the larger the surface, the more extended the low-frequency response. When a boundary microphone is mounted on a surface, the low-frequency response shelves down to a level 6 dB below the mid-frequency level, where the wavelength is about 6 times the boundary dimension. For example, the frequency response of a boundary mike on a 2′ × 2′ (.6 × .6 m) panel shelves at −6 dB below 94 Hz. On a 5″ × 6″ (12 × 15 cm) plate, the response shelves down to −6 dB below about 376 Hz.

For the best bass and flattest frequency response, place the microphone on a large hard boundary such as a floor, wall, table, or baffle, with dimensions measuring at least 2′ × 2′ (.6 × .6 m). A boundary 4′ × 4′ (1.2 × 1.2 m) extends flat response to approximately 40 Hz.

A boundary microphone used on a carpeted floor should be placed on a hard-surfaced panel at least 1′ (.3 m) square for the flattest high-frequency response.

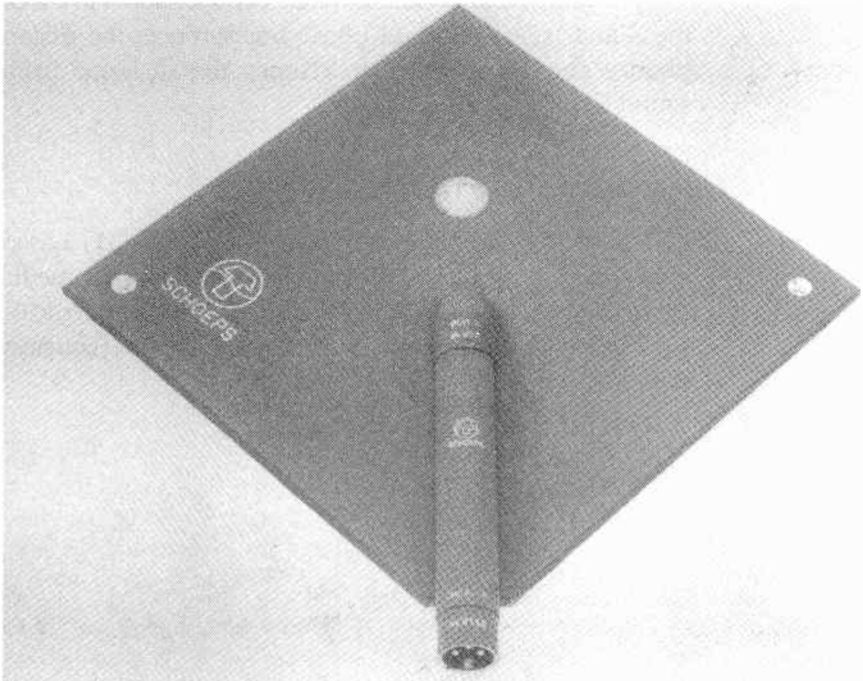


Fig. 8-50. Schoeps BLM-3/CMC. *(Courtesy of Schalltechnik Dr.-Ing. Schoeps GmbH)*

Fig. 8-51. AKG C-562-BL, "The Disc Microphone." *(Courtesy of AKG Acoustics, Inc.)*

Shaping the Polar Pattern

The boundary microphone picks up sounds arriving from any direction above the surface on which it is mounted. However, it is often necessary to eliminate sounds arriving from certain directions. A floor-mounted boundary microphone can be made directional (to reject sounds originating from the rear) by mounting the cantilever in a vertical "corner" boundary made of ¼"- (6 mm-) thick plexiglass. The larger the boundary, the better it discriminates against low-frequency sounds from the rear. Directional boundary microphones, which need no plexiglass barriers to achieve rear rejection, are available.

For a smoother frequency response, a boundary microphone may be taped to the center of a boom-mounted or suspended 2′ × 2′ (.6 × .6 m) or 4′ × 4′ (1.2 × 1.2 m) vertical panel, ¼" (6 mm) thick and placed 4" (10 cm) off center. Using plexiglass makes the panel nearly invisible from a distance. If the edges pick up light, however, tape or paint them black to reduce reflection.

Sounds approaching from any direction at the front or side of the microphone panel are picked up, while sounds approaching the rear of the panel are rejected. The polar pattern of this mike varies from omnidirectional at low frequencies, to super- and hypercardioid at higher frequencies.

Placing the panel in front of or slightly above the performer, with the microphone facing inward will yield good results. Another possibility is to place the panel on the floor, tilted up to aim at the performer, with the capsule as close as possible to the junction of the floor and the panel. For a stereo pickup, mount two mikes on opposite sides of a 2′ × 2′ (.6 × .6 m) or 4′ × 4′ (1.2 × 1.2 m) panel (Fig. 8-52). This forms a bipolar boundary microphone. The edge of the panel should be aimed at the center of the sound source.

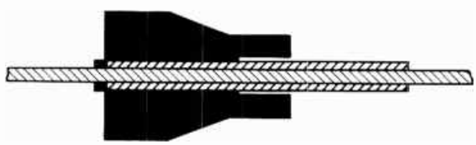


Fig. 8-52. Bipolar (stereo plate) boundary microphone arrangement—edge view.

Boundary mike locations for music production follow.

Acoustic Guitar, Mandolin, Dobro, or Banjo

- On a panel in front, about 2′ (.6 m) away at instrument height.
- On a panel in front and overhead to avoid obscuring the audience's view.
- On the floor (as a solo pickup).
- Use a bipolar boundary setup for stereo effect (adds ambient spaciousness around the solo performer).

String Section

- On panel in front of and above the section.
- For stereo, use a bipolar pickup in the same position.
- On a panel midway between every two instruments, at about a height of 6′ (1.8 m).

Fiddle or Violin

- On a panel in front and overhead.
- On the music stand.

Cello or Acoustic Bass

- On a panel, resting on the floor and tilted toward the performer.
- On a panel in front and above.
- On the floor (for soloist).

String Quartet

- Spaced pair on floor about 6′ (1.8 m) apart.
- Spaced pair on panels in front and above, spaced 3–6′ (.9–1.8 m) apart.
- In front and above for bipolar stereo pickup.

Harp

- On a panel about 2½′ (.8 m) away, aimed toward the treble part of the soundboard.

Sax, Flute, or Clarinet

- On a panel in front and slightly above.
- On the music stand.

Horns, Trumpet, Cornet, Trombone, or Tuba

- On a wall, hard-surfaced flat, or control-room window. Performers play to the wall or flat a few feet away. Their sound bounces off the wall back to them, so they hear each other well enough to produce a natural acoustic balance.
- On a panel in front of and between every two players, 1–2′ (.3–.6 m) away.

- On the music stand.
- For a tuba—on panel overhead.

Grand Piano

- Tape a boundary microphone to the underside of the lid in the middle. For the best sound quality, put the lid on the long stick. To reduce leakage and feedback, put the lid on the short stick or close the lid and cover the piano with a heavy blanket.
- For stereo, use two boundary microphones taped under the lid: one over the treble strings near the hammers and one over the bass strings well away from the hammers. A microphone placed close to the hammers emphasizes attack; one placed far from the hammers yields more tone.
- To pick up the piano and room ambience with a single boundary mike, place it on a panel about 6–8′ (2–4 m) from the piano, 4′ (1.2 m) high. Place the lid on its long stick and face the panel toward the piano. For stereo, use a bipolar mike placed 8–10′ (2.5–3 m) high.
- To add ambience to a close-miked piano, mix in one or two boundary microphones placed on a wall far from the piano.

Amplifier/Speaker for Electric Guitar, Piano, or Bass

- On panel in front of amp.
- On floor a few feet (a meter, or so) in front of the amp.
- Inside the cabinet.

Leslie Organ Cabinet

- Place two boundary microphones on either side of the rotating horn, inside the top of the cabinet. Place another mike inside the bottom cabinet.

Drum Set

- On panel or hard flat, 1–2′ (.3–.6 m) in front of the set, just above the level of the high-toms. Use two microphones 3′ (1 m) apart for stereo. The drummer can balance the sound of the set as he or she plays. A standard or small-plate boundary microphone can be placed in the bass drum against the shell, or hung near the beater, with a pillow or blanket placed against the batter head.

- On panel or a bipolar boundary mike centered overhead about 1′ (.3 m) above the drummer's head. The bass drum may be miked as above.
- Try two boundary microphones placed overhead, each mounted on a 1′- (.3 m-) square panel, angled to form a "V," with the point of the "V" aiming down.
- Two boundary microphones placed on a hard floor, about 2′ (.6 m) one either side of the drummer.
- Tape a boundary microphone to a gauze pad and tape the pad to the bass drum beater head, near the edge. This mike will also pick up the snare.
- Strap a boundary microphone to the drummer's chest and mike the bass drum as above.

Percussion

- Strap a boundary mike to the chest of the player, who carries it when moving among instruments.

Xylophone, Marimba, or Vibraphone

- Use two panels above the instrument, one over the bass and one over the treble side.
- On floor, underneath the instrument. Use two for stereo, one under the bass and one under the treble side. (This arrangement may sound dull and pick up leakage, however.)

Lead Vocal

- In the studio, mount a boundary microphone on a wall, control-room window, a panel, or the floor a few feet (a meter, or so) in front of the performer. The panel can be used in place of a music stand to hold the music. Use a windscreen to prevent "popping."
- To reduce leakage into the vocal mike: (1) overdub the vocal, (2) use flats, or (3) use a well-damped isolation booth with one hard wall for mounting the mike. *Note:* Omnidirectional boundary microphones do not exhibit proximity effect, so console equalization may be used to add extra warmth.
- Place a boundary microphone on the floor, in front of the performer.

Background Harmony Vocals

- On wall, panel, or floor.
- Use a bipolar mike with the singers surrounding the panel.
- Use one or two on both sides of the flat, with singers surrounding the flat.

Combos, Small Groups

For small musical groups with a good natural acoustic blend, such as bluegrass, blues groups, or barbershop quartets:

- On floor, two for stereo.
- On panels in front or on the floor, angled toward the performers.
- Use a bipolar mike in front of the group.
- On rear wall of the stage, with group facing the rear wall.

Drama, Theater, or Opera

- Try one to five unidirectional boundary mikes across the front edge of the stage, about 1′ (.3 m) from the stage edge. One or two microphones are usually sufficient for small stages, and they clearly pick up stage action for dressing room cues. Two microphones may be placed about 20′ (7 m) apart, while three or more should be placed about 15′ (5 m) apart. For maximum clarity and gain-before-feedback, turn up only the microphone nearest the active performer.

  Always be sure performers and custodians know where the microphones are located, so they do not kick or hit the mikes.

  In most cases, the excellent "reach" of boundary microphones provides clear pickup of rear stage action. But if extra reinforcement is needed, additional boundary microphones can be placed on the rear wall, on panels overhead, in a pyramid overhead, on a table under a tablecloth, behind posts, under eves, or on movable scenery (plugged into a wireless transmitter).

Orchestra Pit

- Tape two boundary microphones to the wall on either side of the conductor's podium, about 20′ (7 m) apart, facing into the orchestra.
- Use a separate boundary microphone on a panel for each section of the orchestra.

Orchestra, Marching Band, Jazz Ensemble, or Pipe Organ

Typically, these large sound sources are recorded at a distance, using two microphones for stereo pickup. We'll look at three stereo miking systems here: (1) near-coincident miking using a single panel, (2) near-coincident miking using two panels, and (3) spaced-pair miking.

Near-Coincident Stereo (Using a Single Panel)

With this technique, the stereo effect is created mainly by intensity or level differences between channels and partly by the time differences between channels. The closer the panel is to the musical ensemble, the wider the stereo spread.

- Mount two boundary microphones back-to-back on opposite sides of a panel (forming a bipolar mike). Place the panel between 5′ and 20′ (1.5–6 m) behind the conductor, 14′ (4 m) high, and aim the edge of the panel at the center of the sound source. The further from the ensemble the microphones are placed, the greater is the pickup of hall reverberation or ambience. Adjust the microphone-to-source distance for the desired effect.
- Place the bipolar mike about 20′ (7 m) above the conductor.
- Place the bipolar mike on the floor, on edge, with the microphone elements at the junction of the floor and vertical panel.

Near-Coincident Stereo (ORTF, NOS)

This arrangement employs both intensity and time differences between channels, providing sharp imaging and accurate localization. The ORTF system uses two boundary microphones on panels angled 110° apart and spaced 7″ (18 cm) apart horizontally. The NOS system sets the angle at 90° and the spacing at 12″ (30 cm).

To provide near-coincident stereo with a pair of boundary mikes, proceed as follows:

- Mount an omnidirectional boundary microphone 6″ (15 cm) from the edge of a large panel.
- Similarly, mount another microphone on another panel and tape together the panel edges nearest the microphones, forming a "V."
- Aim the point of the "V" at the center of the sound source. Angle these panels about 70° apart (Fig. 8-53), varying the angle to change the stereo spread.

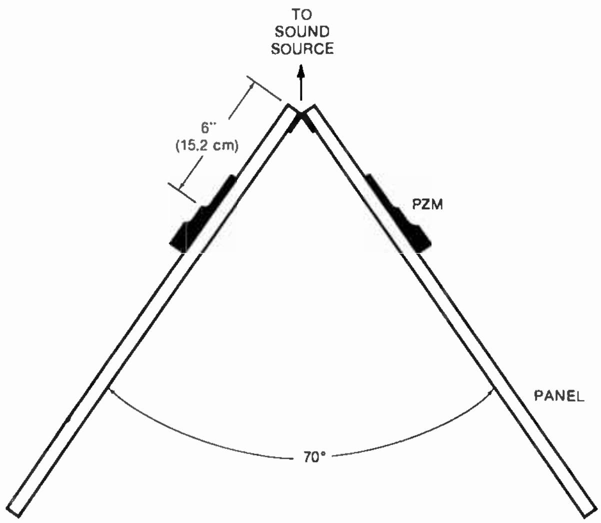


Fig. 8-53. Panel arrangement for ORTF stereo.

Spaced-Pair Stereo

In spaced-pair recording, the stereo effect is created mainly by time differences between channels. The images produced by spaced-pair miking are not as well defined as those of coincident or near-coincident methods. However, spaced-pair miking can sometimes provide a "warm" sense of ambience surrounding the listener.

- Place two panel-mounted boundary microphones from 6′ to 20′ (2–7 m) apart, 14′ (4 m) high and 5′ to 20′ (1.5–7 m) from the first row of players. Place the microphones further apart to widen the stereo spread or closer together to narrow the spread.
- Place two boundary microphones on the floor from 4′ to 10′ (1–3 m) apart and from 10′ to 15′ (3–5 m) from the first row of players. If the stereo spread is exaggerated, add a center-fill microphone midway between the outer pair and mix its output to both channels. This is an inconspicuous arrangement for live concerts.
- Sound reinforcement may require an extra boundary microphone for soloists or for sections that need emphasis. For instance, a "spot" mike for woodwinds can be on the floor, one for brass on the rear stage wall, and one for strings on a panel above the section.
- For noncritical documentary recordings, a boundary microphone can be taped to the proscenium arch, the rear stage wall, or the floor in front of the ensemble.

Choirs

- Use a bipolar boundary mike above and in front of the choir.
- Try two boundary microphones on panels above and in front of the choir. Coverage is wide and the response off-axis is uncolored.
- For sound reinforcement, use one microphone for every hundred singers.
- For small choirs singing in an open area, place the boundary microphone on the floor in front of the group.
- For choirs seated on one side of a church chancel facing the other side of the chancel, mount a boundary microphone on the wall opposite the chancel.

Ambience

- One or two boundary microphones on the walls give an uncolored sound.
- One or two boundary microphones on the walls of an echo chamber provide ambient richness and naturalness.

Other Vocal Miking Needs

Audience

- On panels suspended over left and right sides of the audience.
- On side wall near the front of the auditorium, 10′ to 15′ (3–5 m) up. These arrangements provide clear, realistic pickup of audience reaction.

Altar

- Place a boundary microphone on the altar table (perhaps under the tablecloth) to pick up speech near the altar.

Conferences, Teleconferences, Roundtable Discussions, or Interviews

For maximum clarity, hold the conference in an acoustically "dead" room with carpeting, acoustic tile ceiling, and drapes.

- Place a single boundary microphone in the middle of the table.
- For more control and less pickup of reverberation, use one boundary microphone on the table in the middle of every four to six people. No person should be more than 3′ (1 m) from the nearest microphone.

Lectern

- Place a unidirectional boundary microphone on the lectern shelf top, away from cavities. If the lectern has a raised edge, place the boundary microphone at least twice as far from the edge as the edge is high.

Courtrooms

- A boundary microphone can be permanently mounted on the bench and/or witness stand, permitting freedom of movement without loss of speech. It provides excellent clarity and is less intimidating to witnesses than are traditional microphones.

Sound Effects

- A bipolar boundary mike provides realism in many instances. It "tracks" the motion of moving sound sources more accurately than a spaced pair of microphones does.

Miking Sports Events

Basketball

- On a 2′ × 2′ (.6 × .6 m) panel suspended over center court. Use two mikes for a stereo effect.
- On the floor just outside the court at center court to pick up sneaker squeaks, ball noises, and audience reaction.

Football

- A boundary pyramid (see Appendix C) aimed at the field clearly picks up the quarterback calling the plays.

Bowling

- Place a boundary microphone on the back wall of the alley to pick up ball and pin sounds.

Indoor Sports

- Sports such as weight-lifting or fencing can be picked up with a boundary microphone on the floor.

Summary

In this chapter we saw that musicianship, the overall characteristic of the environment, and the choice and placement of a microphone play an important part in creating a high-quality finished audio product. A deeper understanding of how these factors relate to each other is a continuous learning process, which comes from experience and the desire to create.

9 Microphone Techniques in Video/Film Production

Just as a good auto mechanic without a good set of tools would be severely hindered, proper tools are essential for quality video or film sound recording. When working in the typical sound stage or on-location environment, the recordist's tools include a quality audiotape recorder, a portable mixer, a range of quality microphones and pickup devices, and a wide range of accessories to adapt to various circumstances.

The sound recordist is often faced with the difficult and contradictory task of picking up a quality audio signal from one or more sound sources, while remaining "off-camera." With the demands made on the visual media for high quality and stereo audio, the jobs of the recordist, engineer, and sound mixer are more diverse and creative than ever before, with new techniques and tools regularly entering the professional marketplace.

In this chapter we examine microphone techniques for the visual media, including such topics as on-camera and off-camera pickup techniques, the wireless microphones, and post-production audio.

On-Camera Microphone Techniques

On-camera pickup refers to the placement of a microphone within the field of the camera's vision. The microphone may be hand-held, stand-mounted, or a clip (lavaliere).

In electronic news gathering (ENG), the most common microphone used is a hand-held dynamic, chosen for its ruggedness and simplicity of operation (Fig. 9-1). It may be either omnidirectional or cardioid. An omnidirectional microphone assures even coverage of all on-camera events, as well as background ambience (sounds that add a sense of scenic involvement). A cardioid microphone reduces background noise and increases sound intelligibility.

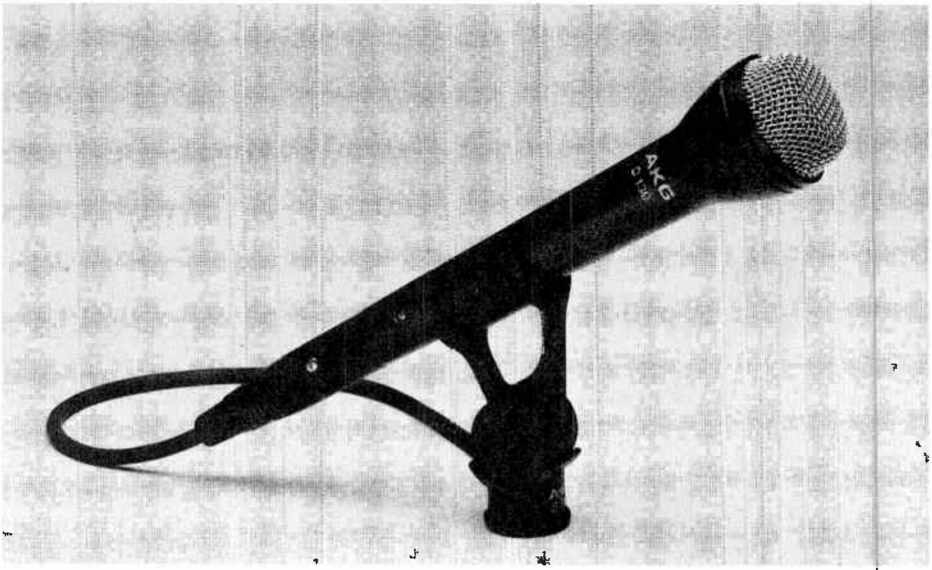


Fig. 9-1. AKG D-130NR omnidirectional, dynamic microphone, designed for film and ENG applications. *(Courtesy of AKG Acoustics, Inc.)*

Stereo microphones, which contain two coincident capsules, are increasingly popular in broadcasting and the visual media. Two examples are the AKG C-522-ENG X/Y stereo microphone and the Sanken CMS-7 M/S stereo microphone.

The AKG C-522-ENG stereo microphone (Fig. 9-2) contains two cardioid condenser capsules oriented to the front at an off-axis angle of ± 45° (90° overall angle), for a simple one-hand operation with the X/Y stereo technique. The C-522 has a built-in rechargeable battery and an integral on-off switch. It may be connected to all balanced or unbalanced inputs, and it may use phantom powering, where available. The microphone is delivered in a reusable carrying case containing the C-522, windscreen, stand adapter, elastic shock mount for boom or fishpole operation, and two cables for balanced and unbalanced recording applications.

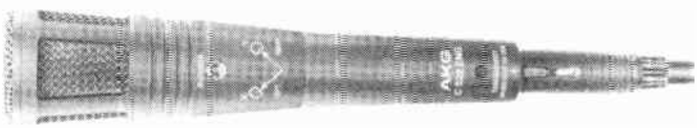


Fig. 9-2. AKG C-522-ENG stereo microphone. *(Courtesy of AKG Acoustics, Inc.)*

The Sanken CMS-7 M/S portable stereo condenser microphone (Fig. 9-3) is designed for use in the radio broadcast, video, and motion picture industries. The CMS-7 employs two fixed condenser capsules, with the forward-facing (mid) capsule cardioid and the perpendicular (side) capsule bidirectional.

When the M and S outputs from both mid and side microphones are added and subtracted by the use of a switchable matrix box, the two outputs (M + S and M − S) reproduce as left and right stereo signals (Fig. 9-4).

Fig. 9-3. The Sanken CMS-7 MS stereo microphone. *(Courtesy of Sanken/Pan Communications, Inc.)*

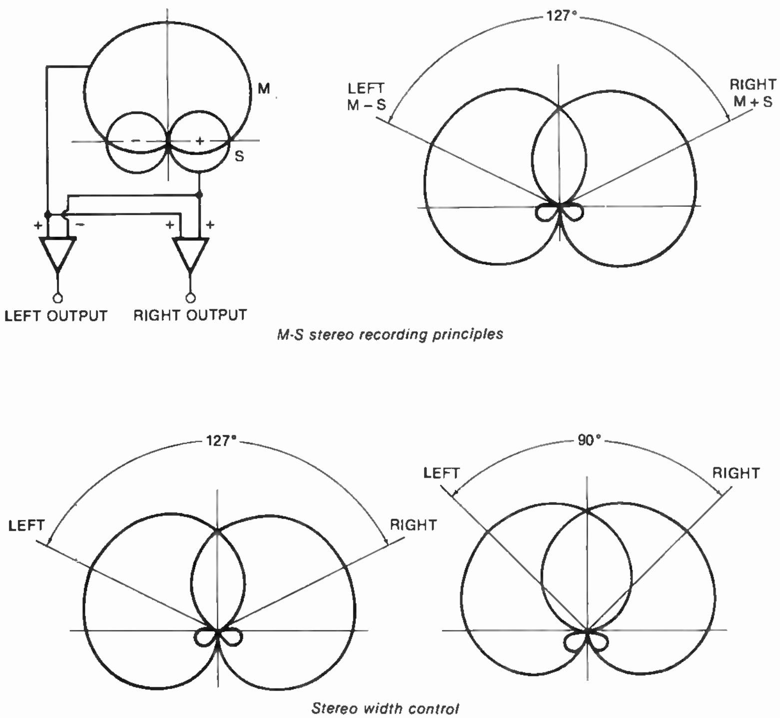
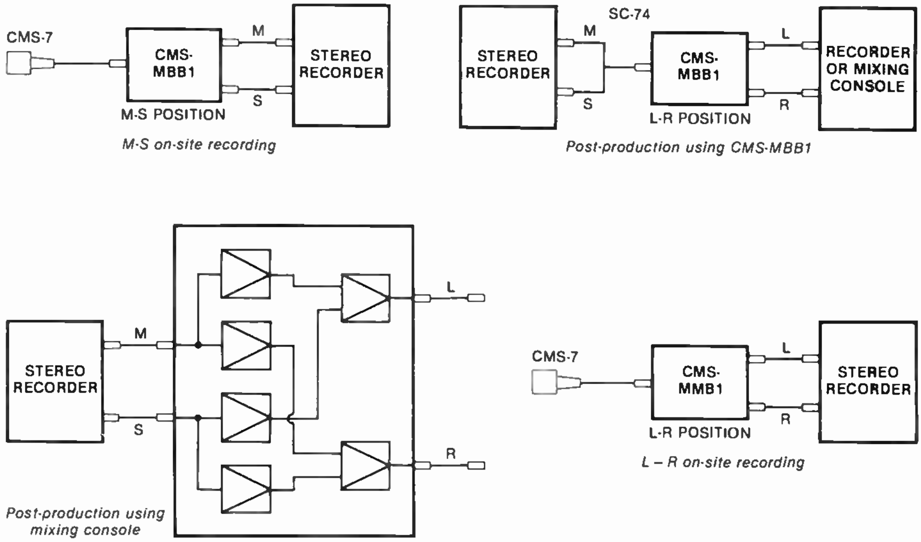


Fig. 9-4. Operating principles of the Sanken CMS-7. *(Courtesy of Sanken/Pan Communications, Inc.)*

Fig. 9-4 (cont.)



The RSM 190i condenser microphone (Fig. 9-5) is a stereo shotgun system with variable directional characteristics, and it consists of two separate capsule assemblies:

- a short shotgun with a pressure-gradient interference transducer to generate the middle (M) signal

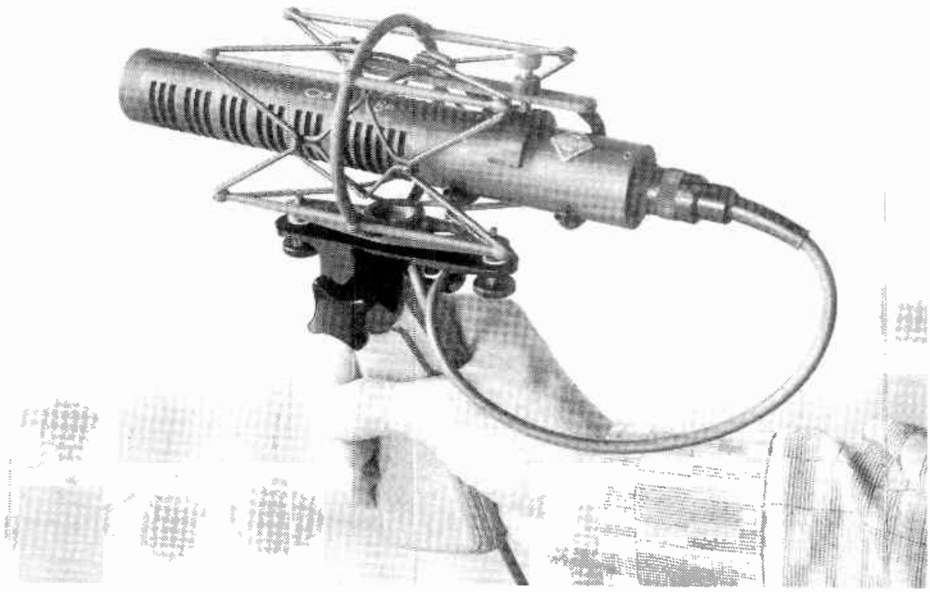


Fig. 9-5. The RSM 190i stereo microphone system. *(Courtesy of Gotham Audio Corporation)*

- a second integrated system, with its axis at a right angle and operating in a figure-eight (bidirectional) characteristic, providing the side (S) information

The MTX 190i matrix amplifier enables gain adjustment of the side signal relative to the middle system in six 3-dB steps, allowing changes in the width of the stereo image. The outputs of this microphone system are available as either mid-side (M-S) information, or can be put through the plus and minus matrix to produce left-right (X/Y) stereo output channels.

The Clip Microphone

The clip mike is receiving wide recognition for its usefulness both on- and off-camera. Its most common use is for the pickup of an announcer or an on-camera talent.

The frequency response of the clip mike is wide and smooth; however, the high frequencies are often emphasized in order to compensate for the high-frequency loss that occurs naturally when the mike is worn on the chest. This effect occurs because the voice is directional at high frequencies. On certain clip models it is possible to restore the microphone response to flat through a minor internal modification.

Options available for the clip mike include belt clip (Fig. 9-6), universal mount, tie-bar mount (Fig. 9-7), and tie-tack mount for locating on clothes, and a windscreen for outdoor use. Mounts available for use with the flat-top Countryman or Crown clip microphone include drum mount, cymbal mount, horn mount, and boundary mount.

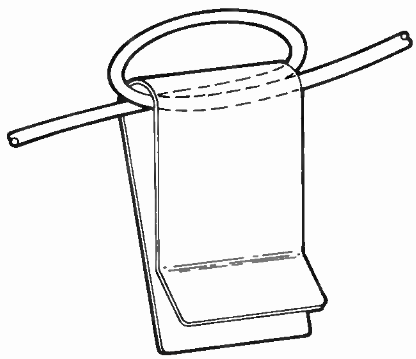


Fig. 9-6. Mike cord relief belt clip. *(Courtesy of Crown International, Inc.)*

Placement Techniques for the Clip Mike

Detailed application notes are available from the mike manufacturer or dealer but Crown International suggests the following placement points.

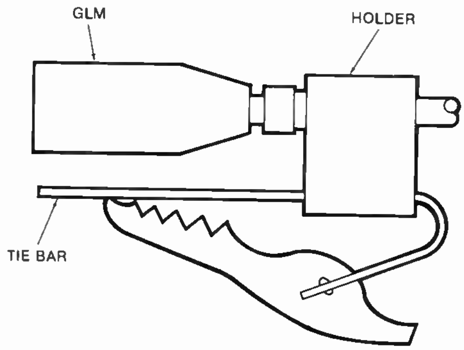


Fig. 9-7. Clip mike tie-bar mount. *(Courtesy of Crown International, Inc.)*

Lavaliere Use

1. For some models, wrap the mike cable around the belt clip (one complete loop) some 2′ (.6 m) from the mike and pull snug.
2. Press the belt clip onto the belt or place it in the pocket.
3. Press the mike into the tie bar or tie tack and attach the mount about 8″ (20 cm) under the chin, aimed toward the mouth.

On Sets

Attach an omnidirectional clip microphone to the back of props close to the action. Tape the mike to the backs of teacups, books, flower pots, tables, and so on.

In an Automobile

Attach an omnidirectional clip microphone to the sun visor, near the centerline of the car.

On Actors

To reduce clothing noise when a clip mike is used on an actor, spray his or her clothing with antistatic solution or water. Spray leather with silicon spray (or WD-40 if the leather can be cleaned). Tape both sides of the cable to the clothing, using adhesive bandages on skin. Make a loop in the cable to act as a strain relief. Place the connector near the actor's foot so that he or she can be "unplugged" between takes.

Wireless Microphone Systems

One other single-mike method of recording to be discussed is the *wireless* system. This method allows the greatest mobility while maintaining a clean, present audio signal.

A wireless audio system (Figs. 9-8 and 9-9) comprises three components: a microphone-transmitter, receiver, and antenna. The microphone-(FM) transmitter is found in one of three combinations: hand-held microphone and body transmitter, lavaliere microphone and body transmitter, or integrated microphone and transmitter. The receiver section picks up the VHF or UHF frequency-modulated signal. It may be supplied as a rack-mounted base station or as a smaller portable device for use with ENG or electronic field production (EFP). The base antenna system may be designed to be either integral or external to the overall wireless system.

Fig. 9-8. Swintek professional wireless microphone system. *(Courtesy of Swintek Telecommunications Division)*

Fig. 9-9. Sony VHF synthesized wireless microphone system. *(Courtesy of Sony Corporation of America)*

The Diversity Antenna System

Multiple signal paths may exist between the transmitter antenna and receiver antenna. Typically, these *multipaths* are caused by surfaces that reflect rf signals. As the transmitter moves around the stage, multipath can cause momentary fades or dropouts (weakening or loss of signal). Often multipath problems can be eliminated by relocating the receiver antenna. A foolproof solution is to use two or more antennas to pick up the rf signal but, unfortunately, you can't just plug two antennas into a receiver and expect to get usable results. Instead, you need a *space diversity* system (Fig. 9-10). Such a system is an important addition to any wireless mike receiver in a production environment where set design and/or location are changed frequently.

Until recently, only two types of diversity system were available. One is known as "Audio Switching Diversity," in which two complete receivers, one

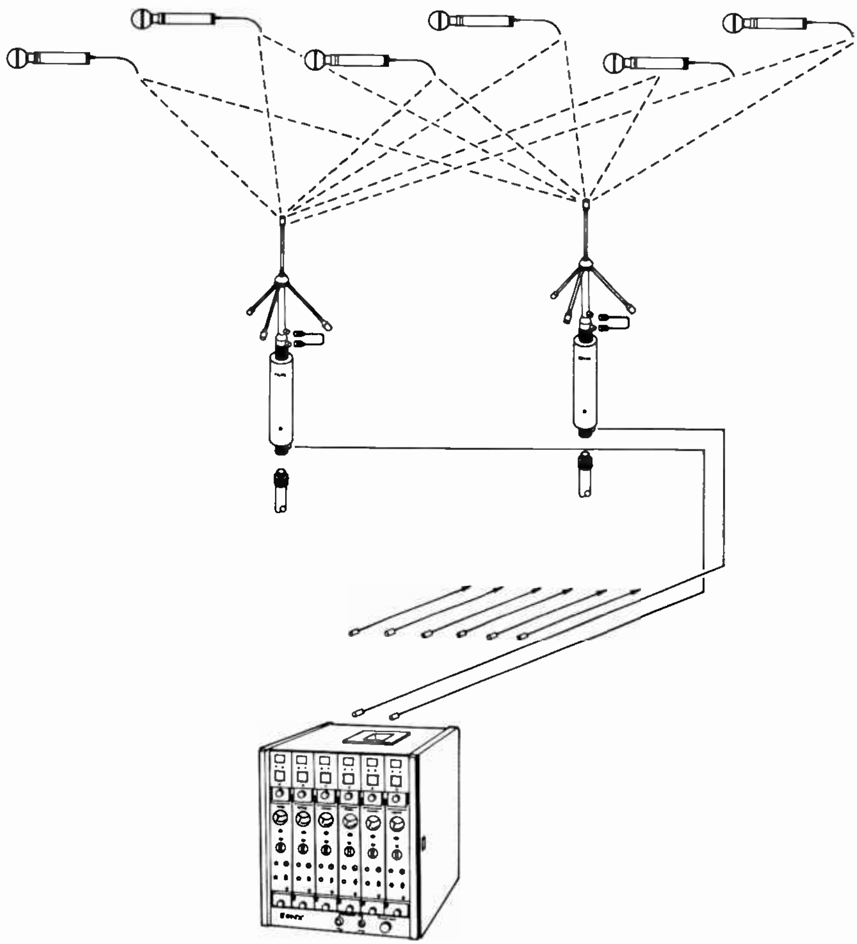


Fig. 9-10. Sony UHF 6-channel diversity reception system. *(Courtesy of Sony Corporation of America)*

for each antenna, are built into a single box with a comparator circuit. The comparator senses which antenna/receiver combination is delivering the strongest radio signal and switches that audio to feed the output. Unfortunately, no benefit is derived from the unused antenna. Also if, due to excessive noise, one channel is louder, the noisy signal is heard. Such systems are also prone to "clicks" as the audio output switches between receivers. The other kind, "Post Detection Combining," uses a more costly proportional comparator. Instead of switching the output between the two antennas and receivers, it blends the audio in proportion to the rf signal strength. Aside from its very high cost, it can have audio signal tracking errors due to the quantizing speed of the logic. Both of these systems require two receivers in order to operate.

Swintek, a manufacturer of wireless microphone systems has developed the "rf Combining Diversity" system, which utilizes two or more antennas and at least one rf combining amplifier. The rf signals from all of the antennas are combined and the resulting output can be fed to any conventional wireless mike receiver.

Off-Camera Microphone Techniques

Off-camera miking techniques use a microphone outside the camera's field of vision to pick up on-camera talent. Off-camera techniques use highly directional microphones, specialized mounting of a mobile microphone boom just outside of the field of vision, or "on-the-set" hidden microphone techniques.

The Porta-Cam

ENG, which is used to gather information for almost any news report, makes use of the portable camera (porta-cam) for on-the-scene production. The porta-cam is extremely portable and easy to operate, which reduces production time and the need for a field crew. EFP, involving on-location shooting, often uses the porta-cam method of video production. This is especially true in the case of a single-camera or limited budget production. Portable systems such as the Sony Betacam are self-contained, over-the-shoulder videotape recorder (VTR), audiotape recorder (ATR), and camera, allowing great versatility in the fields of ENG and EFP.

One of the simplest methods for recording sound for video on-location is to record the original audio *in-camera* (Fig. 9-11). That is, the on-camera sound is picked up by a microphone that is housed directly in the body of the porta-cam itself. In a professional system this microphone is generally directional, often ultradirectional (shotgun), which allows on-camera sounds to be picked up, while much of the off-camera sound is rejected.

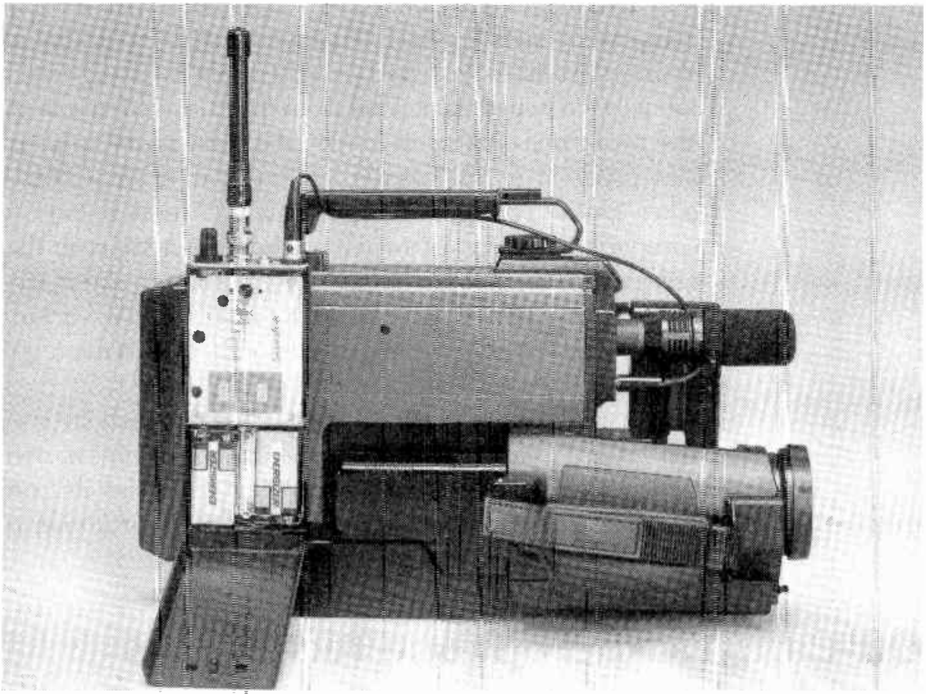


Fig. 9-11. Magnavox porta-cam/Swintek Mark QOC combination. *(Courtesy of Swintek Telecommunications Division)*

The Ultradirectional "Shotgun" Microphone

The ultradirectional mike (Figs. 9-12, -13, and -14), is commonly used in the production of film and video audio. It is characterized by a lobe-shaped, extremely narrow pickup angle (Fig. 9-15). This tight directional pattern is derived from the placement of the microphone element at the end of a slotted tube. Sounds originating on-axis pass through the tube unaffected. Sounds arriving from off-axis, however, enter into the tube through slots that effectively attenuate the signal by acoustic phase cancellation. The effective pickup pattern of the shotgun mike often is hypercardioid at lower frequencies, while its acceptance angle is as little as 15° off-axis at higher frequencies.

Because most shotgun mikes dull the sound off-axis, it is best to use the shotgun microphone only when necessary.

Another effect of a narrow acceptance angle is the accompanying increase in sensitivity. This allows the mike to operate at a greater distance

Fig. 9-12. Neumann KMR 82 ultradirectional microphone. *(Courtesy of Gotham Audio Corporation)*

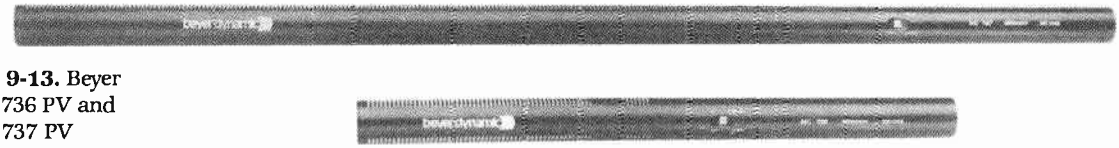


Fig. 9-13. Beyer MC 736 PV and MC 737 PV ultradirectional condenser microphones. *(Courtesy of Beyer Dynamic, Inc.)*

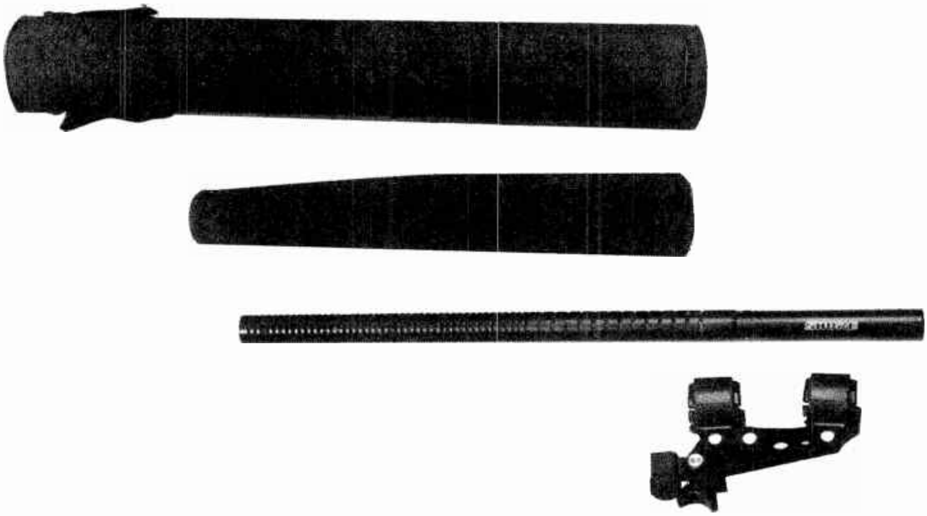


Fig. 9-14. Shure SM 89 condenser shotgun microphone with case, windscreen, and optional A89SM shock mount. *(Courtesy of Shure Brothers Incorporated)*

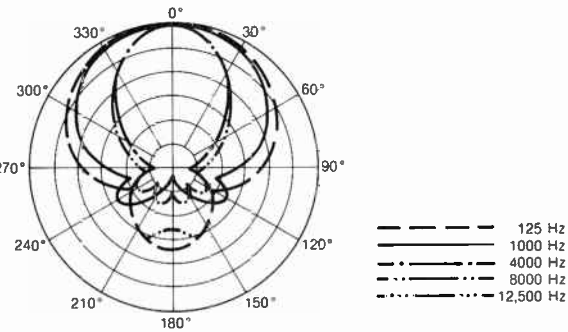


Fig. 9-15. Typical polar pattern of ultradirectional microphone.

while receiving a "present" and close-sounding pickup, thus rejecting unwanted background noise. Such an increase in "reach" may have drawbacks, however. For example, a shotgun microphone, pointed at on-camera talent may also clearly pick up the sound of a passing aircraft as it moves across the axis of the microphone.

There is a wide range of shotgun microphones on the market, designed to suit a number of needs, especially when joined with the wide variety of accessories available. These versatile performers may be handheld with a pistol grip or mounted on a fishpole or boom for extended reach, thus becoming a useful tool in video, film, music recording, sound reinforcement, and theater.

The Microphone Boom

There are many ways to support a microphone out of the camera's line of sight, while allowing mobility. One such support is the *microphone boom* (Fig. 9-16). The boom is often employed in sound-stage productions where the movement of on-camera talent prohibits the use of stationary or hidden microphones on the set.

The microphone boom varies in size from a simple boom support mount to a fully steerable boom mount, consisting of a driveable wheel base, an extendable/retractable boom, and a microphone steering mechanism.

The microphone is most often attached to the boom by an elastic shock mount, which isolates the microphone from floor noises and vibration. Take care not to defeat this isolation by using a stiff microphone cable, which can transmit vibrations directly to the microphone. In this case, a short flexible cable should be employed at the mount in order to absorb these vibrations (Fig. 9-17).

Fig. 9-16. Atlas Sound SB-100W microphone boom. *(Courtesy of Atlas Sound)*

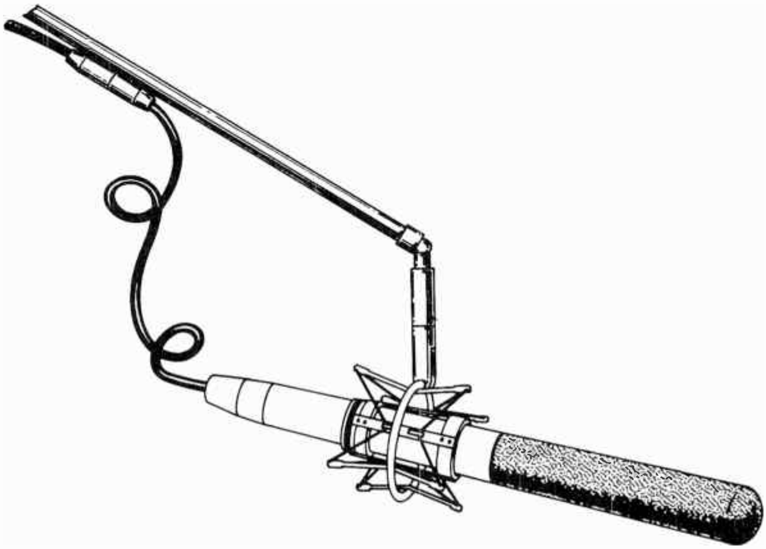


Fig. 9-17. Short, flexible cable helps isolate the microphone from noises and vibration.

Another form of boom is the *fishpole boom* (Fig. 9-18). It is often used in smaller audio-for-visual applications (as in EFP) where additional mobility is needed or in areas too small for a boom stand. The fishpole is made of a hand-held, often retractable pole, containing a mount that is located at the end piece. This boom can be positioned off-camera, either overhead or underneath, pointing toward the source. In the latter case, take care not to allow floor-reflection phase interference.

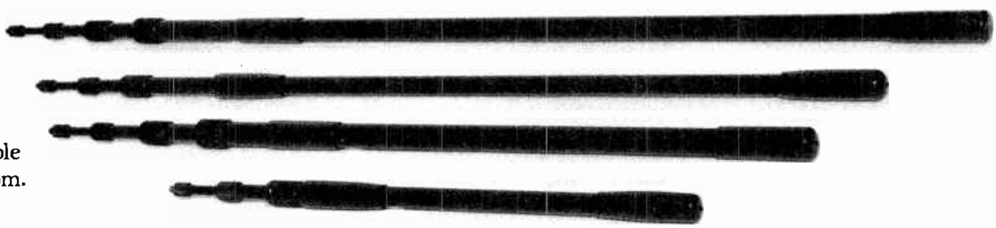


Fig. 9-18. Fishpole microphone boom. *(Courtesy of Keith Monks Ltd.)*

Sound Effects in the Sweetening Process

In video production, the sound effects (SFX) track plays a major role in heightening the impact of a program.

The Foley Sound Stage

In the early 1940s, George Foley, a soundmixer at Warner Brothers, designed a process by which on-camera ambient noise sounds which had been lost through improper on-location pickup or the automatic dialogue replacement (ADR) process were replaced. ADR is the process whereby on-camera dialogue is rerecorded in the post-production phase of film or video.

The Foley process is accomplished through a process similar to ADR, in that the video and audio are played in a continuous loop for repeated practice, and finally for the replacement of these ambient noises by Foley artists. All necessary props which are used to replace synchronized on- and off-camera sounds (e.g., doors, footsteps, clothes rustling, music boxes, etc.) are rerecorded onto separate audio tracks.

A large and slightly acoustically dead studio is often used for Foley replacement in order to maintain the proper distance perspective required by the scene. Also, the Foley studio is often equipped with a variable-surface floor (Fig. 9-19), enabling the replacement of specific types of floor footsteps (such as hardwood, gravel, cement), door slams, broken glass, and so on.

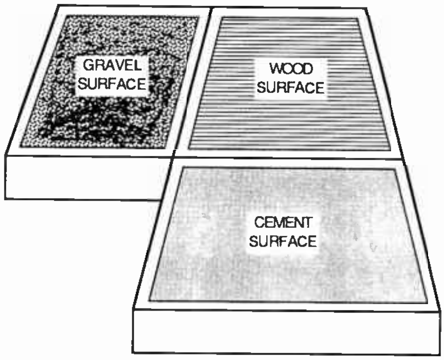


Fig. 9-19. Foley variable floor surface.

Summary

In this chapter we covered basic microphone techniques for both on- and off-camera use. With recent advances of audio production and post-production techniques for the visual media, we may expect changes in the near future in sound-stage and electronic-field production. The most noticeable new techniques are synchronized multitrack audio in the production phase and random-access/sampling techniques in the post-production phases of audio for visual media.

The techniques outlined in this chapter represent present techniques. However, standard microphone and production techniques in audio for film and video continue to evolve and grow more intricate in order to keep pace with more sophisticated audio consumer demands.

10 Speech and Music Reinforcement

Introduction

The primary purpose of a speech- and music-reinforcement system is to amplify high-quality and/or intelligible sound over a greater distance than would otherwise be possible.

Sound System Intelligibility

The intelligibility of a sound system can be measured by the system's ability to relay a signal to the listener, such that the information or intended effect is accurately perceived.

Miking factors that enhance intelligibility in a sound system or listening environment are adequate signal-to-noise ratio and system acoustic gain.

Signal-to-Noise Ratio

Within an acoustic environment, three factors are always in play:

- signal level produced by the sound source
- distance of the listener and microphone from the sound source
- background (ambient) noise level

Each of these factors is important to the intelligibility of a sound source. In the average environment, speech communication generally occurs at levels approaching 65 to 70 dB SPL over distances of 3–5′ (1–1.5 m). These levels drop off as distance increases, in accordance with the *inverse-square law*. This law states that the perceived level of a signal is reduced in direct relation to the square of the distance between the source and the listener, as shown in Figure 10-1.

As the direct-sound level drops in relation to distance, the background or ambient environmental sounds become relatively audible, and they start to mask the desired signal, either distracting from it or rendering it unintelligible. In order to achieve the greatest degree of intelligibility, a signal-to-noise ratio of at least 25 dB within the middle-frequency audio band is desirable. If the ratio is less than 25 dB, some acoustic gain will be necessary in order to provide adequate coverage of the event.

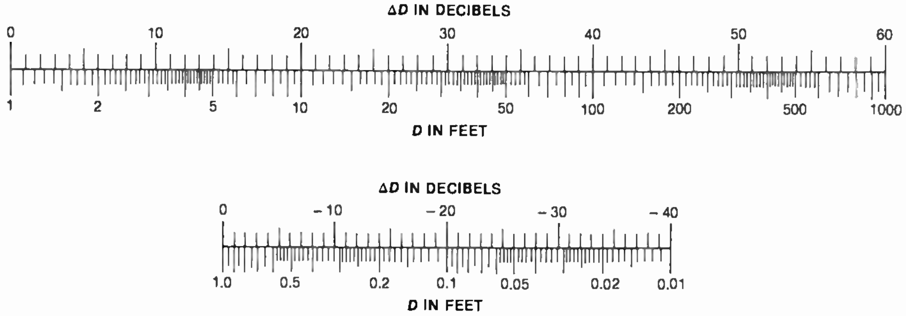


Fig. 10-1. Inverse-square graph showing changes in decibel level ΔD over distance D.

Acoustic Gain

In order for the sound source to be intelligible above the masking effect of background noise, some acoustic gain is required by the sound system. The effect of an increased system gain is that the perceived distance between the sound source and the listener becomes a lesser *equivalent acoustic distance* (EAD) (Fig. 10-2). For example, a sound source placed at a distance (D_0) of 50′ (15 m) from the farthest listener, might have an EAD of only 25′ (7.5 m) with an amplified sound system. That is, the reproduced source would sound as loud as if it were only 25′ away.

A proper degree of *needed acoustic gain* (NAG) is required of the sound amplification system in order to reduce the EAD. To find the NAG of a sound

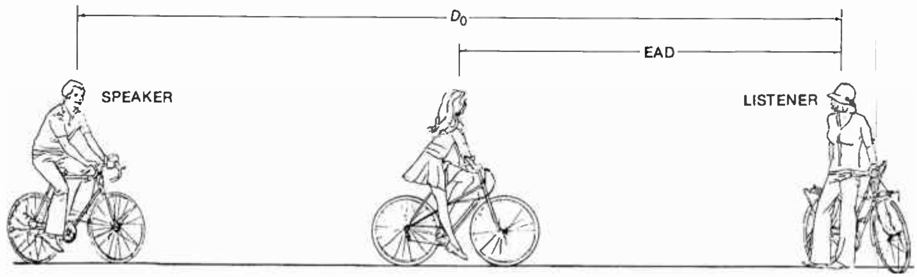


Fig. 10-2. EAD of sound source is brought forward through amplification.

system, we first find the degree of attenuation over D_0 and subtract this from the attenuation over the EAD (equation 10-1).

$$NAG = 20 \log D_0 - 20 \log EAD \tag{10-1}$$

Given our earlier example, this equation would read:

$$NAG = 20 \log 50 - 20 \log 25 = 6 \text{ dB}$$

Thus, in order for this amplified sound system to deliver the perceived impression of having an EAD of 25′ (7.5 m) from the listening position, an acoustic gain of 6 dB must be supplied.

The *number of open microphones* (NOM) also limits the amount of gain that a sound system can deliver without being regenerated into feedback. As a general rule, the gain a system can deliver will be reduced by 3 dB for every doubling in the number of open microphones. Taking this into account, our needed acoustical gain formula for a system using more than one microphone would be read as:

$$NAG = \Delta D_0 - \Delta EAD + 10 \log NOM \tag{10-2}$$

where,

Δ refers to the loss in dB over a given distance D

In "Frequency Characteristics of a Sound Reinforcing System," William B. Snow describes the effects that level and equalization have on the frequency response of a sound system, when the system is increased in level toward the feedback-producing conditions of unity gain. *Unity gain* means that the loudspeaker sound level at the microphone equals the source sound level at the microphone.

Figure 10-3 shows the resulting frequency response of a system operating at levels 10 dB below unity gain and at subsequently lower levels. Note that even at levels far below the feedback point, small irregularities are evident. Such irregularities may be smoothed out through the use of equalization, which, when properly employed, can give a sound system an additional gain of 6 dB or more before feedback occurs. This gain margin, which is rated in dB, is known as the *feedback stability margin* (FSM) of a sound system. Taking this into account, our gain formula becomes:

$$NAG = \Delta D_0 - \Delta EAD + 10 \log NOM + 6 \text{ dB FSM} \tag{10-3}$$

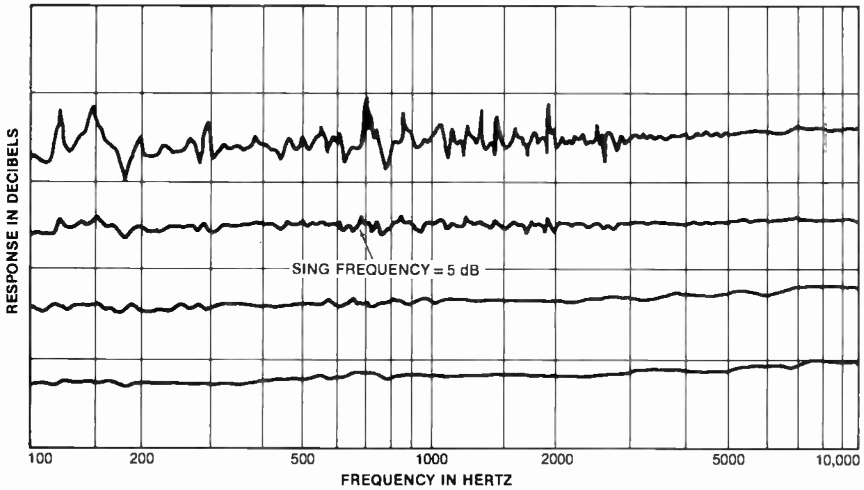


Fig. 10-3. Measured response of sound system at various gain levels below feedback.

Through the use of equalization, a stability margin of 12 dB may be obtained, which is adequate to ensure against feedback. Keep in mind, though, that large corrections in equalization are not needed to reduce the negative effects of feedback-prone "bumps" in the response. Often these irregularities can be corrected at lower levels by less than 1 dB, thus reducing the potential for feedback at that frequency when amplified at higher levels. Boosting a signal through excessive equalization may result in a degraded response at frequencies other than those being equalized. Figure 10-4 shows such nonlinear effects when excessive low- and high-frequency boosting is employed, further increasing the potential for feedback.

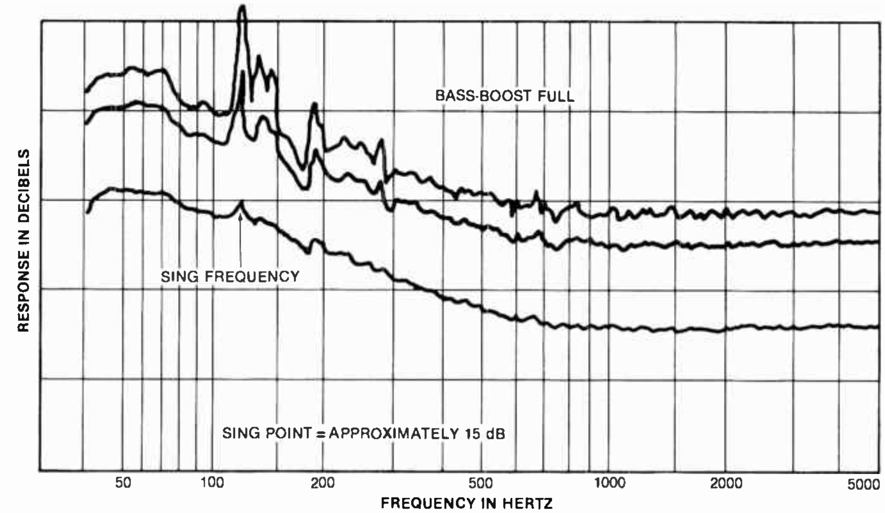
Given a properly equalized indoor system, the amount of *potential acoustic gain* (PAG) that can be attained from a system with one open microphone (Fig. 10-5) can be calculated from equation 10-4. In an intelligible system supplying sufficient amplification, the existing potential acoustic gain is equal to the calculated NAG.

$$PAG = \Delta D_0 + \Delta D_1 - \Delta D_s - \Delta D_2 \qquad (10\text{-}4)$$

where,

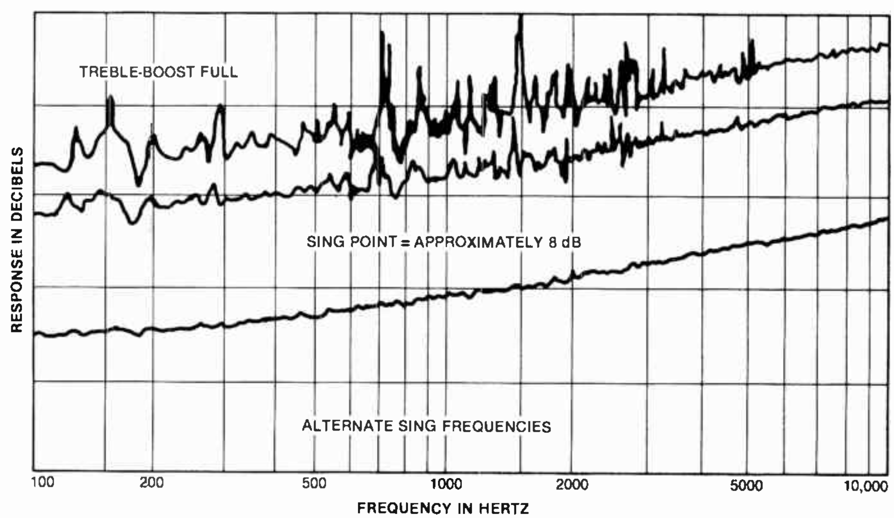
D_0 is the distance from the source to the farthest listener
D_1 is the distance from the loudspeaker to the farthest listener
D_s is the distance from the source to the microphone
D_2 is the distance from the microphone to the loudspeaker

It may be seen from equation 10-4 that the potential gain of a system is limited by the physical separation between the loudspeaker and microphone



(A) Bass boost.

Fig. 10-4. Nonlinear response of sound system employing excessive equalization.



(B) Treble boost.

(D_2). However, the distance between the source and the microphone (D_1) is of equal importance. The smaller the mike-to-source distance, the higher the potential acoustic gain.

$$D_c = D_r \times 10(L_r - L_R)/20 \quad \textbf{(10-5)}$$

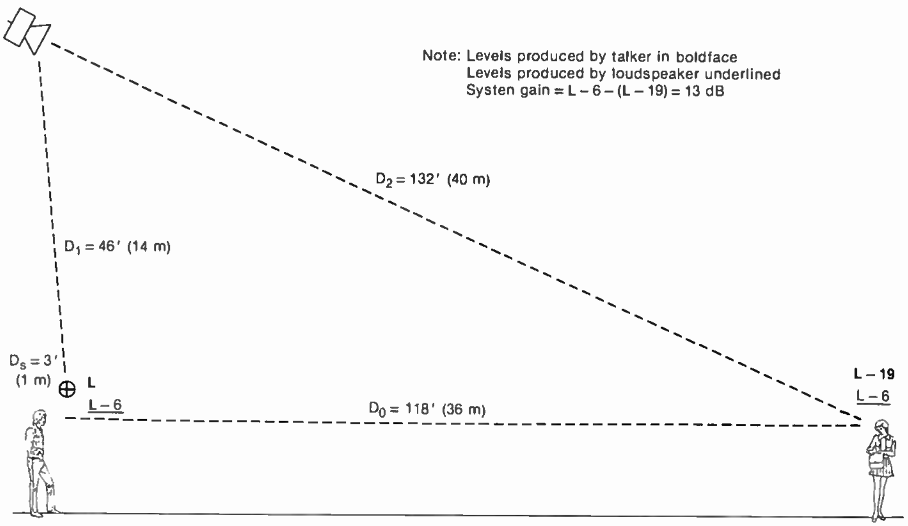


Fig. 10-5. Calculating PAG of indoor sound reinforcement system.

where,

D_r is a distance where the total SPL is at least 10 dB above the reverberant sound-field level

L_r is the direct sound level at D_r

L_R is the reverberant sound level

Large-Scale Indoor/Outdoor Reinforcement Systems

Large-scale reinforcement systems evolved in order to provide intelligible coverage for contemporary music productions of popular, rock, and jazz. Such systems (Fig. 10-6) are often large in scale with multiple-loudspeaker arrays on either side of the stage. Such a system can provide either indoor or outdoor coverage to audiences often numbering well in the thousands.

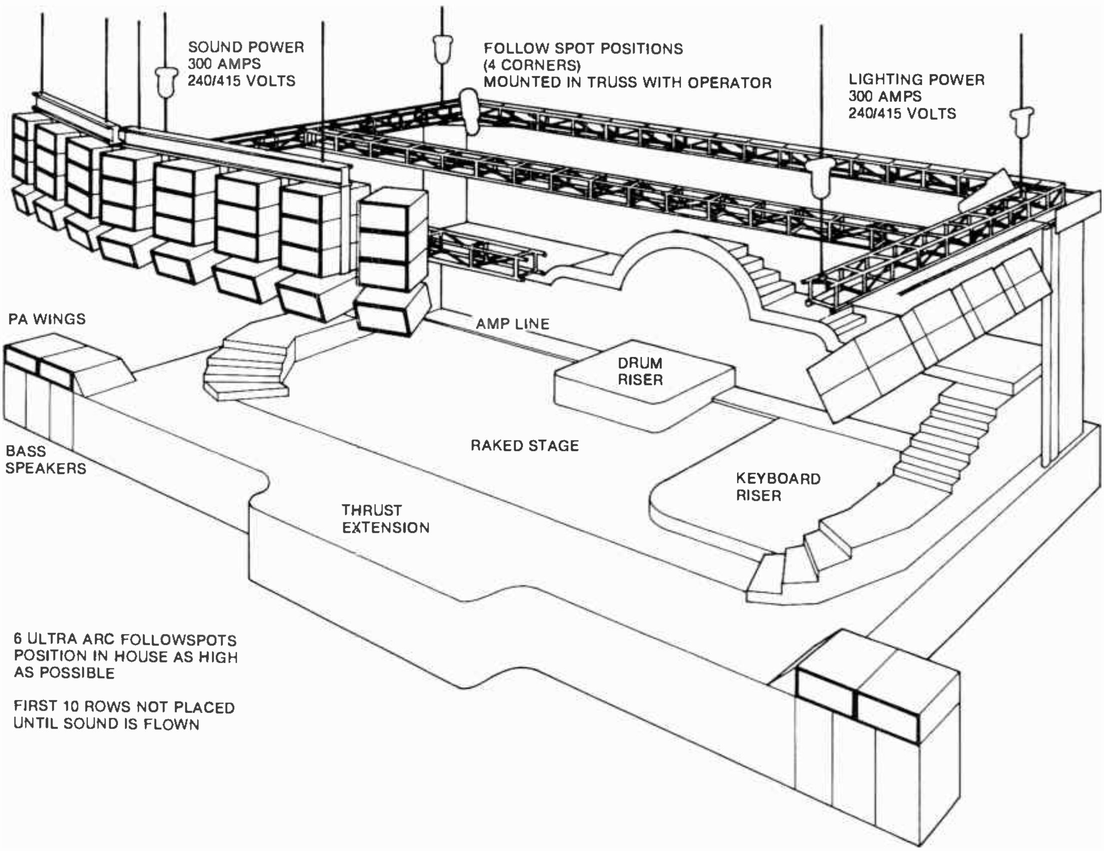
Because of the high-level sound pressure levels (often approaching 110 to 115 dB SPL) of these systems and their extensive on-stage monitoring systems, a high degree of stage gain, in conjunction with a very low critical distance, is present. Given these conditions, the following methods of miking are necessary to reduce or eliminate feedback:

- use of short working distances
- use of contact microphones, which enable a greater freedom of movement over conventional microphones, while maintaining close working distances
- use of direct injection, which is accessible from electric or electronic instruments

Using such direct signals and close working distances effectively eliminates almost all of the natural acoustics surrounding an instrument.

Reinforcement systems are most often used by groups embarked on tours to promote recently released albums. Fans expect to hear a sound closely resembling the balance and effects heard on the group's records. To achieve such a mix, a wide range of effects devices is used by the mixing engineer.

Fig. 10-6. Typical large-scale reinforcement system.



Summary

Our overview here was but a brief summary of the complex field of sound reinforcement. For further reading, an excellent source to consult is *Sound System Engineering* by Don and Carolyn Davis (Howard W. Sams & Company, 1987).

Appendixes

A The Use of Omnidirectional Microphones for Modern Recording

Adapted from "The use of B & K Omnidirectional Microphone for Modern Recording," written by David Rideau and reprinted by permission from Bruel & Kjaer Instruments, Inc.*

The next few pages introduce various mike positions for musical instruments that could help you get the most from your omnidirectional ("omni") microphone. To suggest that these placements are the only possibilities would be ludicrous. Sound and music have very subjective qualities and few people agree with others' impressions of "sound," but the examples given here have been tested and used with success by myself and other engineers. We hope you have the same or better success and are inspired to experiment with other uses.

Applications

Vocals

Undoubtedly, the most popular application for the omnidirectional microphone is the vocal. Once most singers hear the clarity that a high-quality omnidirectional microphone provides, there is no turning back. Recording a lead vocal with an omnidirectional has valuable plusses, including greater

*David Rideau has been working as an engineer for eight years, recording sound for record and film in the United States and Europe. During this time, he aided and advised in the construction of four recording studios, wrote for one of the recording industry's major periodicals, and produced several record albums.

freedom of movement. With a traditional lead-vocal setup (large-diaphragm condenser mike set to the cardioid polar pattern), there can be irritating problems of singer-microphone relationships. As the singer moves closer to the mike there is a buildup in the low-frequency area (the proximity effect). In addition, the physical presence of these usually large mikes causes reflections between the microphone and the singer's face, changing the in situ frequency response of the mike. Generally the artist finds a distance that is most pleasing to the engineer and producer and maintains that distance throughout the performance, or else the sound quality can change drastically. The artist must also be careful of plosives, since the mike is highly susceptible to vocal pops. In comparison, working with the smaller-diaphragm omnidirectional microphone is almost carefree. I generally place the microphone approximately 12–16″ (30–40 cm) directly in front of the artist's lips. In this position, with a respectable amount of compression, the singer has a great deal of freedom of movement in all directions.

Problems can also occur when a cardioid or figure-eight polar pattern is used with a group of "backup" singers; often, they must crowd together to be on the "right" side of the microphone's polar pattern. The signal of a singer who is to one side of the polar pattern often suffers in frequency response and sensitivity. So for background vocals I recommend placing an omnidirectional microphone pointing straight up toward the ceiling (Fig. A-1), at a height just below the chin of the smallest singer. With this placement, regardless of whether there are two or twenty singers, they can all gather around the mike, positioning themselves comfortably, each considering only his or her personal output level relative to the distance to the microphone and the other vocalists. If a particular singer is too quiet relative to the others, that singer should move closer to the microphone.

Acoustic Piano

Acoustic piano is another popular application of the omnidirectional microphone. Some engineers can't understand how a stereo image can be created with omnidirectional mikes, but, as other engineers have proven, it is possible and often beneficial.

Basically, you create a stereo picture by the relationship of the distance between microphones with respect to the distance to the sound source. It is very surprising just how little distance is needed between microphones before a stereo "picture" appears.

When I overdub a concert grand piano I generally place one omni 10″ (25 cm) directly above the hammers at about "high" C. I place a second omni about 12″ (30 cm) toward the lower register of the keyboard and 10″ (25cm) in from the hammers toward the end of the piano. The pair alone creates a

Fig. A-1. A group of background vocalists encircling an omnidirectional mike.

reasonable stereo picture, but for an even more dramatic stereo effect, I sometimes employ a free-hanging baffle (Fig. A-2). The baffle consists of a 14″ × 14″ × 3″ (35 × 35 × 7.5 cm) block of wood with a 1″ (2.5 cm) layer of semirigid insulation on each side. It hangs by string on a third mike stand, between the mikes. The result is a piano sound with great stereo imaging and equal power in every octave. With a small-diaphragm, high-quality omnidirectional microphone, subtle nuances are captured. Headroom and transient response at high and low frequencies are excellent.

Percussion

The omnidirectional microphone is ideal for overdubbing percussion. Generally, when I record percussion such as tambourine, shaker, cowbell, and other small hand-held instruments, I have the musician stand somewhere between 30–60″ (75–150 cm) from the mike, which is placed at about chest level. Musical dynamics can vary wildly, so also there is usually a "fast" compressor in line with the console signal.

For skinned instruments such as congas, bongos, and African drums, I place the mike as close as possible, 10–20″ (25–50 cm) over the drum head, without impairing the drummer's freedom of movement. This placement provides the "slap" that is frequently lost when recording these instruments. For glockenspiel, vibraphone, and marimba, you can also achieve a most faithful reproduction by using the same placement (10–20″ [25–50 cm] over the instrument).

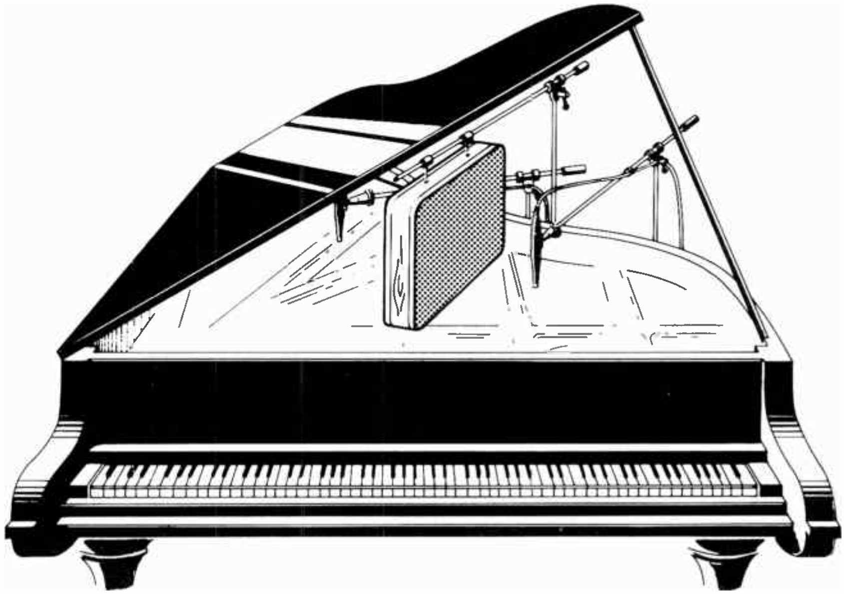


Fig. A-2. Free-hanging acoustic baffle suspended between two microphones to create a more dramatic stereo effect.

Drum Set

The drum set (Fig. A-3) in the modern studio is one of the most difficult challenges for the recording engineer. Often, you are required to create a "fantastic" sound, while maintaining isolation within the drum set itself for later artistic control. This undoubtedly results in a microphone on every tom, or between every two toms (the latter is more desirable, being less prone to phase problems). Also separate mikes are needed on the snare drum, bass drum, and hi-hat. Two overhead mikes and even one or two used as ambience mikes are also used. In this situation it is very hard to do the entire setup with omnidirectional mikes. I usually begin with large-diaphragm condenser mikes set to the cardioid polar pattern and placed 3–6″ (7.5–15 cm) over each tom tom, or one mike placed slightly higher between every two toms. In either case, the mike diaphragm is positioned over the rim of the tom, toward the front of the set. For the snare I usually use a dynamic mike to aid in isolation from the hi-hat. The omnidirectional microphone can be used in many instances for the bass drum, providing a crisp, clean kick sound. Placement can be quite different depending on the drum, but I like the front head off with the mike placed inside the drum, slightly off-center and 10–20″ (25–50 cm) from the back head. Be prepared to use a cover or enclosure for the bass drum itself if more isolation from the rest of the set is needed. I generally use some sort of isolation anyway, even when using a cardioid-pattern mike.

The overhead mike position is where a high-quality omnidirectional microphone can really shine, giving cymbals that "transparent" quality. Place

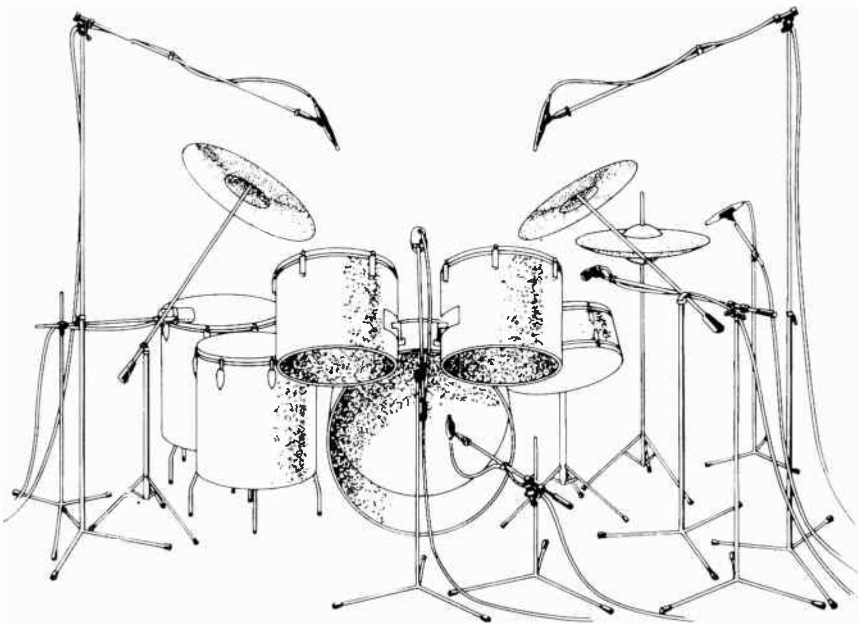


Fig. A-3. Drum set miked with carefully placed array of omnidirectional microphones.

these mikes on either side of the set just outside the drummer's shoulders and 12″ (25 cm) or more over his or her head. Experiment with the "fine tuning," at the same time checking coverage of cymbals, until you achieve a good stereo picture of the set (mikes placed left-right to the monitors). You can then mix in individual tom-tom mikes if needed.

Be aware that the more mikes you use for a particular instrument, the greater the chances are that phase atrocities will occur. Often the omnidirectional mike as overhead can provide a better overall picture in which individual mikes are less essential to the final product. I remember that one of the best drum sounds I've experienced was a session a fellow engineer and I did for a group that needed a quick, "very rough" demo tape. With this in mind, we did a quick, "very rough" setup that consisted of, among other things, dynamic mikes on bass and snare with a single B & K omnidirectional mike over the drummer's head. The clarity and transparency of the toms and cymbals were quite impressive! Alas, it's a stereo world . . . but how much of our sound should we compromise?

String Family

String instruments can benefit greatly from use of the omnidirectional microphone. Almost always, when miking a string instrument or section, you should incorporate the natural ambience of the recording room. Often a large-diaphragm condenser mike (even when set to an omnidirectional polar pattern) produces a "blurred" spatial image of the room's reverberance.

For a "pop" string session, usually consisting of six to twelve violins in pairs (two players reading one piece of music), I place an omnidirectional microphone pointing directly over the two musicians' chairs about 5 to 7′ (1.5−2 m) away, depending on the studio's reverberation characteristics. Violas are also miked in the same manner. With celli, I place the mike just over the bridge at a distance of 10−30″ (25−75 cm). This should produce that "resin" sound of bow movement that is often lacking in the recording of this instrument. The double bass is miked in the same manner.

Wind Instruments

With their wide dynamic range and sometimes brutal high-end transients, brass instruments can be a problem to record. For trumpet, cornet, trombone, fluegelhorn, and so on, the omnidirectional mike does the job magnificently. Placed 6−16″ (15−40 cm) from the bell, a high-quality omnidirectional condenser microphone will reproduce every subtle detail of a musician's tone.

For saxes, I place the omnidirectional mike at about the same distance over the bell of the horn, but slightly favoring the valve side of the instrument. Through experimentation, you can find a musician's "sweet" spot, which varies with a player's personal style.

For clarinet, bass clarinet, flute, and so on, I place the omnidirectional microphone 6−16″ (15−40 cm) over the valve area approximately in the middle of the instrument.

Please keep in mind that these wind and string placements are recommended for the small-ensemble overdub situation. In this setting, close-miking techniques are required to provide isolation between instruments so that the engineer/producer can control any imbalances that exist in the studio. In a classical setting, these balances among instruments or sections should occur naturally, demanding a more distant overall mike placement. But, even for ensemble overdubbing, an overall mike can be essential in creating a natural blend of instruments. Try to place a stereo pair 12−16″ (30−40 cm) apart, just behind the conductor and 2−6′ (.6−1.8 m) above her or his head, depending on the size of the group and the studio ceiling height (Fig. A-4). This signal blended with the individual mikes often produces a very pleasing combination.

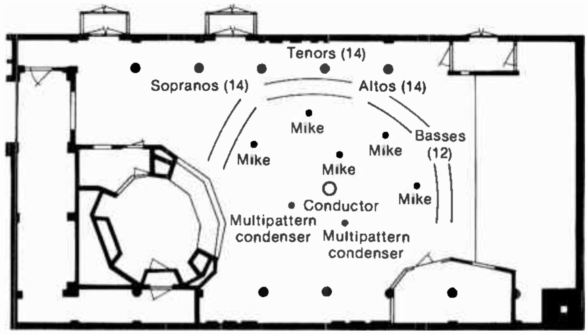


Fig. A-4. Microphone positioning for recording of Royal Theatre Opera Company at Easy Sound Studio, Copenhagen, August 1981.

Guitars

The clarity that can be achieved with a quality omnidirectional on a steel-string acoustic guitar is hard to beat with other mikes. With the mike placed 10–20″ (25–50 cm) in front of the upper fingerboard and angled slightly toward the sound hole, the artist enjoys the same freedom of movement as the lead vocalist. This omnidirectional microphone placement of course applies to the other stringed instruments in the same family: classical guitar (nylon- or gut-strung), mandolin, and so on).

On several occasions when I was using the omnidirectional microphone in this way, I experienced side effects—in the form of a musician's fears. Never before has the musician heard a performance reproduced so perfectly. Every little detail that previously escaped unnoticed now can be heard. When such "perfection" proves undesirable, I place a large-diaphragm condenser microphone set to the cardioid polar pattern approximately twice as far from the instrument as the omnidirectional microphone. A combination of these two signals almost always provides a final product that both the musician and I can agree on.

For electric guitars, I am partial to the "tight," "compact" sound a dynamic mike provides when placed directly in front of the speaker cabinet. This is due largely to its limited frequency response. But I also place an omnidirectional microphone 5–10′ (1.5–3 m) in front of the cabinet (Fig. A-5). The addition of a small amount of this signal to the original signal can produce a nice "open" quality that is very hard to create electronically. For heavy rock-guitar sounds employing a larger cabinet, I even add a third mike at a greater distance if the artist is looking for an "arena"-type ambience.

Electric bass cabinets are another story. They require the flat frequency response that a high-quality omnidirectional microphone can provide. Place the mike directly in front of the loudspeaker at a distance of 6–12″ (15–30 cm) and compress to taste.

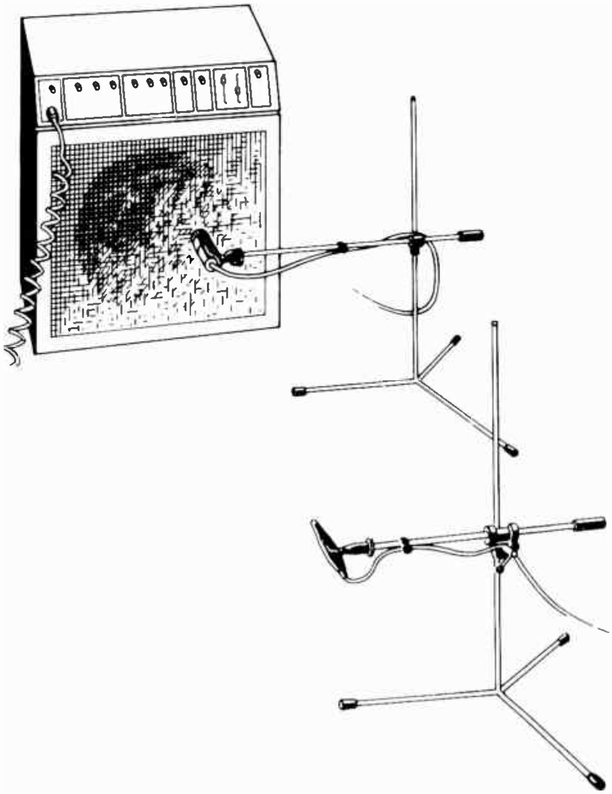


Fig. A-5. For a more open electric guitar sound, try using a second mike at a greater distance.

Special Applications and Considerations

"Good" Leakage

"Leakage" in the recording world has become a bad word. Most of the time engineers will go to great lengths to hold the levels of loud instruments as low as possible in order to avoid these instruments "leaking" into microphones that were not intended for them. In general, this approach is quite correct, but in certain special conditions, leakage can be used to advantage. For instance, I once recorded a jazz singer whose group consisted of drums, sax, piano, and upright bass. The singer understood why I had to use a "drum booth" to keep the drums from "spilling" into all the other mikes, but couldn't understand why she and the other band members couldn't gather cozily around the open grand piano.

I agreed and subsequently used omnidirectional microphones on the piano, sax, and vocal, after unsuccessfully trying several other directional mikes. I placed several gobos in strategic areas, but no one was deprived of direct eye contact. I then brought up the faders and was surprised to find that the leakage situation was not as bad as I had anticipated. Mixing afterwards was a breeze. I simply brought up the faders until I heard a reasonable balance between all the members and that was that. Other than a few subtle moves for solos and more difficult vocal passages, the mix basically took care of itself.

Of course, there was leakage among all the instruments and microphones, but instead of the "boxy" unusable leakage we are too used to hearing, it was a pleasant leakage that could easily blend with the other signals, producing a minimum of phase coloration. Naturally, in these cases positioning is critical, but a little experimentation can bring rewarding results. (I used the theory of equal or even-multiple distances among the various mikes: the piano mikes were 20″ [50 cm] from each other, the vocal mike 40″ [1 m] from the piano mike, the sax mike, 40″ [1 m] from the vocal mike, and so on.)

Diffuse Fields

Owing to its excellent phase response at all angles of incidence, the high-quality omnidirectional microphone is an excellent choice for an ambience mike. Try it in conjunction with drums and electric guitar. It can also be used effectively as a mike in a live chamber. The best suggestions I can make is to avoid the corners of the enclosure where low-end build-up usually occurs and to use your ears.

Adding Dimension to Your "Track"

Omnis can be great tools for overdubbing. Many recordings today can be forced into having a "one dimensional" image. Instruments are almost always close miked (usually with cardioid microphones) in a studio that has a "dead" quality (short reverberation time). Without the aid of artificial reverberation and other effects, this makes for a "one-dimensional" sound. The creative use of the omni mike during the overdub stages often can place an instrument in its own space through the use of natural ambience. This can be a big help in realizing the second- and third-dimension that engineers try to synthesize in the mixing stages of a production.

Omni vs. Directional

Most of the previously mentioned setups are intended for the overdub situation. But I contend that with the judicious use of isolation rooms, gobos, piano covers, and other conventional isolators, the omnidirectional mike can be used in many situations. In general, mike placement should be closer than usual, whenever possible without a compromise of sound quality. An omni has the same isolation characteristic at 3′ (1 m) that a cardioid would have at 5½′ (1.7 m) from the same sound source (Fig. A-6).

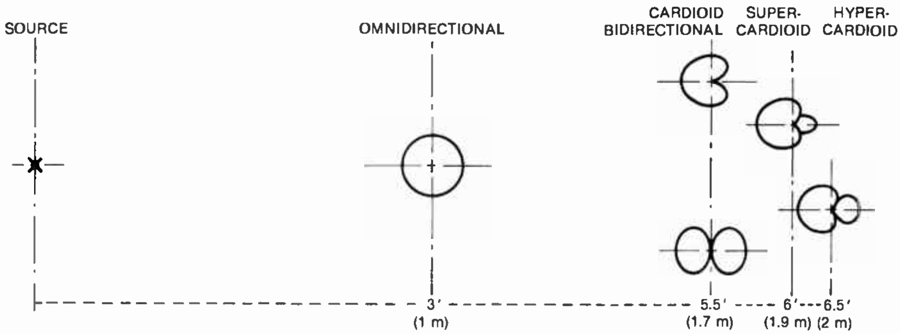


Fig. A-6. Relative positions of various microphone polar patterns for equal degrees of isolation.

Conclusion

A high-quality omnidirectional condenser microphone is often a very accurate device with an excellent dynamic range and "colorless" transient response. The fact that these mikes are omnidirectional should not exclude them from an engineer's arsenal of recording hardware. On the contrary, they can and should be incorporated into our growing pool of electronic resources.

Once you experience their sound, it might even influence some of your basic ideas about recording. I know it has convinced me to often take a "classical" approach to mike placement for overdubbing, finding that point where the balance between direct and reverberant sound is optimal with regard to the basic tracks.

Of course, there will always be conditions where a directional mike must be used, but don't be afraid to experiment with the omnidirectional microphone when you have the opportunity. You could be very pleased with the results as I, and many other engineers, have been.

B Microphone Techniques for Predictable Tonal Balance Control

Written by Bruce Bartlett, and reprinted here by permission of* Recording Engineer/Producer *magazine, Copyright 1982, Intertech Publishing Corp., Overland Park, Md.*

A recording engineer has several ways of creating the desired "sound" or tonal balance of a recorded musical instrument. You can equalize the instrument's sound so that it sounds right. Or you can try to find a microphone that sounds right. Or you can experiment with different microphone positions until you find one that sounds right. To help the reader create the desired sound more efficiently, this article considers the tonal effects of some close microphone placements.

This present study was prompted by an excellent article by W. Woszczyk in the October 1979 issue of *Recording Engineer/Producer*. Woszczyk describes methods of achieving a well-balanced tonal reproduction for the baritone saxophone and viola, using multiple microphones. This present article focuses on single-microphone techniques for the acoustic guitar, piano, electric guitar and voice.

Why Microphone Placement Affects Recorded Tonal Balance

The tonal quality or timbre of a musical instrument is mainly the perception of its spectrum or harmonic structure. Each part of the instrument produces a

*Bruce Bartlett, a microphone project engineer and technical writer for Crown International, has published more than 100 articles, including his popular "Recording Techniques" series. Mr. Bartlett is also author of the *Introduction to Professional Recording Techniques*, published by Howard W. Sams & Company.

different spectrum, and the spectra of all the parts combine into a pleasing composite at some distance from the instrument. A microphone placed there "hears" a blend of the tone qualities of the various parts.

Most musical instruments are built to sound best when heard from some distance away. So, a microphone placed fairly distant from an instrument—say 3′ away—tends to pick up a well-balanced or natural tone quality. But when feedback or leakage forces you to mike an instrument up close, the part of the instrument nearest the microphone is emphasized. Thus, the tonal balance picked up very close may not be representative of the instrument as a whole.

As an analogy, suppose a microphone is placed several feet from a multiple driver, high-fidelity loudspeaker replaying a particular piece of music. A well-balanced sound is usually picked up. But if you move that microphone very close—say next to the mid-range driver—the sound that the microphone "hears" is quite colored. Similarly, an acoustic guitar miked 3′ away sounds reasonably well balanced. Place the microphone close to the sound hole, however, and the guitar will sound bassy, because the sound hole radiates strong low frequencies. In general, close microphone placement can result in tone colorations caused by the local radiation characteristics of the instrument.

Musical instruments radiate different spectra in different directions and from different parts of the instrument. As a result, a microphone used on an instrument picks up a different spectrum depending upon where it is placed. Thus, the recorded timbre and transient character vary greatly with the microphone position.

Spectral Measurements for Acoustic Guitar

Experiments were performed at Shure Brothers to quantify the tonal differences of various close microphone placements. To do this, we recorded the spectrum of a musical instrument at several close microphone positions, and then compared these spectra to the spectrum picked up at a distance from the instrument.

For example, an acoustic guitar was recorded with a microphone placed 3′ away to pick up a well-balanced tone quality or an overall blend. Simultaneously, the guitar was recorded at several typical close-in positions (Fig. B-1). The spectrum picked up at 3′ was compared to the spectra picked up very close to the instrument.

The 3′ position (Fig. B-2) was chosen as a "natural" sounding reference because it was far away enough to pick up a good blend of all the parts of the guitar, aided by early room reflections, but was close enough to minimize the influence of standing waves and reverberation that can color the results.

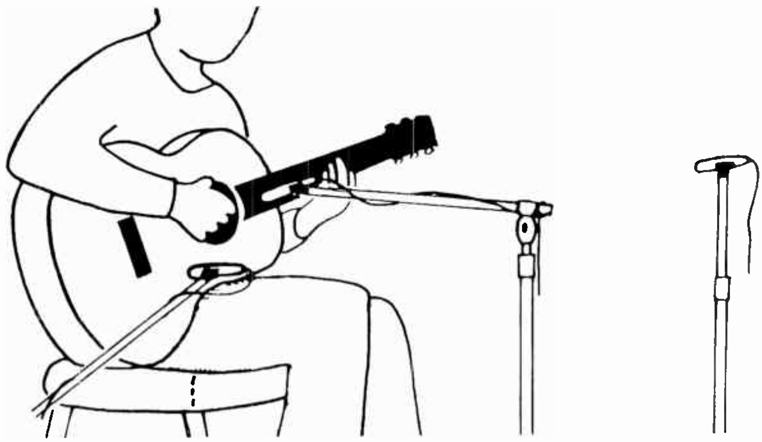


Fig. B-1. Various microphone positions for acoustic guitar.

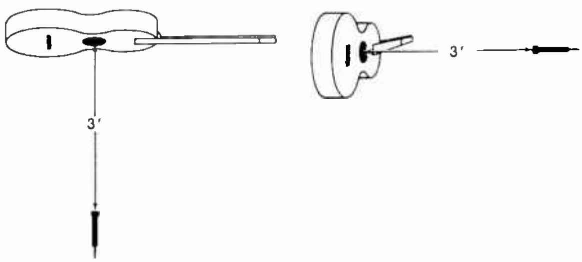


Fig. B-2. The 3′ reference miking position.

There may be other positions that work as well or better; this position is a matter of personal taste.

For the first test, a steel string guitar was recorded simultaneously with two microphones: one placed 3′ in front of the sound hole, and one 3″ from the sound hole. Figure B-3 shows the third-octave spectra picked up in these positions. (Measurement details of these tests are provided in the author's Engineering Report, "Tonal Effects of Close Microphone Placement"; in the October 1981 issue of the *Journal of the Audio Engineering Society*.)

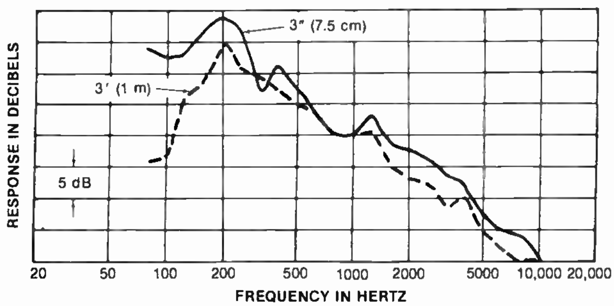
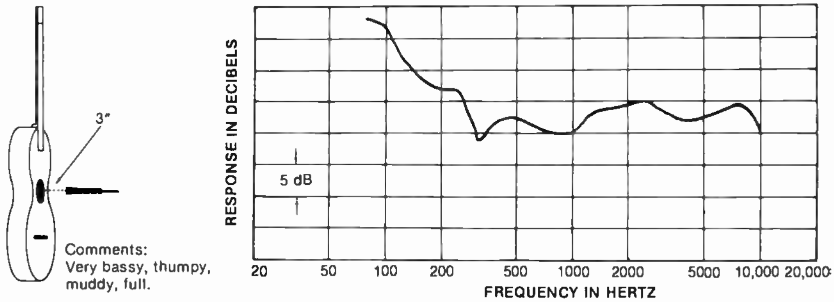


Fig. B-3. Steel string guitar spectra at 3′ and 3″ from the sound hole (Guild D-40).

To make the spectral differences easier to see, Figure B-4 shows the difference between the two spectra. When the guitar is miked three inches from the sound hole, there is a pronounced low-frequency emphasis when compared to the spectrum picked up at 3′. In other words, that is how the spectrum changes as you move the microphone from 3′ to 3″ away. The bass boost is not due to microphone proximity effect, however, since omnidirectional microphones were used to make the recordings. Instead, the bass boost is due to a strong 80-Hz resonance of the sound hole and the air inside the guitar. The resonance is picked up and emphasized by a microphone placed near the sound hole.

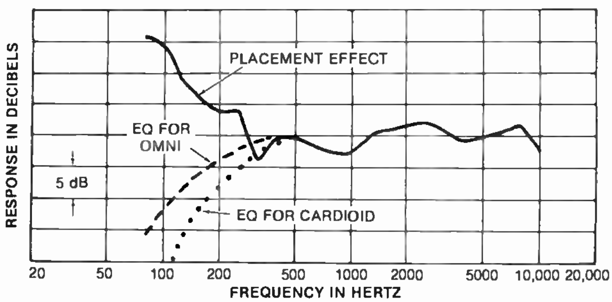
Fig. B-4. Difference between 3″ spectrum and 3′ spectrum of Fig. B-3.



The recordings were also played for a listening panel of eight musicians and audio engineers. In a blind A/B test, the panel compared the tone quality of each close-placed microphone to that of the more distant reference microphone. The majority of the participants considered that a microphone placed next to the sound hole makes the guitar sound bassy, "boomy," "thumpy" and full compared to the timbre picked up at 3′; these comments might be expected in view of the spectrum measurement.

If you want a full, bassy tone quality from a recorded guitar, a good place to put the microphone is near the sound hole. On the other hand, if you want to make the guitar sound more natural or well-balanced when miked close to the sound hole, you can roll off the bass as shown in Figure B-5. The solid line is the spectral difference close to the sound hole that was shown in Figure B-4.

Fig. B-5. Example of tonal correction by equalization.



The dashed line in Figure B-5 is the inverse or mirror image of the spectral curve at low frequencies. It is a suggested equalization to compensate for the bass boost caused by microphone placement close to the sound hole. This equalization is for an omnidirectional microphone with a flat frequency response.

If a cardioid microphone is used instead, it might have an additional inherent bass boost related to close placement called proximity effect. So, to make the guitar sound natural with a cardioid microphone, you must roll off the bass an additional 6 to 12 dB at 100 Hz, as shown by the dotted line in Figure B-5. Such equalization will compensate for proximity effect and the effect of close microphone placement. (It should be noted that some cardioid microphones have a built-in, switchable bass roll-off filter.)

As Woszczyk pointed out in his previous article in *RE/P*, the spectral effects of close microphone placement cannot always be accurately compensated for by equalization. The required equalization is complex and varies from note to note; also, harmonics that are missed by close microphone placement cannot always be recovered by equalization. Still, some general equalization is probably better than none at all.

As an alternative to equalization, you might consider using a microphone with an appropriate frequency response. For example, an omnidirectional microphone with a low frequency roll-off would tend to make a guitar sound natural when placed close to the sound hole. Ideally, the microphone should have a low-frequency response similar to the dashed line of Figure B-5. An example of such a microphone is shown in Figure B-6. It is designed to provide a good starting-point tonal balance when clipped onto the sound hole of a guitar.

Reasons for Equalization

Of course, there are many other reasons for equalizing an instrument besides correcting the tonal effects of close microphone placement. One major use of EQ is for special production effects. Or, you may have to use EQ to compensate for microphone off-axis coloration, microphone frequency response, monitor frequency response or masking by other instruments.

Another reason for equalization has its origins in the psycho-acoustics of our hearing. As discovered by Fletcher and Munsen, the ear is less sensitive to bass and treble frequencies at low volume levels than at high volume levels. Suppose you are listening to a very loud musical instrument, live, such as an amplified electric guitar. If this instrument is recorded with a flat-response microphone and played back at a lower level than you heard live, you probably will hear less lows and highs in the playback than you heard during the

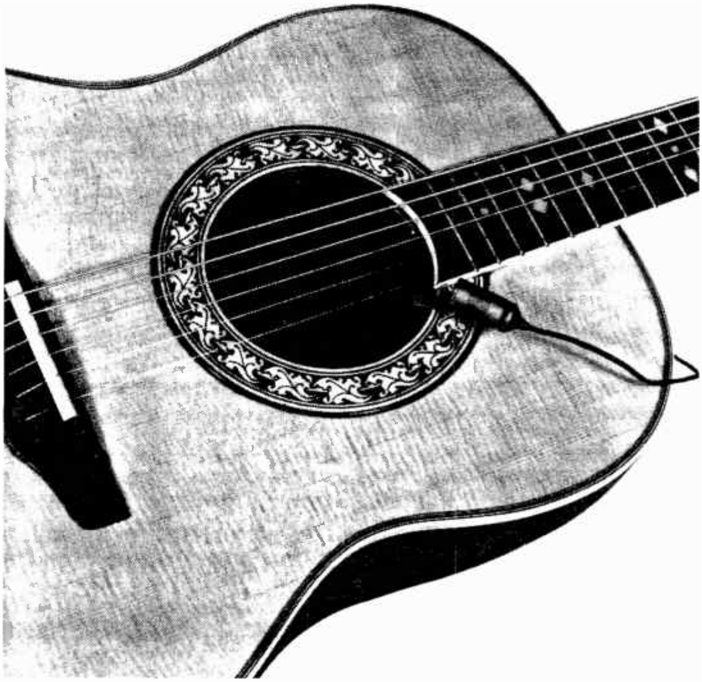


Fig. B-6. Clip microphone near guitar's sound hole.

recording—the sound may be thin and lacking in punch or presence. So, when recording very loud instruments like electric guitars or drums, it may help to use a microphone with bass and treble boost—in other words, proximity effect and a presence peak. This will help compensate for hearing phenomena that occur with playback that is quieter than the live instrument.

Other Acoustic-Guitar Spectral Measurements

We've seen what happens when a guitar is miked close to the sound hole. Let's now look at what occurs at some other close positions.

If the microphone is placed 4″ in front of the bridge (Fig. B-7A), the top-plate vibrational modes starting around 200 Hz are emphasized. The result is a mid-bass boost, producing a somewhat bassy, boomy and dull sound character. In positive terms, you could say that this microphone pickup sounds warm, "woody" and "mellow."

A microphone placement 3″ from the neck where it joins the body of the guitar (Fig. B-7B) de-emphasizes the mid-bass frequencies because the microphone is relatively far from the front plate. The sound hole resonance is still noticeable in this position. Listeners reported that the timbre was bassy

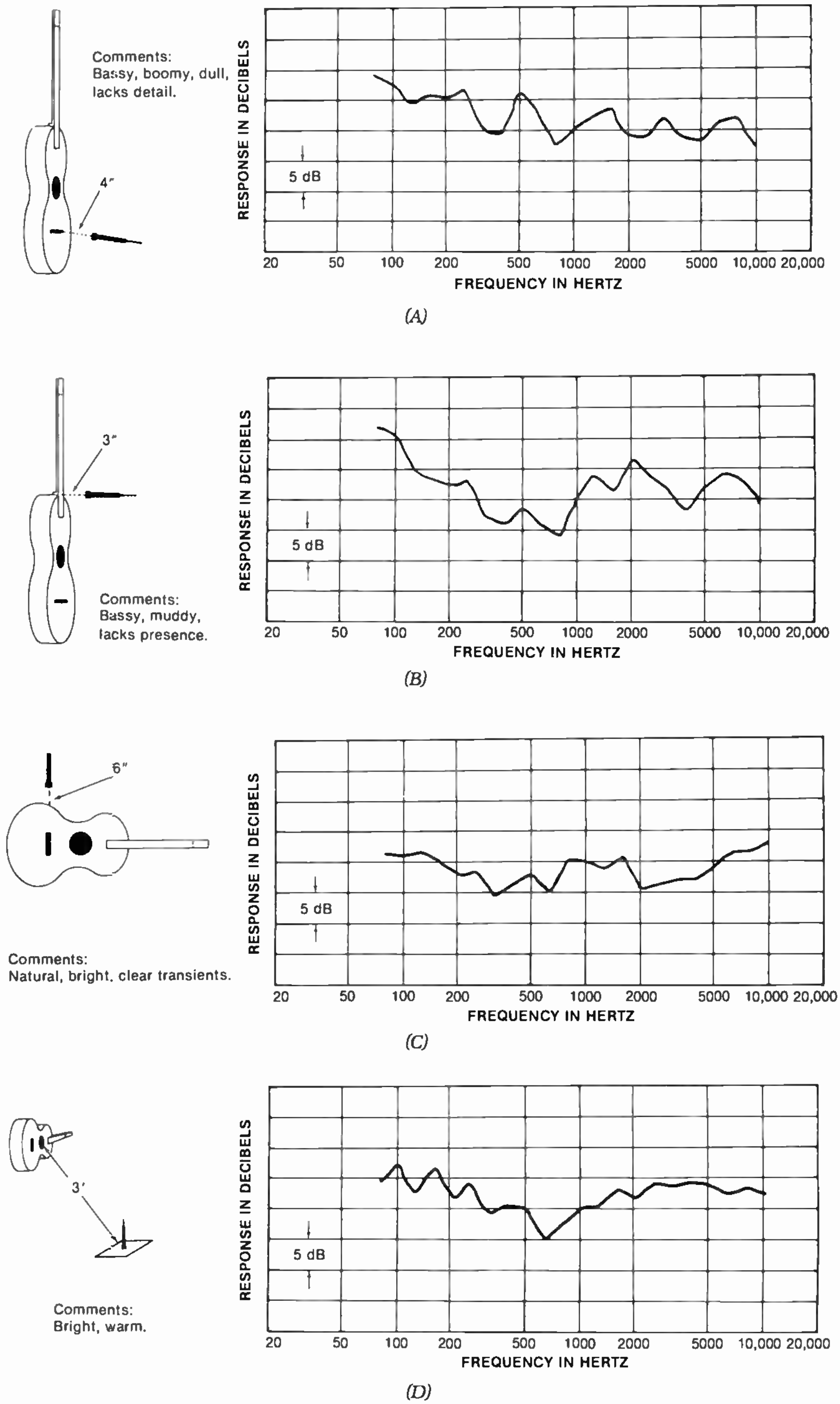


Fig. B-7.
Difference between close-miked spectrum and 3′ spectrum, for steel-string guitar (Guild D-40). Average of results for "E" and "A" chords.

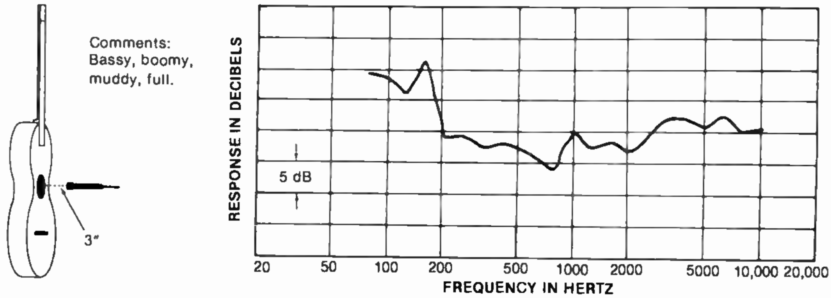
and lacking in presence—perhaps due to low-frequency masking of higher frequencies. With another guitar, the presence range may well be audible.

When the microphone is placed 6″ over the top of the guitar, above the bridge and even with the front plate (Fig. B-7C), the spectrum and timbre picked up at this point are fairly similar to what is picked up 3′ in front. With such an "over-the-top" position, listeners reported that they heard a natural, bright sound with clear transients. The spectra of the various parts of the guitar combine in a pleasing manner in this location.

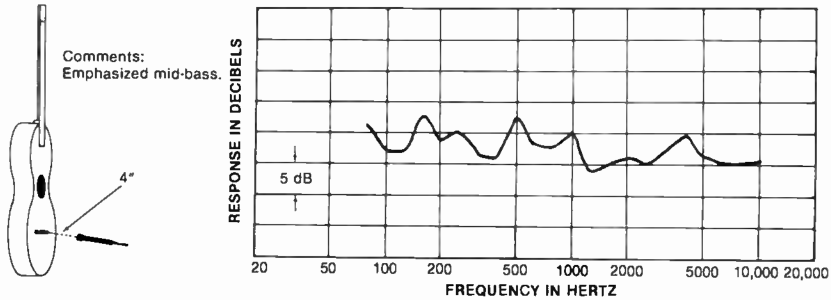
Another realistic-sounding location, according to the listening panel, is shown in Figure B-7D. The microphone diaphragm was positioned 0.05″ from a 1′ × 1′ sound-reflective plate on a carpeted floor (a hard floor could also be used). If feedback or leakage presents no problems, this may be a useful microphone position. (It should be noted that the boundary microphone features a similar principle of operation—*Ed.*)

The same measurements were repeated using a nylon-stringed guitar with generally similar results (Figs. B-8A–E). Differences occurred in isolated frequency bands. Each individual guitar will produce slightly different results; also, the effects vary from note to note. The results shown here are meant to indicate only general trends.

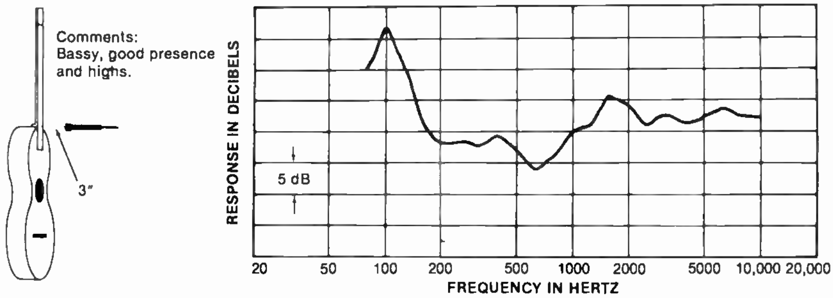
Fig. B-8. Difference between close-miked spectrum and 3′ spectrum, for nylon-string guitar (Sakura). Average of results for "E" and "A" chords.



(A)

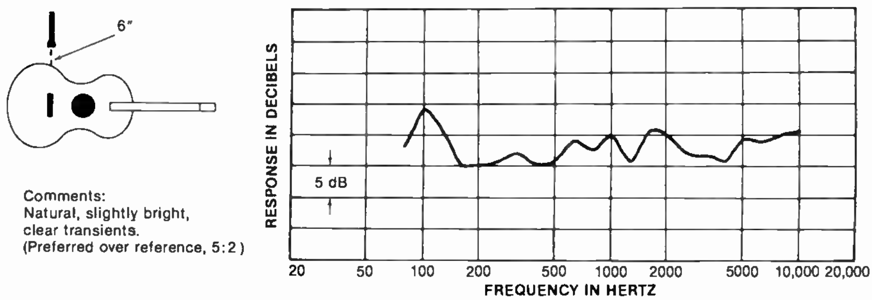


(B)

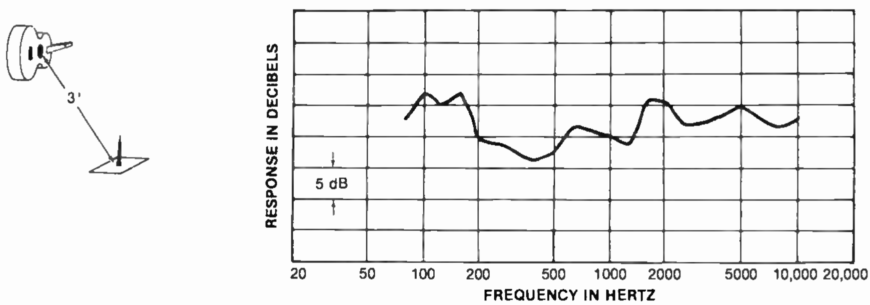


(C)

Fig. B-8 (cont.)



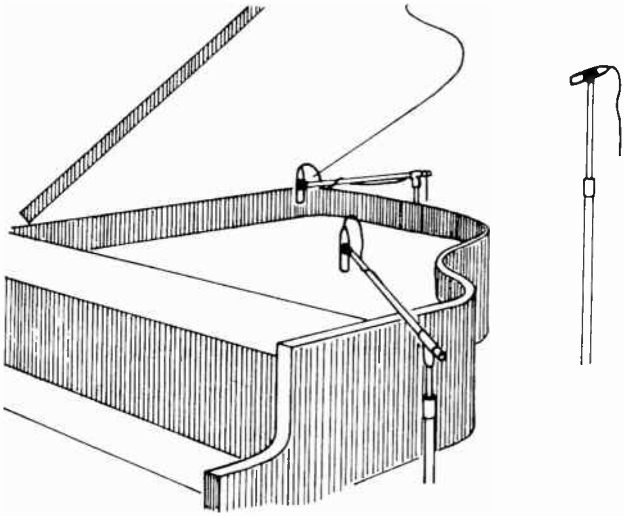
(D)



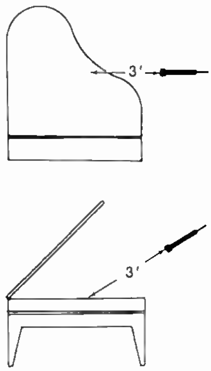
(E)

Grand Piano

Another instrument tested was a 5′ baby grand piano (Fig. B-9A), which was recorded from several typical microphone positions and, simultaneously, 3′ away from the strings (Fig. B-9B) to pick up an overall blend or a reference spectrum. Figures B-10A through F show the difference between the 3′ spectrum and each of the close-miked spectra.

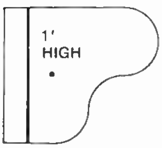


(A)

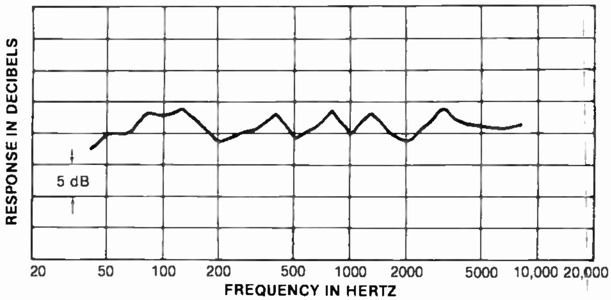


(B)

Fig. B-9. Microphone positions used in pickup of baby grand piano.

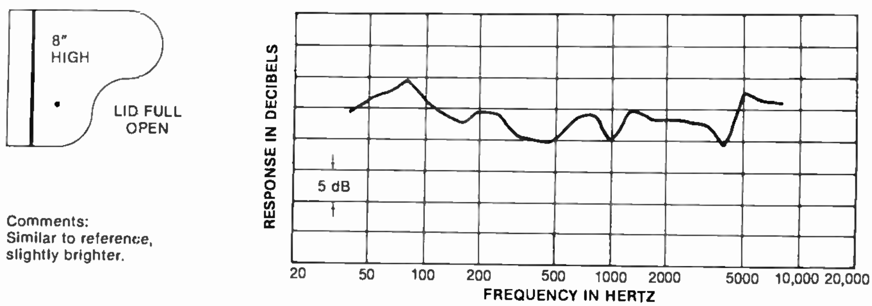


Comments:
Little consensus.
Similar to reference.

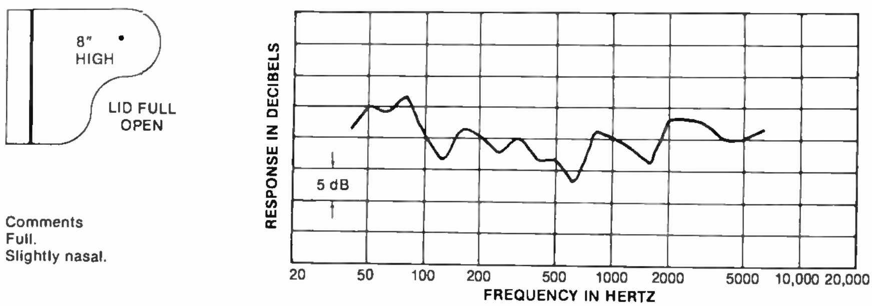


(A)

Fig. B-10. Difference between close-miked spectrum and 3′ spectrum, for grand piano (Baldwin "B"). Nearly all keys were pressed simultaneously.

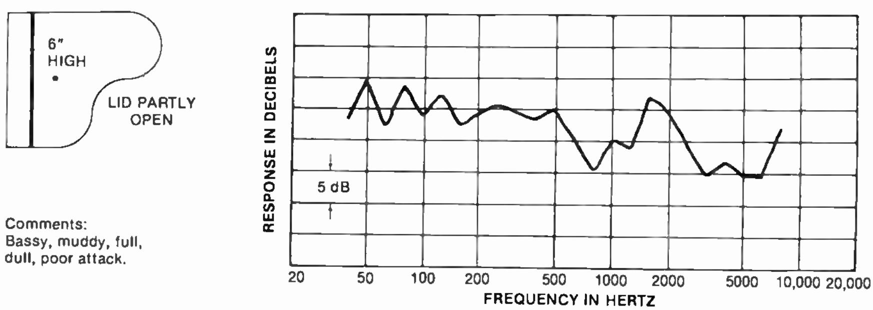


(B)

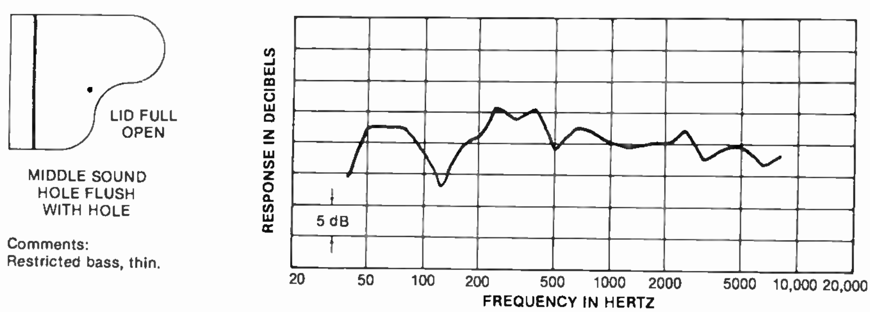


(C)

Fig. B-10 (cont.)

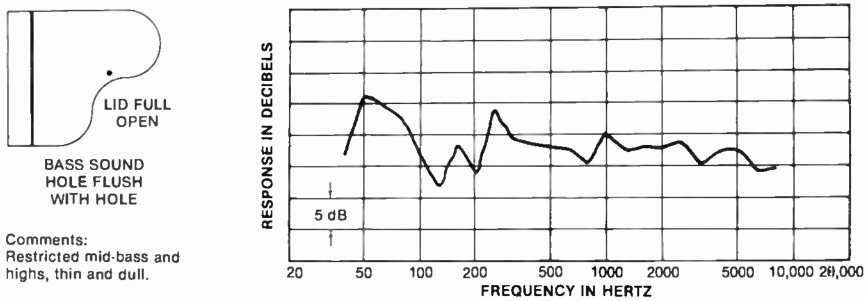


(D)

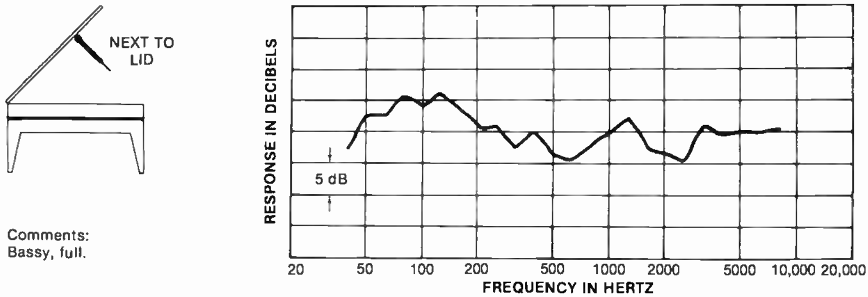


(E)

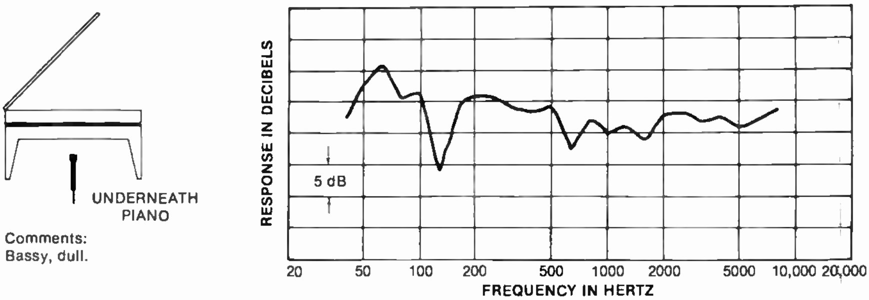
Fig. B-10 (cont.)



(F)



(G)



(H)

In Figure B-10A, the piano is miked 1′ over the middle strings, 8″ horizontally from the hammers, with the lid on the long stick (full open). There are only minor differences between the spectrum picked up in this position and the spectrum picked up 3′ in front. Thus, 1′ over the middle strings is a suggested close-microphone placement for a natural timbre. If the microphone is moved to 8″ over the treble strings (Fig. B-10B), the sound is still well balanced and bright.

Results for other microphone positions are shown in Figure B-10C through H. Two of the positions tested offer good isolation but an unnatural sound:

inside the piano with the lid on the short stick (Fig. B-10D); and located in the sound holes (Fig. B-10E and F).

The spectral effects vary with microphone position because areas close to the microphone are emphasized. In addition, sounds from various areas of the piano and lid combine and cause acoustic phase cancellations that vary with microphone placement. Like the guitar, the sound from a piano can be equalized to complement each spectral curve to approach a natural timbre.

Electric Guitar Amplifier

An electric guitar amplifier/speaker was also recorded. The amplifier was an open-back design with dual 12″ loudspeakers. Pink noise was played through the amplifier, and the acoustic output was equalized flat at the 3′ reference position (Fig. B-11A). This position was selected to minimize acoustic comb-filter effects due to sound arrival-time differences of the two loudspeakers.

Note that sound reflections from the floor still can cause acoustic comb-filter effects at the 3′ microphone position. However, it is unclear whether these effects color the instrument timbre or enhance it. Floor reflections will

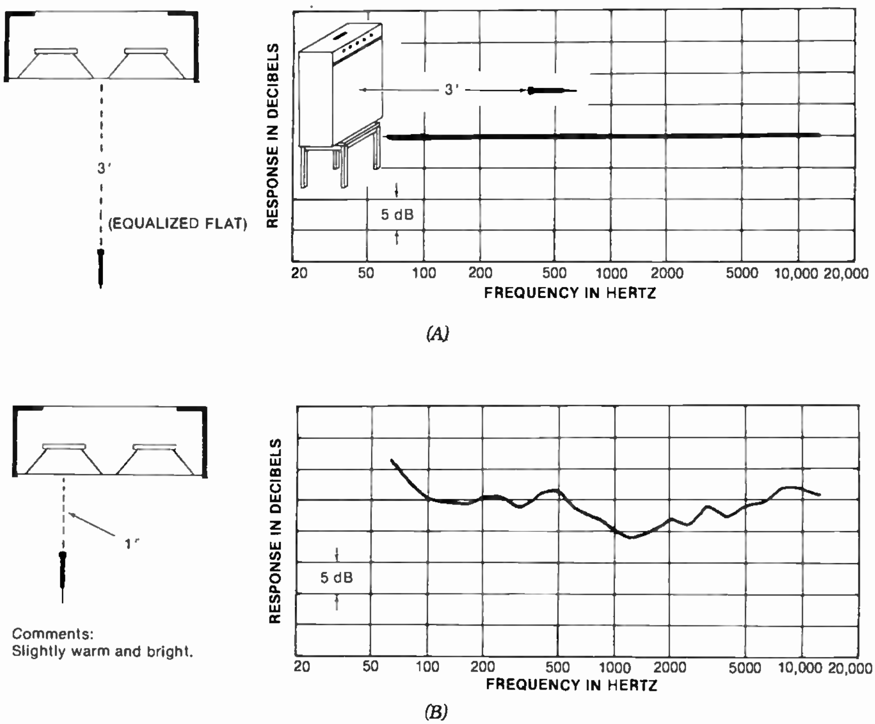


Fig. B-11. Difference between close-miked spectrum and 3′ spectrum for electric guitar amplifier/speaker (Yamaha G-100-212-11). Pink noise input.

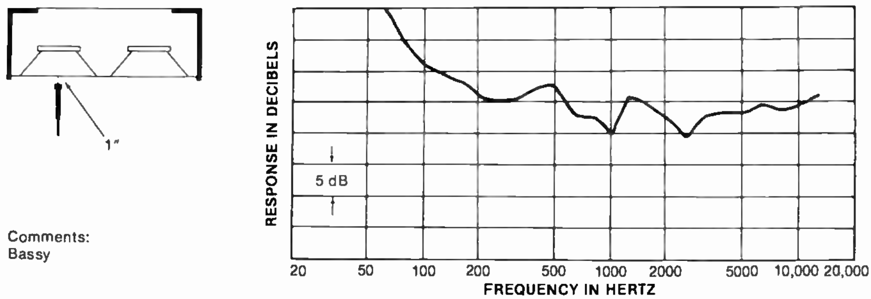
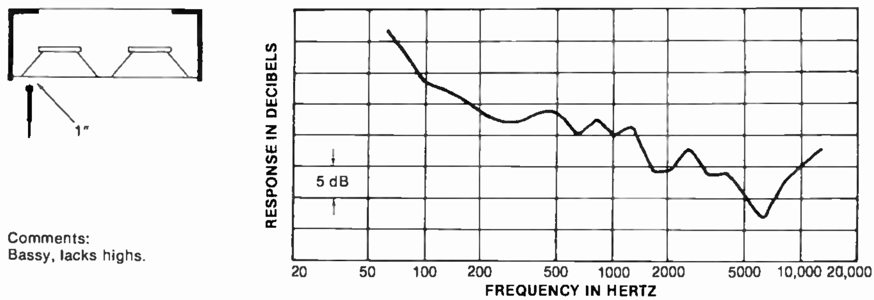


Fig. B-11 (cont.)

(C)



(D)

be heard while listening to a live instrument; they add information about the spectral content of the instrument and its environment. Unfortunately, the comb-filter notch frequencies and notch depths picked up at the microphone position are different from those heard in the audience area. Regardless of whether there is coloration heard occurring at the reference position, it is still a known standard to which other microphone positions can be compared.

At 3′ from the loudspeakers, the rear wave from the speakers tends to cancel the front wave at low frequencies, because the two waves combine in opposite polarity. If the microphone is placed closer, say 1′ away (Fig. B-11B), the front wave is emphasized at the microphone position, resulting in low frequencies not being cancelled so completely. Thus, there is an apparent bass boost compared to the spectrum at 3′.

In Figure B-11C, the microphone is 1″ from the grille cloth, so there is even more apparent bass boost. Remember that, because omnidirectional microphones were used, this boost is not due to microphone proximity effect; it results from loudspeaker radiation characteristics.

Note in Figure B-11D what happens to the high frequencies as the microphone is moved to the edge of the speaker cone. The high end is lost, producing a dull sound quality. This occurs because the microphone is far from the center of the cone, the high-frequency radiating part of the speaker. Pickup of amplifier hiss and crackle can be reduced by this microphone placement.

As these curves have shown, it is easy to control the bass-treble balance of this type of guitar amplifier by varying the microphone distance and position relative to the speaker cone center.

Vocals

To reduce audible reverberation, a voice was recorded with the reference microphone position 1′ away (Fig. B-12A).

Some members of the listening panel experienced difficulty hearing the spectral effects of placing an omnidirectional microphone 1″ from the mouth

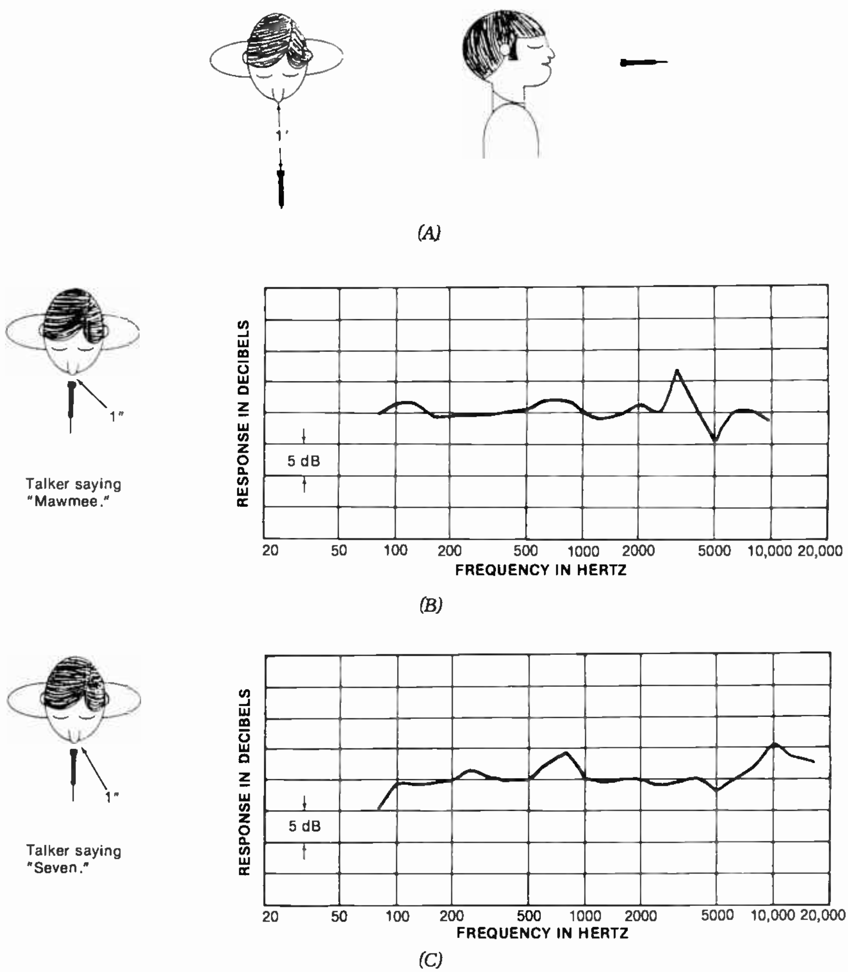
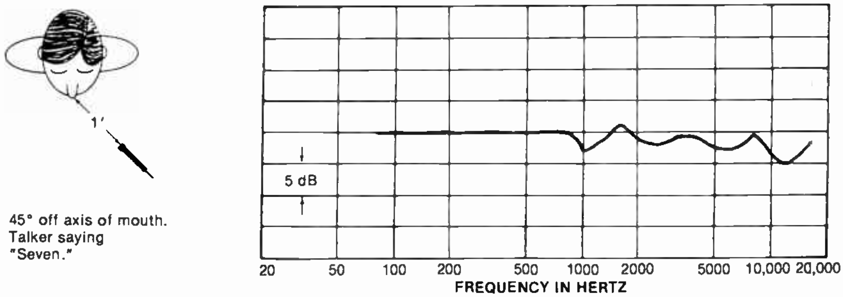


Fig. B-12. Difference between close-miked spectrum and 1′ spectrum for male voice.

Fig. B-12 (cont.)



(D)

(Fig. B-12B and C). Thus, a flat-response omnidirectional microphone placed close to the mouth produces only minor coloration.

A nasal sounding word, "mawnee," was used for one test (Fig. B-12B). When a microphone is placed very close to the mouth, the sound from the mouth is picked up louder than the sound from the nose (especially when the microphone is highly directional). Consequently, the nose sounds "closed," lending a "nasal" coloration to the voice. Note in Figure B-14B the peak at 3 kHz, which often is associated with a "nasal" tone coloration. Possibly the effect would be more audible if a hypercardioid or bidirectional microphone were used, since each one has reduced output off-axis (in the direction of the nose).

To test for sibilance differences, the word "seven" was recorded (Fig. B-12C). The voice is somewhat directional at high frequencies, so sounds that are strong in high frequencies, such as sibilance and lip noises, are slightly emphasized close to the mouth. This can be corrected with a high-frequency roll-off. Also, a 3-dB cut at 800 Hz may make the close-miked voice sound more natural.

In Figure B-12D, the microphone is 1′ away like the reference microphone, but is positioned 45° off-axis to the mouth. Again, since the voice is directional at high frequencies, the microphone placed to one side picks up a duller-sounding spectrum than the same microphone placed in front. This position, or one farther to the side, can be used to minimize sibilance.

Summary and Conclusions

The spectral plots presented in this article suggest some starting microphone placements to achieve particular tonal effects. According to the listening panel, the 3′ reference microphone positions generally were judged to provide a more realistic and natural-sounding tone quality than the close positions tested. Such observations suggest that a natural sound is best obtained by relatively distant microphone placement—assuming, of course, that the room acoustics are suitable.

Some close microphone positions were discovered that pick up a tonal balance similar to that picked up 3′ away. For acoustic guitar, the best-matching position tested was 6″ over the top of the guitar, above the bridge and even with the front face. For grand piano, a good match was found with the microphone located 1′ above the middle strings, or 8″ above the treble strings, 8″ horizontally from the hammers, with the lid on the long stick. Remember that these positions are for flat-response omnidirectional microphones. Cardioid microphones may sometimes add bass boost caused by proximity effect.

Other locations not tested in this study may work as well or better.

If a natural sound is desired, but you are forced to place the microphone in a bad-sounding position, the instrument can be equalized as suggested in this article as a beginning—or use a microphone with an appropriate frequency response. Final adjustments should be done by the ear to suit the particular instrument and application.

Surprisingly, a flat-response microphone does not always produce the most natural reproduced sound, because close microphone placement itself can color the tone quality. So it is necessary to experiment with various microphones and microphone positions to find the best compromise.

The bass-and-treble balance of an open-back guitar amplifier can be controlled easily by varying the microphone distance and position relative to the speaker-cone center. If a voice is miked up close with a flat-response omnidirectional microphone, only minor coloration occurs.

The purpose of this article has been to indicate the general tonal effects that can be expected in various microphone positions. Whether or not these effects are desirable is up to the engineer and the producer. The more you know about microphones, musical instruments and the interface between them, the easier it is to achieve the desired end results.

C The PZM Boundary Booklet— A Basic Primer and Experimenter's Guide

Written by Bruce Bartlett, and reprinted here by permission of Crown International, Inc.

You can greatly broaden your range of PZM (abbreviation for Pressure Zone Microphone) applications by mounting PZMs on one or more boundaries. A boundary is a stiff, non-absorbent surface such as a floor, table or plexiglass panel. PZM boundaries are usually constructed of clear acrylic plastic (plexiglass) to make them less conspicuous, but any stiff, sound reflective material can be used. By adding boundaries to a PZM, you can tailor the microphone's frequency response and directional pattern.

This guidebook explains the theory, benefits and drawbacks of single and multiple boundaries. Also covered are construction methods for several types of PZM boundary assemblies.

Credit is due to Ken Wahrenbrock for his pioneering work in multiple-boundary experiments and for many of the boundary array suggestions in this booklet.

A PZM is designed to be mounted very near a boundary to prevent acoustic phase cancellations. The boundaries mentioned in this guide will degrade the frequency response and polar patterns of conventional microphones. Only the boundary-type microphone can be used effectively with multiple boundaries.

The size, shape and number of boundaries all have profound effects on the performance of a PZM mounted on those boundaries. Let's discuss these effects in detail.

Sensitivity Effects

Imagine a PZM mike capsule in open space, away from any boundaries. This microphone has a certain sensitivity in this condition.

Now suppose the PZM capsule is placed very near (within .020″ of) a single large boundary, such as a wall. Incoming sound reflects off the wall. The reflected sound wave adds to the incoming sound wave in the "pressure zone" next to the boundary. This coherent addition of sound waves doubles the sound pressure at the microphone, effectively increasing the microphone sensitivity or output by 6 dB.

In short, adding one boundary increases sensitivity by 6 dB. This is *free gain*.

Now suppose the PZM capsule is placed at the junction of *two* boundaries at right angles to each other, such as the floor and a wall. The wall increases sensitivity by 6 dB, and the floor increases sensitivity by another 6 dB. Thus, adding two boundaries at right angles increases sensitivity 12 dB.

Now let's place the PZM element at the junction of *three* boundaries at right angles, such as in the corner of the floor and two walls. Microphone sensitivity will be 18 dB higher than it was in open space. This is increased gain with no increase in noise!

Note that the *acoustic* sensitivity of the microphone rises as boundaries are added, but the *electronic* noise of the microphone stays constant. Thus the effective signal-to-noise ratio of the microphone improves 6 dB every time a boundary is added at right angles to previous boundaries.

If the PZM is in the corner of three boundaries that are *not* at right angles to each other, the sensitivity increases *less than* 6 dB per boundary. For example a PZM-2.5 boundary is built with two panels at 135°. This panel assembly is at right angles to a base plate. The net gain in sensitivity from these three boundaries is approximately 16 dB, rather than 18 dB.

Direct-to-Reverb Ratio Effects

We mentioned that sensitivity increases 6 dB per boundary added. That phenomenon applies to *direct* sound reaching the microphone. *Reverberant* or *random incident* sound increases only 3 dB per boundary added. Consequently, the direct-to-reverb ratio increases 3 dB whenever a boundary is added at right angles to previous angles.

A high direct-to-reverb ratio sounds close and clear; a low direct-to-reverb ratio sounds distant or muddy. Adding boundaries increases the direct-to-reverb ratio, so the subjective effect is to make the sound source audibly closer or clearer. That is, "reach" is enhanced by adding boundaries.

Frequency-Response Effects

The size of the boundary the PZM is mounted on affects the PZM's low-frequency response: The bigger the boundary, the better the bass. Specifically, the response shelves down 6 dB at and below the frequency F, where: F[−6] = 188/D′ and D is the boundary dimension in feet. For example, if the boundary is 2′ square, the 6 dB down point is: F[−6] = 188/D = 188/2 = 94 Hz. (See Fig. C-1.)

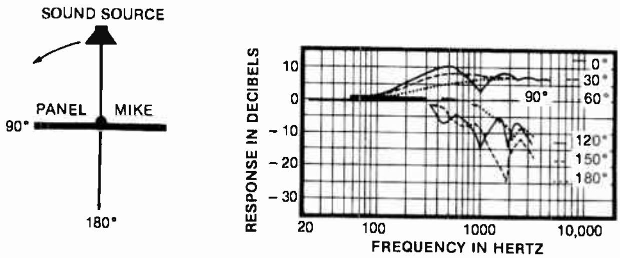


Fig. C-1. Frequency response of PZM mounted in center of 2′ square panel, at various angles of incidence.

Below 94 Hz, the response is a constant 6 dB below the upper mid-frequency level. Note that there is a response shelf, not a roll-off.

If a PZM is mounted on a 4′ square boundary, the response is down 6 dB at and below F[−6] = 188/D = 188/4 = 47 Hz. This result has been loosely called the "4 foot/40 Hz" rule.

What if the PZM is on a rectangular boundary? Let's call the long side "D_{max}" and the short side "D_{min}." The response is down 3 dB at 188/D_{max}, and it is down another 3 dB at 188/D_{min}.

The low-frequency shelf varies with the angle of the sound source to the panel. The shelf starts to disappear when the source is closer than a panel dimension away. If the source is very close to the PZM mounted on a panel, there is no low frequency shelf; the frequency response is flat.

If the PZM is at the junction at two or more boundaries at right angles to each other, the response shelves down 6 dB *per boundary* at the frequency mentioned above. For example, a two-boundary unit made of 2′-square panels shelves down 12 dB at and below 94 Hz. The cause of the low-frequency shelf is explained in the Sub-Appendix.

There are other frequency-response effects in addition to the low-frequency shelf. For sound sources on-axis to the boundary, the response rises about 10 dB above the shelf at the frequency where the wavelength equals the boundary dimension.

For a square panel F(peak) = .88C/D, where C = the speed of sound (1130 ft/sec) and D = the boundary dimension in feet. For a circular panel F(peak) = C/D. As an example, a 2′-square panel has a 10-dB rise above the shelf at .88C/D = .88 × 1130/2 = 497 Hz. (.88 × 344/.6096 m = 497 Hz).

Note that this response peak is only for the direct sound of an on-axis source. If the sound field at the panel is partly reverberant, or if the sound wave strikes the panel at an angle, the effect is much less. The peak is also reduced if the mike capsule is placed off-center on the boundary. An explanation of the response anomalies is given in the Sub-Appendix.

Figure C-1 shows the frequency response of a PZM mounted on a 2′-square panel, at various angles of sound incidence. Note several phenomena shown in the figure:

1. The low-frequency shelf (most visible at 30° and 60°).
2. The lack of low-frequency shelving at 90 degrees (grazing incidence).
3. The 10 dB rise in response at 497 Hz.
4. The edge-interference peaks and dips above 497 Hz (most visible at 0° or normal incidence).
5. The lessening of interference at increasing angles.
6. The greater rear rejection at high frequencies than low frequencies.

Directional Effects

The polar pattern of a PZM on a large surface is hemispherical. The microphone picks up equally well in any direction above the surface plane, at all frequencies.

By adding boundaries adjacent to this PZM, you can shape its directional pattern. Boundaries make the PZM reject sound coming from behind the boundaries. In addition, making the PZM directional increases its gain-before-feedback in live-reinforcement applications. Directional PZMs also pick up a higher ratio of direct sound to reverberant sound, so the resulting audio sounds "closer" and "clearer."

In general, sound pickup is fairly constant for sound sources at any angle in front of the boundaries, and drops off rapidly when the source moves behind the boundaries.

For sounds approaching the rear of the panel, low frequencies are rejected least and high frequencies are rejected most.

A small boundary makes the PZM directional only at high frequencies. Low frequencies diffract or bend around a small boundary as if it isn't there. The bigger you can make the boundary assembly, the more directional the microphone will be across the audible band.

The bigger the boundary, the lower the frequency at which the PZM becomes directional. A PZM on a square panel is omnidirectional at very low frequencies and starts to become directional above the frequency F, where F = 188/D and D is the boundary dimension in feet. Sound familiar? That's the same equation used to predict the 6-dB-down point in the frequency response.

Boundaries create different polar patterns at different frequencies. For example, a 2′-square panel is omnidirectional at and below 94 Hz. At mid-frequencies the polar pattern becomes supercardioid. At high frequencies, the polar pattern approaches a hemisphere (as in Fig. C-2). Two boundaries are more directional than one and three are more directional than two.

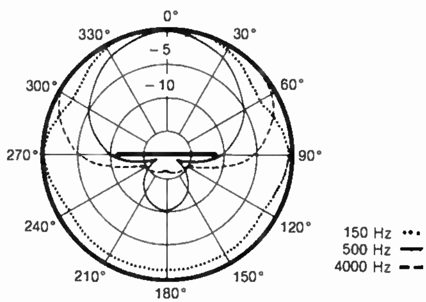


Fig. C-2. Polar response of a 2′-square boundary.

With multiple boundaries, the shape of the pickup pattern approximates the shape of the boundary assembly. For example, a V-shaped boundary produces a polar pattern with a lobe whose sides are defined by the sides of the "V." Note, however, that the polar pattern varies with frequency.

This "V"-shaped boundary works like a horn loudspeaker in reverse. Speaker horn theory applies to microphone horns. For instance, if you want a constant-directivity boundary horn, the horn must flare out like a well-designed loudspeaker horn.

Disadvantages of Boundaries

Boundaries must be large to be effective. Their size and weight makes them cumbersome to mount or hang. Large boundaries are also visually conspicuous, but this problem is reduced by using clear plastic.

Many users claim that the sound quality and flexibility of multiple-boundary PZMs outweigh the disadvantages. For those users who need a directional PZM but prefer not to use boundaries, Crown makes the model PCC-160, a supercardioid surface-mounted microphone. It uses a directional mike capsule, rather than boundaries, to make the microphone directional.

Summary

- Microphone sensitivity increases 6 dB for every boundary added at right angles to previous boundaries (less than 6 dB if not at right angles).
- For a flat panel, the frequency response shelves down 6 dB at and below the frequency F = 188/D, where D = the boundary dimension in feet. This shelf disappears if the sound source is at the side of the panel, or if the source is very close to the microphone (less than a panel dimension away).
- For a square panel, the frequency response rises about 10 dB above the low-frequency shelf at the frequency F = .88C/D, where C = the speed of sound (1130 ft/sec) and D = the boundary dimension in feet (or meters).
- The PZM/panel assembly is omnidirectional at and below the frequency F = 188/D, where D = the boundary dimension in feet. The panel becomes increasingly directional as frequency increases.
- Use the biggest boundaries that are not visually conspicuous. Big boundaries provide flatter response, better bass and more directionality than small boundaries.
- For flattest response with a single panel, place the PZM ⅓ of the way off-center (say 4″ off-center for a 2′ panel). For flattest response on multiple boundaries, place the tip of the PZM cantilever touching against the boundaries (leaving the usual gap under the mike capsule).
- To increase directionality and reach, increase boundary size or add more boundaries.

Construction Tips

You can obtain clear acrylic plastic (plexiglass) from a plastic vendor or plastic fabrication company listed in the Yellow Pages of major cities. Plastic ¼″ thick is recommended for good sound rejection. Many vendors can heat and shape the plastic according to your specifications. They use their own adhesives which are usually proprietary.

Cyanacrylate adhesive ("Supper Glue") or RTV ("Sealastic") have worked well in some instances. Or you can join several pieces of plastic with metal brackets, bolts and nuts.

If you intend to hang or "fly" the boundary assembly, drill holes in the plastic for tying nylon line. To prevent cracks in the plastic, start with small drill-bit sizes and work up. You may want to paint the boundary edges flat black to make them less visible.

When making a multiple-boundary assembly, be sure to mount the PZM mike capsule as close as possible to the junction of the boundaries. Let the tip of the cantilever touch the plastic, but leave the usual gap under the mike capsule.

Note: Some early PZMs include a small block of foam under the mike capsule for acoustical adjustment. If your PZM has this foam block, trap it under the mike capsule before screwing the PZM cantilever to the boundary.

The PZM models used for multi-boundary assembly are the PZM-6 series. . . . When drilling the holes for a PZM-6 series cantilever, make them 5/32″ diameter, .562″ center-to-center and countersunk .250″ × 90°.

Sub-Appendix: Frequency-Response Anomalies Caused by Boundaries

When sound waves strike a boundary, pressure-doubling occurs at the boundary surface, but does not occur outside the boundary. Thus, there is a pressure difference at the edge of the boundary. This pressure difference creates sound waves.

These sound waves generated at the edge of the boundary travel to the microphone in the center of the boundary. At low frequencies, these edge waves are opposite in polarity to the incoming sound waves. Consequently, the edge waves cancel the pressure-doubling effect.

Thus at low frequencies pressure-doubling does not occur; but at mid to high frequencies, pressure doubling does occur. The net effect is a mid-to-high frequency boost, or looked at another way, a low-frequency loss or shelf.

Incoming waves having wavelengths about six times the boundary dimensions are cancelled by edge effects; waves of wavelengths much smaller than the boundary dimensions are not cancelled by edge effects.

Waves having wavelengths on the order of the boundary dimensions are subject to varying interference vs. frequency (i.e., peaks and dips in the frequency response).

At the frequency where the wavelength equals the boundary dimension, the edge wave is in phase with the incoming wave. Consequently, there is a response rise (about 10 dB above the low-frequency shelf) at that frequency. Above that frequency, there is a series of peaks and dips that decrease in amplitude with frequency.

The edge-wave interference decreases if the incoming sound waves approach the boundary at an angle. Interference is also reduced by placing the mike capsule off-center. This randomizes the distances from the edges to the mike capsule, resulting in a smoother response.

Flat 2′-Square Panel

This boundary (Fig. C-3) is most often used for directional pickup of solo instruments, choirs, orchestras and bands. Two PZMs back-to-back on a panel form a "bipolar" PZM for coincident stereo miking. With this arrangement, the panel is usually hung just behind the conductor, about 14′ above the stage floor.

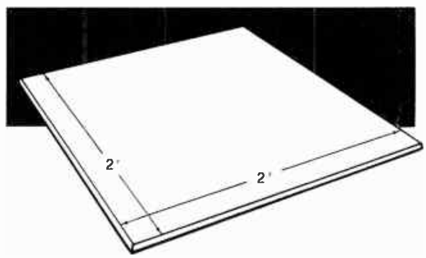


Fig. C-3. 2′-square flat panel.

For near-coincident stereo miking, place two panels with edges touching to form a "V." Aim the point of the "V" at the sound source. Mount a PZM about 4″ off-center on each panel, towards the point of the "V" for better stereo imaging. This assembly provides a higher direct-to-reverb ratio (a closer perspective) than the bipolar PZM mentioned above. It also rejects sounds approaching the rear of the panels.

Crown makes a model A240 Boundary, a 2′-square plexiglass panel with an adjustable microphone stand adapter. The panel includes a mike clip on either side of the panel and holes for suspending the panel. A 2′-diameter disk can be hung horizontally over choir sections. The disk is suspended by the mike cable as in Figure C-4.

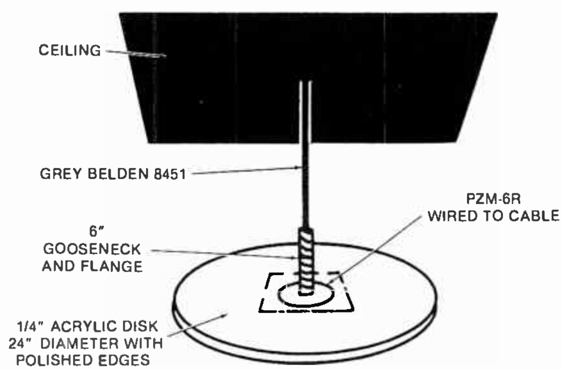


Fig. C-4. Circular boundary to be hung over choir sections.

The frequency response of a flat panel is the smoothest of all boundary assemblies in this booklet. For a 2′-square panel there is a 10 dB rise above the low-frequency shelf at 497 Hz for direct sound at normal incidence. F[−6] = 94 Hz. The polar pattern is omnidirectional at low frequencies, supercardioid at mid frequencies and hemispherical at high frequencies. Random Energy Efficiency = −3 dB at high frequencies. The assembly has 3 dB less reverb pickup than an omnidirectional microphone in open space at the same distance.

Distance factor = 1.41. That is, the microphone panel can be placed 1.41 times as far as the source as an omnidirectional microphone for the same direct-to-reverb ratio.

PZM-2

This model (Fig. C-5) uses two panels at right angles to each other. One of the panels is placed on a large, flat surface, such as a table or floor. One configuration uses a 1′ × 2′ vertical panel. When this vertical panel is placed on a horizontal surface, the vertical panel is "reflected" in the horizontal surface. The panel and its reflection appear to be a 2′ × 2′ panel with a 94 Hz shelving frequency.

Random Energy Efficiency = −6 dB. The assembly has 6 dB less reverb pickup than an omnidirectional mike in open space at the same distance.

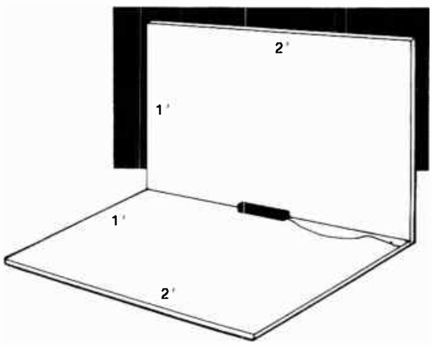


Fig. C-5. PZM-2.

This model (Fig. C-6) provides about 10 dB of forward gain at mid frequencies compared to a PZM on the floor. The assembly is placed on a large horizontal surface such as a stage floor. The 18″-tall unit works well for pickup of drama, musicals and opera. The taller units have been used to pick up cello, string bass or kick drum.

F[−6] = 160 Hz for 12″ tall model. Polar pattern (12″ model): see Figures C-7 and 8.

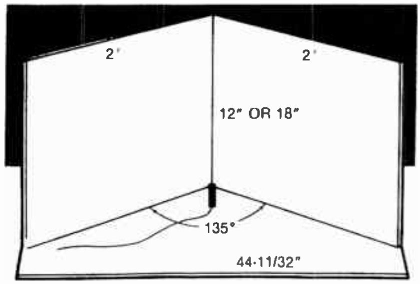


Fig. C-6. PZM-2.5.

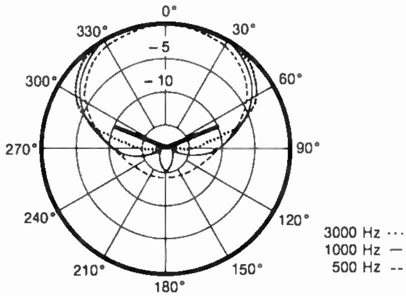


Fig. C-7. 12″ tall PZM-2.5: horizontal-plane polar response.

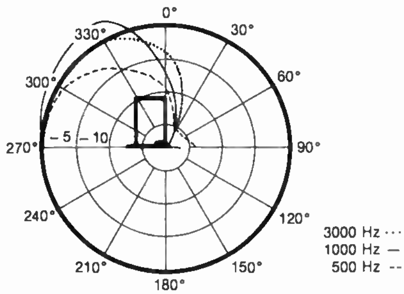


Fig. C-8. 12″ tall PZM-2.5: vertical-plane polar response.

PZM-3

This model (Fig. C-9) has a tighter polar pattern than the PZM-2.5, so it can be used to isolate soloists. Again, the assembly is placed on a large, horizontal surface such as a stage floor. F[−12] (two 1′-square panels on floor) = 94 Hz.

Random Energy Efficiency = −9 dB. The assembly has 9 dB less reverb pickup than an omnidirectional mike in open space at the same distance.

PZM Pyramid

This model can be made of three or four sides (Fig. C-10). It emphasizes mid frequencies and is recommended only for speech. Its highly directional pattern makes it useful for long-distance pickup of quarterback calls. Pyramids

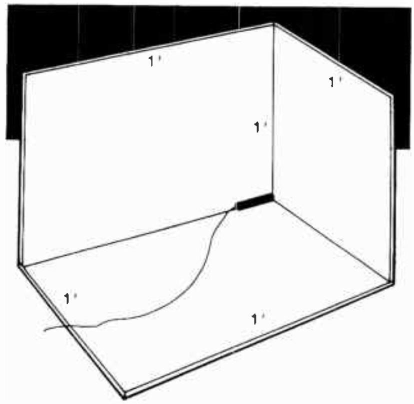


Fig. C-9. PZM-3

have also been hung over stages for pickup of rear-stage dialogue. Since a plexiglass pyramid can be quite heavy, you may want to make it out of sheet metal.

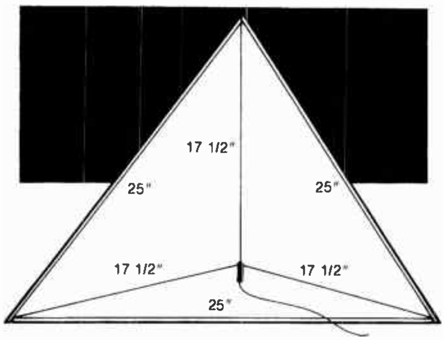


Fig. C-10. PZM pyramid.

PZM Dish

The PZM dish (Fig. C-11) has an uneven response on-axis, but is useful for its excellent directionality at mid-to-high frequencies. Dishes have been used over orchestral sections for isolation and for long-reach speech applications.

The dish is not a parabolic microphone. The PZM is placed on the dish, rather than at the focus of a parabolic surface. The dish obtains its directionality from diffraction (blocking sound waves from certain directions), while a parabolic microphone obtains its directionality by focusing sound energy from a particular direction on the mike capsule.

F[−12] = 250 Hz. Frequency response: see Figure C-12. Polar pattern: see Figure C-13.

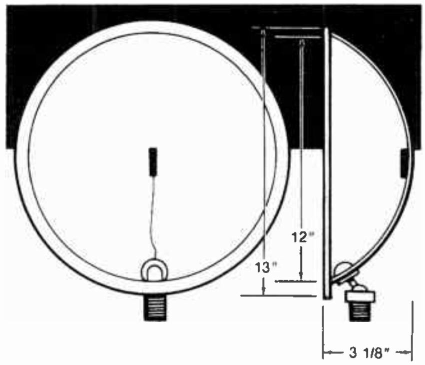


Fig. C-11. PZM dish.

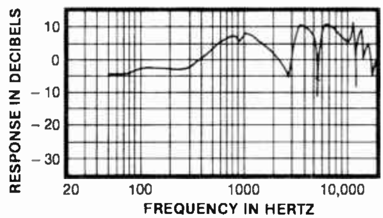


Fig. C-12. PZM dish frequency response.

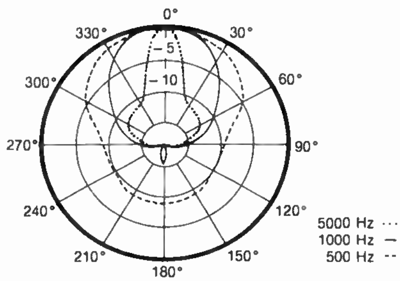


Fig. C-13. PZM dish polar response.

PZM Cone

This model (Fig. C-14) is highly directional and emphasizes mid frequencies. It has been used as a "follow" mike for a roving TV camera and provides a close-up audio perspective.

Random Energy Efficiency (for a cone with a 90° included angle) = −8.3 dB. The cone rejects reverb by 8.3 dB compared to an omnidirectional microphone in open space at the same distance. Distance Factor: 2.6. That is, the cone can be placed 2.6 times as far from the source as an omnidirectional mike can for the same direct/reverb ratio.

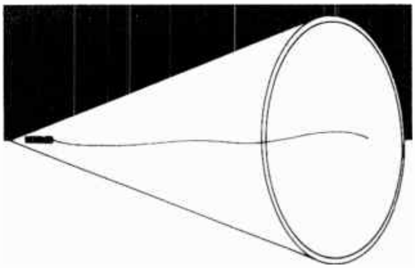


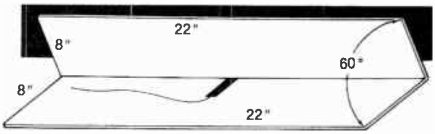
Fig. C-14. PZM cone (1′ long, 1′ diameter).

1560, 2260, 4060, 7260

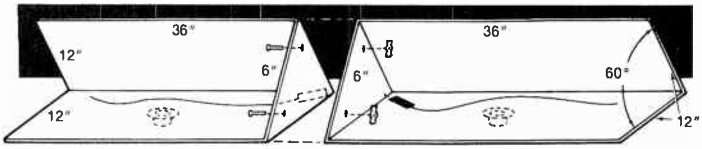
These models have the same basic shape (Fig. C-15), two panels angled 60° apart, but have different sizes. In general, the bigger the panels, the better the low-end response and the lower in frequency the directivity extends.

Fig. C-15.

(A) 2260.



(B) 7260.



The 1560, at one time made by Crown, is typically used on lecterns. Its response and polar patterns are shown in Figures C-16, -17, and -18. The 7260 has been used for stereo pickup of xylophones or brass sections. It is assembled in two halves for easier transport.

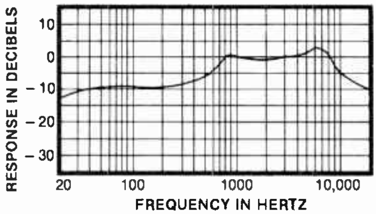


Fig. C-16. Frequency response of PZM-6FS on 1560 boundary.

1560 with Side Boundaries

This is a basic 1560 modified with two side boundaries at 45° on each side (Fig. C-19). The side boundaries provide additional discrimination of loudspeakers to either side of the lectern.

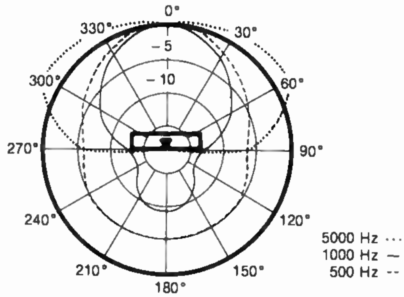


Fig. C-17. 1560 horizontal-plane polar response.

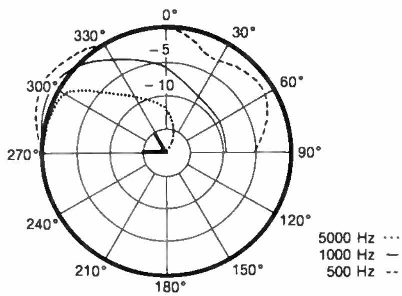


Fig. C-18. 1560 vertical-plane polar response.

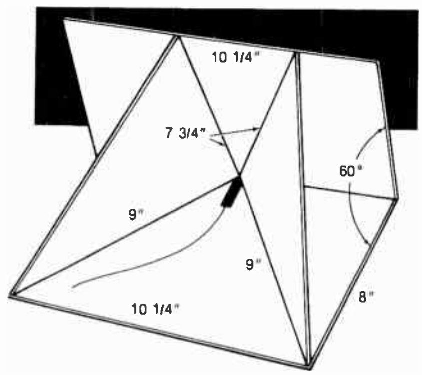


Fig. C-19. 1560 with side boundaries.

L² Array

This multi-purpose array (Fig. C-20) was designed by Mike Lamm of Dove and Note Recording in Houston, Texas. Mike has used this array extensively for overall stereo and quad pickup of large musical ensembles. The hinged, sliding panels can be adjusted to obtain almost any stereo pickup pattern. A complete description of the L² array is in AES preprint 2025 (C-9). "The Use of Boundary Layer Effect Microphones in Traditional Stereo Miking Techniques," presented at the 74th Convention of the Audio Engineering Society, October 1983. The frequency response is shown in Figure C-21 and the polar response is shown in Figures C-22 and -23.

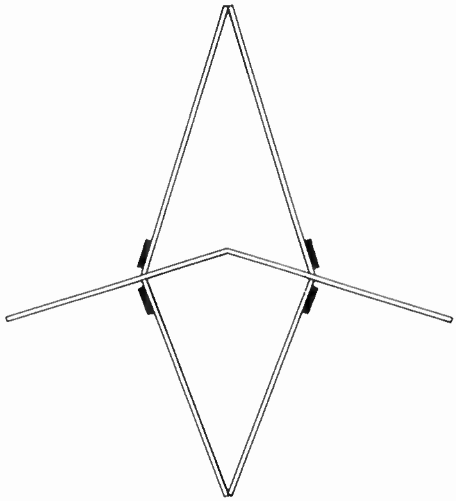


Fig. C-20. L^2 array.

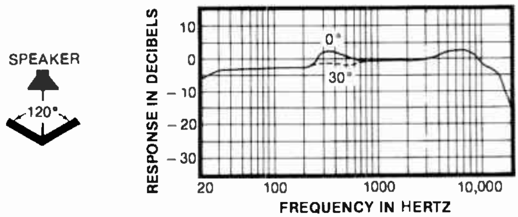


Fig. C-21. Frequency response of the L^2 array (with PZM-6FS capsule), 120° between boundaries.

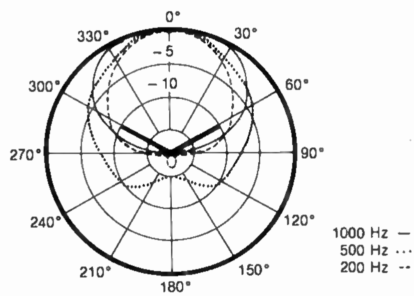


Fig. C-22. Polar response of L^2 array, 120° between boundaries.

L^2 Floor Array

Here's another PZM stereo array (Fig. C-24) designed by Mike Lamm and John Lehmann of Dove and Note Recording of Houston, Texas. It simulates the O.R.T.F. stereo mike technique. According to one user, "You can take this array, set it down and just roll. You get a very close approximation of the real event."

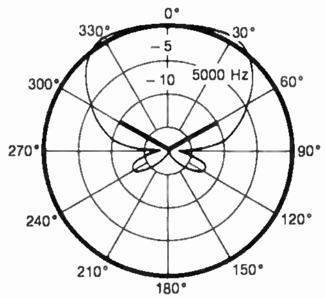


Fig. C-23. Polar response of the L² array, 120° between boundaries.

Suspending the inverted array results in less bass and more highs, while placing it on the floor reverses the balance. When this array is used on a stage floor, the construction shown in Figure C-25 is useful. It has decreased side pickup and increased pattern overlap. The axes of the left and right polar patterns may be any desired angle, just so the 120° boundary angle and 6.7″ capsule spacing are maintained.

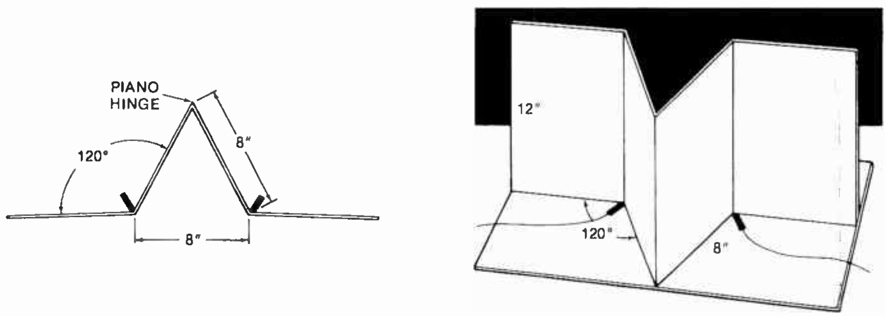


Fig. C-24. L² floor arrays.

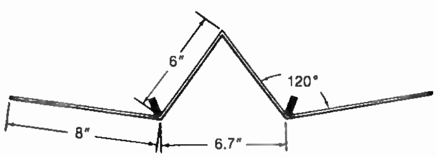


Fig. C-25. Another version of the L² floor array.

PZM Wedge or Axe

This stereo PZM array (Fig. C-26) has been used extensively by Mike Lamm and John Lehmann of Dove and Note Recording of Houston, Texas. It simulates the O.R.T.F. stereo microphone technique. Stereo imaging is precise and coverage is even.

Place the mike capsules 2½″ below the center of the panels to smooth the frequency response. To compensate for the bass shelving of the panels, boost the bass +6 dB at and below 141 Hz.

A panel containing a ⅝″-27 Atlas flange can be fastened to the bottom of the array for stand mounting. Aim the point of the wedge at the center of the sound source and raise the array about 14′ off the floor.

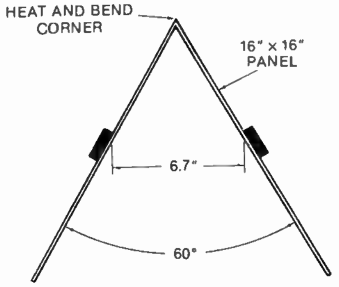


Fig. C-26. PZM wedge or axe (top view).

Pillon Stereo PZM Array

This stereo PZM array (Fig. C-27) was devised by Gary Pillon, a sound mixer at General Television Network of Detroit, Michigan. The assembly can be stand-mounted from the backside or hand-held if necessary.

The stereo image, which is partly a result of the 8″-capsule spacing, is designed to be like that produced by a binaural recording, but with more realistic playback over loudspeakers. Ideally, this device would mount on a Steadicam platform and give an excellent match between audio and video perspectives.

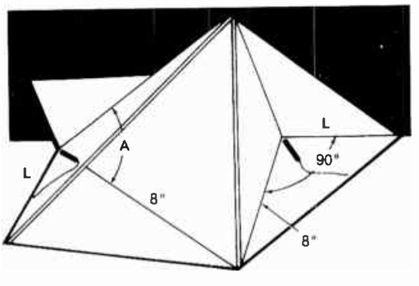


Fig. C-27. Pillon stereo PZM. L = 8″ and A = 90° for speech use (ENG) or L = 12″ and A = 120° for music use.

D Popular Professional Microphones

The following catalog outlines the technical specifications of many of the microphones used in professional audio production.

Shure SM57

The SM57 (Fig. D-1) is widely used by touring sound companies and artists' staff engineers for instrumental and remote recording applications. The SM57's mid-range presence peak and good low-frequency response make it an ideal microphone for use with vocals, snare drums, toms, bass drums, electric guitars, and keyboards.

Specifications

Transducer type:	moving-coil dynamic
Polar response:	cardioid
Frequency response:	40–15,000 Hz
Sensitivity:	−75.5 dB (0 dB = 1 V/microbar)

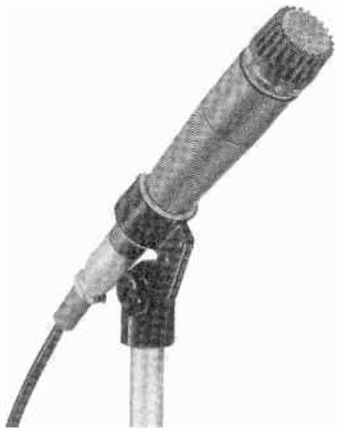
Shure SM63L

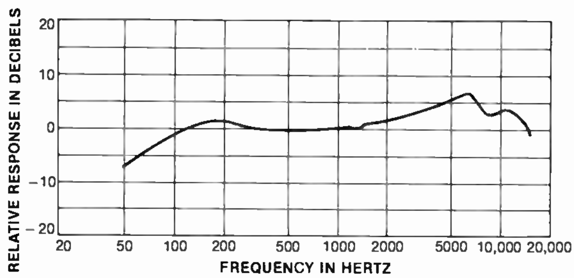
The SM63L (Fig. D-2) is particularly well suited to hand-held vocal and ENG applications. The smooth, extended frequency response provides a clear, crisp sound and a low frequency roll-off gives a natural sounding pickup without the presence of any "boominess."

Specifications

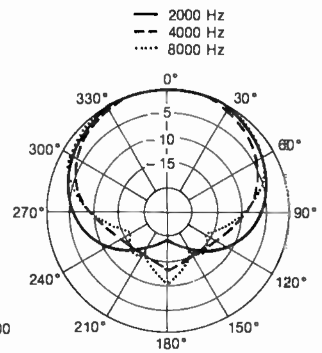
Transducer type:	moving coil dynamic
Polar response:	omnidirectional
Frequency response:	50–20,000 Hz
Sensitivity:	−76 dB (0 dB = 1 V/microbar)

Fig. D-1. Shure SM57. *(Courtesy of Shure Brothers, Inc.)*



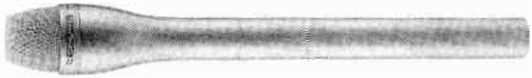


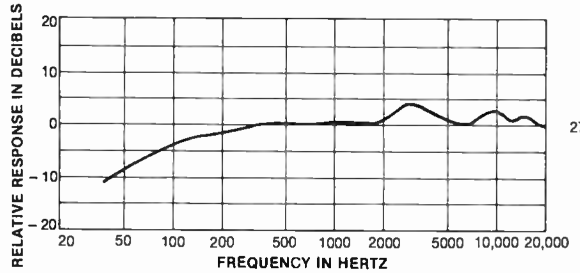
Typical frequency response



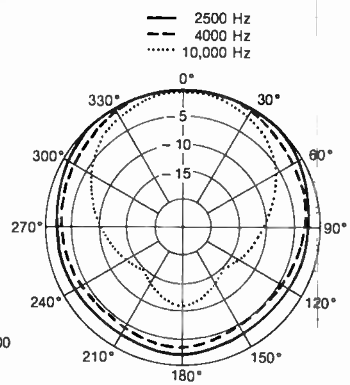
Typical polar pattern

Fig. D-2. Shure SM63L. *(Courtesy of Shure Brothers, Inc.)*





Typical frequency response



Typical polar pattern

Electro-Voice RE20

Specifications

Transducer type:	moving coil dynamic
Polar response:	cardioid
Frequency response:	45–18,000 Hz
Sensitivity:	−57 dB (150-Ω output)

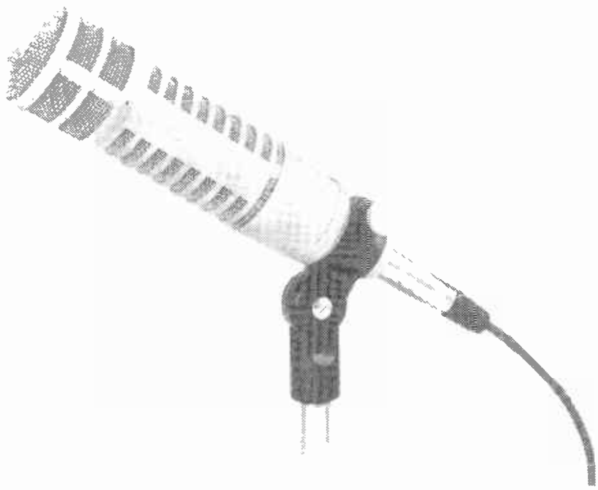
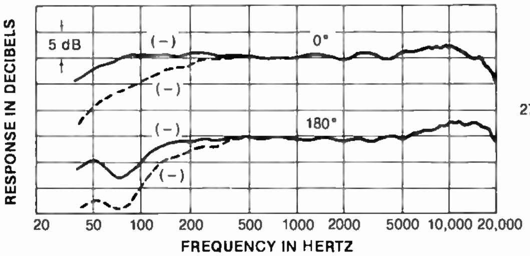
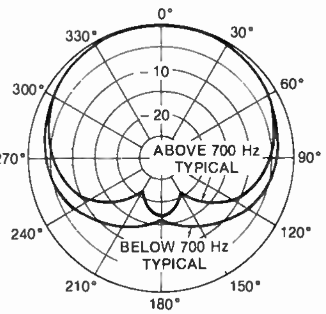


Fig. D-3. The Electro-Voice RE20. *(Courtesy of Electro-Voice, Inc.)*



Typical frequency response



Typical polar pattern

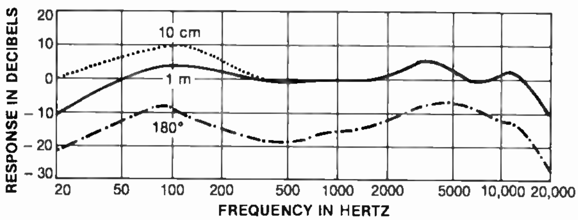
AKG D-112

The AKG D-112 (Fig. D-4) is the direct successor to the D-12E bass microphone. A new type of dynamic transducer enables the extremely punchy and powerful bass sounds to be accurately reproduced up to incredibly high sound pressure levels (a theoretical 168 dB). An additional design philosophy was to make a microphone that delivers a good bass sound without additional processing. This was attained by giving the D-112 a presence lift at 4 kHz for an extra punchy sound and sparkle.

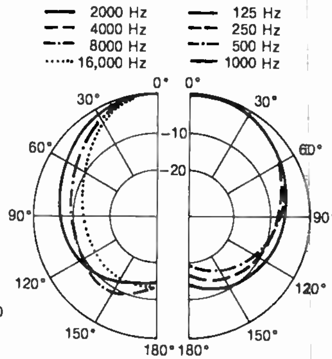
Specifications

Transducer type:	moving-coil dynamic
Polar response:	cardioid
Frequency response:	30–17,000 Hz
Sensitivity:	1.8 mV/Pa
Impedance:	greater than 600 Ω
Maximum SPL rating:	unmeasurable (calculated at 168 dB)

Fig. D-4. AKG D-112 Microphone. *(Courtesy of AKG Acoustics, Inc.)*



Typical frequency response



Typical polar pattern

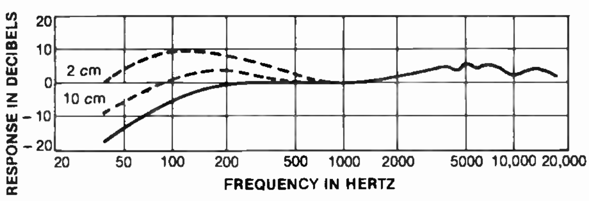
Beyer Dynamic M 69

For high-quality music and vocal productions. Wide frequency response and unusually high sensitivity. Very low feedback. Flexibly usable as instrumentalist's microphone, in electro-acoustical installations (sound reinforcement for churches and halls) and for outside broadcasts. Universal microphone for the demanding nonprofessional (Fig. D-5).

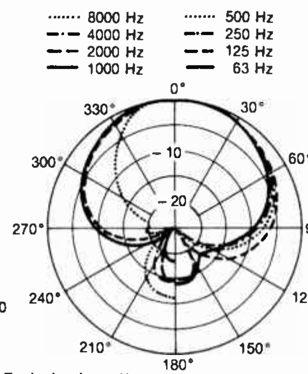
Specifications

Transducer type:	moving-coil dynamic
Polar response:	hypercardioid
Frequency response:	50–16,000 Hz
Sensitivity:	−52 dB (0 dB = 1 mW/Pa)
Equivalent input noise:	−145 dB (0 dB = 1 mW/2.10[−5] Pa)
Output impedance:	200 Ω

Fig. D-5. Beyer Dynamic M 69. *(Courtesy of Beyer Dynamic, Inc.)*



Typical frequency response



Typical polar pattern

Sennheiser MD 421

Specifications

Transducer type:	moving-coil dynamic
Polar response:	cardioid
Frequency response:	30–17,000 Hz
Sensitivity:	−54 dB ±3 dB re. 1 V/microbar

Sennheiser MD 441

Specifications

Transducer type:	moving-coil dynamic
Polar response:	supercardioid
Frequency response:	40–20,000 Hz
Sensitivity:	−53 dBm 2mV/Pa. at 1 kHz

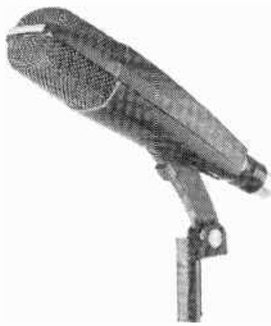
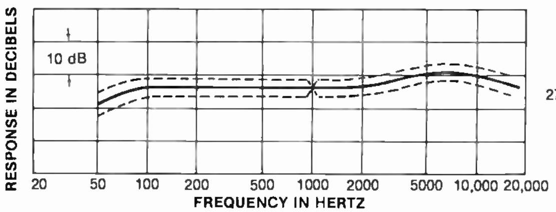
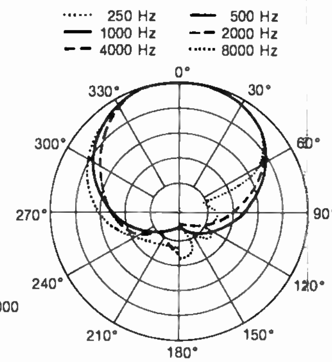


Fig. D-6. Sennheiser MD 421. *(Courtesy of Sennheiser Electronic Corporation [N.Y.])*



Typical frequency response



Typical polar pattern

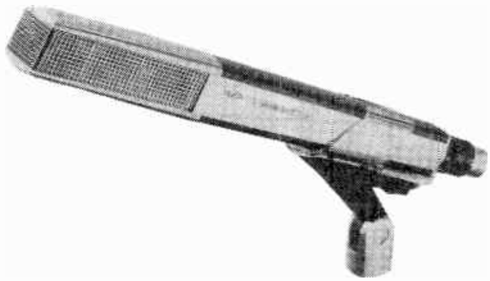
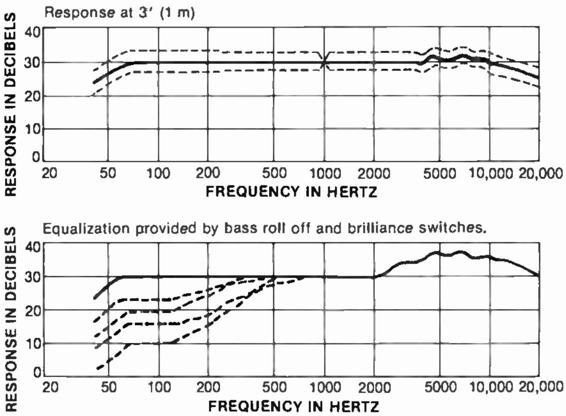
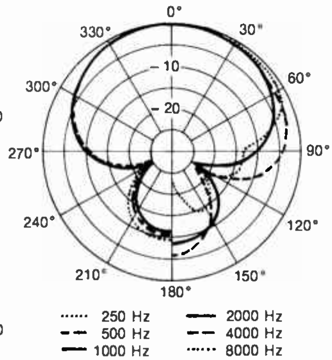


Fig. D-7. Sennheiser MD-441. *(Courtesy of Sennheiser Electronic Corporation [N.Y.])*



Typical frequency response



Typical polar pattern

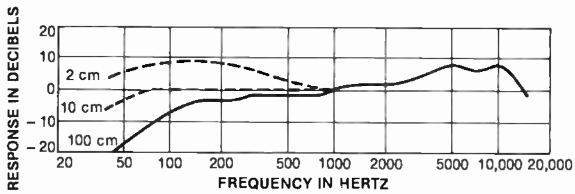
Beyer Dynamic M 500

Widely used, high-quality ribbon microphone for vocal soloists and announcers (Fig. D-8). Rising frequency-response curve for optimum voice reproduction and vocal presence. Extremely low feedback. Built-in pop screen.

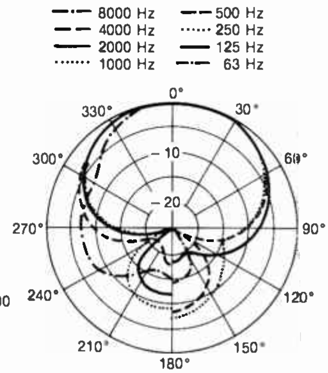
Specifications

Transducer type:	ribbon dynamic
Polar response:	hypercardioid
Frequency response:	40–18,000 Hz
Equivalent input noise:	−150 dB (0 dB = 1 mW/2.10[−5] Pa)
Output impedance:	200 Ω

Fig. D-8. Beyer M 500. *(Courtesy of Beyer Dynamic, Inc.)*



Typical frequency response



Typical polar pattern

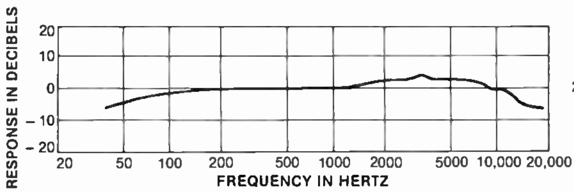
Beyer M 160

This double-ribbon microphone (Fig. D-9) excels through the transparency inherent in ribbon microphones. Very wide frequency response and extremely low feedback. Suited for string instruments and, particularly, piano.

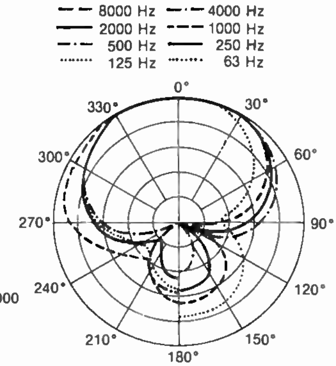
Specifications

Transducer type:	ribbon dynamic
Polar response:	hypercardioid
Frequency response:	40–18,000 Hz
Sensitivity:	−52 dB (0 dB = 1 mW/Pa)
Equivalent input noise:	−145 dB (0 dB = 1 mW/2.10[−5] Pa)
Output impedance:	200 Ω

Fig. D-9. Beyer M 160. *(Courtesy of Beyer Dynamic, Inc.)*



Typical frequency response



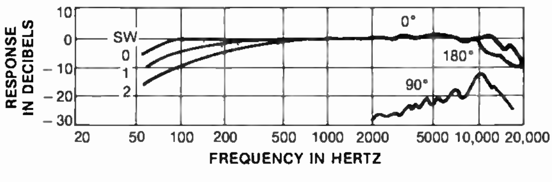
Typical polar pattern

Fostex M88 RP

Specifications

Transducer type:	printed ribbon
Polar response:	bidirectional
Frequency response:	40–18,000 Hz
Sensitivity:	–56 dB, –1.6 mV/Pa (0 dB = 1 V/Pa)
Output impedance:	250 Ω

Fig. D-10. Fostex M88 RP printed ribbon microphone. *(Courtesy of Fostex Corporation)*

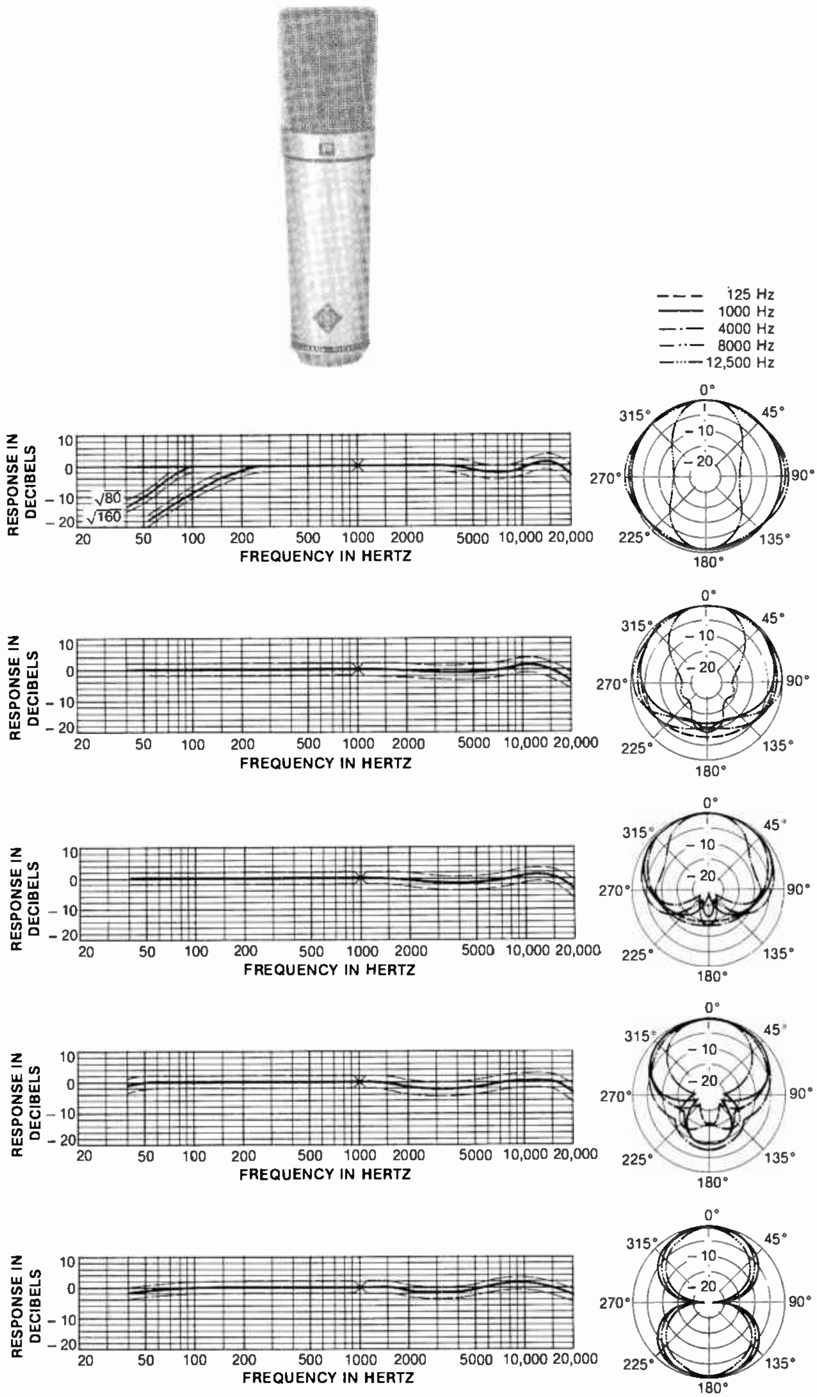


Neumann U-89 i

The U-89 i microphone (Fig. D-11), similar in shape but smaller than the U-87 i, is a studio microphone with switchable directional patterns. Its grille houses a dual-membrane capsule with a particularly linear frequency response for all polar patterns. The amplifier allows sound pressure levels of up to 134 dB to be reproduced without distortion (140 dB with −6 dB pad). A high-pass filter inserted ahead of the output transformer provides a roll-off in sensitivity at either 80 Hz or 160 Hz.

Specifications

Transducer type:	condenser
Polar response:	omni/wide-angle cardioid/cardioid/ hypercardioid/figure-8
Frequency response:	40−18,000 Hz
Sensitivity:	8 mV/Pa ±1 dB
Self-noise:	24 dB
Maximum SPL for 0.5% THD at 1 kHz:	134 dB/100 Pa



Typical frequency responses

Typical polar patterns

Fig. D-11.
Neumann U-89 i.
(Courtesy of Gotham Audio Corp.)

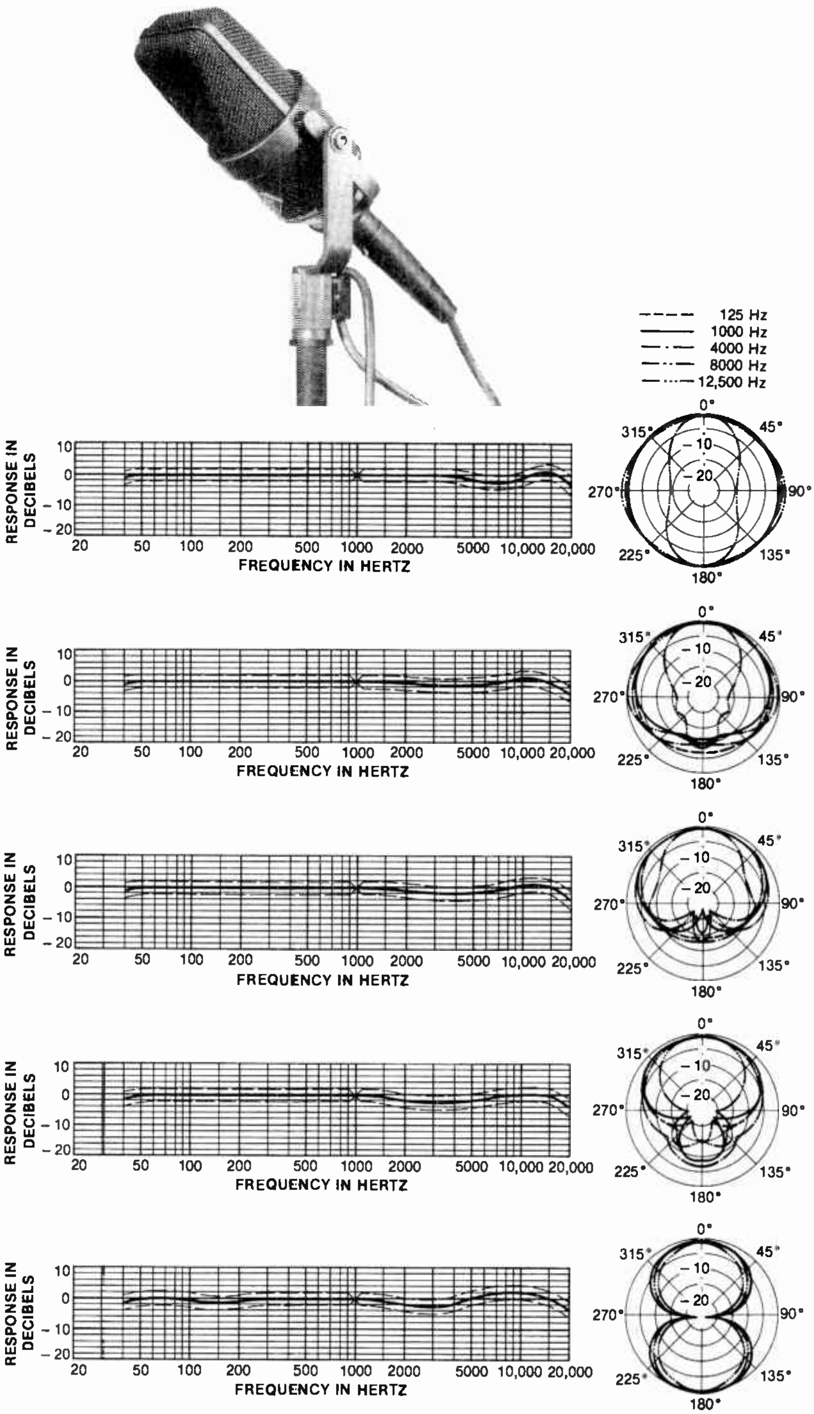
Neumann TLM-170 i

The TLM-170 i condenser microphone (Fig. D-12) has no transformer. Its direct, balanced output signal is achieved through the use of a completely new kind of electronic circuit, yet it maintains a high degree of freedom from interference and consumes little current. This mike may be powered identically by either a 48-V or 24-V phantom power supply. The TLM-170 i is equipped with a tiltable, elastically suspended mounting bracket, which effectively isolates the microphone against mechanical noise interference.

Specifications

Transducer type:	condenser
Polar response:	omni/wide angle cardiod/cardiod/ hypercardioid/figure-8
Frequency response:	40–18,000 Hz
Sensitivity:	8 mV/Pa
Self-noise:	21 dB (DIN 45 405)
Maximum SPL for 0.5% THD at 1 kHz:	150 dB

Fig. D-12.
Neumann
TLM-170 i.
(Courtesy of Gotham Audio Corp.)



Typical frequency responses *Typical polar patterns*

AKG C-414B/ULS

The C-414B/ULS (Fig. D-13) is a FET condenser microphone with a large-diaphragm capsule, providing four selectable patterns, with a three-position bass roll-off switch (flat, 75 Hz, 150 Hz) and 0, −10 dB, and −20 dB attenuator switch. For 9−52 V low-current drain (1 mA) phantom powering. Complete with W-26 windscreen, SA-18/3 clamp stand adapter, and case.

Specifications

Transducer type:	condenser
Polar response:	omni/cardioid/hypercardioid/figure-8
Frequency response:	20−20,000 Hz
Sensitivity at 1 kHz:	9 mV/Pa, −40 dBV

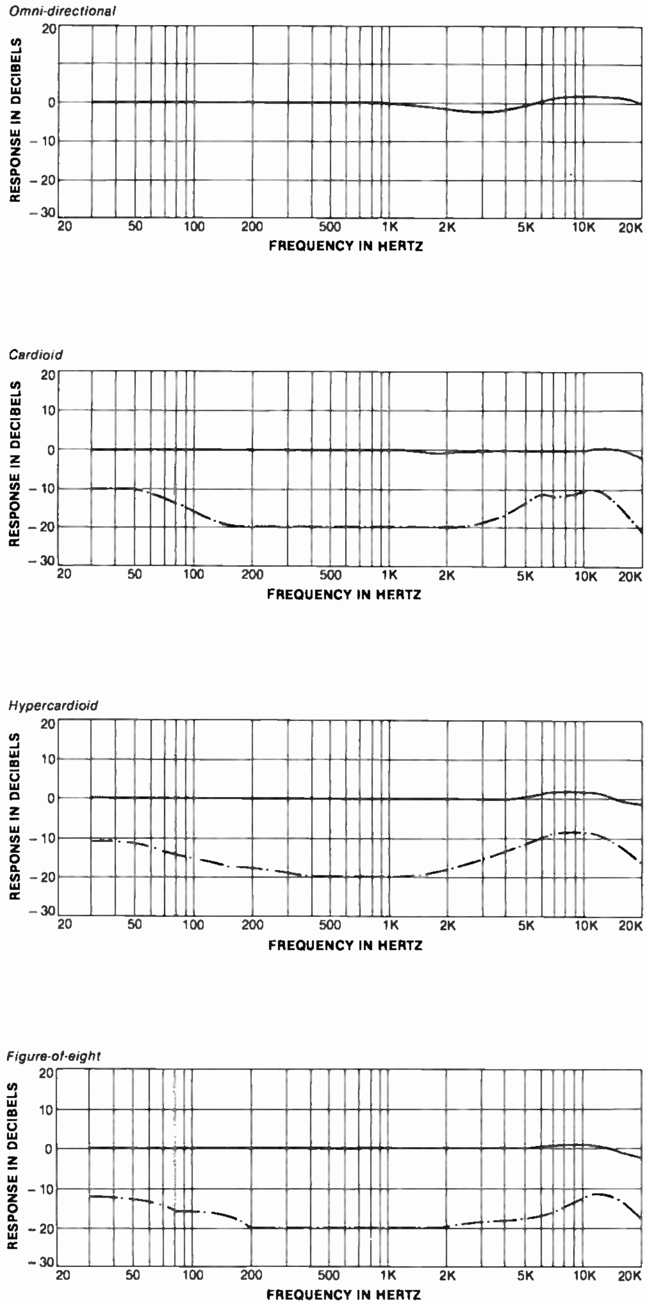
AKG C-460B

The AKG C-460B (Fig. D-14) is a 12−52 V cardioid-condenser microphone combination. With switchable bass roll-off in four positions: flat, 70 Hz, 150 Hz, and 10 dB attenuation. Consists of C-460B preamplifier and CK-61 cardioid capsule, satin black finish, W-32 foam windscreen, SA-40 stand adapter, and foam-lined case.

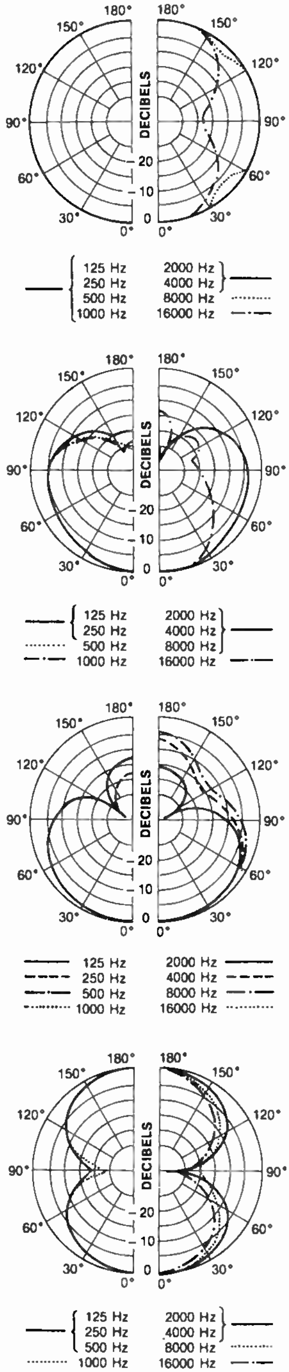
Specifications

Transducer type:	condenser
Polar response:	CK-1 cardioid, CK-2 omni, CK-3 hypercardioid
Frequency response:	CK-1,2 20−20,000 Hz
	CK-3 20−16,000 Hz
Maximum SPL rating:	140 dB SPL

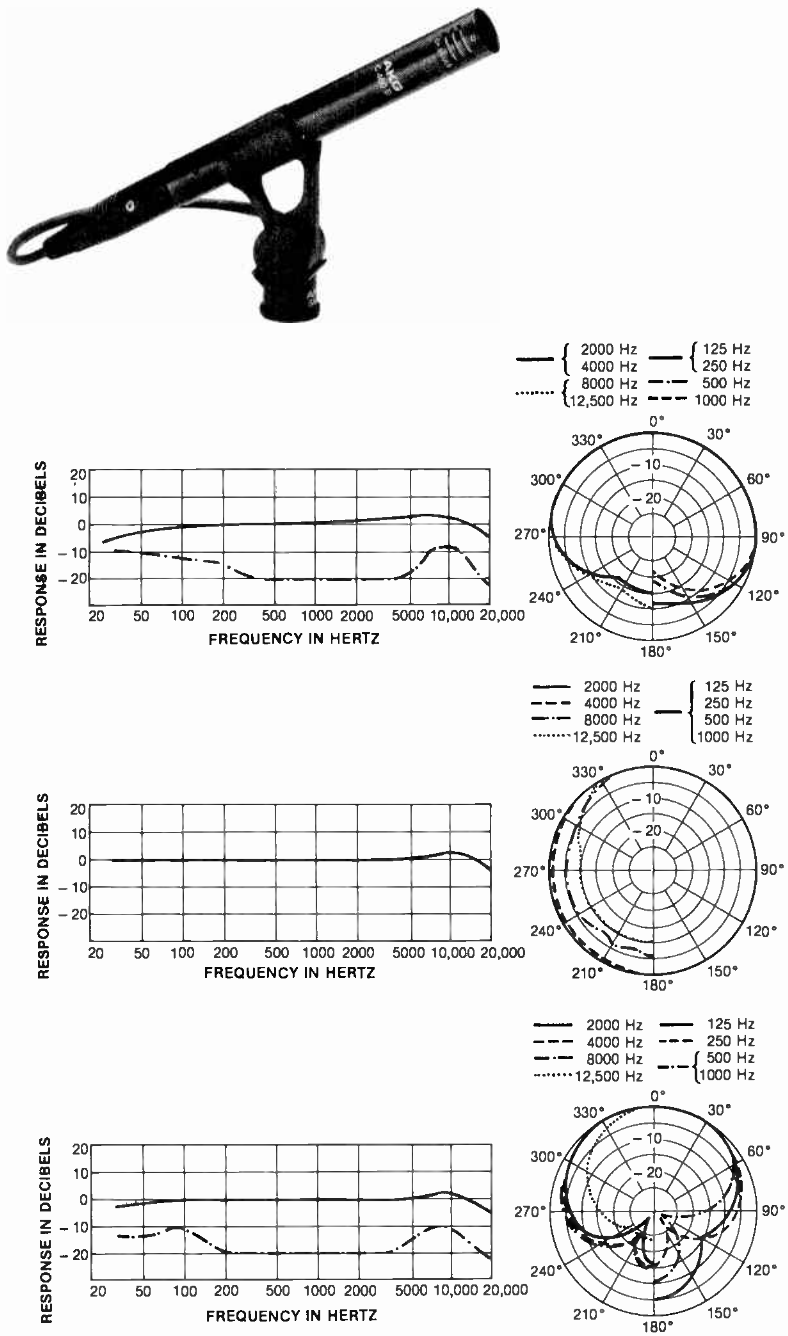
Fig. D-13. AKG C-414B/ULS. *(Courtesy of AKG Acoustics, Inc.)*



Typical frequency responses



Typical polar patterns



Typical frequency responses

Typical polar patterns

Fig. D-14. AKG C-460B. *(Courtesy of AKG Acoustics, Inc.)*

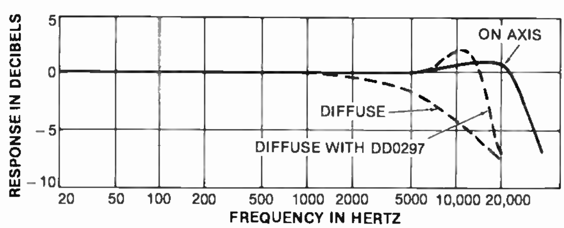
Bruel & Kjaer Types 4003, 4006

These microphones are specifically designed for low-level applications and are ideal for general recording of soloists and ensembles. For use under predominantly reverberant conditions, the microphones are supplied with an additional protective grille that gives a linear diffuse-field response up to 15 kHz by boosting the on-axis response approximately 5 dB in the 10–12-kHz range.

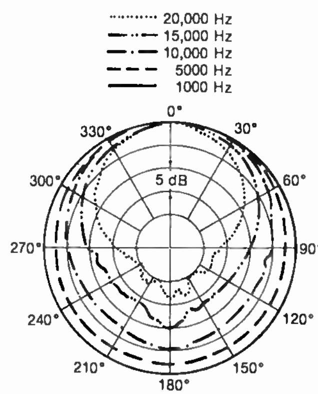
Specifications

Transducer type:	condenser
Polar response:	omnidirectional
Frequency response:	20–20,000 Hz
Sensitivity at 250 Hz:	4003 50 mV/Pa, 4006 12.5 mV/Pa
Self-noise:	typically 15 dB(A)
Maximum SPL rating:	4003 154 dB SPL, 4006 143 dB SPL

Fig. D-15. Bruel & Kjaer Types 4003 and 4006. *(Courtesy of Bruel & Kjaer Instruments, Inc.)*



Typical frequency response



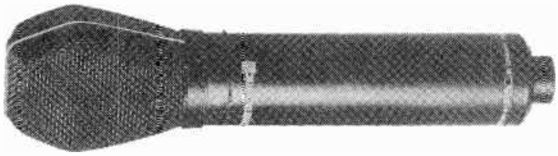
Typical polar pattern

Beyer Dynamic MC 740 N(C) P48

Studio-quality condenser microphone, shown in Figure D-16, with frequency-independent, switch-selectable directional pattern: omni, wide cardioid, cardioid, hypercardioid, or figure-8. Uniform frequency response is independent of polar pattern. For 48-V phantom powering. Switch-controlled preattenuation of 10 dB for high sound pressure levels. Available in the standard 3-pin XLR version and also in a special 5-pin XLR version for remote control of directional pattern from the MSG-740 power pack.

Specifications

Transducer type:	condenser
Polar response:	omni/wide cardioid/cardioid/hypercardioid
Frequency response:	40–20,000 Hz
Sensitivity:	10 mV/Pa
Self-noise:	approx. 17 dB
Maximum SPL rating:	134 dB, 144 dB (pad)



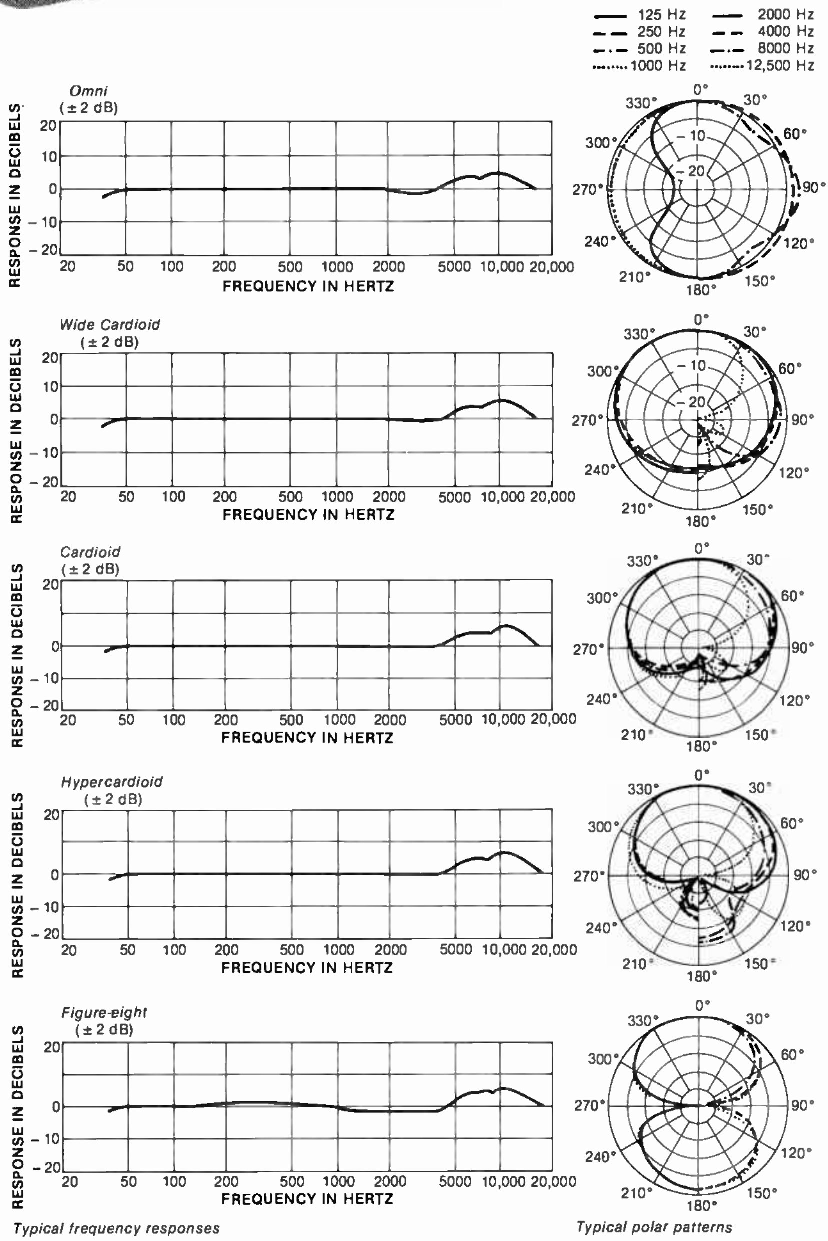


Fig. D-16. Beyer Dynamic MC 740 N(C) P48. *(Courtesy of Beyer Dynamic, Inc.)*

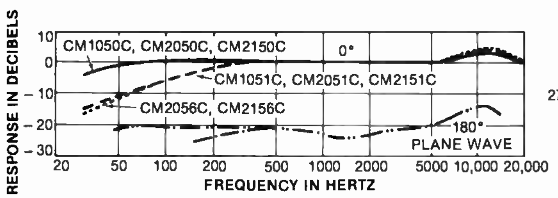
Calrec 1050

The diaphragm of this economical, high-quality condenser microphone (Fig. D-17) is made of a light, aluminum-coated polyester, producing a very smooth, virtually flat frequency response and an excellent transient response. These reliable microphones are immune to heat, dampness, and mechanical shock.

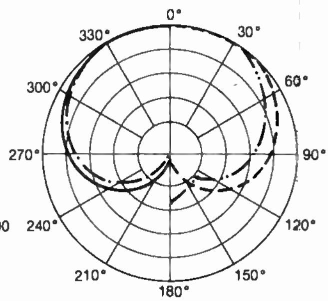
Specifications

Transducer type:	condenser
Polar response:	cardioid
Frequency response:	30–20,000 Hz
Sensitivity:	0.8 mV/microbar
Self-noise:	approx. 17 dB(A)
Maximum SPL rating:	130 dB

Fig. D-17. Calrec 1050. *(Courtesy of Calrec by Advanced Music Systems)*



Typical frequency response



Typical polar pattern

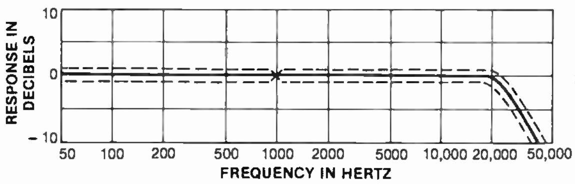
Sanken CU-41

Suited for a great variety of applications in recording, broadcasting, video, and film, the double-capsule Sanken CU-41 (Fig. D-18) delivers a transparency and distortion-free performance, with a flat response ±1 dB from 20–20,000 kHz.

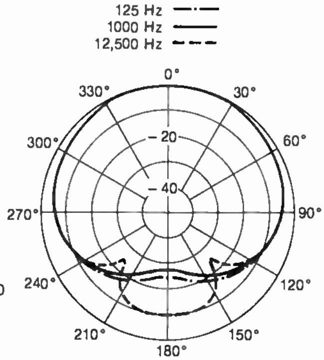
Specifications

Transducer type:	double-capsule condenser
Polar response:	cardioid
Frequency response:	20–20,000 kHz
Sensitivity at 1 kHz:	7 mV/Pa
Self-noise:	15 dB
Maximum SPL rating:	134 dB

Fig. D-18. Sanken CU-41. *(Courtesy of Sanken/Pan Communications, Inc.)*



Typical frequency response



Typical polar pattern

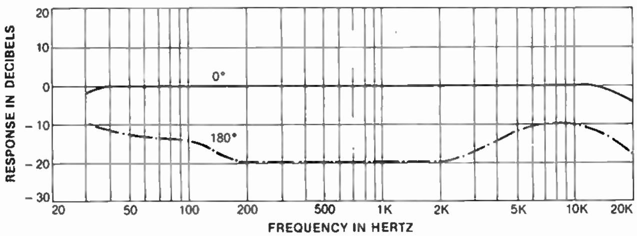
AKG Tube

The AKG Tube (Fig. D-19) is a large-diaphragm condenser that uses the low-noise 6072 vacuum tube, combined with new circuit design, providing the classic "tube" sound. Nine polar patterns and three bass roll-off positions (flat, 75 Hz, and 150 Hz) can be selected via remote control. 0, −10 and −20 dB preattenuator.

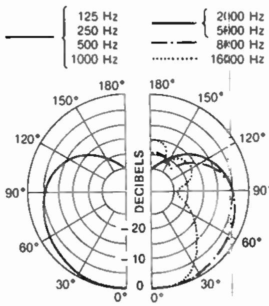
Specifications

Transducer type:	condenser
Polar response:	variable
Frequency response:	30−20,000 Hz
Sensitivity at 1 kHz:	10 mV/Pa
Self-noise:	25 dB
Maximum SPL rating:	128 dB SPL

Fig. D-19. AKG Tube. *(Courtesy of AKG Acoustics, Inc.)*



Typical frequency response



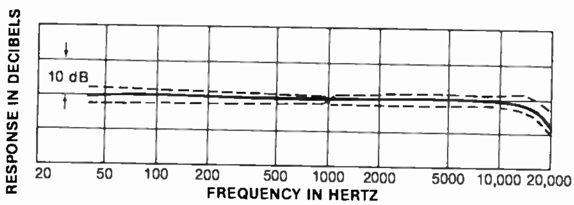
Typical polar pattern

Sennheiser MKH 40 P48

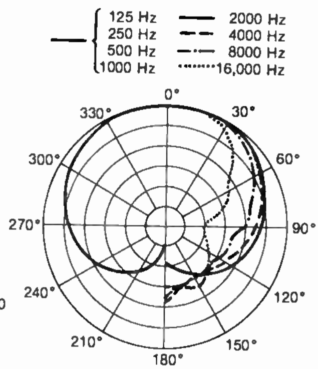
Specifications

Transducer type:	condenser
Polar response:	cardioid
Frequency response:	40–20,000 Hz
Sensitivity:	25 mV/Pa ±1 dB
Self-noise:	12 dBA
Maximum SPL rating:	134 dB SPL

Fig. D-20. Sennheiser MKH 40 P48. *(Courtesy of Sennheiser Electronic Corporation [N.Y.])*



Typical frequency response



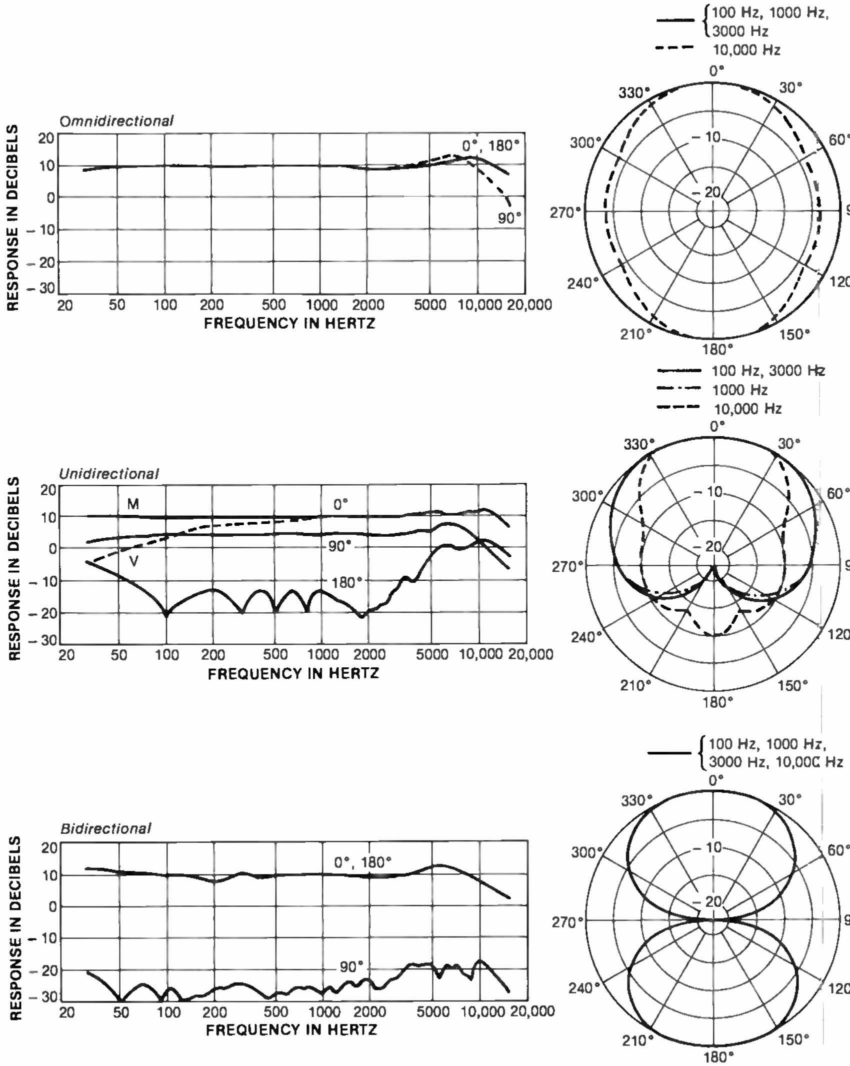
Typical polar pattern

Sony C-48

Specifications

Transducer type:	condenser
Polar response:	variable
Frequency response:	30–16,000 Hz
Maximum SPL rating:	126 dB SPL

Fig. D-21. Sony C-48. *(Courtesy of Sony Corporation of America)*



Typical frequency responses

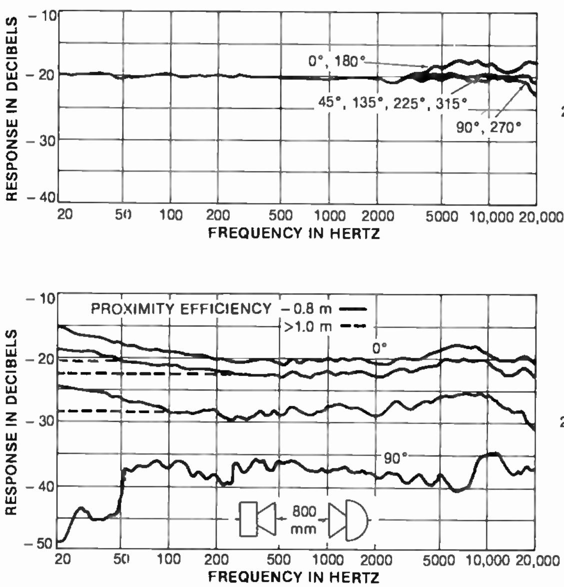
Typical polar patterns

Pearl TL-4

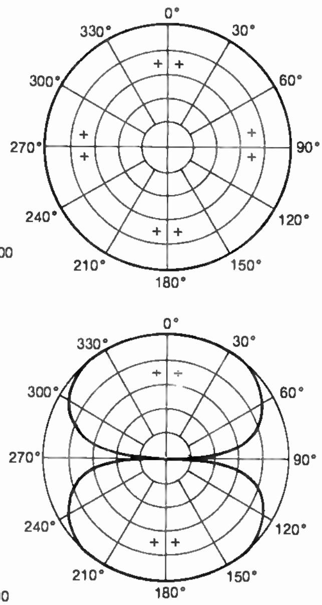
Specifications

Transducer type:	condenser
Polar response:	variable
Frequency response:	16–20,000 Hz
Sensitivity:	120 mV/Pa
Self-noise:	less than 20 dBA
Maximum SPL rating:	126 dB SPL

Fig. D-22. Pearl TL-4. *(Courtesy of Karlberg Enterprises)*



Typical frequency response



Typical polar pattern

E Microphone Terms

A-Weighting Network: An electronic filter network designed to accentuate or attenuate signal levels at certain frequencies so that the response of a system corresponds to the results of subjective ear-sensitivity tests.

Absorption: The ability of a surface to absorb sound. The *absorption coefficient* of a material is a figure between 0 and 1, representing its degree of absorption.

Acoustics: The study of sound-wave motion and sound behavior both within an enclosure and out of doors. Acoustics also includes the study of various materials and their effect on the character of sound.

Ambience: Reverberation and early reflections. The characteristic sound of a location that tells the ear it is listening in a particular room, concert hall, etc.

Ambient Noise: The noise associated with a location in the absence of foreign excitation.

Amplitude: The strength or level of an acoustic waveform (dB SPL) or electrical signal (voltage or wattage) measured either at a particular instant or as an average over time.

Anechoic Chamber: A room in which the walls effectively absorb all direct sound, creating a free-field condition for measurement purposes. A room in which there is little reverberant sound is said to be acoustically "dead."

Attenuation: The process of reducing an acoustical or electrical signal level by a prescribed amount (e.g., −10 dB, −20 dB).

Axis: An imaginary line that is perpendicular to the front face of the microphone diaphragm. Those sounds reaching the microphone at an angle to the axis are said to be off-axis.

Bidirectional Polar Pattern: A pure pressure-gradient microphone designed to accept signals arriving from in front (0°) and rear (180°) of the diaphragm(s) and to reject signals arriving at ±90° off-axis.

Capsule: The transducer element that transforms acoustic energy into a corresponding electrical waveform.

Carbon Microphone: A microphone that operates by directly varying applied pressure to carbon granules (by a diaphragm/carbon-button arrangement). In this manner, the carbon acts as a variable resistor that modulates an applied dc source.

Cardioid Polar Pattern: A polar pattern made of equal pressure and pressure-gradient characteristics. Such a pattern displays a heart-shaped response. It is sensitive to those signals arriving from the front and rejects signals arriving from 180° off-axis.

Ceramic Microphone: A piezoelectric microphone in which a diaphragm is directly coupled to a crystal (barium titinate). When deformed by the motion of the diaphragm, the crystal produces a voltage across its surface, thus providing an output signal.

Clip Microphone: A small microphone (usually an electret condenser) that can be attached directly to the sound source (e.g., on the tie of the lecturer or the bridge of a violin).

Close Miking: Placement of a microphone near a sound source (1″ (25 mm) – 3′ (1 m)), effectively eliminating all but the direct sound of a source.

Coincident Microphone: A microphone containing two or more microphone capsules mounted in close proximity, on a common vertical axis, but angled apart.

Coloration: Nonuniformity in frequency response, resulting in a distortion of the tonal quality of the source.

Condenser Microphone: An electrostatic microphone, in which a static charge is built up across two capacitive plates (diaphragm and backplate). When powered by an external polarizing voltage or internal static charge, a corresponding output voltage is created in relation to the changes in capacitance across these plates (in direct relation to the acoustic forces acting upon the diaphragm). The high-impedance signal output is fed directly to a preamplifier, where the impedance is reduced and the output level is adjusted to a standard operating value.

Crosstalk: Any leakage of signals between two channels.

dB: *See* Decibel.

dBm: Signal level relative to 1 milliwatt (1 mW), measured in dB.

dB SPL: Acoustic level relative to 0.0002 microbar (one-millionth of normal atmospheric pressure) or the softest sound that an average individual can hear 50% of the time.

dBV: Signal level relative to one volt.

Decibel (dB): A unit of measurement of changes in sound intensity or ratios of electrical quantities. The human ear hears in a logarithmic fashion and the decibel complements this scale, for easily handled measurements. The decibel is defined as 20 times the common logarithm to the base 10 of a ratio of linear quantities and 10 times the common logarithm to the base 10 of a ratio of squared quantities.

Example

the voltage E and SPL are linear quantities

dB = 20 log (E1/E2)

dBspl = 20 log (SPL1/SPL2)

the power P is a squared quantity

dB = 10 log (P1/P2)

Diaphragm: The moving element of a microphone that converts sound-wave energy into mechanical energy.

Diffuse Field: A sound field in which the sound pressure level is the same everywhere and the flow of energy is equally probable in all directions.

Directional Microphone: The measured response of a microphone to sounds arriving from various angles. The sensitivity is plotted as a function of the angle of incidence at various angles. *See* Polar Pattern.

Direct Sound: The sound that arrives directly at the reception point (without reflections). Sound lacking in any reverberation.

Distant Miking: Placement of a microphone at a distance (3′–5′ [1–1.5 m] or greater) from the source, to pick up a larger portion of the overall instrument and/or reflected sound.

Dynamic Microphone: A microphone that produces an electrical signal when sound causes a conductor (coil or ribbon) to vibrate in a magnetic field.

Dynamic Range: The range between the quietest level and loudest level an instrument can produce. For a microphone or measuring system, dynamic range is normally specified as the range between the inherent noise level and the sound pressure level causing a specified amount of distortion.

Echo: One or several distinct repetitions of a sound.

Electret-Condenser Microphone: A condenser microphone in which the required polarizing charge is permanently fixed in an electret material in one of the capacitive plates. This eliminates the need for an external polarizing voltage; however, such microphones often require an internal battery or phantom power for powering an internal FET preamplifier.

Equalization (EQ): The modifying of the balance of frequencies within a sound. Electronic alteration of frequency response.

Feedback: An electrical or audible signal that is fed from a system's or a device's output back into its input, repeatedly increasing in gain until a maximum distortion level is reached. Feedback is most likely to occur when a microphone is placed too close to a PA loudspeaker.

Flat: A movable frame or set of frames serving as an acoustic barrier. Flats are used to surround a sound source in order to reduce leakage within a multi-microphone recording setup. Also known as a *gobo*. Also, a system with *flat* frequency response exhibits equal response at all frequencies; its graph of level vs. frequency is a horizontal line (flat).

Free Field: A sound field in which there are no reflecting or obstructing boundaries. The sound field consists of uniformly progressing plane waves.

Frequency: The number of complete cycles of sound pressure or voltage oscillation in one second, measured in hertz (Hz). When the frequency is high enough, frequency is expressed in kHz (thousand hertz).

Frequency Response: The range between the upper and lower limits that a microphone or audio system will adequately transmit (within a range, such as ±3 dB).

Front-to-Back Discrimination: In a directional microphone, the ratio of the signal produced by a sound on-axis, compared to that produced by the same sound at the angle of maximum rejection). It is rated in decibels of attenuation (e.g., a typical front-to-back ratio might be −23 dB at 180°).

Handling Noise: The noise created by mechanical vibrations or shocks picked up by a microphone.

Headroom: The difference in decibels between the standard operating level of an audio device and the level at which a specified amount of distortion occurs.

Hypercardioid Polar Pattern: A polar pattern that is formed of 25% pressure and 75% pressure-gradient characteristics. Such a pattern is sensitive to signals arriving from the front and rejects signals arriving from 110° off-axis.

Impedance: The resistance that a device or transmission line will show to an ac or dc current at a specified frequency.

Inherent Noise (Self-Noise): The noise signal generated inside a system in the absence of external excitation. Inherent noise is usually expressed as an equivalent sound pressure level that would produce an output voltage equal to the noise voltage.

Lavaliere Microphone: A miniature microphone that is designed to be suspended around the user's neck or attached to the user's clothing via a clip (e.g., a tie clip).

Line-Matching Transformer: A transformer that matches the impedance of one device to that of another. (e.g., 250-Ω microphone to 50-kΩ input).

Magnetic Induction: The generation of an electrical signal in a conductor caused by relative motion between the conductor and external magnetic lines of force. When a conductor (wire) moves within a magnetic field and cuts across the magnetic lines of force, a voltage will be set up or "induced" across this conductor. Also, when a stationary conductor is placed in an oscillating magnetic field, a voltage is induced across the conductor.

Moving-Coil Microphone: An electromagnetic transducer, in which a coil of wire is attached to a diaphragm. The coil is suspended within a high magnetic flux field. As the diaphragm displaces the coil within this field, an output voltage is generated across the coil, providing a signal output. The moving-coil microphone is often called a *dynamic microphone*.

Noise-Cancelling Microphone: A microphone designed for speech communication in high-level environments, whereby close-proximity signals are reinforced and distant (ambient) signals are cancelled out.

Omnidirectional Polar Pattern: A circular polar pattern, pressure-created. A microphone with an omni pattern is equally sensitive to signals arriving from all angles. Due to the directionality of high frequencies reflecting off the microphone case, such a microphone may become increasingly directional with frequency.

On-Axis Response: The frequency response of a microphone measured in a free-field, with the sound source on the axis of the microphone.

P-Popping: A microphone noise produced when the diaphragm is struck by a puff of air that is forced out of a speaker's or singer's mouth during pronunication of plosive sounds (p, b, t).

Phantom Powering: A technique for supplying power to a condenser microphone, in which the dc voltage is applied equally to each signal conductor (within a balanced microphone cable) and is returned to the voltage source via the cable shield. Commonly, this voltage is 12-V dc to 48-V dc.

Phase: The measurement (in degrees) of how far a wave has traveled through one complete cycle.

Phase Shift: In the phase relation between one wave and another, phase shift occurs when one wave has been advanced or retarded through its cycle relative to a similar waveform.

Polarization Voltage: The (normally high) dc voltage applied to the backplate/diaphragm capacitor of a condenser microphone via a high resistance, thus setting up a fixed-charge condition. Changes in the backplate/diaphragm distance due to pressure variations result in a varying output voltage.

Prepolarization: A technique of depositing a fixed charge-carrying layer on either the diaphragm or backplate of a condenser microphone, thus eliminating the need for an external polarization voltage.

Pink Noise: Electronically generated noise that has equal energy per octave.

Polar Pattern: The graphic representation of the sensitivity of a microphone over all incident angles at a rated frequency.

Pop Filter: A wind screen (often made of open-cell foam) that fits over the exterior of the diaphragm/microphone casing to reduce the pressure levels of plosive vocal sounds which cause a popping effect.

Pressure-Gradient Microphone: A microphone in which both sides of the diaphragm are exposed to the incident sound; therefore, the microphone is responsive to the pressure differential (gradient) between the two sides of the membrane. In a figure-eight or bidirectional microphone, sound incident parallel to the plane of the diaphragm (90° off-axis) produces no pressure differential and very little output.

Pressure Microphone: A microphone in which only one side of the diaphragm is exposed to the impinging sound. The diaphragm responds uniformly to the pressure variations and so pressure microphones are inherently omnidirectional.

Proximity Effect: An inherent characteristic of pressure-gradient microphones, resulting in a boost at the low-frequency response when a microphone is in close proximity to a source. The effect becomes significant when the source-to-microphone distance is approximately the same as the wavelength of the impinging sound.

Resonance: Certain systems, whether they are the sounding board of a grand piano or the air cavities within a microphone, have a natural frequency at which they vibrate. When the movement of air or vibrations born by the structure hit this frequency the system mass operates in "sympathy" and the system is said to be in resonance. Resonances can also reinforce certain frequencies because they are in sympathetic vibration with them.

Reverberation: Repetitions of a reflected sound that arrive at the listener at intervals so brief that the individual echoes cannot be discerned by the listener.

Reverberation Time: The time it takes for a reverberated signal to decrease by 60 dB within an enclosed space; also known as a room's *RT60*.

Ribbon Microphone: An electromagnetic microphone, in which a thin, corrugated ribbon (diaphragm) is suspended within a high magnetic-flux field. As the difference in acoustic pressure between the front and rear of the ribbon displaces the ribbon, an output voltage is generated across it and thus provides a signal output.

Roll-Off: A gradual, continuous decrease in low- or high-frequency response in order to correct for the bass boost of proximity effect or excessive sibilance. A roll-off switch may be located in the microphone itself, or a roll-off can be done at the console equalizers.

Sensitivity: The output voltage or power, expressed in dBV or dBm, of a microphone, when it is exposed to a specific sound pressure level.

Shield: In an audio cable, a conductive cylinder around one or more center conductors that protects against unwanted electrostatic fields that could induce a signal, heard as a hum or buzz, across the conductors of the cable.

Shock Mounting: Any mounting or suspension system that mechanically isolates equipment from unwanted vibration.

Sibilance: Emphasis of s, sh, or ch sounds.

Sound Pressure: The pressure above and below normal atmospheric pressure caused by sound vibration at a point. As a sound wave travels through the air it brings molecules closer together and then pulls them further apart. This periodic change in the atmospheric pressure is termed the sound pressure. Its greatest value is called the *peak amplitude*.

Sound Pressure Level (SPL): The expression of sound pressure as a dB level, referenced to a pressure (SPL_{REF}) of 0.0002 microbar (0 dBSPL). SPL is thus defined: $SPL = 20 \log (SPL/SPL_{REF})$. The suffix "A" (e.g., 23 dB[A]) indicates that the SPL is A-weighted.

Supercardioid Polar Pattern: The polar pattern formed of 38% pressure and 62% pressure-gradient characteristics. Such a pattern is most sensitive to those signals arriving from the front and rejects most signals arriving from 125° off-axis.

Total Harmonic Distortion: Distortion in which harmonic components (integer multiples of a fundamental frequency) are produced. THD is normally expressed as a percentage of the fundamental and includes all the distortion components.

Trackability: The comparative phase response of a microphone pair.

Transducer: A device that changes one form of energy into another, corresponding form of energy (e.g., microphone, loud-speaker, phono cartridge).

Transient Response: The ability of a transducer to accurately "track" or respond to a given waveform.

Unidirectional Polar Pattern: A polar pattern of a microphone that is most sensitive to sound in one direction (as opposed to omnidirectional and bidirectional).

Working Distance: The distance between the sound source and the microphone.

F Bibliography

Chapter 1

Bartlett, Bruce. *Introduction to Professional Recording Techniques*. Indianapolis: Howard W. Sams & Company, 1987.

Huber, David Miles. *Audio Production Techniques for Video*. Indianapolis: Howard W. Sams & Company, 1987.

Runstein, Robert E. and David Miles Huber. *Modern Recording Techniques*. Indianapolis: Howard W. Sams & Company, 1986.

Chapter 2

Bore, Gerhart. *Microphones for Professional and Semi-Professional Applications*. Scahlltechnik Dr.-Ing. Schoeps GmbH.

Sank, Jon R. "Microphones." *Journal of the Audio Engineering Society*. July/August, 1985.

Chapter 3

Bauch, F. W. O. "New High-Grade Condenser Microphones." *Wireless World*. February, 1953.

Collette Program Product Information. Dr.-Ing.

Peus, Stephen. "Microphones and Transients." *db-The Sound Engineering Magazine*. May, 1977.

Sank, Jon R. "Microphones." *Journal of the Audio Engineering Society*. July/August, 1985.

Chapter 4

Bore, Gerhart. *Microphones for Professional and Semi-Professional Applications*. Dr.-Ing.

Burroughs, Lou. *Microphones: Design and Application*. Plainview, New York: Sagamore Publishing Company, 1974.

Eargle, John. *The Microphone Handbook*. Plainview, New York: Elar Publishing, 1981.

Smolen, Michael. "Equalizers." *Stereo Review*. December, 1986. 82–86.

Woram, John M. *The Recording Studio Handbook*. Plainview, New York: Elar Publishing, 1982.

Chapter 7

Eargle, John. *The Microphone Handbook*. Plainview, New York: Elar Publishing, 1981.

Huber, David Miles. *Audio Production Techniques for Video*. Indianapolis: Howard W. Sams & Company, 1987.

MS-38 & MS-380 brochure, Audio Engineering Associates, 1987.

Streicher, Ron and Wes Dooley. "Basic Stereo Microphone Perspectives: A Review." *Journal of the Audio Engineering Society*. vol. 33, no. 7/8. July/August 1985. 548–555.

Sunier, John. "A History of Binaural Sound." *Audio Magazine*. March, 1986.

Chapter 8

Bartlett, Bruce. *Introduction to Professional Recording Techniques*. Indianapolis: Howard W. Sams & Company, 1987.

Microphone Application Notes. Crown International, Inc., 1986.

Orban, Robert. "How to Choose Equalizers." *Applications Guide*, 1985.

Runstein, Robert E. and David Miles Huber. *Modern Recording Techniques*. Second Edition. Indianapolis: Howard W. Sams & Company, 1986.

Smolen, Michael. "Equalizers." *Stereo Review*. December, 1986. Beyer Dynamic, Inc., 1986.

What Every Drummer Should Know About Miking Drums. Technical paper.

Chapter 9

Antenna Systems. Swintek Technical Specifications, 1986.

Huber, David Miles. *Audio Production Techniques for Video*. Indianapolis: Howard W. Sams & Company, 1987.

Isomax Application Notes. Countryman Associates, Inc., 1985.

Microphone Application Notes: The GLM Microphone. Crown International, Inc., 1986.

Runstein, Robert E. and David Miles Huber. *Modern Recording Techniques*. Second Edition. Indianapolis: Howard W. Sams & Company, 1986.

Peterson, George. "Shotgun Microphones: Tools of the Trade." *Mix Magazine*. April, 1987.

Sony Wireless Microphone Systems. Application notes. Sony Corporation of America, Inc., 1986.

Chapter 10

Davis, Don and Carolyn Davis. *Sound System Engineering*. Second Edition. Indianapolis: Howard W. Sams & Company, 1987.

Eargle, John. *The Microphone Handbook*. Plainview, New York: Elar Publishing, 1981.

Snow, William B. "Frequency Characteristics of a Sound Reinforcing System." *Journal of the AES*. April, 1955.

Index

A

B

C

D

K

L

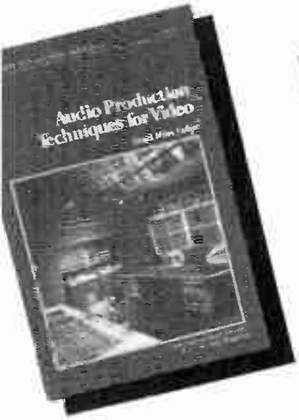
M

N

O

R

S



Audio Production Techniques for Video

David Miles Huber

'his reference book examines the im-
portant role that audio plays in video
production. It is a well-rounded assess-
nent of the equipment, techniques, and
echnology required to understand and
create in today's world of media
production.

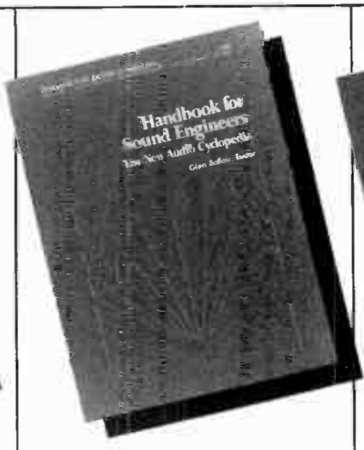
Bridging the gap between the currently
merging technologies of audio and video
production, this book outlines modern
audio production and post-production
echniques for video. It thoroughly
covers the often confusing and
misunderstood time code, electronic
editing, digital audio, multi-track audio,
and live broadcast stereo.

Unlike any other book on the market, this book addresses the specific needs of the audio track in video tape production and the new audio-for-video standards set for the industry.

Topics covered include:

- The Audio Tape Recorder/Video Tape Recorder
- Synchronization
- Audio Production for Video
- Audio Post-Production for Video
- Introductory Electronic Editing Techniques
- Appendices Include: Preventing and Troubleshooting Time Code Problems, Definitions and Explanations of SMPTE, VITC, and Pilotone

352 Pages, 7½ x 9¾, Softbound
ISBN: 0-672-22518-2
No. 22518, $29.95



Handbook for Sound Engineers: The New Audio Cyclopedia

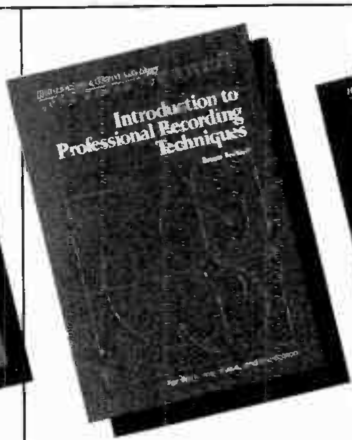
Glen Ballou, Editor

The most authoritative audio reference on the market, this book offers the professional engineer or technician a one-stop guide to the complete field.

The 31 chapters are combined into 7 parts as follows:

- Acoustics—Fundamentals; Psychoacoustics; Small Room Acoustics; Common Factors; Acoustic Design; Recording Studio Design; Rooms for Speech, Music, and Cinema; Open Plan Rooms
- Electronic Components for Sound Engineering—Resistors, Capacitors, and Inductors; Transformers; Tubes, Discrete Solid-State Devices, and Integrated Circuits; Heat Sinks, Wire, and Relays
- Electroacoustic Devices—Microphones; Loudspeakers, Enclosures, and Headphones
- Audio Electronic Circuits and Equipment—Amplifiers; Attenuators; Filters and Equalizers; Delay; Power Supplies; Constant- and Variable-Speed Devices; VU and Volume Indicator Meters and Devices; Consoles and Systems
- Recording and Playback—Disk, Magnetic, and Digital
- Design Applications—Sound System Design; Systems for the Hearing Impaired, the Broadcast Chain; Image Projection
- Measurements—Audio Measurements; Fundamentals and Units of Measurements

1,264 Pages, 8 x 10, Hardbound
ISBN: 0-672-21983-2
No. 21983, $79.95



Introduction to Professional Recording Techniques

Bruce Bartlett,
The John Woram Audio Series

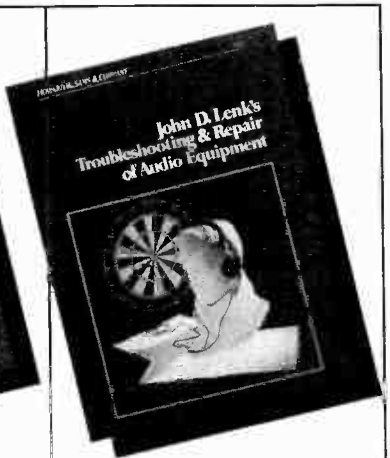
This all-inclusive introduction to the equipment and techniques for state-of-the-art recording—whether in residences or professional studios or on location—offers a wealth of valuable information on topics not found in other books on audio recording.

Geared primarily for the audio hobbyist or aspiring professional, this book delivers a comprehensive discussion of recording engineering and production techniques, including special coverage of microphones and microphone techniques, sampling, sequencing, and MIDI. It provides up-to-date coverage of monitoring, special effects, hum prevention, and spoken-word recording, as well as special sections on recognizing good sound and troubleshooting bad sound.

Topics covered include:

- The Recording and Reproduction Chain
- Simple Home Recording
- Setting Up the Studio
- Microphones and Microphone Techniques
- Control-Room Techniques
- On-Location Recording
- Judging the Recording
- Appendices Include: dB or not dB, Introduction to SMPTE Time Code, and Further Education

416 Pages, 7½ x 9¾, Softbound
ISBN: 0-672-22574-3
No. 22574, $22.95



John D. Lenk's Troubleshooting & Repair of Audio Equipment

John D. Lenk

This manual provides the most up-to-date data available and a simplified approach to practical troubleshooting and repair of major audio devices. It will enable both the beginning and the intermediate level technician or hobbyist to apply tips and tricks to any specific equipment.

This book also includes such time-saving hints as circuit-by-circuit troubleshooting based on failure or trouble symptoms, universal step-by-step procedures, and actual procedures recommended by manufacturers' service personnel.

Topics covered include:

- Introduction to Modern Audio Equipment Troubleshooting
- Troubleshooting and Repair of Amplifiers and Loudspeakers
- Troubleshooting and Repair of Linear-Tracking Turntables
- Troubleshooting and Repair of Audio Cassette Decks
- Troubleshooting and Repair of AM/FM Stereo Tuners
- Troubleshooting and Repair of CD Players

John D. Lenk is the author of over 60 books in electronics. Over one million of his books have been sold in six different languages.

208 Pages, 8½ x 11, Softbound
ISBN: 0-672-22517-4
No. 22517, $21.95

Visit your local book retailer, use the order form provided, or call 800-428-SAMS.

Modern Recording Techniques, Second Edition

Runstein and Huber

For both engineers and students, this revised edition is a valuable guide to state-of-the-art developments and practices in the recording industry. It reflects all the latest equipment, controls, acoustics, and digital effect devices being used in modern recording studios. It also explores the marriage of video and audio multi-track studios and illustrates sound and studio capabilities and limitations.

Topics covered include:

- Sound and Hearing
- Studio Acoustics
- Microphones and Pickups
- Microphone Techniques
- Magnetic Tape Recording
- Amplifiers
- Signal-Processing Equipment
- Consoles
- Noise-Reduction Devices
- Interlocking Tape Machines
- Automated Mixdown
- Speakers and Monitors
- Product Manufacture
- Studio Session Procedures
- Tomorrow's Industry

384 Pages, 7½ x 9¾, Softbound
ISBN: 0-672-22451-8
No. 22451, $18.95

Musical Applications of Microprocessors, Second Edition

Hal Chamberlin

This expanded and revised edition covers analog, digital, and microprocessor sound and music synthesis. Its nonmathematical language makes the material accessible to musicians and computer users, as well as engineers.

New synthesis techniques, nonlinear waveshaping, Vosim, and the Fourier transform are covered and supported with program listings in BASIC and 68000 assembly language.

An entirely new section examines the practical application of synthesis theory in actual synthesis products, including professional and studio equipment, novelty products using modern synthesis techniques, and sound generation circuits.

Topics covered include:

- Music Synthesis Principles
- Sound Modification Methods
- Direct Computer Synthesis Methods
- Computer-Controlled Analog Synthesis
- Digital-to-Analog and Analog-to-Digital Converters
- Control Sequence Display and Editing
- Digital Synthesis and Sound Modification
- Source-Signal Analysis
- Product Applications and the Future

816 Pages, 6¼ x 9¼, Hardbound
ISBN: 0-672-45768-7
No. 45768, $39.95

Sound System Engineering, Second Edition

Don and Carolyn Davis

Like the first edition, this comprehensive text will provide you with useful information for the day-to-day work of designing sound systems—with much more material and in-depth coverage of subjects. It is a practical manual that carefully examines methods of accurately predicting such variables as acoustic gain, clarity of sound, and required electrical input power while plans are still on the drawing board.

Discover problems that might occur in a sound system as it evolves through design, installation, equalization, operation, and maintenance.

Topics covered include:

- Audio Systems
- Mathematics for Audio Systems
- Using the Decibel
- Impedance Matching
- Interfacing Electrical and Acoustics Systems
- Loudspeaker Directivity and Coverage
- The Acoustic Environment
- Large-Room Acoustics
- Small-Room Acoustics
- Designing for Speech Intelligibility
- Designing for Acoustic Gain
- Microphones
- Loudspeakers and Loudspeaker Arrays
- Using Delay Devices
- Installing the Sound System
- Equalizing the Sound System
- Audio and Acoustic Instrumentation
- Putting it All Together
- Specifications

688 Pages, 7½ x 9¾, Hardbound
ISBN: 0-672-21857-7
No. 21857, $39.95

Principles of Digital Audio

Ken C. Pohlmann

Starting with the fundamentals of numbers, sampling and quantizing, you'll get a look at a complete audio digitization system and its various components. You'll also explore storage mediums and learn how digital data is read and processed through demodulation, error detection and correction, digital/audio conversion and output filtering.

Principles of Digital Audio presents a clear, concise overview with many interesting illustrations and formulas. And a vast majority of the data has never been published before. Whether you're an electronics novice or skilled audio engineer, you'll find this book to be an entertaining and comprehensive look into the world of digital audio.

Topics covered include:

- Digital Audio Basics
- Fundamentals of Digital Audio
- Digital Audio Recording
- Digital Audio Reproduction
- Digital Audio Media
- Error Protection
- The Compact Disc
- Music and Technology

Ken C. Pohlmann is Associate Professor and Director of the Music Engineering program at the University of Miami.

288 Pages, 7½ x 9¼, Softbound
ISBN: 0-672-22388-0
No. 22388, $19.95

Visit your local book retailer, use the order form provided, or call 800-428-SAMS.

Audio IC Op-Amp Applications, Third Edition

Walter G. Jung

The classic book in its field, this new update includes devices such as the OP-27/37, as well as new applications circuitry to illustrate current trends of use. Included is a brief section on differential input/output IC devices, IC applications in car stereo chips and audio testing, with over 70 pages of industry data.

As with previous editions, this one provides you with a thorough understanding of basic op-amp theory, the unique characteristics of the various devices, and the technical barriers to performance. You will learn how to apply op-amp configurations to audio applications and the advantages of using IC op amps for the highest-quality audio applications.

Topics covered include:

- IC Op-Amp Parameters Important in Audio Applications
- The Basic Op-Amp Configurations Translated to Audio Applications
- Practical Audio Circuits Using IC Op Amps
- Equalized Amplifiers and Active Filters
- Miscellaneous Audio Circuits
- Appendices: Device Data Sheets, Op Amp Manufacturers and Devices

336 Pages, 5½ x 8½, Softbound
ISBN: 0-672-22452-6
No. 22452, $17.95

How to Build Speaker Enclosures

Badmaieff and Davis

A practical guide to the whys and hows of constructing high quality, top performance speaker enclosures. A wooden box alone is not a speaker enclosure—size, baffling, sound insulation, speaker characteristics, and crossover points must all be carefully considered.

The book contains many detailed drawings and instructions for building the various basic types of enclosures, including the infinite-baffle, the bass-reflex, and the horn-projector types, as well as different combinations of these. This practical book covers both the advantages and disadvantages of each enclosure type and includes a discussion of speaker drivers, crossover networks, and hints on the techniques of construction and testing.

Topics covered include:

- Speaker Enclosures
- Drivers for Enclosures
- Infinite Baffles
- Bass-Reflex or Phase-Inversion Enclosures
- Horn Enclosures
- Combination Enclosures
- Crossover Networks
- Cosntruction and Testing Techniques

144 Pages, 5½ x 8½, Softbound
ISBN: 0-672-20520-3
No. 20520, $6.95

Visit your local book retailer, use the order form provided, or call 800-428-SAMS.