

CHAPTER 4

Microphones: Design and Application

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THE MICROPHONE: AN INTRODUCTION

A *microphone* (often called a *mic*) is usually the first device in a recording chain. Essentially, a mic is a transducer that changes one form of energy (sound waves) into another corresponding form of energy (electrical signals). The quality of its pickup will often depend on external variables (such as placement, distance, instrument and the acoustic environment), as well as on design variables (such as the microphone's operating type, design characteristics and quality). These interrelated elements tend to work together to affect the overall sound quality.

In order to deal with the wide range of musical, acoustic and situational circumstances that might come your way (not to mention your own personal taste), a large number of mic types, styles and designs can be pulled out of our "sonic toolbox." Because the particular characteristics of a mic might be best suited to a specific range of applications, engineers and producers use their artistic talents to get the best possible sound from an acoustic source by carefully choosing the mic or mics that fit the specific pickup application at hand.

The road to considering microphone choice and placement is best traveled by considering a few simple rules:

Rule 1: *There are no rules, only guidelines.* Although guidelines can help you achieve a good pickup, don't hesitate to experiment in order to get a sound that best suits your needs or personal taste.

Rule 2: *The overall sound of an audio signal is no better than the weakest link in the signal path.* If a mic or its placement doesn't sound as good as it could, make the changes to improve it *BEFORE* you commit it to tape, disc or whatever. More often than not, the concept of "fixing it later in the mix" will often put you in the unfortunate position of having to correct a situation after the fact, rather than recording the best sound and/or performance during the initial session.

THE "GOOD RULE"

Good musician + good instrument
+ good performance + good acoustics
+ good mic + good placement
= good sound.

Rule 3: *Whenever possible, use the “Good Rule”: Good musician + good instrument + good performance + good acoustics + good mike + good placement = good sound.* This rule refers to the fact that a music track will only be as good as the performer, instrument, mic placement and the mic itself. If any of these elements falls short of its potential, the track will suffer accordingly. However, if all of these links are the best that they can be, the recording will almost always be something that you’ll be proud of!

The miking of vocals and instruments (both in the studio and onstage) is definitely an art form. It’s often a balancing act to get the most out of the Good Rule. Sometimes you’ll have the best of all of the elements; at others, you’ll have to work hard to make lemonade out of a situational lemon. The best rule of all is to use common sense and to trust your instincts.

Before delving into placement techniques and facts that deal with the finer points of microphone technology, I’d like to take a basic look at how microphones (and their operational characteristics) work. Why do I put this in the book? Well, from a personal standpoint, having a basic understanding of what happens “under the hood” has helped me to get a mental image of how a particular mic or mic technique will work in a certain situation. Basically it helps to make judgments that can be combined with my own intuition to make the best artistic judgment at the time ... I hope it will help you, as well.

MICROPHONE DESIGN

A microphone is a device that converts acoustic energy into corresponding electrical voltages that can be amplified and recorded. In audio production, three transducer mic types are used:

- Dynamic mic
- Ribbon mic
- Condenser mic.

THEORY OF ELECTROMAGNETIC INDUCTION

Whenever an electrically conductive metal cuts across the flux lines of a magnetic field, a current of a specific magnitude and direction will be generated within that metal.

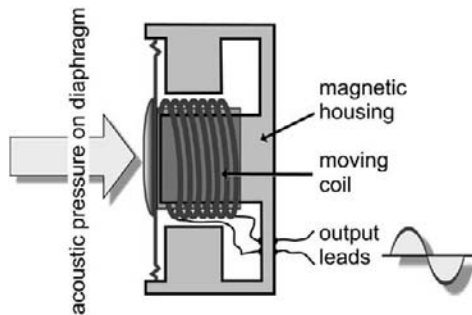
The dynamic microphone

In principle, the *dynamic mic* (Figure 4.1) operates by using electromagnetic induction to generate an output signal. The simple *theory of electromagnetic induction* states that whenever an electrically conductive metal cuts across the flux lines of a magnetic field, a current of a specific magnitude and direction will be generated within that metal.

Dynamic mic designs (Figure 4.2) generally consist of a stiff Mylar diaphragm of roughly 0.35-mil thickness. Attached to the diaphragm is a finely wrapped core of wire (called a *voice coil*) that’s precisely suspended within a high-level magnetic field. Whenever an acoustic pressure

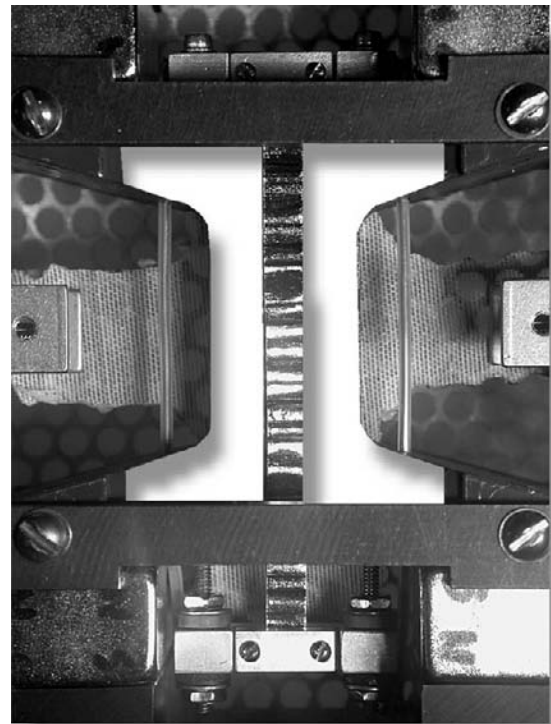
**FIGURE 4.1**

The Shure Beta 58A dynamic mic. (Courtesy of Shure Incorporated, www.shure.com.)

**FIGURE 4.2**

Inner workings of a dynamic microphone.

wave hits the diaphragm's face (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 that's supplied by a permanent magnet (C). In doing so, an analogous electrical signal (of a specific magnitude and direction) is induced into the coil and across the output leads, thus producing an analog audio output signal.

**FIGURE 4.3**

Cutaway detail of a ribbon microphone. (Courtesy of Audio Engineering Associates, www.ribbonmics.com.)

The ribbon microphone

Like the dynamic microphone, the *ribbon mic* also works on the principle of electromagnetic induction. Older ribbon design types, however, use a diaphragm of extremely thin aluminum ribbon (2 microns). Often, this diaphragm is corrugated along its width and is suspended within a strong field of magnetic flux (Figure 4.3). Sound-pressure variations between the front and the back of the diaphragm cause it to move and cut across these flux lines, inducing a current into the ribbon that's proportional to the amplitude and frequency of the acoustic waveform. Because the ribbon generates a small output signal (when compared to the larger output that's generated by the multiple wire turns of a moving coil), its output signal is too low to drive a microphone input stage



FIGURE 4.4
The AEA A440 ribbon mic. (Courtesy of Audio Engineering Associates, www.ribbonmics.com.)



FIGURE 4.5
Cascade Microphones Fat Head ribbon mic. (Courtesy of Cascade Microphones, www.cascademicrophones.com.)



FIGURE 4.6
The Beyerdynamic M160 ribbon mic. (Courtesy of Beyerdynamic, www.beyerdynamic.com.)

directly; thus, a step-up transformer must be used to boost the output signal and impedance to an acceptable range.

Until recently, traditional ribbon technology could be only found on the original, vintage mics (such as the older RCA and Cole ribbon mics); however, with the skyrocketing price of vintage mics and a resurgence in the popularity of the smooth, transient quality of the “ribbon sound,” newer mics that follow the traditional design philosophies have begun to spring up on the market (Figures 4.4 and 4.5).

RECENT DEVELOPMENTS IN RIBBON TECHNOLOGY

During the past several decades, certain microphone manufacturers have made changes to original ribbon technologies by striving to miniaturize and improve their basic operating characteristics. The popular M160 (Figure 4.6) and M260 ribbon mics from Beyerdynamic use a rare-earth magnet to produce a capsule that’s small enough to fit into a 2-inch grill ball (much smaller than a traditional ribbon-style mic). The ribbon (which is corrugated along its length to give it added strength and at each end to give it flexibility) is 3 microns thick, about 0.08 inch wide, 0.85 inch long and weighs only 0.000011 ounce. A plastic throat is fitted above the ribbon, which houses a pop-blast filter. Two additional filters and the grill greatly reduce the ribbon’s potential for blast and wind damage, a feature that has made these designs suitable for outdoor and handheld use.

Another relatively recent advance in ribbon technology has been the development of the printed ribbon mic. In principle, the printed ribbon operates in precisely the same manner as the conventional ribbon pickup; however, the rugged diaphragm is made from a polyester film that has a spiral aluminum ribbon printed onto it. Ring magnets are then placed at the diaphragm's front and back, thereby creating a wash of magnetic flux that makes the electromagnetic induction process possible.

Other alterations to traditional ribbon technology make use of phantom power to supply power to an active, internal amplifier (Figure 4.6), so as to boost the mic's output to that of a dynamic or condenser mic, without the need for a passive transformer (an explanation of phantom power can be found in the next section on condenser mics).

The condenser microphone

Condenser mics (like the capsules which are shown in Figures 4.7 and 4.8) operate on an *electrostatic principle* rather than the electromagnetic principle used by a dynamic or ribbon mic. The capsule of a basic condenser mic consists of two plates: one very thin movable diaphragm and one fixed backplate. These two plates form a capacitor (or condenser, as it is still called in the UK and in many parts of the world). A capacitor is an electrical device that's capable of storing an electrical charge. The amount of charge that a capacitor can store is determined by its capacitance value and the voltage that's applied to it, according to the formula:

$$1\ Q = CV$$

where Q is the charge (in coulombs), C is the capacitance (in farads), and V is the voltage (in volts).

At its most basic level, a condenser mic operates when a regulated DC power supply is applied between its diaphragm plates to create a capacitive charge. When sound acts upon the movable diaphragm, the varying distance between the plates will likewise create a change in the device's capacitance (Figure 4.9). According to the above equation, if Q (the power supply charge) is constant and C (the diaphragm's capacitance) changes, then V (voltage across the diaphragm) will change in a proportional and inverse fashion. In other words:

FIGURE 4.7
Exposed example of a condenser diaphragm. (Courtesy of ADK, www.adkmic.com; photograph by K. Bujack.)

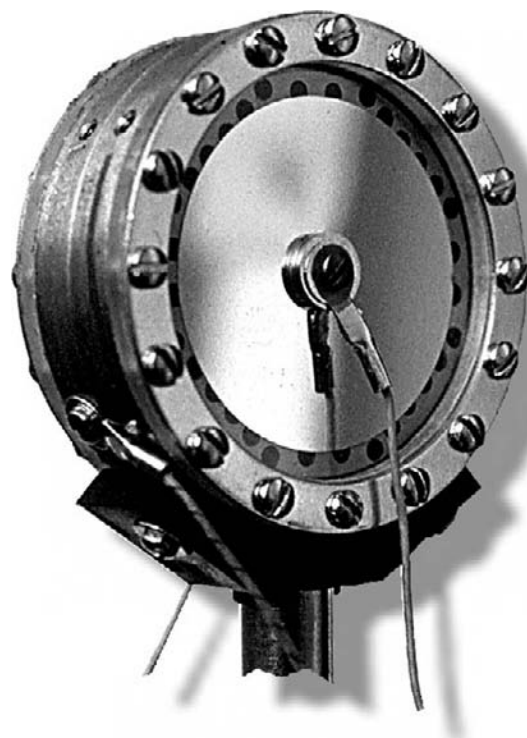




FIGURE 4.8
Inner detail of an
AKG C3000B
condenser mic.
(Courtesy of AKG
Acoustics, Inc., www.ake.com.)

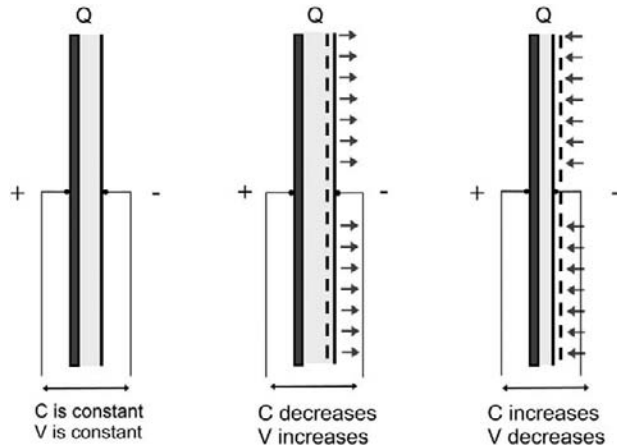


FIGURE 4.9
Output and potential
relationships as a
result of changing
capacitance.

Since the charge (Q) is known to be constant and the diaphragm's capacitance (C) changes with differences in sound pressure, the voltage (V) must change in inverse

proportion. Given that the capsule's voltage now changes in proportion to the sound waves that act upon it, *voilà* ... we have a condenser mic!

The next trick is to tap into the circuit to capture the changes in output voltage, by placing a high-value resistor across the circuit. Since the voltage across the resistor will change in inverse proportion to the capacitance across the capsule plates, this signal will then become the mic's output signal (Figure 4.10).

Since the resulting signal has an extremely high impedance, it must be fed through a preamplifier in order to preserve the mic's frequency response characteristics. Since this amp must be placed at a point just following the resistor (often at a distance of 2 inches or less), it is almost always placed within the mic's body in order to prevent hum, noise pickup and signal-level losses. In addition to the need for a polarizing voltage, the preamp is another reason why conventional condenser microphones require a supply voltage in order to operate.

PHANTOM POWER

Most modern professional condenser (and some ribbon) mics don't require internal batteries, external battery packs or individual AC power supplies in order to operate. Instead, they are designed to be powered directly from the console through the use of a *phantom power* supply. Phantom power works by supplying a positive DC supply voltage of +48 V through both audio conductors (pins 2 and 3) of a balanced mic line to the condenser capsule and preamp. This voltage is equally distributed through identical value resistors, so that no differential exists between the two leads. The -48-V side of the circuit is supplied to the capsule and preamp through the cable's grounding wire (pin 1).

Since the audio is only affected by potential differences between pins 2 and 3 (and not the ground signal on pin 1), the carefully matched +48-V potential at these leads is therefore not electrically "visible" to the input stage of a balanced mic preamp. Instead, only the balanced, alternating audio signal that's being simultaneously carried along the two audio leads will be detected (Figure 4.11).

The resistors (R) used for distributing power to the signal leads should be $1/4$ -W resistors with a $\pm 1\%$ tolerance and have the following values for the following supply voltages (because some mics can also be designed to work at voltages lower than 48 V): 6.8 k Ω for 48 V, 1.2 k Ω for 24 V, and 680 Ω for a 12-V supply. In addition to precisely matching the supply voltages, these resistors also provide a degree of power isolation between other mic inputs on a console. If a signal lead were accidentally shorted to ground (which could happen if defective cables or unbalanced XLR cables were used), the power supply should still be able to deliver power to other mics in the system. If two or more inputs were accidentally shorted, however, the phantom voltage could drop to levels that would be too low to be usable.

Although most modern condensers use some form of a field effect transistor (FET) to reduce the capsule impedance, an increasing number of original era and "revival" models use an internally housed vacuum tube to amplify and change the impedance of the condenser capsule. These mics are generally valued by studios and collectors alike for their "tube" sound, which results from

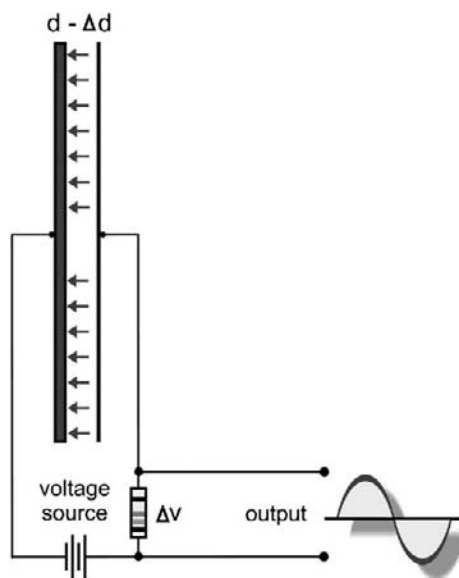


FIGURE 4.10

As a sound wave decreases the condenser spacing by d the capacitance will increase, causing the voltage to proportionately fall (and vice versa).

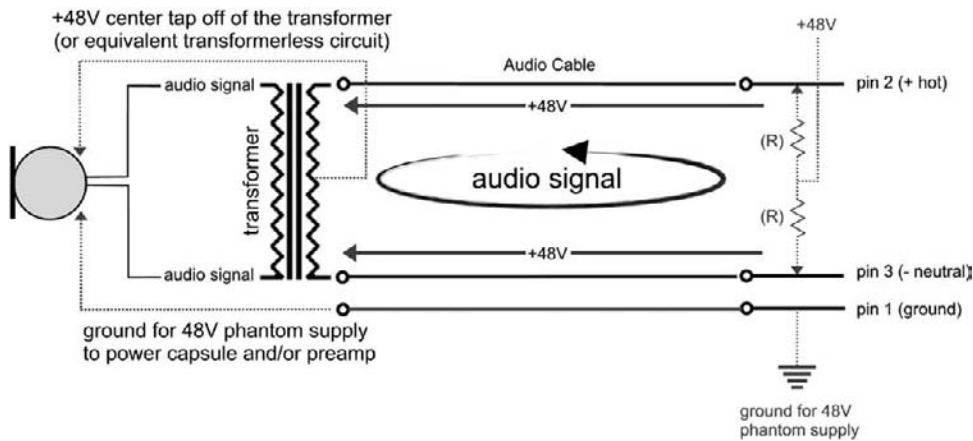


FIGURE 4.11
Schematic drawing
of a phantom power
system.

even-harmonic distortion and other sonic characteristics that occur whenever tubes are used.

THE ELECTRET-CONDENSER MICROPHONE

Electret-condenser mics work on the same operating principles as their externally polarized counterparts, with the exception that a static polarizing charge has been permanently set up between the mic's diaphragm and its backplate. Since the charge (Q) is built into the capsule, no external source is required to power the diaphragm. However, as with a powered condenser mic, the capsule's output impedance is so high that a preamp will still be required to reduce it to a standard value. As a result, a battery, external powering source or standard phantom supply must be used to power the low-current amp.



Tutorial: Mic Types

1. Go to the tutorial section of www.modrec.com, click on “Mic Types” and download the soundfiles (which include examples of each mic operating type).
2. Listen to the tracks. If you have access to an editor or digital audio workstation (DAW), import the files and look at the waveform amplitudes for each example. If you'd like to DIY, then ...
3. Pull out several mics from each operating type and plug them in (if you don't have several types, maybe a studio, your school or a friend has a few you can take out for a spin). Try each one on an instrument and/or vocal. Are the differences between operating types more noticeable than between models in the same family?

MICROPHONE CHARACTERISTICS

To handle the wide range of applications that are encountered in studio, project and on-location recording, microphones will often differ in their overall sonic, electrical and physical characteristics. The following section highlights many of these characteristics in order to help you choose the best mic for a given application.

Directional response

The *directional response* of a mic refers to its sensitivity (output level) at various angles of incidence with respect to the front (on-axis) of the microphone (Figure 4.12). This angular response can be graphically charted in a way that shows a microphone's sensitivity with respect to direction and frequency over 360°. Such a chart is commonly referred to as the mic's *polar pattern*. Microphone directionality can be classified into two categories:

- Omnidirectional polar response
- Directional polar response.

The *omnidirectional mic* (Figure 4.13) is a pressure-operated device that's responsive to sounds that emanate from all directions. In other words, the diaphragm will react equally to all sound-pressure fluctuations at its surface, regardless of the source's location. Pickups that display *directional* properties are pressure-gradient devices, meaning that the pickup is responsive to relative differences in pressure between the front, back and sides of a diaphragm. For example, a purely pressure-gradient mic will exhibit a *bidirectional* polar pattern (commonly called a *figure-8 pattern*), as shown in Figure 4.14. Many of the older ribbon mics exhibit a bidirectional pattern. Since the ribbon's diaphragm is often exposed to sound waves from both the front and rear axes, it's equally sensitive to sounds that emanate from either direction. Sounds from the rear will produce a signal that's 180° out of phase with an equivalent on-axis signal (Figure 4.15a). Sound waves arriving 90° off-axis produce equal but opposite pressures at both the front and rear of the ribbon (Figure 4.15b), resulting in a cancellation at the diaphragm and no output signal.

Figure 4.16 graphically illustrates how the acoustical combination (as well as electrical and mathematical combination, for that matter) of a bidirectional

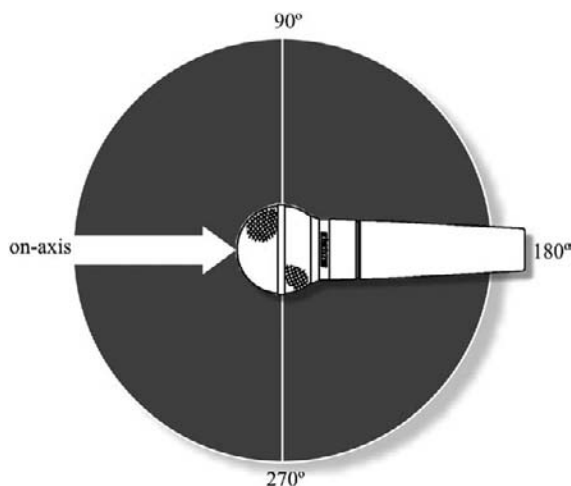


FIGURE 4.12
Directional axis of a microphone.



FIGURE 4.13
Graphic representation of a typical omnidirectional pickup pattern.



FIGURE 4.14
Graphic representation of a typical bidirectional pickup pattern.

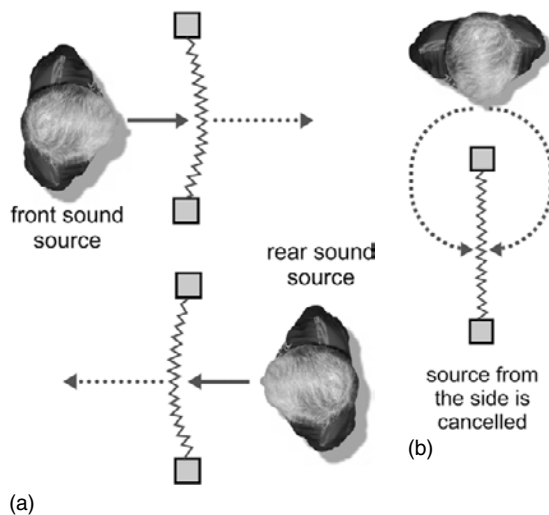


FIGURE 4.15
Sound sources on-axis and 90° off-axis at the ribbon's diaphragm. (a) The ribbon is sensitive to sounds at the front and rear. (b) Sound waves from 90° and 270° off-axis are canceled.

(pressure-gradient) and omnidirectional (pressure) pickup can be combined to obtain other directional pattern types. Actually, an infinite number of directional patterns can be obtained from this mixture, with the most widely known patterns being the *cardioid*, *supercardioid* and *hypercardioid* polar patterns (Figure 4.17).

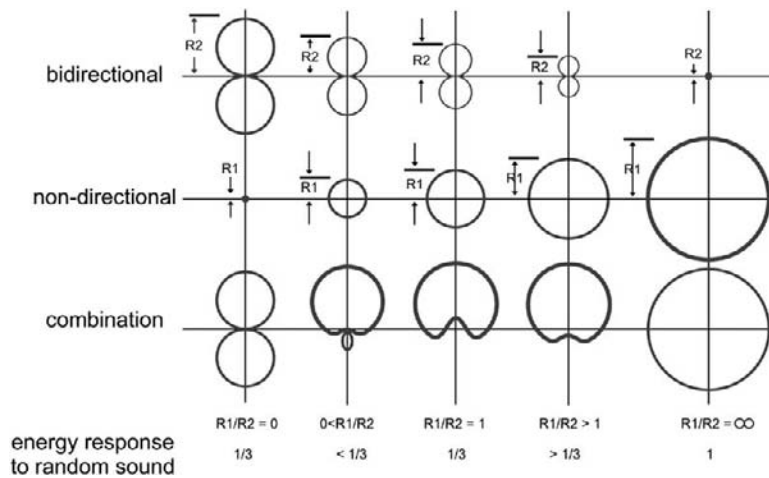


FIGURE 4.16
Directional combinations of various bidirectional and nondirectional pickup patterns.

Often, dynamic mics achieve a cardioid response (named after its heart-shaped polar chart, as shown in Figure 4.18) by incorporating a rear port into their design. This port serves as an acoustic labyrinth that creates an acoustic resistance (delay). In Figure 4.19a, a dynamic pickup having a cardioid polar response is shown receiving an on-axis (0°) sound signal. In effect, the diaphragm receives two signals: the incident signal, which arrives from the front, and an acoustically delayed rear signal. In this instance, the on-axis signal exerts a positive pressure on the diaphragm and begins its travels 90° to a port located on the side of the pickup. At this point, the signal is delayed by another 90° (using an internal, acoustically resistive material or labyrinth). In the time it takes for the delayed signal to reach the rear of the diaphragm (180°), the on-axis signal moves on to the negative portion of its acoustic cycle and then begins to exert a negative pressure on the diaphragm (pulling it outward). Since the delayed rear signal is 180° out of phase at this point in time, it will also begin to push the diaphragm outward, resulting in an output signal.

Conversely, when a sound arrives at the rear of the mic, it begins its trek around to the mic's front. As the sound travels 90° to the side of the pickup, it is again delayed by another 90° before reaching the rear of the diaphragm. During this delay period, the sound continues its journey around to the front of the mic—a delay shift that's also equal to 90° . Since the acoustic pressures at the diaphragm's front and rear sides are equal and opposite, the sound is being simultaneously pushed inward and outward with equal force, resulting in little or no movement ... and therefore will have little or no output signal (Figure 4.19b). The attenuation of such an off-axis signal, with respect to an equal on-axis signal, is known as its *front-to-back discrimination* and is rated in decibels.

Certain condenser mics can be electrically switched from one pattern to another by using a second capsule that's mounted on both sides of a central backplate. Configuring these dual-capsule systems electrically in phase will create an

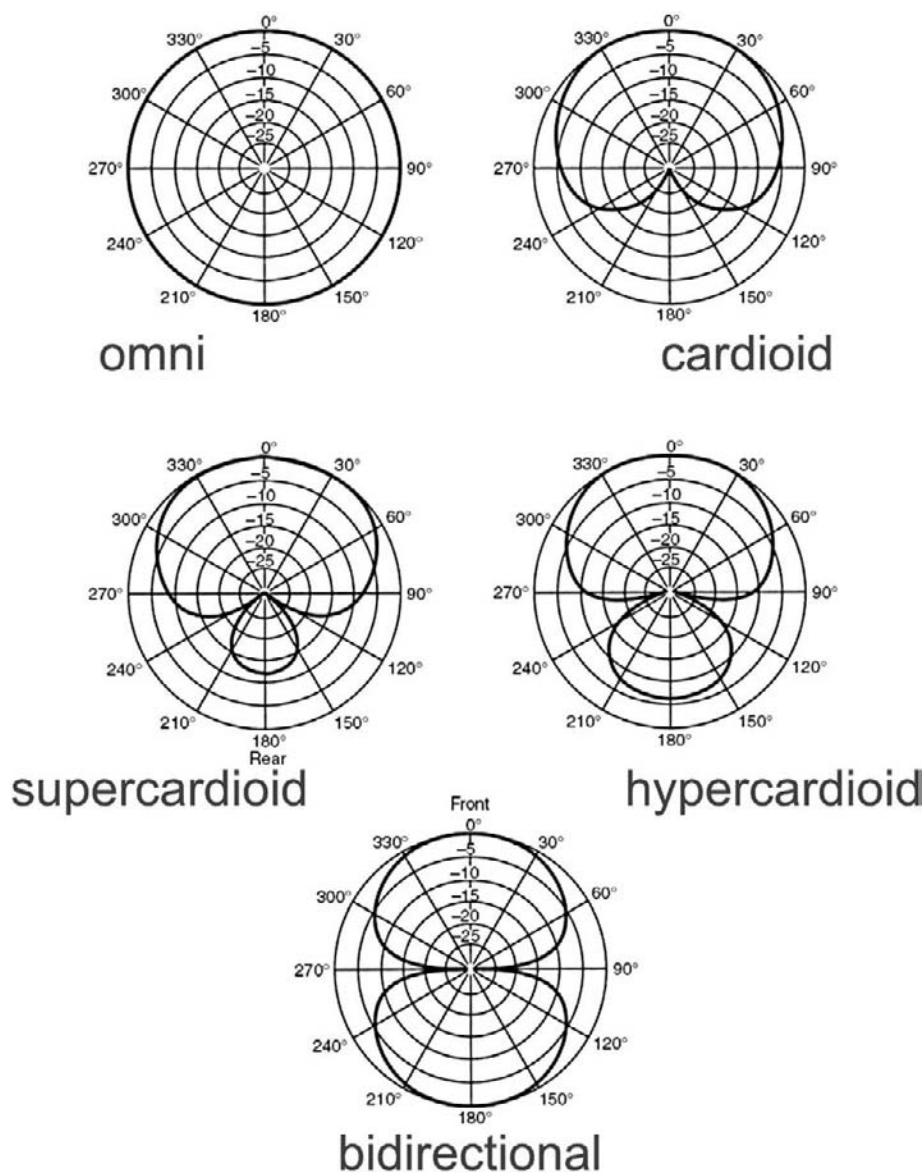


FIGURE 4.17
Various polar
patterns with output
sensitivity plotted
versus angle of
incidence.

omnidirectional pattern, while configuring them out of phase results in a bidirectional pattern. A number of intermediate patterns (such as cardioid and hypercardioid) can be created by electrically varying between these two polar states (in either continuous or stepped degrees), as was seen earlier in Figure 4.16.

Frequency response

The on-axis *frequency-response curve* of a microphone is the measurement of its output over the audible frequency range when driven by a constant, on-axis input signal. This response curve (which is generally plotted in output level [dB] over the 20- to 20,000-Hz frequency range) will often yield valuable information and can give clues as to how a microphone will react at specific frequencies. It should be noted that a number of other variables also determine how a mic will sound, some of which have no measurement standards—the final determination should always be your own ears.

A mic that's designed to respond equally to all frequencies is said to exhibit a flat frequency response (shown as the top curve in Figure 4.20a). Others can be made to emphasize or de-emphasize the high-, mid- or low-end response of the audio spectrum (shown as the boost in the high-end curve in Figure 4.20b) so as to give it a particular sonic character. The solid frequency-response curves (as shown in both parts a and b) were measured on-axis and exhibit an acceptable response. However, the same mics might exhibit a "peaky" or erratic curve when measured off-axis. These signal colorations could affect their sound when operating in an area where off-axis sound (in the form of leakage) arrives at the pickup (shown as the dotted curves in both parts a and b), and will often result in a tone quality change, when the leaked signal is mixed in with other properly miked signals.

At low frequencies, *rumble* (high-level vibrations that occur in the 3- to 25-Hz region) can be easily introduced into the surface of a large unsupported floor space, studio or hall from any number of sources (such as passing trucks, air conditioners, subways or fans). They can be reduced or eliminated in a number of ways, such as:

- Using a shock mount to isolate the mic from the vibrating surface and floor stand.



FIGURE 4.18
Graphic representation of a typical cardioid pickup pattern.

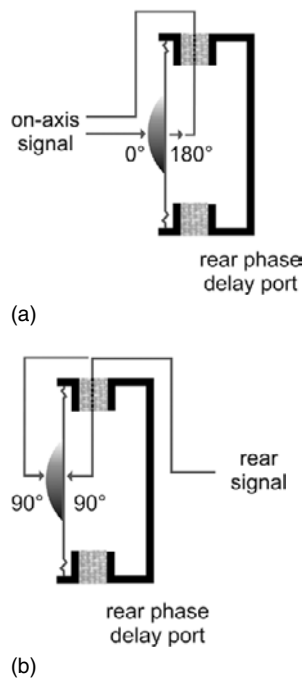
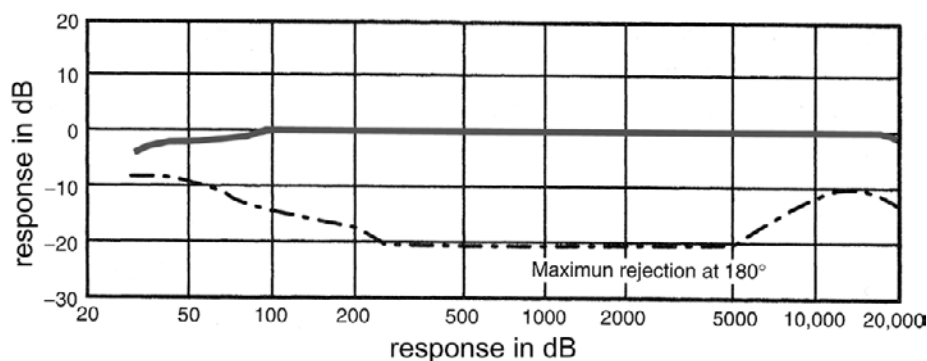
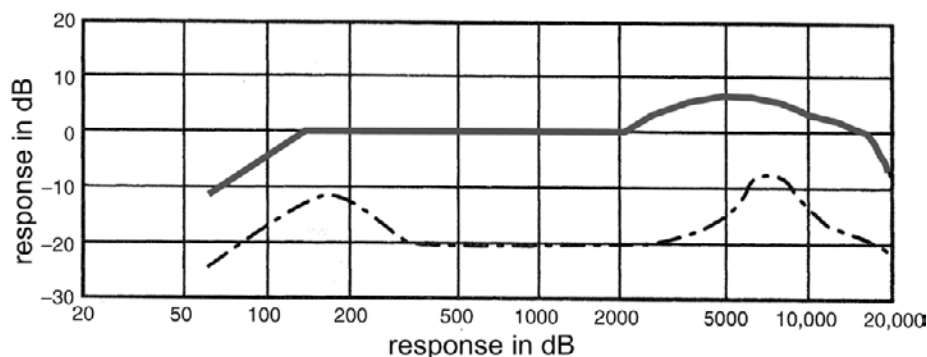


FIGURE 4.19
The directional properties of a cardioid microphone. (a) Signals arriving at the front (on-axis) of the diaphragm will produce a full output level. (b) Signals arriving at the rear of the diaphragm (180°) will cancel each other out, resulting in a greatly reduced output.



(a)



(b)

FIGURE 4.20

Frequency response curves: (a) response curve of the AKG C460B/CK61 ULS; (b) response curve of the AKG D321. (Courtesy of AKG Acoustics, Inc., www.akg-acoustics.com.)

- Choosing a mic that displays a restricted low-frequency response
- Restricting the response of a wide-range mic by using a low-frequency roll-off filter.

Another low-frequency phenomenon that occurs in most directional mics is known as *proximity effect*. This effect causes an increase in bass response whenever a directional mic is brought within 1 foot of the sound source. This bass boost (which is often most noticeable on vocals) proportionately increases as the distance decreases. To compensate for this effect (which is somewhat greater for bidirectional mics than for cardioids), a low-frequency roll-off filter switch (which is often located on the microphone body) can be used. If none exists, an external roll-off or equalizer can be used to reduce the low end. Any of these tools can be used to help restore the bass response to a flat and natural-sounding balance. Another way to reduce or eliminate proximity effect and its associated “popping” of the letters “p” and “b” is to replace the directional microphone with an omnidirectional mic when working at close distances. On a more positive note, this increase in bass response has long been appreciated by vocalists and DJs for their ability to give a full, “larger-than-life” quality to voices that are otherwise thin. In many cases, the use of a directional mic has become an important part of the engineer, producer and vocalist’s toolbox.



Tutorial: Proximity Effect

1. Pull out omnidirectional, cardioid and bidirectional mics (or one that can be switched between these patterns).
2. Move in on each mic pattern type from distances of 3 feet to 6 inches (being careful of volume levels and problems that can occur from popping).
3. Does the bass response increase as the distance is decreased with the cardioid? ... the bidirectional? ... the omni?

Transient response

A significant piece of data (which currently has no accepted standard of measure) is the *transient response* of a microphone (Figure 4.21). Transient response is the measure of how quickly a mic's diaphragm will react when it is hit by an acoustic wavefront. This figure varies wildly among microphones and is a major reason for the difference in sound quality among the three pickup types. For example, the diaphragm of a dynamic mic can be quite large (up to 2.5 inches). With the additional weight of the coil of wire and its core, this combination can be a very large mass when compared to the power of the sound wave that drives it. Because of this, a dynamic mic can be very slow in reacting to a waveform—often giving it a rugged, gutsy, and less accurate sound.

By comparison, the diaphragm of a ribbon mic is much lighter, so its diaphragm can react more quickly to a sound waveform, resulting in a clearer sound. The condenser pickup has an extremely light diaphragm, which varies in diameter from 2.5 inches to less than $\frac{1}{4}$ inch and has a thickness of about 0.0015 inch. This means that the diaphragm offers very little mechanical resistance to a sound-pressure wave, allowing it to accurately track the wave over the entire frequency range.

Output characteristics

A microphone's *output characteristics* refer to its measured sensitivity, equivalent noise, overload characteristics, impedance and other output responses.

SENSITIVITY RATING

A mic's *sensitivity rating* is the output level (in volts) that a microphone will produce, given a specific and standardized acoustic signal at its input (rated in dB SPL). This figure will specify the amount of amplification that's required to raise the mic's signal to line level (often referenced to -10 dBv or +4 dBm) and allows us to judge the relative output levels between any two mics. A

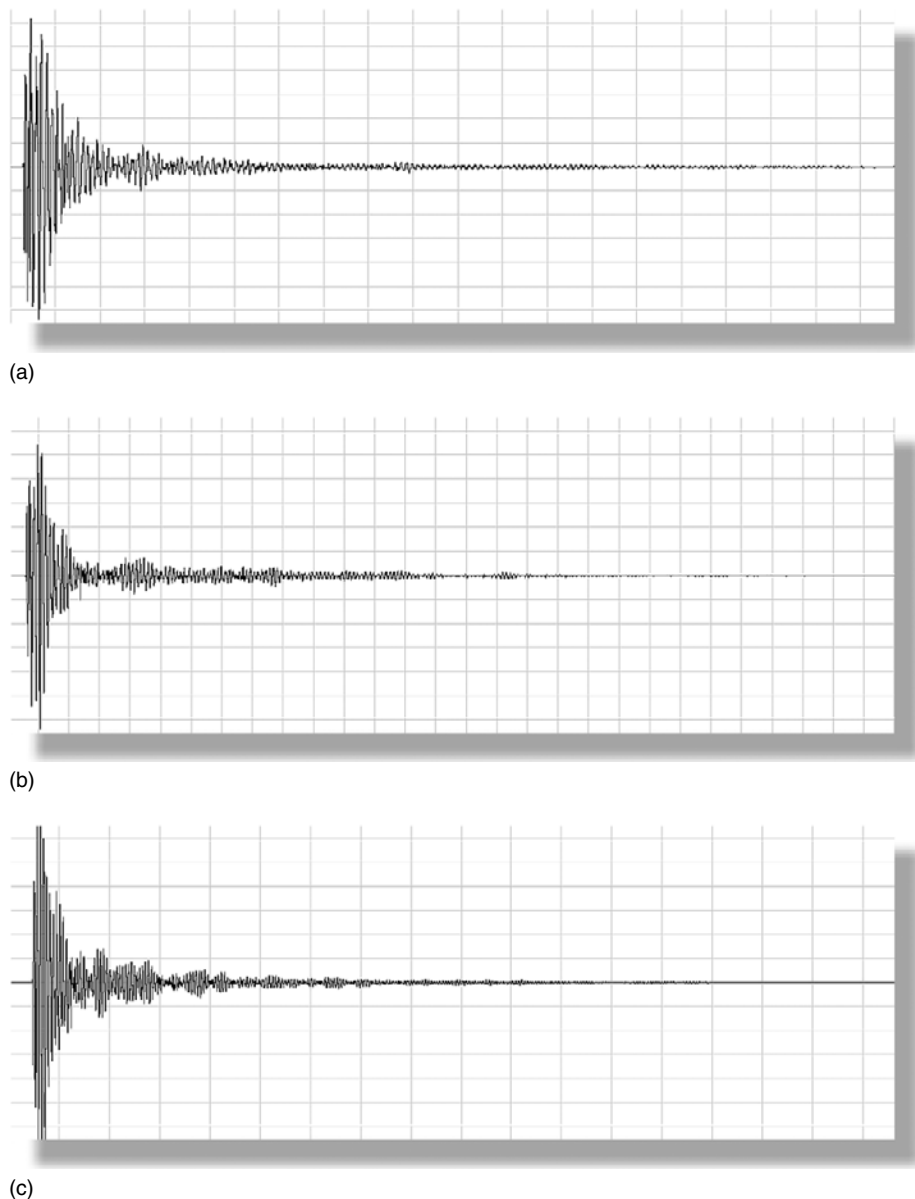


FIGURE 4.21
Transient response characteristics of a percussive wood-block using various microphone types: (a) Shure SM58 dynamic; (b) RCA 44BX ribbon; (c) AKG C3000 condenser.

microphone with a higher sensitivity rating will produce a stronger output signal voltage than one with a lower sensitivity.

EQUIVALENT NOISE RATING

The *equivalent noise rating* of a microphone can be viewed as the device's electrical self-noise. It is expressed in dB SPL or dBA (a weighted curve) as a signal

that would be equivalent to the mic's self-noise voltage. As a general rule, the mic itself doesn't contribute much noise to a system when compared to the mixer's amplification stages, the recording system or media (whether analog or digital). However, with recent advances in mic preamp/mixer technologies and overall reductions in noise levels produced by digital systems, these noise ratings have become increasingly important. Interestingly enough, the internal noise of a dynamic or ribbon pickup is actually generated by the electrons that move within the coil or ribbon itself. Most of the noise that's produced by a condenser mic is generated by the built-in preamp. It almost goes without saying that certain microphone designs will have a higher degree of self-noise than will others; thus, care should be taken in your microphone choices for critical applications (such as with distant classical recording techniques).

OVERLOAD CHARACTERISTICS

Just as a microphone is limited at low levels by its inherent self-noise, it's also limited at high sound-pressure levels (SPLs) by *overload distortion*. In terms of distortion, the dynamic microphone is an extremely rugged pickup, often capable of an overall dynamic range of 140 dB. Typically, a condenser microphone won't distort, except under the most severe sound-pressure levels; however, the condenser system differs from the dynamic in that at high acoustic levels the capsule's output might be high enough to overload the mic's preamplifier. To prevent this, most condenser mics offer a switchable attenuation pad that immediately follows the capsule output and serves to reduce the signal level at the preamp's input, thereby reducing or eliminating overload distortion. When inserting such an attenuation pad into a circuit, keep in mind that the mic's signal-to-noise ratio will be degraded by the amount of attenuation; therefore, it's always wise to remove the inserted pad when using the microphone under normal conditions.

MICROPHONE IMPEDANCE

Microphones are designed to exhibit different *output impedances*. Output impedance is a rating that's used to help you match the output resistance of one device to the rated input resistance requirements of another device (so as to provide the best-possible level and frequency response matching).

Impedance is measured in ohms (with its symbol being Ω or Z). The most commonly used microphone output impedances are 50, 150 and 250 Ω (low) and 20 to 50 k Ω (high). Each impedance range has its advantages. In the past, high-impedance mics were used because the input impedances of most tube-type amplifiers were high. A major disadvantage to using high-impedance mics is the likelihood that their cables will pick up electrostatic noise (like those caused by motors and fluorescent lights). To reduce such interference, a shielded cable is necessary, although this begins to act as a capacitor at lengths greater than 20 to 25 feet, which serves to short out much of the high-frequency information that's picked up by the mic. For these reasons, high-impedance microphones are rarely used in the professional recording process.

Most modern-day systems, on the other hand, are commonly designed to accept a low-impedance microphone source. The lines of very-low-impedance mics ($50\ \Omega$) have the advantage of being fairly insensitive to electrostatic pickup. They are, however, sensitive to induced hum pickup from electromagnetic fields (such as those generated by AC power lines). This extraneous noise can be greatly reduced through the use of a twisted-pair cable, because the interference that's magnetically induced into the cable will flow in opposite directions along the cable's length and will cancel out at the console or mixer's balanced microphone input stage. Mic lines of 150 to $250\ \Omega$ are less susceptible to signal losses and can be used with cable lengths of up to several thousand feet. They're also less susceptible to electromagnetic pickup than the $50\text{-}\Omega$ lines but are more susceptible to electrostatic pickup. As a result, most professional mics operate with an impedance of $200\ \Omega$, use a shielded twisted-pair cable and have reduced noise through the use of a balanced signal line.

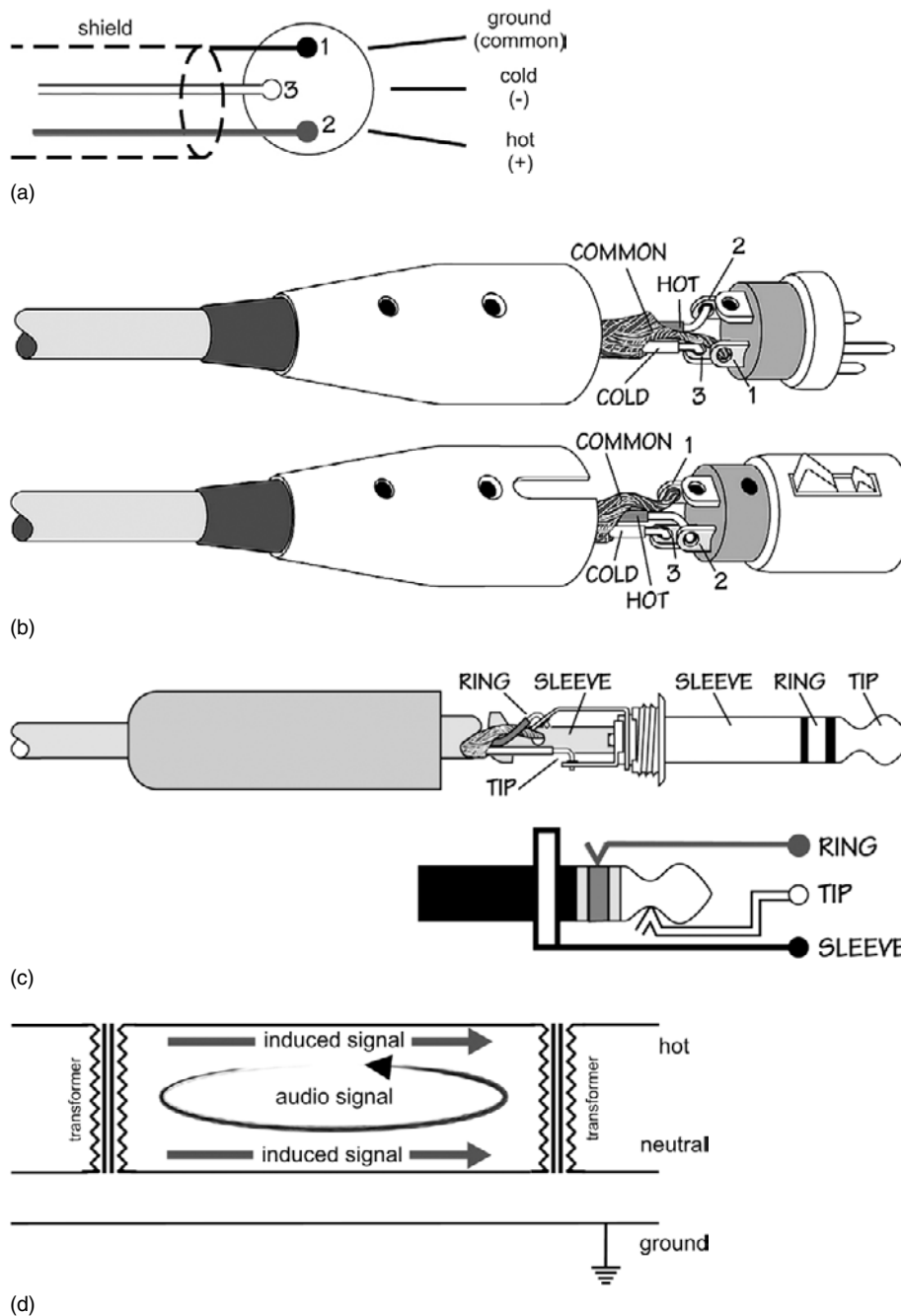
BALANCED/UNBALANCED LINES

In short, a *balanced line* uses three wires to properly carry the audio signal. Two of the wires are used to carry the signal voltage, while a third lead is used as a neutral ground wire. Since neither of the two signal conductors of a balanced line is directly connected to the signal ground, the alternating current of an audio signal will travel along the two independent wires. From a noise standpoint, whenever an electrostatic or electromagnetic signal is induced across the audio leads, it will be induced into both of the audio leads at an equal level (Figure 4.22). Since the input of a balance device will only respond to the alternating voltage potentials between the two leads, the unwanted noise (which is equal and opposite in polarity) will be canceled.

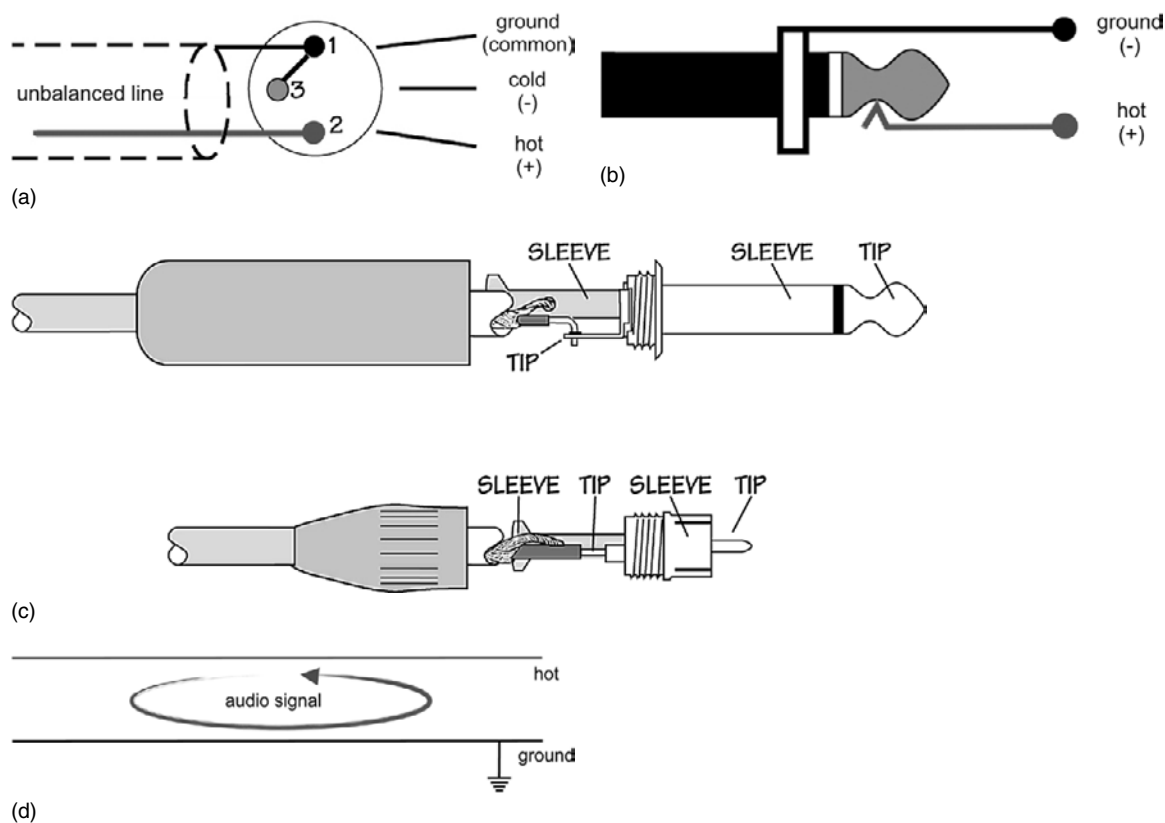
The standard that has been widely adopted for the proper polarity of two-conductor, balanced, XLR connector cables specifies pin 2 as being positive (+ or

hot) and pin 3 as being negative (– or neutral), with the cable ground being connected to pin 1.

If the hot and neutral pins of balanced mic cables are haphazardly pinned in a music or production studio, it's possible that any number of mics (and other equipment, for that matter) could be wired in opposite, out-of-phase polarities. For example, if a single instrument were picked up by two mics using two improperly phased cables, the instrument might totally or partially cancel when mixed to mono. For this reason, it's always wise to use a phase tester or volt-ohm meter to check the cable wiring throughout a pro or project studio complex.

**FIGURE 4.22**

Wiring detail of a balanced microphone cable (courtesy of Loud Technologies Inc., www.mackie.com): (a) diagram for wiring a balanced microphone (or line source) to a balanced XLR connector; (b) physical drawings; (c) diagram for wiring a balanced 1/4-inch phone connector; (d) equivalent circuit, where the induced signals travel down the wires in equal polarities that cancel at the transformer, whereby the AC audio signals are of opposing polarities that generate an output signal.

**FIGURE 4.23**

Unbalanced microphone circuit (courtesy of Loud Technologies Inc., www.mackie.com): (a) diagram for wiring an unbalanced microphone (or line source) to a balanced XLR connector; (b) diagram for wiring an unbalanced 1/4-inch phone connector; (c) physical drawings; (d) equivalent circuit.

High-impedance mics and most line-level instrument lines use *unbalanced lines* (Figure 4.23) to transmit signals from one device to another. In an unbalanced circuit, a single signal lead carries a positive current potential to a device, while a second, grounded shield (which is tied to the chassis ground) is used to complete the circuit's return path. When working at low signal levels (especially at mic levels), any noises, hums, buzzes or other types of interference that are induced into the signal path will be amplified along with the input signal.

MICROPHONE PREAMPS

Since the output signals of most microphones are at levels far too low to drive the line-level input stage of most recording systems, a mic preamplifier must be used to boost its signal to acceptable levels (often by 30 to 70 dB). With the advent of improved technologies in analog and digital console design, hard-disk

recorders, DAWs, signal processors and the like, low noise and distortion figures have become more important than ever. To many professionals, the stock mic pres (pronounced “preeze”) that are designed into many console types don’t have that special “sound,” aren’t high enough in quality to be used in critical applications or don’t have enough of a special, boutique cache for that special application. As a result, outboard mic preamps are chosen instead (Figures 4.24 through 4.27) for their low-noise, low-distortion specs and/or their unique sound. These devices might make use of tube, FET and/or integrated circuit technology, and offer advanced features in addition to the basic variable input gain, phantom power and high-pass filter controls. As with most recording tools, the sound, color scheme, retro style, tube or transistor type and budget level are up to the individual, the producer and the artist ... it’s totally a matter of personal style and taste. Note that mic pres have tapped into the growing



FIGURE 4.24
PreSonus TubePre.
(Courtesy of
PreSonus Audio
Electronics, www.presonus.com.)



FIGURE 4.25
Grace M101 mic
preamplifier.
(Courtesy of Grace
Design, www.gracedesign.com.)



FIGURE 4.26
Millennia HV-3D mic
preamp. (Courtesy of
Millennia Music &
Media Systems,
www.mil-media.com.)



FIGURE 4.27
Xlogic SuperAnalogue
mic amp. (Courtesy
of Solid State Logic,
www.solid-state-logic.com.)

market of those systems that are based around a DAW, which doesn't need a console or mixer but does require a quality pre (or set of pres) for plugging mic signals directly into the interface.

MICROPHONE TECHNIQUES

Most microphones have a distinctive sound character that's based on its specific type and design. A large number of types and models can be used for a variety of applications, and it's up to the engineer to choose the right one for the job. Over the years, I've come to the realization that there are two particular paths that one can take when choosing the types and models of microphones for a studio's production toolbox. These can basically be placed into the categories of:

- Selecting a limited range of mics that are well suited for a wide range of applications
- Acquiring a larger collection of mics that are commonly perceived as being individually suited for a particular instrument or situation.

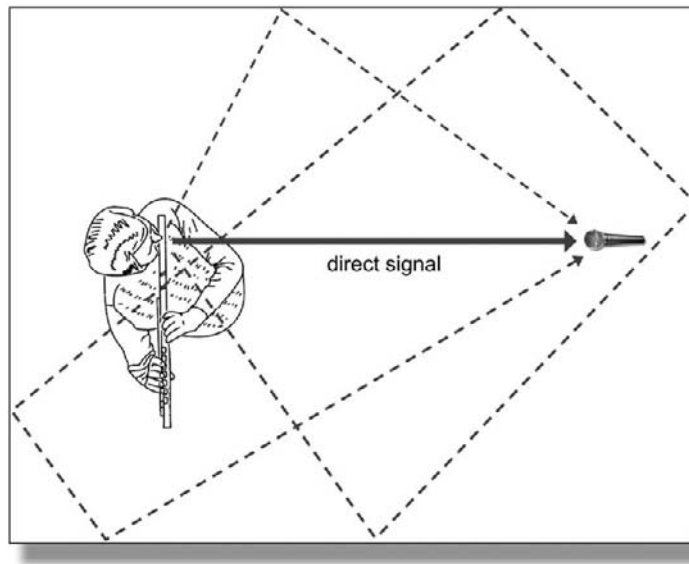
The first approach is ideal for the project studio and those who are just starting out and are on a limited budget. It is also common practice among seasoned professionals who swear by a limited collection of their favorite mics that are chosen to cover a wide range of applications. These dynamic and/or condenser mics can be used both in the project studio and in the professional studio to achieve the best possible sound on a budget.

The second approach (I often refer to it as the "Alan Sides" approach) is better suited to the professional studio (and to personal collectors) who actually have a need or desire to amass their own "dream collection" and offer it to their clients. In the end, both approaches have their merits. ... Indeed, it's usually wise to keep an open mind and choose a range of mic types that best fit your needs, budget and personal style.

Choosing the appropriate mic, however, is only half the story. The placement of a microphone will often play just as important a role, and is one of the engineer's most valued tools. Because mic placement is an art form, there is no right or wrong. Placement techniques that are currently considered "bad" might easily be the accepted as being standard practice five years from now ... and as new musical styles develop, new recording techniques will also tend to evolve, helping to breathe new life into music and production. The craft of recording should always be open to change and experimentation—two of the strongest factors that keep the music and the biz of music alive and fresh.

Pickup characteristics as a function of working distance

In studio and sound-stage recording, four fundamental styles of microphone placement are directly related to the working distance of a microphone from its sound source. These extremely important placement styles are as important as any tool in the toy box:

**FIGURE 4.28**

Example of an overall distant pickup.

- Distant miking
- Close miking
- Accent miking
- Ambient miking.

DISTANT MICROPHONE PLACEMENT

With distant microphone placement (Figure 4.28), one or more mics can be positioned at a distance of 3 feet or considerably more from the intended signal source. This technique (whose distance will vary with room and instrument size) will often yield the following results:

- It can pick up a large portion of a musical instrument or ensemble, thereby preserving the overall tonal balance of that source. Often, a natural tone balance can be achieved by placing the mic at a distance that's roughly equal to the size of the instrument or sound source.
- It allows the room's acoustic environment to be picked up (and naturally mixed in) with the direct sound signal.

Distant miking is often used to pick up large instrumental ensembles (such as a symphony orchestra or choral ensemble). In this application, the pickup will largely rely on the acoustic environment to help achieve a natural, ambient sound. The mic should be placed at a distance so as to strike an overall balance between the ensemble's direct sound and the room's acoustics, giving a balance that's determined by a number of factors, including the size of the sound source, its overall volume level, mic distance and placement and the reverberant characteristics of the room.

This technique tends to add a live, open feeling to a recorded sound; however, it could put you at a disadvantage if the acoustics of a hall, church or studio aren't particularly good. Improper or bad room reflections can create a muddy or poorly defined recording. To avoid this, the engineer might take one of the following actions:

- Temporarily correct for bad or excessive room reflections by using absorptive and/or offset reflective panels (to break up the problematic reflections).
- Place the mic closer to its source and add a degree of artificial ambience.

If a distant mic is used to pick up a portion of the room sound, placing it at a random height can result in a hollow sound due to phase cancellations that occur between the direct sound and delayed sounds that are reflected off the floor and other nearby surfaces (Figure 4.29). If these delayed reflections arrive at the mic at a time that's equal to one-half a wavelength (or at odd multiples thereof), the reflected signal will be 180° out of phase with the direct sound. This could produce dips in the signal's pickup response that could adversely

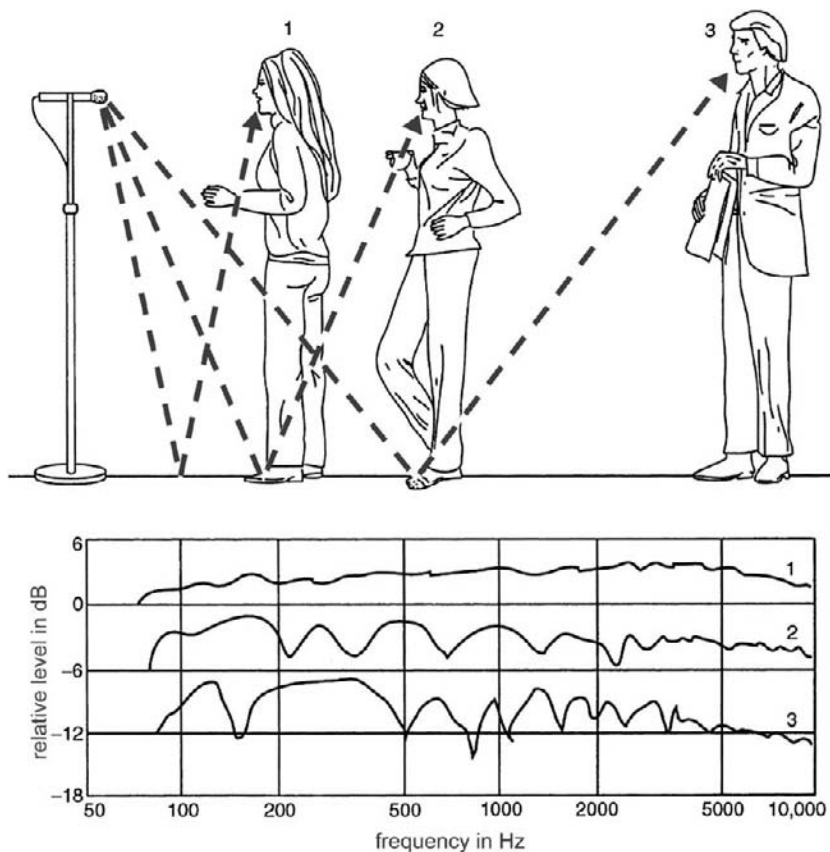


FIGURE 4.29
Resulting frequency response from a microphone that receives a direct and delayed sound from a single source.

color the signal. Since the reflected sound is at a lower level than the direct sound (as a result of traveling farther and losing energy as it bounces off a surface), the cancellation will only be partially complete. Raising the mic will have the effect of reducing reflections (due to the increased distances that the reflected sound must travel), while moving the mic close to the floor will conversely reduce the path length and raise the range in which the frequency cancellation occurs. In practice, a height of 1/8 to 1/16 inch will raise the cancellation above 10 kHz. One such microphone design type, known as a *boundary microphone* (Figures 4.30 and 4.31), places an electret-condenser or condenser diaphragm well within these low height restrictions. For this reason, this mic type might be a good choice for use as an overall distant pickup, when the mics need to be out of sight (i.e., when placed on a floor, wall or large boundary).

CLOSE MICROPHONE PLACEMENT

When a *close microphone* placement is used, the mic is often positioned about 1 inch to 3 feet from a sound source. This commonly used technique generally yields two results:

- It creates a tight, present sound quality.
- It effectively excludes the acoustic environment.

Because sound diminishes with the square of its distance from the sound source, a sound that originates 3 inches from the pickup will be much higher in level than one that originates 6 feet from the mic (Figure 4.32). Therefore, whenever *close miking* is used, only the desired on-axis sound will be recorded—extraneous, distant sounds (for all practical purposes) won't be picked up. In effect, the distant pickup will

FIGURE 4.30

The boundary microphone system.

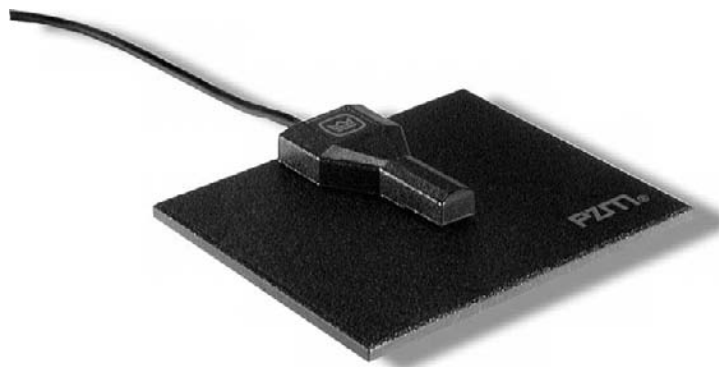
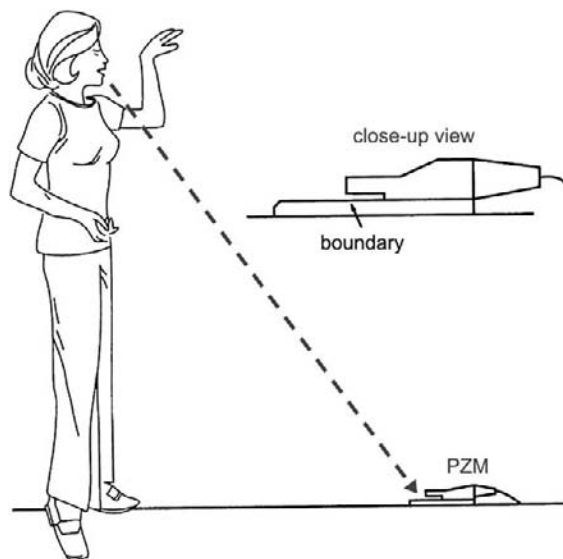


FIGURE 4.31

The PZM-6D boundary microphone. (Courtesy of Crown International, Inc., www.crownaudio.com.)

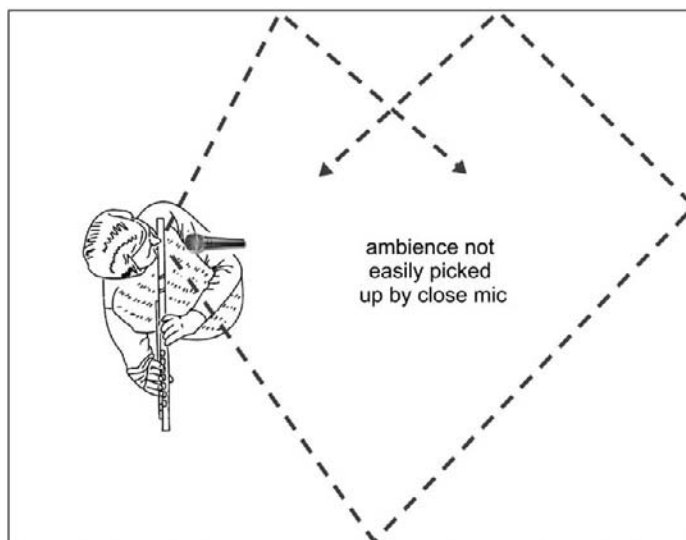


FIGURE 4.32
Close miking reduces the effects of the acoustic environment.

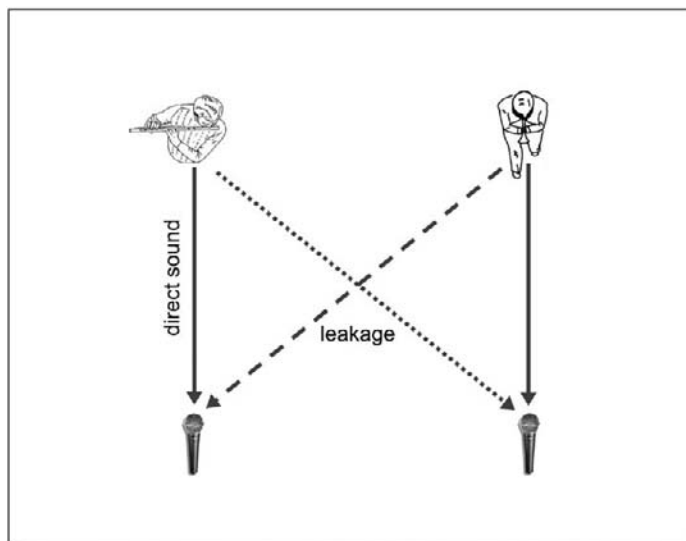


FIGURE 4.33
Leakage due to indirect signal pickup.

be masked by the closer sounds and/or will be reduced to a relative level that's well below the main pickup.

Whenever an instrument's mic also picks up the sound of a nearby instrument, a condition known as *leakage* occurs (Figure 4.33). Whenever a signal is picked up by both its intended mic and a nearby mic (or mics), it's easy to see how the signals could be combined together within the mixdown process. When

this occurs, level and phase cancellations often make it more difficult to have control over the volume and tonal character of the involved instruments within a mix.

To avoid the problems that can be associated with leakage, try the following:

- Place the mics closer to their respective instruments (Figure 4.34a).
- Use directional mics.
- Place an acoustic barrier (known as a flat, gobo, or divider) between the instruments (Figure 4.34b). Alternatively, mic/instruments can be surrounded on several sides by sound baffles and (if needed) a top can be draped over them.
- Spread the instruments farther apart.
- An especially loud (or quieter) instrument can be isolated by putting it in an unused iso-room or vocal or instrument booth. Electronic amps that are played at high volumes can also be recorded in such a room. An amp and the mic can be covered with a blanket or other flexible sound-absorbing material, so that there's a clear path between the amplifier and the mic.
- Separation can be achieved by plugging otherwise loud electronic instruments directly into the console via a direction injection (DI) box, thereby bypassing the miked amp.

Obviously, these examples can only suggest the number of possibilities that can occur during a session. For example, you might choose not to isolate the instruments and instead, place them in an acoustically “live” room. This approach will require that you carefully place the mics in order to control leakage; however, the result will often yield a live and present sound. As an engineer, producer and/or artist, the choices belong to you. Remember, the idea is to work out the kinks beforehand and to simplify technology as much as possible in the studio because Murphy's law is always alive and well in any production facility.

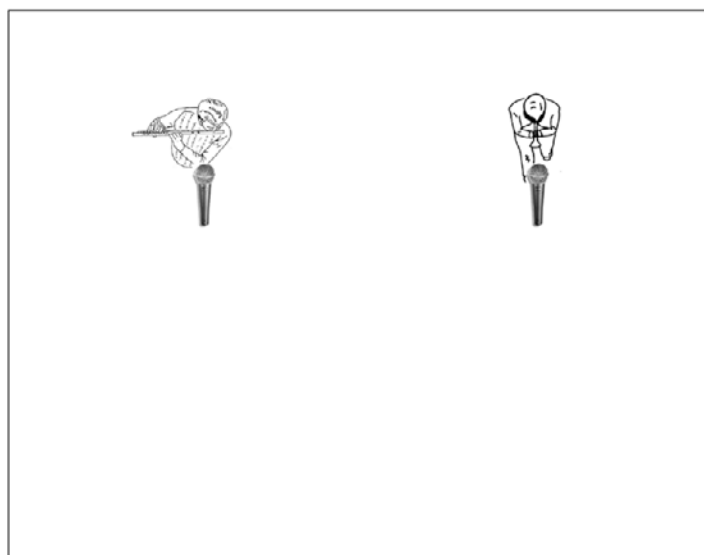
Whenever individual instruments are being miked close (or semi-close), it's generally wise to follow the *3:1 distance rule*.

3:1 Distance Rule

To reduce leakage and maintain phase integrity, this rule states that for every unit of distance between a mic and

its source, a nearby mic (or mics) should be separated by at least three times that distance (Figure 4.35).

Some err on the side of caution and avoid leakage even further by following a 5:1 distance rule. As always, experience will be your best teacher. Although the close miking of a sound source offers several advantages, a mic should be placed only as close to the source as is necessary, not as close as possible. Miking too

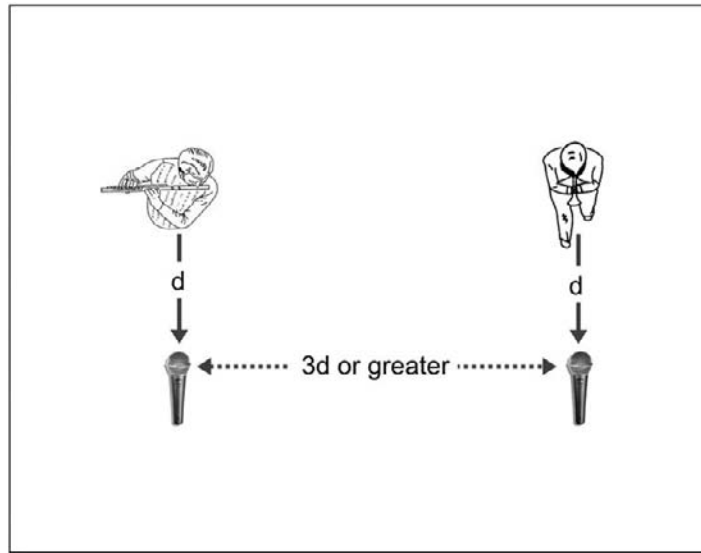


(a)



(b)

FIGURE 4.34 a-b
Two methods for reducing leakage:
(a) Place the microphones closer to their sources.
(b) Use an acoustic barrier to reduce leakage.

**FIGURE 4.35**

Example of the 3:1 microphone distance rule: “For every unit of distance between a mic and its source, a nearby mic (or mics) should be separated by at least three times that distance.”

close can color the recorded tone quality of a source, unless care is taken and careful experimentation is done.

It should be noted, however, that a bit of “bleed” (a slang word for leakage) between mics just might be a good thing. With semi-distant and even multiple mics that are closely spaced, the pickup of a source by several pickups can add a sense of increased depth and sonic space. Having an overall distant set of mics in the studio can add a dose of natural ambience that can actually help to “glue” a mix together. The concept of minute phase cancellations and leakage in a mix isn’t always something to be feared; it’s simply important that you be aware of the effects that it can have on a mix ... and use that knowledge to your advantage.

Because close mic techniques commonly involve distances of 1 to 6 inches, the tonal balance (timbre) of an entire sound source often can’t be picked up; rather, the mic might be so close to the source that only a small portion of the surface is actually picked up, giving it a tonal balance that’s very area specific (much like hearing the focused parts of an instrument through an acoustic microscope). At these close distances, moving a mic by only a few inches can easily change the pickup tonal balance. If this occurs, try using one or more of the following remedies:

- Move the microphone along the surface of the sound source until the desired balance is achieved.
- Place the mic farther back from the sound source to allow for a wider angle (thereby picking up more of the instrument’s overall sound).
- Change the mic.
- Equalize the signal until the desired balance is achieved.



Tutorial: Close Mic Experimentation

1. Mic an acoustic instrument (such as a guitar or piano) at a distance of 1 to 3 inches.
 2. Move (or have someone move) the mic over the instrument's body as it's being played, while listening
- to variations in the sound. Does the sound change? What are your favorite and least favorite positions?

In addition to all of the above considerations, the placement of musicians and instruments will often vary from one studio and/or session to the next because of the room, people involved, number of instruments, isolation (or lack thereof) among instruments, and the degree of visual contact that's needed for creative communication. If additional isolation (beyond careful microphone placement) is needed, flats and baffles can be placed between instruments in order to prevent loud sound sources from spilling over into other open mikes. Alternatively, the instrument or instruments could be placed into separate isolation (iso) rooms and/or booths, or they could be overdubbed at a later time.

During a session that involves several musicians, the setup should allow them to see and interact with each other as much as possible. It's extremely important that they be able to give and receive visual cues and otherwise "feel the vibe." The instrument/mic placement, baffle arrangement, and possibly room acoustics (which can often be modified by placing absorbers in the room) will depend on the engineer's and artists' personal preferences, as well as on the type of sound the producer wants.

ACCENT MICROPHONE PLACEMENT

Often, the tonal and ambient qualities will sound very different between a distant- and close-miked pickup. Under certain circumstances, it's difficult to obtain a naturally recorded balance when mixing the two together. For example, if a solo instrument within an orchestra needs an extra mic for added volume and presence, placing the mic too close would result in a pickup that sounds overly present, unnatural and out of context with the distant, overall orchestral pickup. To avoid this pitfall, a compromise in distance should be struck. A microphone that has been placed within a reasonably close range to an instrument or section within a larger ensemble (but not so close as to have an unnatural sound) is known as an *accent pickup* (Figure 4.36). Whenever accent miking is used, care should be exercised in placement and pickup choices. The amount of accent signal that's introduced into the mix should sound natural relative to the overall pickup, and a good accent mic should only add presence to a solo passage and not stick out as separate, identifiable pickup.

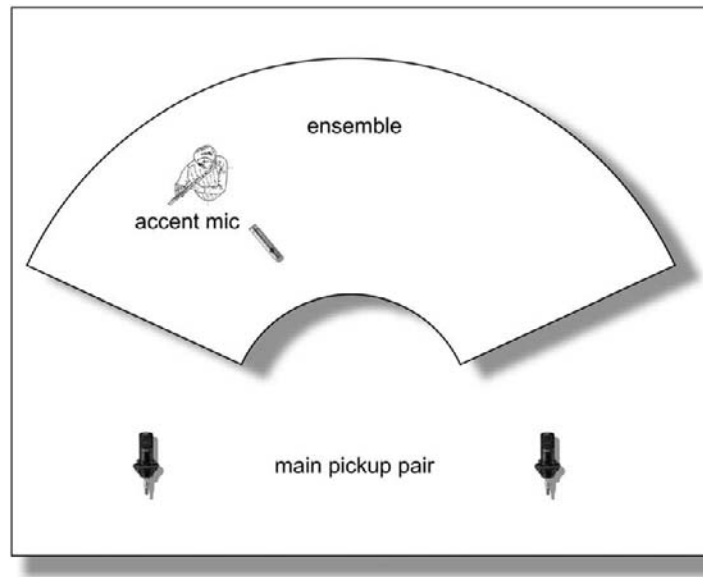


FIGURE 4.36
Accent microphone
placed at proper
compromise
distance.

AMBIENT MICROPHONE PLACEMENT

Ambient miking places the pickup at such a distance that the reverberant or room sound is equally or more prominent than the direct signal. The ambient pickup is often a cardioid stereo pair or crossed figure-8 (Blumlein) pair that can be mixed into a stereo or surround-sound production to provide a natural reverb and/or ambience. To enhance the recording, you can use ambient mic pickups in the following ways:

- In a live concert recording, ambient mics can be placed in a hall to restore the natural reverberation that is often lost with close miking techniques.
- In a live concert recording, ambient microphones can be placed over the audience to pick up their reaction and applause.
- In a studio recording, ambient microphones can be used in the studio to add a sense of space or natural acoustics back into the sound.



Tutorial: Ambient Miking

1. Mic an instrument or its amp (such as an acoustic or electric guitar) at a distance of 6 inches to 1 foot.
2. Place a stereo mic pair (in an X/Y and/or spaced configuration) in the room, away from the instrument.
3. Mix the two pickup types together. Does it “open” the sound up and give it more space? Does it muddy the sound up or breathe new life into it?

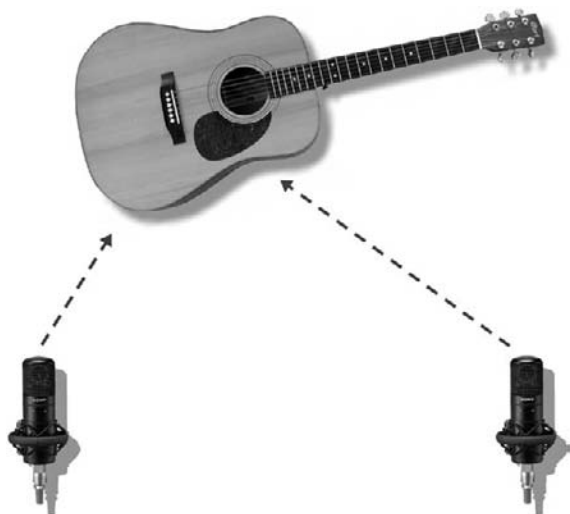


FIGURE 4.37
Spaced stereo miking technique.

Stereo miking techniques

For the purpose of this discussion, the term *stereo miking technique* refers to the use of two microphones in order to obtain a coherent stereo image. These techniques can be used in either close or distant miking of single instruments, vocals, large or small ensembles, within on-location or studio applications ... in fact, the only limitation is your imagination. The four fundamental stereo miking techniques are:

- Spaced pair
- X/Y
- M/S
- Decca tree.

SPACED PAIR

Spaced microphones (Figure 4.37) can be placed in front of an instrument or ensemble (in a left/right fashion) to obtain an overall stereo image. This technique places the two mics (of the same type, manufacturer and model) anywhere from only a few feet to more than 30 feet apart (depending on the size of the instrument or ensemble) and uses time and amplitude cues in order to create a stereo image. The primary drawback to this technique is the strong potential for phase discrepancies between the two channels due to differences in a sound's arrival time at one mic relative to the other. When mixed to mono, these phase discrepancies could result in variations in frequency response and even the partial cancellation of instruments and/or sound components in the pickup field.

X/Y

X/Y stereo miking is an intensity-dependent system that uses only the cue of amplitude to discriminate direction. With the X/Y coincident-pair technique (Figure 4.38), two directional microphones of the same type, manufacture and model are placed with their grills as close together as possible (without touching) and facing at angles to each other (generally between 90° and 135°). The midpoint between the two mics is pointed toward the source, and the mic outputs are equally panned left and right. Even though the two mics are placed together, the stereo imaging is excellent—often better than that of a spaced pair. In addition, due to their proximity, no appreciable phase problems arise. Most commonly, X/Y pickups use mics that have a cardioid polar pattern, although the Blumlein technique is being increasingly used. This technique (which is named after the unheralded inventor, Alan Dower Blumlein) uses two crossed bidirectional mics that are offset by 90° to each other. This simple technique often yields excellent ambient results for the pickup of the overall ambience within a studio or concert hall, while also being a good choice for picking up sources that are placed “in the round.”



FIGURE 4.38
X/Y stereo miking technique using an X/Y crossed cardioid pair.

Stereo microphones that contain two diaphragms in the same case housing are also available on the new and used market. These mics are either fixed (generally in a 90° or switchable X/Y pattern) or are designed so that the top diaphragm can be rotated by 180° (allowing for the adjustment of various coincident X/Y angles).

M/S

Another coincident-pair system, known as the M/S (or mid-side) technique (Figure 4.39), is similar to X/Y in that it uses two closely spaced, matched pickups. The M/S method differs from the X/Y method, however, in that it requires the use of an external transformer, active matrix, or software plug-in in order to work. In the classic M/S stereo miking configuration, one of the microphone capsules is designated the M (mid) position pickup and is generally a cardioid pickup pattern that faces forward, toward the sound source. The S (side) capsule is generally chosen as a figure-8 pattern that's oriented sideways (90° and 270°) to the on-axis pickup (i.e., with the null facing to the side, away from the cardioid's main axis). In this way, the mid capsule picks up the direct sound, while the side figure-8 capsule picks up ambient and reverberant sound. These outputs are then combined through a sum-and-difference decoder matrix either electrically (through a transformer matrix) or mathematically (through a digital M/S plug-in), which then resolves them into a conventional X/Y stereo signal: ($M + S = \text{left}$) and ($M - S = \text{right}$).

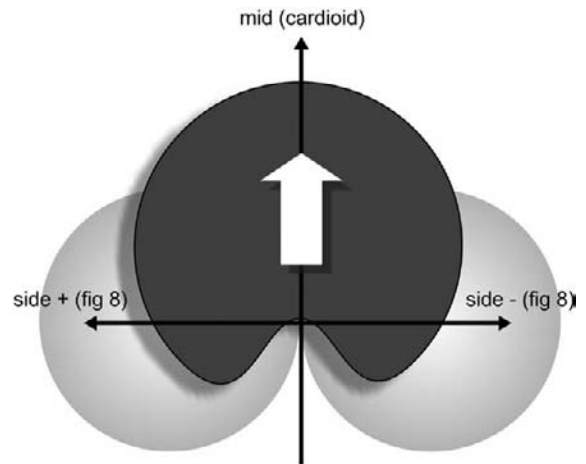


FIGURE 4.39
M/S stereo microphone technique.

One advantage of this technique is its absolute monaural compatibility. When the left and right signals are combined, the sum of the output will be $(M + S) + (M - S) = 2M$. That's to say, the side (ambient) signal will be canceled, but the mid (direct) signal will be accentuated. Since it is widely accepted that a mono signal loses its intelligibility with added reverb, this tends to work to our advantage. Another amazing side benefit of using M/S is the fact that it lets us continuously vary the mix of mid (direct) to side (ambient) sound that's being picked up either during the recording (from the console location) ... or even at a later time during mixdown, after it's been recorded! These are both possible by simply mixing the ratio of mid to side that's being sent to the decoder matrix (Figure 4.40). In a mixdown scenario, all that's needed is to record the mid on one track and the side on another. (It's often best to use a digital recorder, because phase delays associated with the analog recording process can interfere with decoding.) During mixdown, routing the M/S tracks



(a)

FIGURE 4.40

M/S decoder matrix:
(a) AEA MS-38 Mark II dual-mode stereo width controller and Matrix MS processor (courtesy of Audio Engineering Associates, www.ribbonmics.com);
(b) Waves S1 Stereo Imager plug-in includes True Blumlein shuffling and MS/LR processing (courtesy of Waves, www.waves.com).



(b)

**FIGURE 4.41**

Decca tree
microphone array.
(Courtesy of Audio
Engineering
Associates, [www.
ribbonmics.com](http://www.ribbonmics.com).)

to the decoder matrix allows you to make important decisions regarding stereo width and depth at a later, more controlled date.

DECCA TREE

Although not as commonly used as the preceding stereo techniques, the *Decca tree* is a time-tested, classical miking technique that uses both time and amplitude cues in order to create a coherent stereo image. Attributed originally to Decca engineers Roy Wallace and Arthur Haddy in 1954, the Decca tree (Figure 4.41) consists of three omnidirectional mics (originally, Neumann M50 mics were used). In this arrangement, a left and right mic pair is placed 3 feet apart, and a third mic is placed 1.5 feet out in front and panned in the center of the stereo field. Still favored by many in orchestral situations as a main pickup pair, the Decca tree is most commonly placed on a tall boom, above and behind the conductor. According to lore, when Haddy first saw the array, he remarked, “It looks like a bloody Christmas tree!” The name stuck.

Surround miking techniques

With the advent of 5.1 surround-sound production, it’s certainly possible to make use of a surround console or DAW to place sources that have been recorded in either mono or stereo into a surround image field. Under certain situations, it’s also possible to consider using multiple-pickup surround miking techniques in order to capture the actual acoustic environment and then translate that into a surround mix. Just as the number of techniques and personal styles increases when miking in stereo compared to mono, the number of placement and technique choices will likewise increase when miking a source in surround. Although guidelines have been and will continue to be set, both placement and mixing styles are definitely an art and not a science.

AMBIENT SURROUND MICS

A relatively simple, yet effective way to capture the surround ambience of a live or studio session is to simply place a spaced or coincident mic pair out in the studio at a distance from the sound source. These can be facing toward or away from the sound source, and placement is totally up to experimentation. During a surround mixdown, placing distant mics into the studio or hall can work wonders to add a sense of space to an ensemble group, drum set or instrument overdub.



Tutorial: Ambient Surround Mics

1. Mic an instrument or ensemble group using traditional close pickup techniques.
2. Place a spaced or Blumlein pickup pair in the room at a considerable distance from the source.
3. Record the ambient mics to a spare set of tracks and place them into a stereo mix.
4. If you're lucky enough to be surround-capable, place the ambient tracks to the rear. Does it add an extra dimension of space? Does it alter the recording's definition?

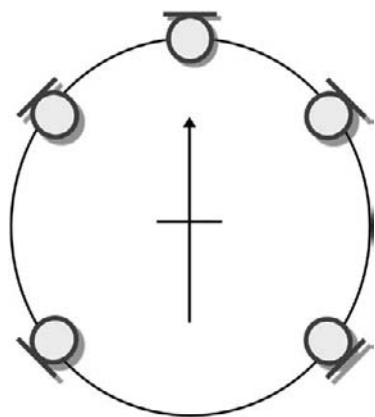


FIGURE 4.42
Five cardioid microphones can be arranged in a circular pattern (with the center microphone facing toward the source) to create a modified, mini-surround Decca tree. A four-microphone (quad) approach to surround miking can be easily made by simply eliminating the center pickup.

SURROUND DECCA TREE

One of the most logical techniques for capturing an ensemble or instrument in a surround setting places five mics onto a modified Decca tree. This ingenious and simple system adds two rear-facing mics to the existing three-mic Decca tree system. Another simpler approach is to place five cardioid mics in a circle, such that the center channel faces toward the source, thereby creating a simple setup that can be routed L-C-R-SL-SR (Figure 4.42).

One last approach (which doesn't actually fall under the Decca tree category) involves the use of four cardioid mics that are spaced at 90° angles, representing L-R-SL-SR, with the on-axis point being placed 45° between the L and R mics. This "quad" configuration can be easily made by mounting the mics on two stereo bars that are offset by 90°. Note that Samson's H2 handheld flash memory recorder uses four mics to affordably and simply record in this fashion.

RECORDING DIRECT

As an alternative, the signal of an electric or electronic instrument (guitar, keyboard, etc.) can be directly "injected" into a console, recorder or DAW without the use of a microphone. This option often produces a cleaner, more present sound by bypassing the distorted components of a head/amp combination. It also reduces leakage into other mics by eliminating room sounds. In the project or recording studio, the *direct injection* (DI) box (Figure 4.43) serves to interface an instrument with an analog output signal to a console or recorder in the following ways:

- It reduces an instrument's line-level output to mic level for direct insertion into the console's mic input jack.
- It changes an instrument's unbalanced, high-source impedance line to a balanced, low-source impedance signal that's needed by the console's input stage.

- It often can electrically isolate the audio signal paths between the instrument and mic/line preamp stages (thereby reducing the potential for ground-loop hum and buzzes).

Most commonly, the instrument's output is plugged directly into the DI box (where it's stepped down in level and impedance), and the box's output is then fed into the mic pre of a console or DAW. If a "dirtier" sound is desired, certain boxes will allow high-level input signals to be taken directly from the amp's speaker output jack. It's also not uncommon for an engineer, producer and/or artist to combine the punchy, full sound of a mic with the present crispness of a direct sound. These signals can then be combined onto a single tape track or recorded to separate tracks (thereby giving more flexibility in the mixdown stage). The ambient image can be "opened up" even further by mixing a semi-distant or distant mic (or stereo pair) with the direct (and even with the close miked amp) signal. This ambient pickup can be either mixed into a stereo field or at the rear of a surround field to fill out the sound.

When recording a guitar, the best tone and lowest hum pickup for a direct connection occurs when the instrument volume control is fully turned up. Because guitar tone controls often use a variable treble roll-off, leaving the tone controls at the treble setting and using a combination of console EQ and different guitar pickups to vary the tone will often yield the maximum amount of control over the sound. Note that if the treble is rolled off at the guitar, boosting the highs with EQ will often increase pickup noise.

REAMPING IT IN THE MIX

Another way to alter the sound of a recorded track or to inject a new sense of acoustic space into an existing take is to reamp a track. The "reamp" process (originally conceived in 1993 by recording engineer John Cuniberti; www.reamp.com) lets us record a guitar's signal directly to a track using a DI during the recording session and then play this cleanly recorded track back through a miked guitar amp/speaker, allowing it to be re-recorded to new tracks at another time (Figure 4.44).

The re-recording of an instrument that has been recorded directly gives us total flexibility for changing the final, recorded amp and mic sound at a later time. For example, it's well known that it's far easier to add an effect to a "dry" track that doesn't have effects during mixdown than to attempt to remove an effect after it's been printed to track. Whenever reamping is used at a later time, it's



FIGURE 4.43
Radial JDI passive direct box. (Courtesy of Radial Engineering, www.radialeng.com.)

FIGURE 4.44

Example of how a direct recording can be “reamped” in a studio, allowing for complete tonal, mic placement, and acoustical control ... after the fact! (Courtesy of John Cuniberti, www.reamp.com.)



possible to audition any number of amps, using any number of effects and/or mic settings, until the desired sound has been found. This process allows the musician to concentrate solely on getting the best recorded performance, without having to spend extra time getting the perfect guitar, amp, mic and room sound. Leakage problems in the studio are also reduced, because no mikes are used in the process.

Although the concept of recording an instrument directly and playing the track back through a miked amp at a later time is relatively new, the idea of using a room's sound to fill out the sound of a track or mix isn't. The reamp concept takes this idea a bit further by letting you go as wild as you like. For example, you could use the process to re-record a single, close-miked guitar amp and then go back and layer a larger stack at a distance. An electronic guitarist could take the process even further by recording his or her MIDI guitar both directly and to a sequenced MIDI track. In this way, the reamp and patch combinations would be virtually unlimited.

MICROPHONE PLACEMENT TECHNIQUES

The following sections are meant to be used as a general guide to mic placement for various acoustic and popular instruments. It's important to keep in mind that these are only guidelines. Several general application and characteristic notes are detailed in Table 4.1, and descriptions of several popular mics are

Table 4.1 Microphone selection guidelines.

| Needed Application | Required Microphone Choice and/or Characteristic |
|---|--|
| Natural, smooth tone quality | Flat frequency response |
| Bright, present tone quality | Rising frequency response |
| Extended lows | Dynamic or condenser with extended low-frequency response |
| Extended highs (detailed sound) | Condenser |
| Increased “edge” or midrange detail | Dynamic |
| Extra ruggedness | Dynamic or modern ribbon/condenser |
| Boosted bass at close working distances | Directional microphone |
| Flat bass response up close | Omnidirectional microphone |
| Reduced leakage, feedback, and room acoustics | Directional microphone, or omnidirectional microphone at close working distances |
| Enhanced pickup of room acoustics | Place microphone or stereo pair at greater working distances |
| Reduced handling noise | Omnidirectional, vocal microphone, or directional microphone with shock mount |
| Reduced breath popping | Omnidirectional or directional microphone with pop filter |
| Distortion-free pickup of very loud sounds | Dynamic or condenser with high maximum SPL rating |
| Noise-free pickup of quiet sounds | Condenser with low self-noise and high sensitivity |

outlined toward the end in the “Microphone Selection” section to help give insights into placement and techniques that might work best in a particular application.

As a general rule, choosing the best mic for an instrument or vocal will ultimately depend on the sound you’re searching for. For example, a dynamic mic will often yield a “rugged” or “punchy” character (which is often further accentuated by the proximity of bass boost that’s generally associated with a directional mic). A ribbon mic will often yield a mellow sound that ranges from being open and clear to slightly “croony” ... depending on the type and distances involved. Condenser mics are often characterized as having a clear, present and full-range sound that varies with mic design, grill options and capsule size. Before jumping into this section, I’d like to again take time to point out the “Good Rule” to anyone who wants to be a better engineer, producer and/or musician:

As a rule, starting with an experienced, rehearsed and ready musician who has a quality instrument that's well tuned is the best insurance toward getting the best possible sound. Let's think about this for a moment. Say that we have a live rhythm session that involves drums, piano, bass guitar and scratch vocals. All of the players are the best around, except for the drummer, who is new to the studio process. Unfortunately, you've now signed on to teach the drummer the ropes of proper drum tuning, studio interaction and playing under pressure. It goes without saying that the session might go far less smoothly than it otherwise would, as you'll have to take the extra time to work with the player to get the best possible sound. Once you're rolling, it'll also be up to you or the producer to pull a professional performance out of someone who's new to the field.

Don't get me wrong, musicians have to start somewhere ... but an experienced studio musician who comes into the studio with a great instrument that's tuned and ready to go (and who might even clue you in on some sure-fire mic and placement techniques for the instrument) is simply a joy from a sound, performance, time and budget-saving standpoint. Simply put, if you and/or the project's producer have prepared enough to get all your "goods" lined up, the track will have a much better chance of being something that everyone can be proud of. Just as with the art of playing an instrument, careful mic choice, placement and "style" in the studio are also subjective ... and are a few of the fundamental calling cards of a good engineer. Experience simply comes with time and the willingness to experiment. Be patient, learn, listen and have fun ... and you too will eventually rise to the professional occasion.

Brass instruments

The following sections describe many of the sound characteristics and miking techniques that are encountered in the brass family of instruments.

TRUMPET

The fundamental frequency of a trumpet ranges from E3 to D6 (165 to 1175 Hz) and contains overtones that stretch upward to 15 kHz. Below 500 Hz, the sounds emanating from the trumpet project uniformly in all directions; above 1500 Hz, the projected sounds become much more directional; and above 5 kHz, the dispersion emanates at a tight 30° angle from in front of the bell. The formants of a trumpet (the relative harmonic and resonance frequencies that give an instrument its specific character) lie at around 1 to 1.5 kHz and at 2 to 3 kHz. Its tone can be radically changed by using a mute (a cup-shaped dome that fits directly over the bell), which serves to dampen frequencies above 2.5 kHz. A conical mute (a metal mute that fits inside the bell) tends to cut back on frequencies below 1.5 kHz while encouraging frequencies above 4 kHz. Because of the high sound-pressure levels that can be produced by a trumpet (up to 130 dB SPL), it's best to place a mic slightly off the bell's center at a distance of 1 foot or more (Figure 4.45). When closer placements are needed, a -10- to -20-dB pad can help prevent input overload at the mic or console

preamp input. Under such close working conditions, a windscreen can help protect the diaphragm from windblasts.

TROMBONE

Trombones come in a number of sizes; however, the most commonly used “bone” is the tenor, which has a fundamental note range spanning from E2 to C5 (82 to 523 Hz) and produces a series of complex overtones that range from 5 kHz (when played medium loud) to 10 kHz (when overblown). The trombone’s polar pattern is nearly as tight as the trumpet’s: Frequencies below 400 Hz are distributed evenly, whereas its dispersion angle increases to 45° from the bell at 2 kHz and above. The trombone most often appears in jazz and classical music. The *Mass in C Minor* by Mozart, for example, has parts for soprano, alto, tenor and bass trombones. This style obviously lends itself to the spacious blending that can be achieved by distant pickups within a large hall or studio. On the other hand, jazz music often calls for closer miking distances. At 2 to 12 inches, for example, the trombonist should play slightly to the side of the mic to reduce the chance of overload and wind blasts. In the miking of a trombone section, a single mic might be placed between two players, acoustically combining them onto a single channel and/or track.



FIGURE 4.45
Typical microphone placement for a single trumpet.

TUBA

The bass and double-bass tubas are the lowest pitched of the brass/wind instruments. Although the bass tuba’s range is actually a fifth higher than the double bass, it’s still possible to obtain a low fundamental of B (31 Hz). A tuba’s overtone structure is limited; its top response ranges from 1.5 to 2 kHz. The lower frequencies (around 75 Hz) are evenly dispersed; however, as frequencies rise, their distribution angles reduce. Under normal conditions, this class of instruments isn’t miked at close distances. A working range of 2 feet or more, slightly off-axis to the bell, will generally yield the best results.

FRENCH HORN

The fundamental tones of the French horn range from B1 to B5 (62 to 700 Hz). Its “oo” formant gives it a round, broad quality that can be found at about 340 Hz, with other frequencies falling between 750 Hz and 3.5 kHz. French horn players often place their hands inside the bell to mute the sound and promote a formant at about 3 kHz. A French horn player or section is traditionally placed at the rear of an ensemble, just in front of a rear, reflective stage wall.

This wall serves to reflect the sound back toward the listener's position (which tends to create a fuller, more defined sound). An effective pickup of this instrument can be achieved by placing an omni- or bidirectional pickup between the rear, reflecting wall and the instrument bells, thereby receiving both the direct and reflected sound. Alternatively, the pickups can be placed in front of the players, thereby receiving only the sound that's being reflected from the rear wall.

Guitar

The following sections describe the various sound characteristics and techniques that are encountered when miking the guitar.

ACOUSTIC GUITAR

The popular steel-strung, acoustic guitar has a bright, rich set of overtones (especially when played with a pick). Mic placement and distance will often vary from instrument to instrument and may require experimentation to pick up the best tonal balance. A balanced pickup can often be achieved by placing the mic (or an X/Y stereo pair) at a point slightly off-axis and above or below the sound hole at a distance of between 6 inches and 1 foot (Figure 4.46). Condenser mics are often preferred for their smooth, extended frequency response and excellent transient response. The smaller-bodied classical guitar is normally strung with nylon or gut and is played with the fingertips, giving it a warmer, mellower sound than its steel-strung counterpart. To make sure that the instrument's full range is picked up, place the mic closer to the center of the bridge, at a distance of between 6 inches and 1 foot.

Miking near the sound hole

The sound hole (located at the front face of a guitar) serves as a bass port, which resonates at the lower frequencies (around 80 to 100 Hz). Placing a mic too



FIGURE 4.46
Typical microphone
placement for the
guitar.

close to the front of this port might result in a boomy and unnatural sound; however, miking close to the sound hole is often popular 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).

Surround guitar miking

An effective way to translate an acoustic guitar to the wide stage of surround (if a big, full sound is what you're after) is to record the guitar using X/Y or spaced techniques stereo (panned front L/R) ... and pan the guitar's electric pickup (or added contact pickup) to the rear center of the surround field. Extra ambient surround mics can also be used in an all-acoustic session.

THE ELECTRIC GUITAR

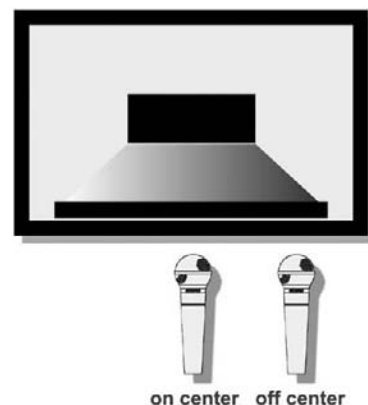
The fundamentals of the average 22-fret guitar extend from E2 to D6 (82 to 1174 Hz), with overtones that extend much higher. All of these frequencies might not be amplified, because the guitar chord tends to attenuate frequencies above 5 kHz (unless the guitar has a built-in low impedance converter or low-impedance pickups). The frequency limitations of the average guitar loud-speaker often add to this effect, because their upper limit is generally restricted to below 5 or 6 kHz.

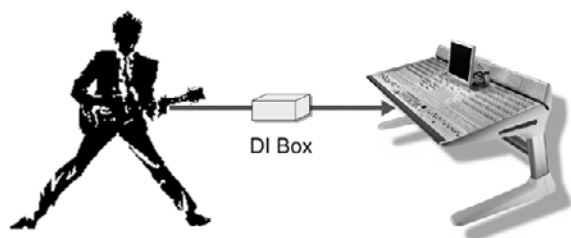
Miking the guitar amp

The most popular guitar amplifier used for recording is a small practice-type amp/speaker system. These high-quality amps often help the guitar's suffering high end by incorporating a sharp rise in the response range at 4 to 5 kHz, thus helping to give it a clean, open sound. High-volume, wall-of-sound speaker stacks are less commonly used in a session, because they're harder to control in the studio and in a mix. By far the most popular mic type for picking up an electric guitar amp is the cardioid dynamic. A dynamic tends to give the sound a full-bodied character without picking up extraneous amplifier noises. Often guitar mics will have a pronounced presence peak in the upper frequency range, giving the pickup an added clarity. For increased separation, a microphone can be placed at a working distance of 2 inches to 1 foot. When miking at a distance of less than 4 inches, mic/speaker placement becomes slightly more critical (Figure 4.47). For a brighter sound, the mic should face directly into the center of the speaker's cone. Placing it off the cone's center tends to produce a more mellow sound while reducing amplifier noise.

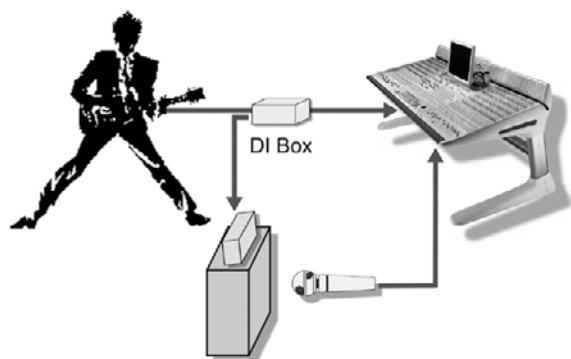
Isolation cabinets have also come onto the market that are literally sealed boxes that house a speaker or guitar amp/cabinet system, as well as an internal mic mount. These systems are used to reduce leakage and to provide greater control over instrument levels within a recording studio or control room during a session.

FIGURE 4.47
Miking an electric guitar cabinet directly in front of and off-center to the cone.





(a)



(b)

FIGURE 4.48
Direct recording of
an electric guitar:
(a) direct recording;
(b) combined direct
and miked signal.

Recording direct

A DI box is often used to feed the output signal of an electric guitar directly into the mic input stage of a recording console or mixer. By routing the direct output signal to a track, a cleaner, more present sound can be recorded (Figure 4.48a). This technique also reduces the leakage that results from having a guitar amp in the studio and even makes it possible for the guitar to be played in the control room or project studio. A combination of direct and miked signals often results in a sound that adds the characteristic fullness of a miked amp to the extra “bite” that a DI tends to give. These may be combined onto a single track or, whenever possible, can be assigned to separate tracks, allowing for greater control during mixdown (Figure 4.48b). During an overdub, the ambient image can be “opened up” even further by mixing a semidistant or distant mic (or stereo pair) with the direct mic (and even with the close miked amp signal). This ambient pickup can be either mixed into a stereo field or at the rear of a surround field to fill out the sound.

THE ELECTRIC BASS GUITAR

The fundamentals of an electric bass guitar range from about E1 to F4 (41.2 to 343.2 Hz). If it’s played loudly or with a pick, the added harmonics can range upward to 4 kHz. Playing in the “slap” style or with a pick gives a brighter, harder attack, while a “fingered” style will produce a mellower tone. In modern music production, the bass guitar is often recorded direct for the cleanest possible sound. As with the electric guitar, the electric bass can be either miked at the amplifier or picked up through a DI box. If the amp is miked, dynamic mics usually are chosen for their deep, rugged tones. The large-diaphragm dynamic designs tend to subdue the high-frequency transients. When combined with a boosted response at around 100 Hz, these large diaphragm dynamics give a warm, mellow tone that adds power to the lower register. Equalizing a bass can sometimes increase its clarity, with the fundamental being affected from 125 to 400 Hz and the harmonic punch being from 1.5 to 2 kHz. A compressor is commonly used on electric and acoustic basses. It’s a basic fact that the signal output from the instrument’s notes often varies in level, causing some notes to stand out while others dip in volume. A compressor having a smooth input/output ratio of roughly 4:1, a fast attack (8 to 20 milliseconds), and a slower release time ($\frac{1}{4}$ to $\frac{1}{2}$ second) can often smooth out these levels, giving the instrument a strong, present and smooth bass line.

Keyboard instruments

The following sections describe the various sound characteristics and techniques that are encountered when miking keyboard instruments.

GRAND PIANO

The grand piano is an acoustically complex instrument that can be miked in a variety of ways, depending on the style and preferences of the artist, producer and/or engineer. The overall sound emanates from the instrument's strings, soundboard and mechanical hammer system. Because of its large surface area, a minimum miking distance of 4 to 6 feet is needed for the tonal balance to fully develop and be picked up; however, leakage from other instruments often means that these distances aren't practical or possible. As a result, pianos are often miked at distances that favor such instrument parts as:

- *Strings and soundboard*, often yielding a bright and relatively natural tone
- *Hammers*, generally yielding a sharp, percussive tone
- *Soundboard holes alone*, often yielding a sharp, full-bodied sound.

In modern music production, two basic grand piano styles can be found in the recording studio: the concert grand, which traditionally has a rich and full-bodied tone (often used for classical music and ranging in size up to 9 feet in length), and the studio grand, which is more suited for modern music production and has a sharper, more percussive edge to its tone (often being about 7 feet in length).

Figure 4.49 shows a number of miking positions that can be used in recording a grand piano. Although several mic positions are illustrated, it's important to

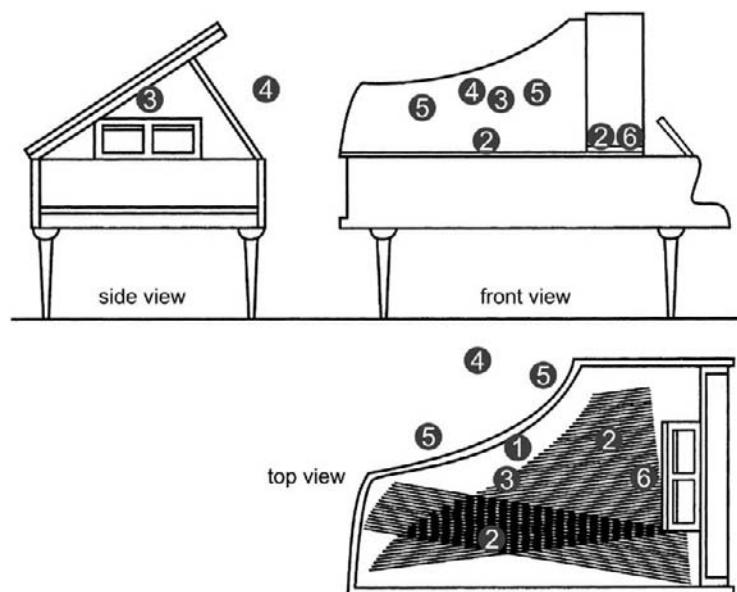


FIGURE 4.49
Possible miking combinations for the grand piano.

keep in mind that these are only guidelines from which to begin. Your own personal sound can be achieved through mic choice and experimentation with mic placement.

Position 1: The mic is attached to the partially or entirely open lid of the piano.

The most appropriate choice for this pickup is the boundary mic, which can be permanently attached or temporarily taped to the lid. This method uses the lid as a collective reflector and provides excellent pickup under restrictive conditions (such as on stage and during a live video shoot).

Position 2: Two mics are placed in a spaced stereo configuration at a working distance of 6 inches to 1 inch. One mic is positioned over the low strings and one is placed over the high strings.

Position 3: A single mic or coincident stereo pair is placed just inside the piano between the soundboard and its fully or partially open lid.

Position 4: A single mic or stereo coincident pair is placed outside the piano, facing into the open lid (this is most appropriate for solo or accent miking).

Position 5: A spaced stereo pair is placed outside the lid, facing into the instrument.

Position 6: A single mic or stereo coincident pair is placed just over the piano hammers at a working distance of 4 to 8 inches to give a driving pop or rock sound.

A condenser or extended-range dynamic mic is most often the preferred choice when miking an acoustic grand piano, as those types of mics tend to accurately represent the transient and complex nature of the instrument. Should excessive leakage be a problem, a close-miked cardioid (or cardioid variation) can be used; however, if leakage isn't a problem, backing away to a compromise distance (3 to 6 feet) can help capture the instrument's overall tonal balance.

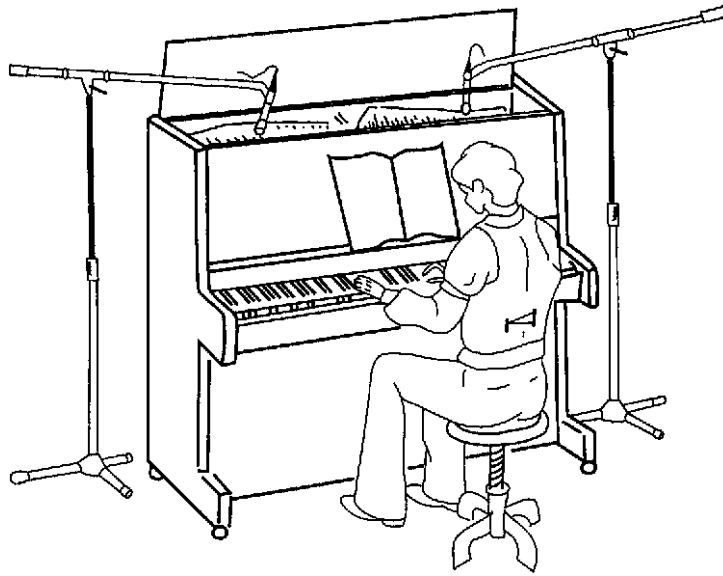
Separation

Separation is often a problem associated with the grand piano whenever it is placed next to noisy neighbors. Separation, when miking a piano, can be achieved in the following ways:

- Place the piano inside a separate isolation room.
- Place a flat (acoustic separator) between the piano and its louder neighbor.
- Place the mics inside the piano and lower the lid onto its short stick. A heavy moving or other type of blanket can be placed over the lid to further reduce leakage.
- Overdub the instrument at a later time. In this situation, the lid can be removed or propped up by the long stick, allowing the mics to be placed at a more natural-sounding distance.

UPRIGHT PIANO

You would expect the techniques for this seemingly harmless piano type to be similar to those for its bigger brother. This is partially true. However, because

**FIGURE 4.50**

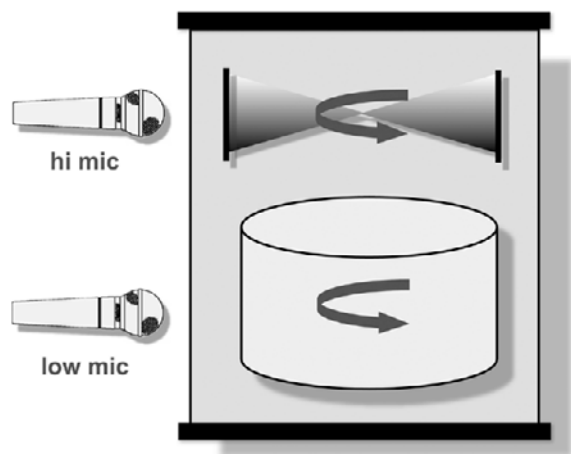
One possible pickup combination places the mics over the top of an upright piano.

this instrument was designed for home enjoyment and not performance, the mic techniques are often very different. Since it's often more difficult to achieve a respectable tone quality when using an upright, you might want to try the following methods (Figure 4.50):

- *Miking over the top:* Place two mics in a spaced fashion just over and in front of the piano's open top, with one over the bass strings and one over the high strings. If isolation isn't a factor, remove or open the front face that covers the strings in order to reduce reflections and, therefore, the instrument's characteristic "boxy" quality. Also, to reduce resonances you might want to angle the piano out and away from any walls.
- *Miking the kickboard area:* For a more natural sound, remove the kickboard at the lower front part of the piano to expose the strings. Place a stereo spaced pair over the strings (one each at a working distance of about 8 inches over the bass and high strings). If only one mic is used, place it over the high-end strings. Be aware, though, that this placement can pick up excessive foot-pedal noise.
- *Miking the upper soundboard area:* To reduce excessive hammer attack, place a microphone pair at about 8 inches from the soundboard, above both the bass and high strings. In order to reduce muddiness, the soundboard should be facing into the room or be moved away from nearby walls.

ELECTRONIC KEYBOARD INSTRUMENTS

Signals from most electronic instruments (such as synthesizers, samplers and drum machines) are often taken directly from the device's line-level output(s) and inserted into a console, either through a DI box or directly into a channel's



(a)

FIGURE 4.51

A Leslie speaker cabinet creates a unique vibrato effect by using a set of rotating speaker baffles that spin on a horizontal axis. (a) Miking the rotating speakers of a Leslie cabinet; (b) modern portable rotary amp with built-in microphones and three XLR outputs. (Courtesy of Motion Sound, www.motion-sound.com.)



(b)

line-level input. Alternatively, the keyboard's output can be plugged directly into the recorder or interface line-level inputs. The approach to miking an electronic organ can be quite different from the techniques just mentioned. A good Hammond or other older organ can sound wonderfully "dirty" through miked loudspeakers. Such organs are often played through a Leslie cabinet (Figure 4.51), which adds a unique, Doppler-based vibrato. Inside the cabinet is a set of rotating speaker baffles that spin on a horizontal axis and, in turn, produce a pitch-based vibrato as the speakers accelerate toward and away from the mics. The upper high-frequency speakers can be picked up by either one or two mics (each panned left and right), with the low-frequency driver being picked up by one mic. Motor and baffle noises can produce quite a bit of wind, possibly creating the need for a windscreen and/or experimentation with placement.

Percussion

The following sections describe the various sound characteristics and techniques that are encountered when miking drums and other percussion instruments.

DRUM SET

The standard drum kit (Figure 4.52) is often at the foundation of modern music, because it provides the "heartbeat" of a basic rhythm track; consequently, a

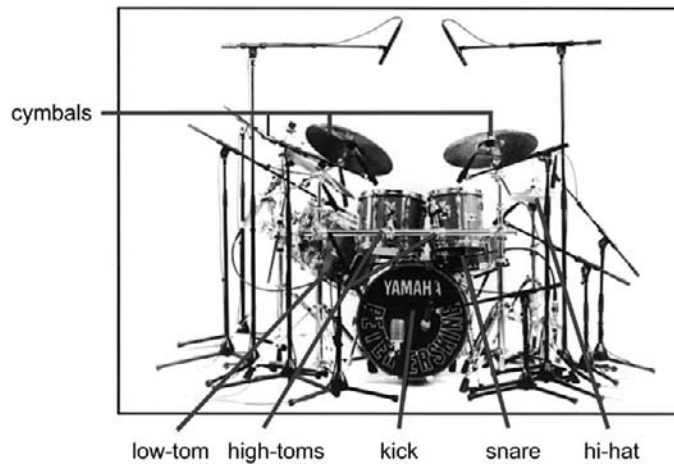


FIGURE 4.52
Peter Erskine's studio
drum kit. (Courtesy
of Beyerdynamic,
www.beyerdynamic.com.)

proper drum sound is extremely important to the outcome of most music projects. Generally, the drum kit is composed of the kick drum, snare drum, high-toms, low-tom (one or more), hi-hat and a variety of cymbals. Since a full kit is a series of interrelated and closely spaced percussion instruments, it often takes real skill to translate the proper spatial and tonal balance into a project. The larger-than-life driving sound of the acoustic rock drum set that we've all become familiar with is the result of an expert balance among playing techniques, proper tuning and mic placement.

During the past several decades, drums have undergone a substantial change with regard to playing technique, miking technique and choice of acoustic recording environment. In the 1960s and 1970s, the drum set was placed in a small isolation room called a drum booth. This booth acoustically isolated the instrument from the rest of the studio and had the effect of tightening the drum sound because of the limited space (and often dead acoustics). The drum booth also physically isolated the musician from the studio, which often caused the musician to feel removed and less involved in the action. Today, many engineers and producers have moved the drum set out of smaller iso-rooms and back into larger open studio areas where the sound can fully develop and combine with the studio's own acoustics. In many cases, this effect can be exaggerated by placing a distant mic pair in the room (a technique that often produces a fuller, larger-than-life sound, especially in surround).

Before a session begins, the drummer should tune each drum while the mics and baffles for the other instruments are being set up. Each drumhead should be adjusted for the desired pitch and for constant tension around the rim by hitting the head at various points around its edge and adjusting the lugs for the same pitch all around the head. Once the drums are tuned, the engineer should listen to each drum individually to make sure that there are no buzzes, rattles, or resonant after-rings. Drums that sound great in live performance may not sound nearly as good when being close miked. In a live performance, the rattles

and rings are covered up by the other instruments and are lost before the sound reaches the listener. Close miking, on the other hand, picks up the noises as well as the desired sound.

If tuning the drums doesn't bring the extraneous noises or rings under control, duct or masking tape can be used to dampen them. Pieces of cloth, dampening rings, paper towels, or a wallet can also be taped to a head in various locations (which is determined by experimentation) to eliminate rings and buzzes. Although head damping has been used extensively in the past, present methods use this damping technique more discreetly and will often combine dampening with proper design and tuning styles (all of which are the artist's personal call).

During a session, it's best to remove the damping mechanisms that are built into most drum sets, because they apply tension to only one spot on the head and unbalance its tension. These built-in dampeners often vibrate when the head is hit and are a chief source of rattles. Removing the front head and placing a blanket or other damping material inside the drum (so that it's pressing against the head) can often dampen the kick drum. Adjusting the amount of material can vary the sound from being a resonant boom to a thick, dull thud. Kick drums are usually (but not always) recorded with their front heads removed, while other drums are recorded with their bottom heads either on or off. Tuning the drums is more difficult if two heads are used because the head tensions often interact; however, they will often produce a more resonant tone. After the drums have been tuned, the mikes can be put into position. It's important to keep the mics out of the drummer's way, or they might be hit by a stick or moved out of position during the performance.

Miking the drum set

After the drum set has been optimized for the best sound, the mics can be placed into their pickup positions (Figure 4.53). Because each part of the drum set is so different in sound and function, it's often best to treat each grouping as an individual instrument. In its most basic form, the best place to start when miking a drum set is to start with the fundamental "groups." These include

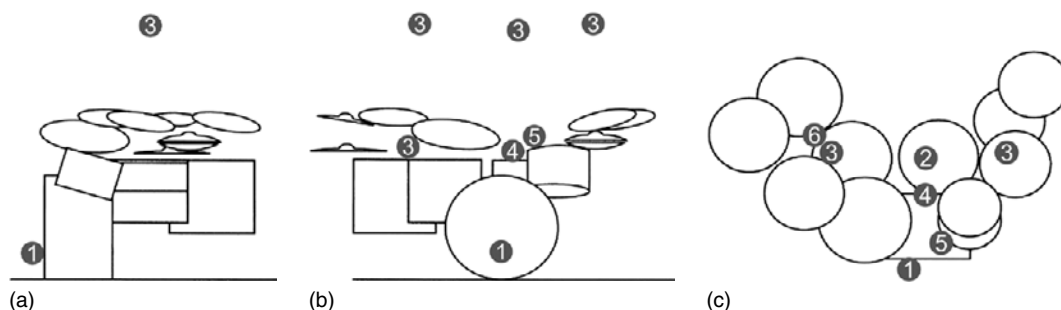


FIGURE 4.53

Typical microphone placements for a drum set: (a) side view; (b) front view; (c) top view.

placing a mic on the kick (1) and on the snare drum (4). At an absolute minimum, the entire drum set can be adequately picked up using only four mics by adding two overhead pickups, either spaced (3) or coincident (4). In fact, this “bare bones” placement was (and continues to be) commonly used on many classic jazz recordings. If more tracks are available (or required), additional mics can be placed on the various toms, hi-hat and even individual cymbals.

A mic’s frequency response, polar response, proximity effect and transient response should be taken into account when matching it to the various drum groups. Dynamic range is another important consideration when miking drums. Since a drum set is capable of generating extremes of volume and power (as well as softer, more subtle sounds), the chosen mics must be able to withstand strong peaks without distorting, and yet still be able to capture the more delicate nuances of a sound.

Since the drum set usually is one of the loudest sound sources in a studio setting, it’s often wise to place it on a solidly supported riser. This reduces the amount of low-end “thud” that can otherwise leak through the floor into other parts of the studio. Depending on the studio layout, the following drum scenarios may occur:

- The drums could be placed in their own room, isolated from other instruments.
- To achieve a bigger sound, the drums could be placed in the large studio room while the other instruments are placed in smaller iso-rooms or are recorded direct.
- To reduce leakage, the drums could be placed in the studio, while being enclosed by 4-foot (or higher) divider flats.

Kick drum

The kick drum adds a low-energy drive or “punch” to a rhythm groove. This drum has the capability to produce low frequencies at high sound-pressure levels, so it’s necessary to use a mic that can both handle and faithfully reproduce these signals. Often the best choice for the job is a large-diaphragm dynamic mic. Since proximity effect (bass boost) occurs when using a directional mic at close working distances and because the drum’s harmonics vary over its large surface area, even a minor change in placement can have a profound effect on the pickup’s overall sound. Moving the mic closer to the head (Figure 4.54) can add a degree of warmth and fullness, while moving it farther back often emphasizes the high-frequency “click.” Placing the mic closer to the beater emphasizes the hard “thud” sound, whereas an off-center pickup captures more of the drum’s characteristic skin tone. A dull and loose kick sound can be tightened to produce a sharper, more defined transient sound by placing a blanket or other damping material inside the drum shell firmly against the beater head. Cutting back on the kick’s equalization at 300 to 600 Hz can help reduce the dull “cardboard” sound, whereas boosting from 2.5 to 5 kHz adds

FIGURE 4.54

Placing the microphone at a distance just outside the kick drumhead to bring out the low end and natural fullness.

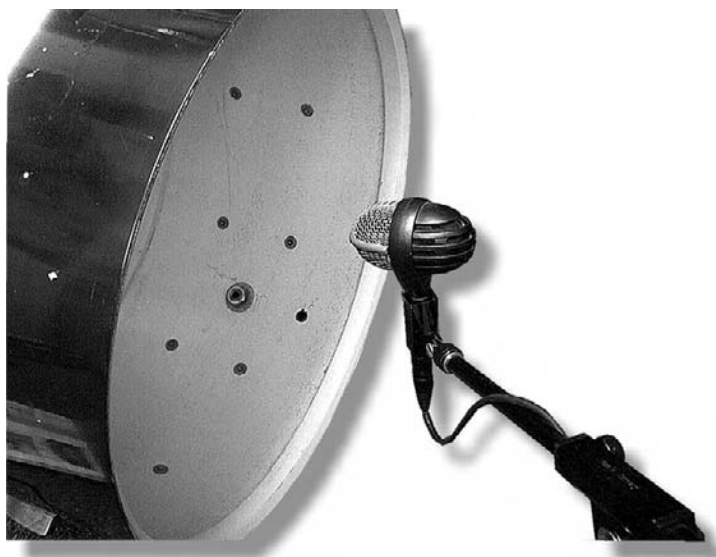


FIGURE 4.55

Typical microphone positioning for the snare drum.



a sharper attack, “click” or “snap.” It’s also often a good idea to have a can of WD-40® or other light oil handy in case squeaks from some of the moving parts (most often the kick pedal) gets picked up by the mics.

Snare drum

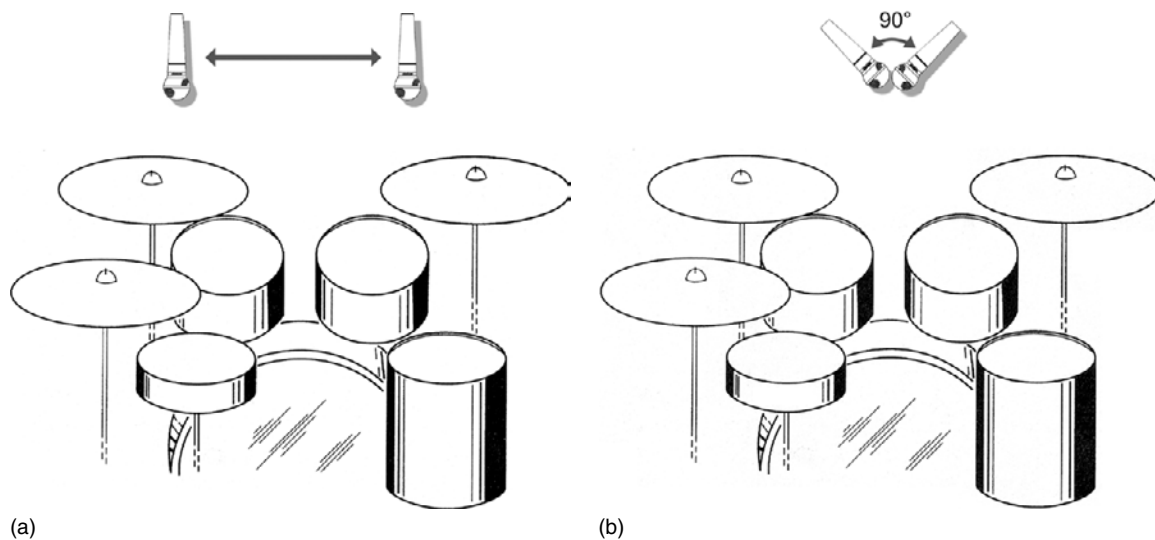
Commonly, a snare mic is aimed just inside the top rim of the snare drum at a distance of about 1 inch (Figure 4.55). The mic should be angled for the best

possible separation from other drums and cymbals. Its rejection angle should be aimed at either the hi-hat or rack-toms (depending on leakage difficulties). Usually, the mic's polar response is cardioid, although bidirectional and super-cardioid responses might offer a tighter pickup angle. With certain musical styles (such as jazz), you might want a crisp or "bright" snare sound. This can be achieved by placing an additional mic on the snare drum's bottom head and then combining the two mics onto a single track. Because the bottom snare head is 180° out of phase with the top, it's generally a wise idea to reverse the bottom mic's phase polarity. When playing in styles where the snare springs are turned off, it's also wise to keep your ears open for snare rattles and buzzes that can easily leak into the snare mic (as well as other mics). The continued ringing of an "open" snare note (or any other drum type, for that matter) can be dampened in several ways. Dampening rings, which can be purchased at music stores, are used to reduce the ring and to deepen the instrument's tone. If there are no dampening rings around, the tone can be dampened by taping a billfold or similar-sized folded paper towel to the top of a drumhead, a few inches off its edge.

Overheads

Overhead mics are generally used to pick up the high-frequency transients of cymbals with crisp, accurate detail while also providing an overall blend of the entire drum kit. Because of the transient nature of cymbals, a condenser mic is often chosen for its accurate high-end response. Overhead mic placement can be very subjective and personal. One type of placement is the spaced pair, whereby two mics are suspended above the left and right sides of the kit. These mics are equally distributed about the L/R cymbal clusters so as to pick up their respective instrument components in a balanced fashion (Figure 4.56a). Another

FIGURE 4.56
Typical stereo
overhead pickup
positions: (a) spaced
pair technique;
(b) X/Y coincident
technique.



placement method is to suspend the mics closely together in a coincident fashion (Figure 4.56b). This often yields an excellent stereo overhead image with a minimum of the phase cancellations that might otherwise result when using spaced mics. Again, it's important to remember that there are no rules for getting a good sound. If only one overhead mic is available, place it at a central point over the drums. If you're using a number of pickups to close mic individual components of a kit, there might be times when you won't need overheads at all (the leakage spillover just might be enough to do the trick).

Rack-toms

The upper rack-toms can be miked either individually (Figure 4.57) or by placing a single mic between the two at a short distance (Figure 4.58). When miked individually, a "dead" sound can be achieved by placing the mic close to the drum's top head (about 1 inch above and 1 to 2 inches in from the outer rim). A sound that's more "live" can be achieved by increasing the height above the head to about 3 to 6 inches. If isolation or feedback is a consideration, a hypercardioid pickup pattern can be chosen. Another way to reduce leakage and to get a deep, driving tone (with less attack) is to remove the tom's bottom head and place the mic inside, 1 to 6 inches away from the top head.

Floor-tom

Floor-toms can be miked similarly to the rack-toms (Figure 4.59). The mic can be placed 2 to 3 inches above the top and to the side of the head, or it can be placed inside 1 to 6 inches from the head. Again, a single mic can be placed above and between the two floor-toms, or each can have its own mic pickup (which often yields a greater degree of control over panning and tonal color).



FIGURE 4.57
Individual miking of a
rack-tom.

**FIGURE 4.58**

Single microphone placement for picking up two toms.

**FIGURE 4.59**

Typical microphone placement for the floor-tom.

Hi-hat

The “hat” usually produces a strong, sibilant energy in the high-frequency range, whereas the snare’s frequencies often are more concentrated in the midrange. Although moving the hat’s mic won’t change the overall sound as much as it would on a snare, you should still keep the following three points in mind:

- Placing the mic above the top cymbal will help pick up the nuances of sharp stick attacks.
- The open and closing motion of the hi-hat will often produce rushes of air; consequently, when miking the hat’s edge, angle the mic slightly above or below the point where the cymbals meet.

- If only one mic is available (or desired), both the snare and hi-hat can be simultaneously picked up by carefully placing the mic between the two, facing away from the rack-toms as much as possible. Alternatively, a figure-8 mic can be placed between the two with the null axis facing toward the cymbals and the kick.

TUNED PERCUSSION INSTRUMENTS

The following sections describe the various sound characteristics and techniques that are encountered when miking tuned percussion instruments.

Congas and hand drums

Congas, tumbas and bongos are single-headed, low-pitched drums that can be individually miked at very close distances of 1 to 3 inches above the head and 2 inches in from the rim, or the mics can be pulled back to a distance of 1 foot for a fuller, “live” tone. Alternatively, a single mic or X/Y stereo pair can be placed at a point about 1 foot above and between the drums (which are often played in pairs). Another class of single-headed, low-pitched drums (known as hand drums) isn’t necessarily played in pairs but is often held in the lap or strapped across the player’s front. Although these drums can be as percussive as congas, they’re often deeper in tone and often require that the mic(s) be backed off in order to allow the sound to develop and/or fully interact with the room. In general, a good pickup can be achieved by placing a mic at a distance of 1 to 3 feet in front of the hand drum’s head. Since a large part of the drum’s sound (especially its low-end power) comes from its back hole, another mic can be placed at the lower port at a distance of 6 inches to 2 feet. Since the rear sound will be 180° out of phase from the front pickup, the mic’s phase should be reversed whenever the two signals are combined.

Xylophone, vibraphone and marimba

The most common way to mic a tuned percussion instrument is to place two high-quality condenser or extended-range dynamic pickups above the playing bars at a spaced distance that’s appropriate to the instrument size (following the 3:1 general rule). A coincident stereo pair can help eliminate possible phase errors; however, a spaced pair will often yield a wider stereo image.

Stringed instruments

Of all the instrumental families, stringed instruments are perhaps the most diverse. Ethnic music often uses instruments that range from being single stringed to those that use highly complex and developed systems to produce rich and subtle tones. Western listeners have grown accustomed to hearing the violin, viola, cello and double bass (both as solo instruments and in an ensemble setting). Whatever the type, stringed instruments vary in their design type and in construction to enhance or cut back on certain harmonic frequencies. These variations are what give a particular stringed instrument its own characteristic sound.

VIOLIN AND VIOLA

The frequency range of the violin runs from 196 Hz to above 10 kHz. For this reason, a good mic that displays a relatively flat frequency response should be used. The violin's fundamental range is from G3 to E6 (196 to 1300 Hz), and it is particularly important to use a mic that's flat around the formant frequencies of 300 Hz, 1 kHz, and 1200 Hz. The fundamental range of the viola is tuned a fifth lower and contains fewer harmonic overtones. In most situations, the violin or viola's mic should be placed within 45° of the instrument's front face. The distance will depend on the particular style of music and the room's acoustic condition. Miking at a greater distance will generally yield a mellow, well-rounded tone, whereas a closer position might yield a scratchy, more nasal quality ... the choice will depend on the instrument's tone quality. The recommended miking distance for a solo instrument is between 3 and 8 feet, over and slightly in front of the player (Figure 4.60). Under studio conditions, a closer mic distance of between 2 and 3 feet is recommended. For a fiddle or jazz/rock playing style, the mic can be placed at a close working distance of 6 inches or less, as the increased overtones help the instrument to cut through an ensemble. Under PA (public address) applications, distant working conditions are likely to produce feedback (since less amplification is needed). In this situation, an electric pickup, contact, or clip-type microphone can be attached to the instrument's body or tailpiece.



FIGURE 4.60
Example of a typical
microphone
placement for the
violin.

CELLO

The fundamental range of the cello is from C2 to C5 (56 to 520 Hz), with overtones up to 8 kHz. If the player's line of sight is taken to be 0° , then the main direction of sound radiation lies between 10° and 45° to the right. A quality mic can be placed level with the instrument and directed toward the sound holes. The chosen microphone should have a flat response and be placed at a working distance of between 6 inches and 3 feet.

DOUBLE BASS

The double bass is one of the orchestra's lowest-pitched instruments. The fundamentals of the four-string type reach down to E1 (41 Hz) and up to around middle C (260 Hz). The overtone spectrum generally reaches upward to 7 kHz, with an overall angle of high-frequency dispersion being $\pm 15^\circ$ from the player's line of sight. Once again, a mic can be aimed at the *f* holes at a distance of between 6 inches and 1.5 feet.

Voice

From a shout to a whisper, the human voice is a talented and versatile sound source that displays a dynamic and timbral range that's matched by few other instruments. The male bass voice can ideally extend from E2 to D4 (82 to 294 Hz) with sibilant harmonics extending to 12 kHz. The upper soprano voice can range upward to 1050 Hz with harmonics that also climb to 12 kHz.

When choosing a mic and its proper placement, it's important to step back for a moment and remember that the most important "device" in the signal chain is the vocalist. Let's assume that the engineer/producer hasn't made the classic mistake of waiting until the last minute (when the project goes over budget and/or into overtime) to record the vocals. ... Good, now the vocalist can relax and concentrate on a memorable performance. Next step is to concentrate on the vocalist's "creature comforts": How are the lighting and temperature settings? Is the vocalist thirsty? Once done, you can go about the task of choosing your mic and its placement to best capture the performance.

The engineer/producer should be aware of the following traps that are often encountered when recording the human voice:

- *Excessive dynamic range:* This can be solved either by mic technique (physically moving away from the mic during louder passages) or by inserting a compressor into the signal path. Some vocalists have dynamics that range from whispers to normal volumes to practically screaming . . . all in a single passage. If you optimize your recording levels during a moderate-volume passage and the singer begins to belt out the lines, then the levels will become too "hot" and will distort. Conversely, if you set your recording levels for the loudest passage, the moderate volumes will be buried in the music. The solution to this dilemma is to place a compressor in the mic's signal path. The compressor automatically "rides" the signal's gain and reduces excessively loud passages to a level that the system can effectively handle. (See Chapter 12 for more information about compression and devices that alter dynamic range.)
- *Sibilance:* This occurs when sounds such as *f*, *s* and *sh* are overly accentuated. This often is a result of tape saturation and distortion at high levels or slow tape speeds. Sibilance can be reduced by inserting a frequency-selective compressor (known as a de-esser) into the chain or through the use of moderate equalization.
- *Excessive bass boost due to proximity effect:* This bass buildup often occurs when a directional mic is used at close working ranges. It can be reduced or compensated for by increasing the working distance between the source and the mic, by using an omnidirectional mic (which doesn't display a proximity bass buildup), or through the use of equalization.

MIC TOOLS FOR THE VOICE

Some of the most common tools in miking are used for fixing problems that relate to picking up the human voice and to room isolation.

Explosive popping *p* and *b* sounds often result when turbulent air blasts from the mouth strike the mic diaphragm. This problem can be avoided or reduced by:

- Placing a pop filter over the mic
- Placing a mesh windscreen between the mic and the vocalist
- Taping a pencil in front of the mic capsule, so as to break up the “plosive” air blasts
- Using an omnidirectional mic (which is less sensitive to popping, but might cause leakage issues).

Reducing problems due to leakage and inadequate isolation can be handled in any number of situational ways, including:

- Choice of directional pattern (i.e., choosing a tighter cardioid or hypercardioid pattern can help reduce unwanted leakage)
- Isolating the singer with a flat or portable isolation device
- Isolating the singer in a separate iso-booth
- Overdubbing the vocals at a later time, keeping in mind that carefully isolated “scratch” vocals can help glue the band together and give the vocalist a better feel for the song.

Woodwind instruments

The flute, clarinet, oboe, saxophone and bassoon combine to make up the woodwind class of instruments. Not all modern woodwinds are made of wood nor do they produce sound in the same way. For example, a flute’s sound is generated by blowing across a hole in a tube, whereas other woodwinds produce sound by causing a reed to vibrate the air within a tube.

Opening or covering finger holes along the sides of the instrument controls the pitch of a woodwind by changing the length of the tube and, therefore, the length of the vibrating air column. It’s a common misunderstanding that the natural sound of a woodwind instrument radiates entirely from its bell or mouthpiece. In reality, a large part of its sound often emanates from the finger holes that span the instrument’s entire length.

CLARINET

The clarinet commonly comes in two pitches: the B clarinet, with a lower limit of D3 (147 Hz), and the A clarinet, with a lower limit of C3 (139 Hz). The highest fundamental is around G6 (1570 Hz), whereas notes an octave above middle C contain frequencies of up to 150 Hz when played softly. This spectrum can range upward to 12 kHz when played loudly. The sound of this reeded woodwind radiates almost exclusively from the finger holes at frequencies between 800 Hz and 3 kHz; however, as the pitch rises, more of the sound emanates from the bell. Often, the best mic placement occurs when the pickup is aimed toward the lower finger holes at a distance of 6 inches to 1 foot (Figure 4.61).

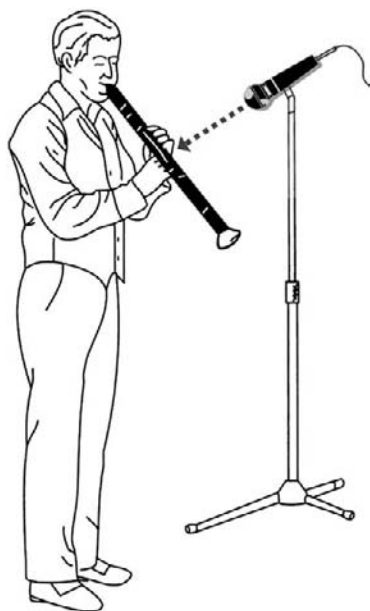


FIGURE 4.61
Typical microphone
position for the
clarinet.

FLUTE

The flute's fundamental range extends from B3 to about C7 (247 to 2093 Hz). For medium loud tones, the upper overtone limit ranges between 3 and 6 kHz. Commonly, the instrument's sound radiates along the player's line of sight for frequencies up to 3 kHz. Above this frequency, however, the radiated direction often moves outward 90° to the player's right. When miking a flute, placement depends on the type of music being played and the room's overall acoustics. When recording classical flute, the mic can be placed on-axis and slightly above the player at a distance of between 3 and 8 feet. When dealing with modern musical styles, the distance often ranges from 6 inches to 2 feet. In both circumstances, the microphone should be positioned at a point $1/3$ to $1/2$ the distance from the instrument's mouthpiece to its footpiece. In this way, the instrument's overall sound and tone quality can be picked up with equal intensity (Figure 4.62). Placing the mic directly in front of the mouthpiece will increase the level (thereby reducing feedback and leakage); however, the full overall body sound won't be picked up and breath noise will be accentuated. If mobility is important, an integrated contact pickup can be used or a clip mic can be secured near the instrument's mouthpiece.

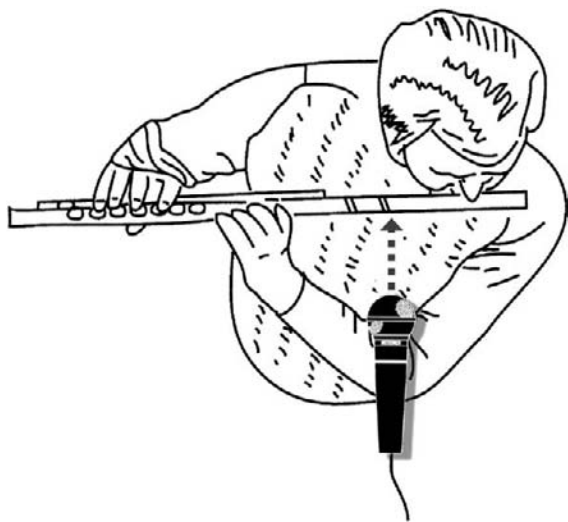


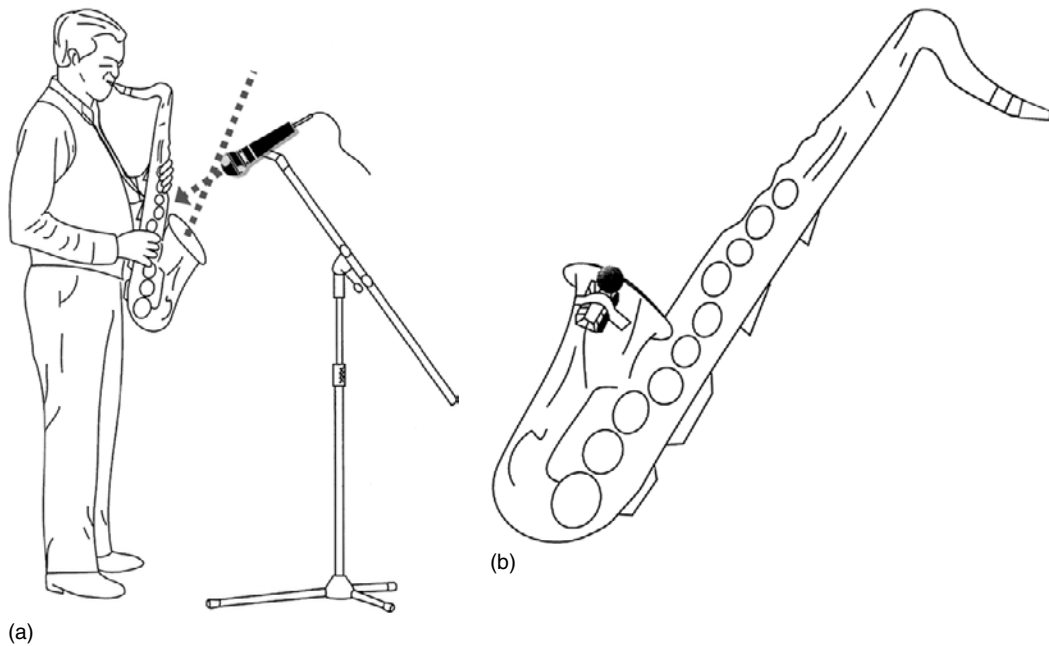
FIGURE 4.62
Typical microphone
position for the flute.

SAXOPHONE

Saxophones vary greatly in size and shape. The most popular sax for rock and jazz is the S-curved B-flat tenor sax, whose fundamentals span from B2 to F5 (117 to 725 Hz), and the E-flat alto, which spans from C3 to G5 (140 to 784 Hz). Also within this family are the straight-tubed soprano and sopranino, as well as the S-shaped baritone and bass saxophones. The harmonic content of these instruments ranges up to 8 kHz and can be extended by breath noises up to 13 kHz. As with other woodwinds, the mic should be placed roughly in the middle of the instrument at the desired distance and pointed slightly toward the bell (Figure 4.63). Keypad noises are considered to be a part of the instrument's sound; however, even these can be reduced or eliminated by aiming the microphone closer to the bell's outer rim.

HARMONICA

Harmonicas come in all shapes, sizes and keys ... and are divided into two basic types: the diatonic and the chromatic. Their pitch is determined purely by the

**FIGURE 4.63**

Typical microphone positions for the saxophone: (a) standard placement; (b) typical “clip-on” placement.

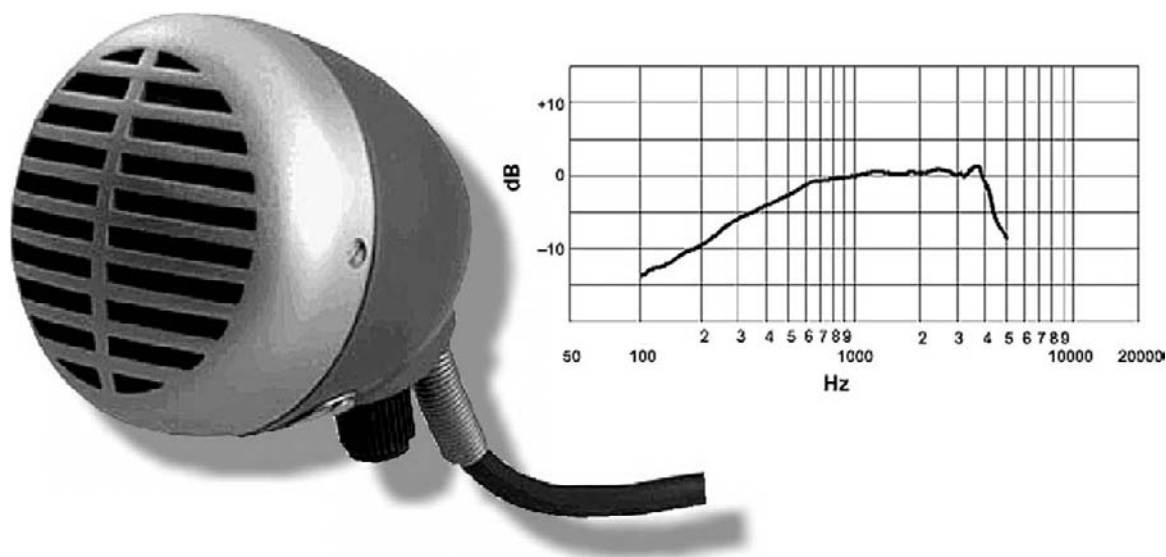
length, width and thickness of the various vibrating metal reeds. The “harp” player’s habit of forming his or her hands around the instrument is a way to mold the tone by forming a resonant cavity. The tone can be deepened and a special “wahing” effect can be produced by opening and closing a cavity that’s formed by the palms; consequently, many harmonica players carry their preferred microphones with them (Figure 4.64) rather than being stuck in front of an unfamiliar mic and stand.

MICROPHONE SELECTION

The following information is meant to provide insights into a limited number of professional mics that are used for music recording and professional sound applications. This list is by no means complete, as literally hundreds of mics are available, each with its own particular design, sonic character and application.

Shure SM57

The SM57 (Figure 4.65) is widely used by engineers, artists, touring sound companies, etc., for instrumental and remote recording applications. The SM57’s midrange presence peak and good low-frequency response make it useful for use with vocals, snare drums, toms, kick drums, electric guitars and keyboards.

**FIGURE 4.64**

The Shure 520DX “Green Bullet” microphone, a preferred harmonica pickup for many musicians. (Courtesy of Shure Brothers, Inc., www.shure.com.)

Specifications:

- *Transducer type:* moving-coil dynamic
- *Polar response:* cardioid
- *Frequency response:* 40 to 15,000 Hz
- *Equivalent noise rating:* -7.75 dB (0 dB = 1 V/microbar).

AKG D112

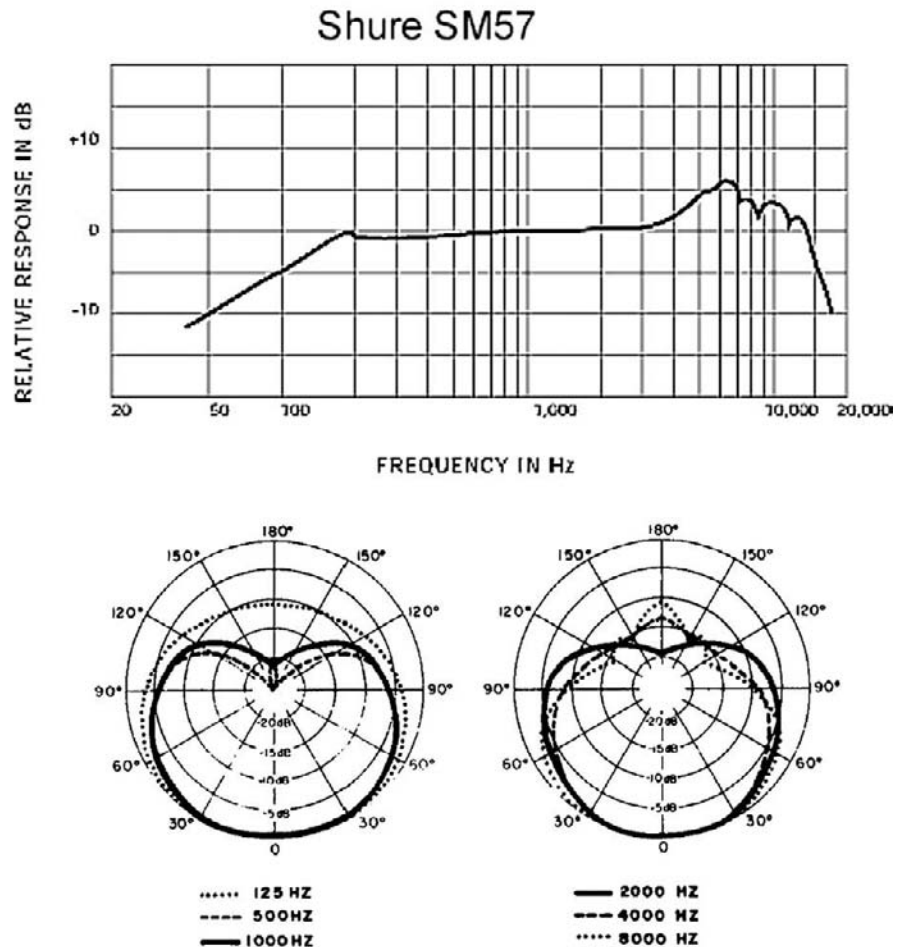
Large-diaphragm cardioid dynamic mics, such as the AKG D112 (Figure 4.66), are often used for picking up kick drums, bass guitar cabinets, and other low-frequency, high-output sources.

Specifications

- *Transducer type:* moving-coil dynamic
- *Polar response:* cardioid
- *Frequency response:* 30 to 17,000 Hz
- *Sensitivity:* -54 dB \pm 3 dB re. 1 V/microbar.

Beyerdynamic M160

The Beyer M160 ribbon microphone (Figure 4.67) is capable of handling high sound-pressure levels without sustaining damage while providing the transpar-

**FIGURE 4.65**

Shure SM57 dynamic microphone. (Courtesy of Shure Brothers, Inc., www.shure.com.)

ency that often is inherent in ribbon mics. Its hypercardioid response yields a wide-frequency response/low-feedback characteristic for both studio and stage.

Specifications

- *Transducer type:* ribbon dynamic
- *Polar response:* hypercardioid
- *Frequency response:* 40 to 18,000 Hz
- *Sensitivity:* 52 dB (0 dB = 1 mW/Pa)
- *Equivalent noise rating:* -145 dB
- *Output impedance:* 200 Ω .

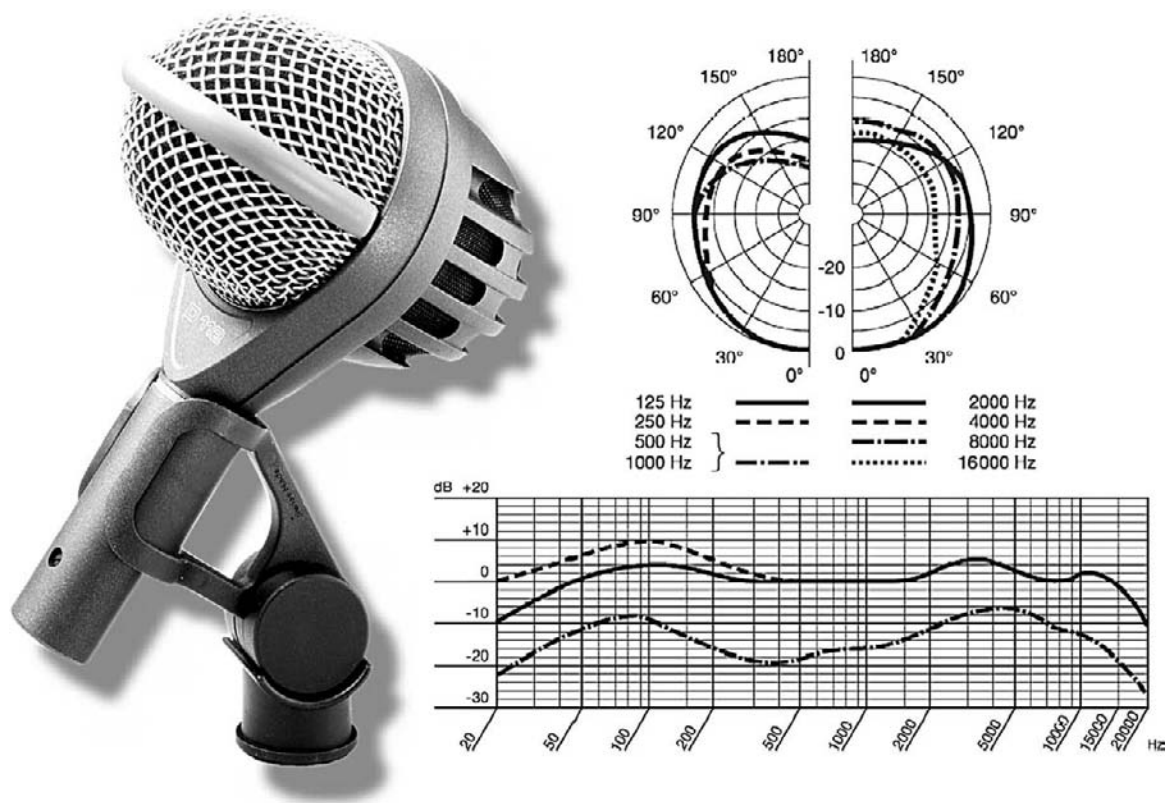


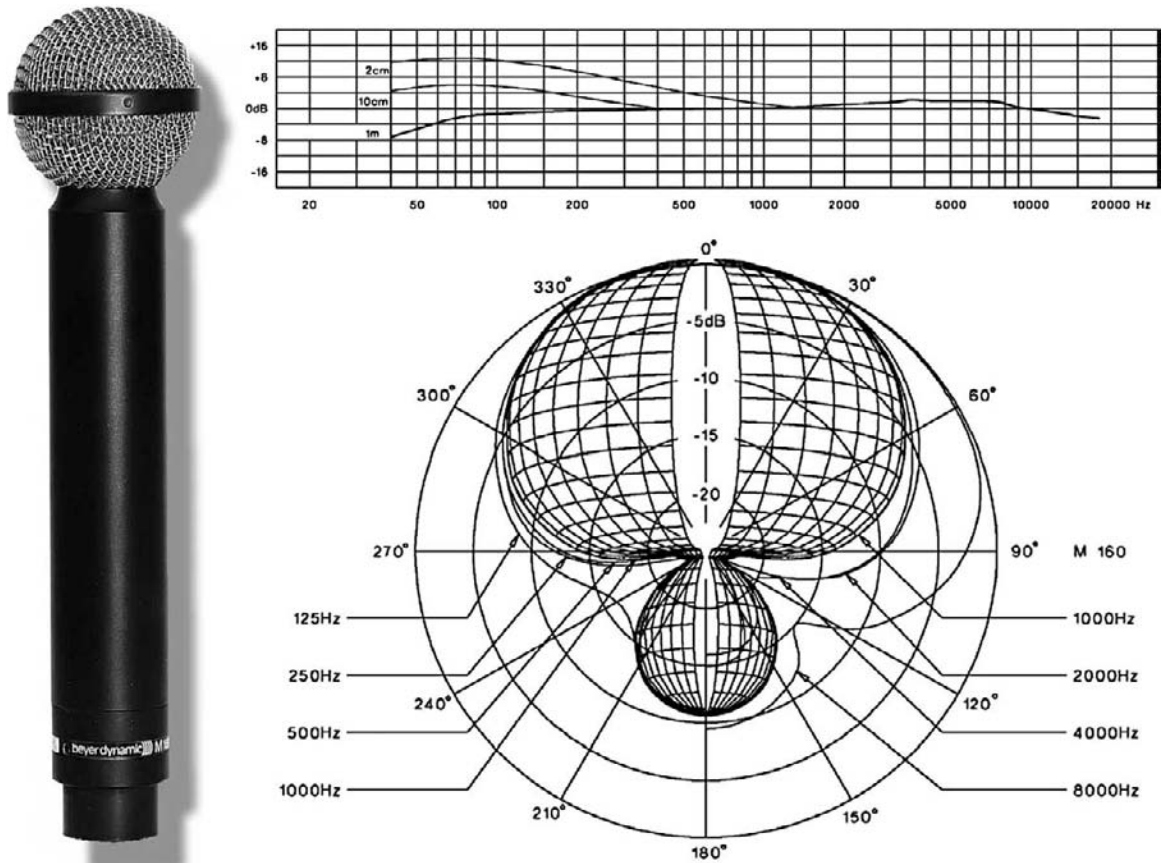
FIGURE 4.66
AKG D112 dynamic microphone. (Courtesy of AKG Acoustics, Inc., www.akg.com.)

Royer Labs R-121

The R-121 is a ribbon mic with a figure-8 pattern (Figure 4.68). Its sensitivity is roughly equal to that of a good dynamic mic, and it exhibits a warm, realistic tone and flat frequency response. Made using advanced materials and cutting-edge construction techniques, its response is flat and well balanced; the low end is deep and full without getting boomy, mids are well defined and realistic, and the high-end response is sweet and natural sounding.

Specifications

- *Acoustic operating principle:* electrodynamic pressure gradient
- *Polar pattern:* figure 8
- *Generating element:* 2.5-micron aluminum ribbon
- *Frequency response:* 30 to 15,000 Hz \pm 3 dB
- *Sensitivity:* -54 dBV re. 1 V/Pa \pm 1 dB
- *Output impedance:* 300 Ω at 1 K (nominal); 200 Ω optional
- *Maximum SPL:* >135 dB.

**FIGURE 4.67**

Beyerdynamic M160 ribbon microphone. (Courtesy of Beyerdynamic, www.beyerdynamic.com.)

Neumann KM 180 Series

The 180 Series consists of three compact miniature microphones (Figure 4.69): the KM 183 omnidirectional and KM 185 hypercardioid microphones and the successful KM 184 cardioid microphone. All 180 Series microphones are available with either a matte black or nickel finish and come in a folding box with a windshield and two stand mounts that permit connection to the microphone body or the XLR-connector.

Specifications

- *Transducer type:* condenser
- *Polar response:* cardioid (183), cardioid (184) and hypercardioid (185)
- *Frequency response:* 20 to 20 kHz
- *Sensitivity:* 12/15/10 mV/Pa
- *Output impedance:* 50 Ω
- *Equivalent noise level:* 16/16/18 dB -A.

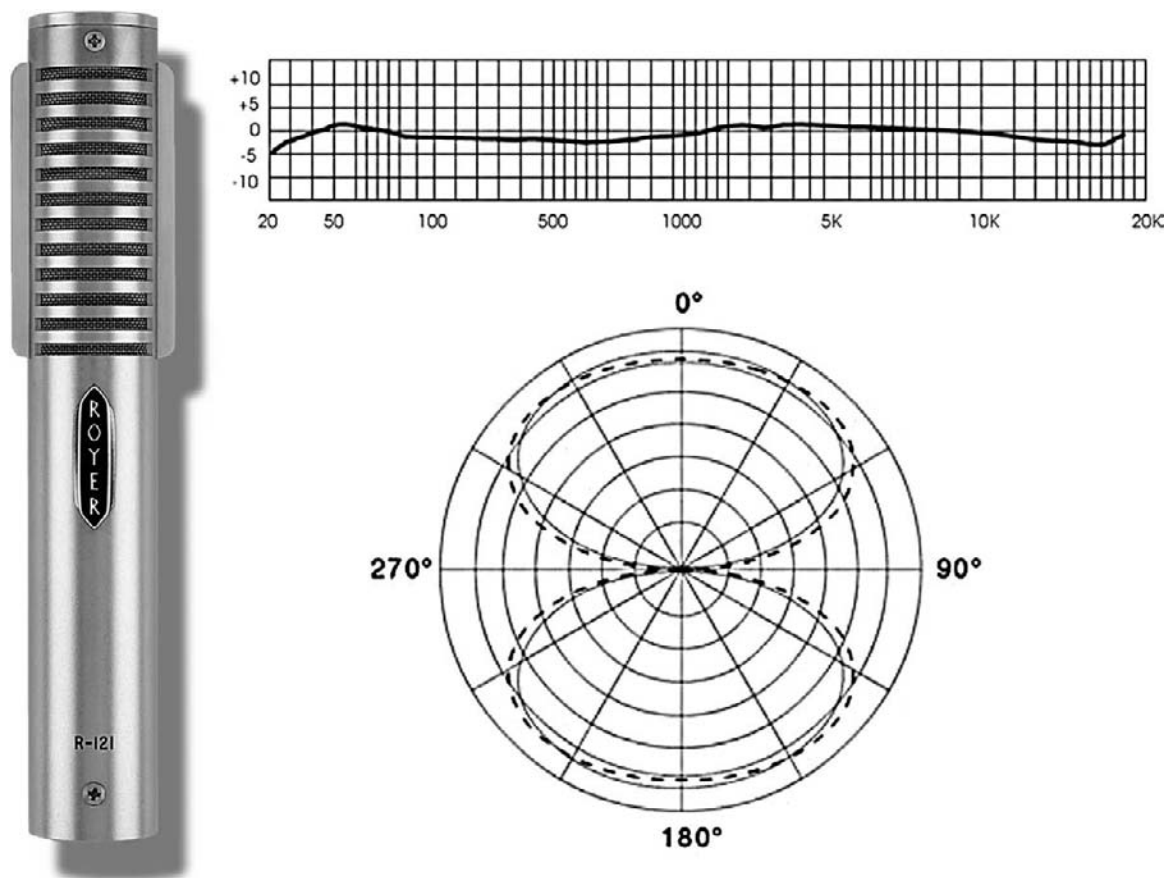


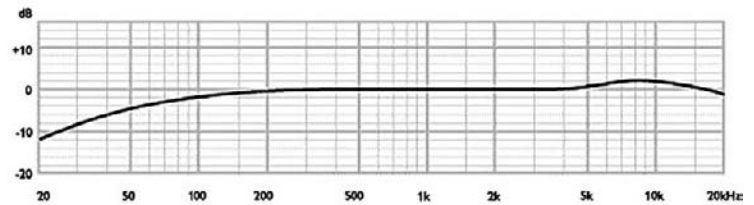
FIGURE 4.68
Royer Labs R-121 ribbon microphone. (Courtesy of Royer Labs, www.royerlabs.com.)

AKG C3000B

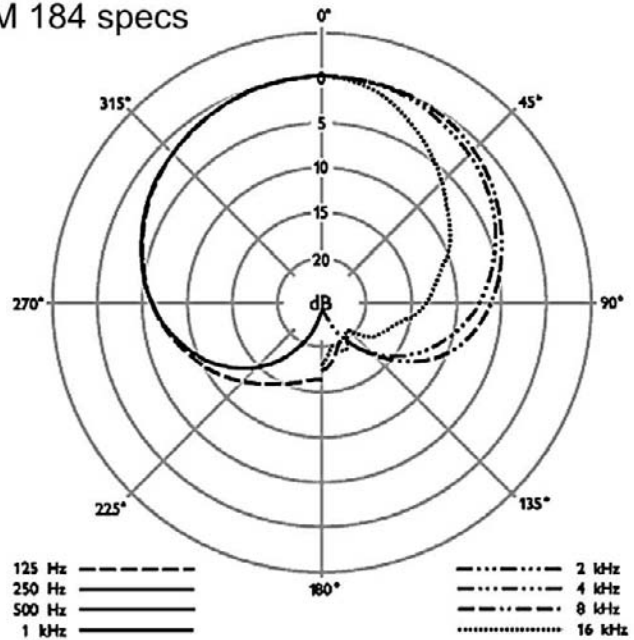
The AKG C3000B (Figure 4.70) is a low-cost, large-diaphragm condenser mic. Its design incorporates a bass roll-off switch, a -10-dB pad and a highly effective internal windscreen. The mic's dual-diaphragm capsule design is floated in an elastic suspension for improved rejection of mechanical noise.

Specifications

- *Transducer type:* condenser
- *Polar response:* cardioid
- *Frequency response:* 20 to 20,000 Hz
- *Sensitivity:* 25 mV/Pa (-32 dBv).



KM 184 specs

**FIGURE 4.69**

Neumann KM 180 Series condenser microphones. (Courtesy of Georg Neumann GMBH, www.neumann.com.)

MXL V67i

The MXL V67i's design (Figure 4.71) includes two selectable diaphragms, with the front side having a warm sound, and the mic's back side being used to produce a brighter more airy sound. A bright red LED shines through the grill indicating which capsule is energized.

Specifications

- *Type:* selectable capsule condenser microphone
- *Frequency range:* 30 to 20 kHz
- *Polar pattern:* cardioid
- *Sensitivity:* 15 mV/Pa
- *Impedance:* 200 Ω
- *Signal-to-noise ratio:* 74 dB (ref. 1 Pa A-weighted)
- *Equivalent noise level:* 20 dB (A-weighted IEC 268-4)
- *Max SPL for 0.5% THD:* 140 dB

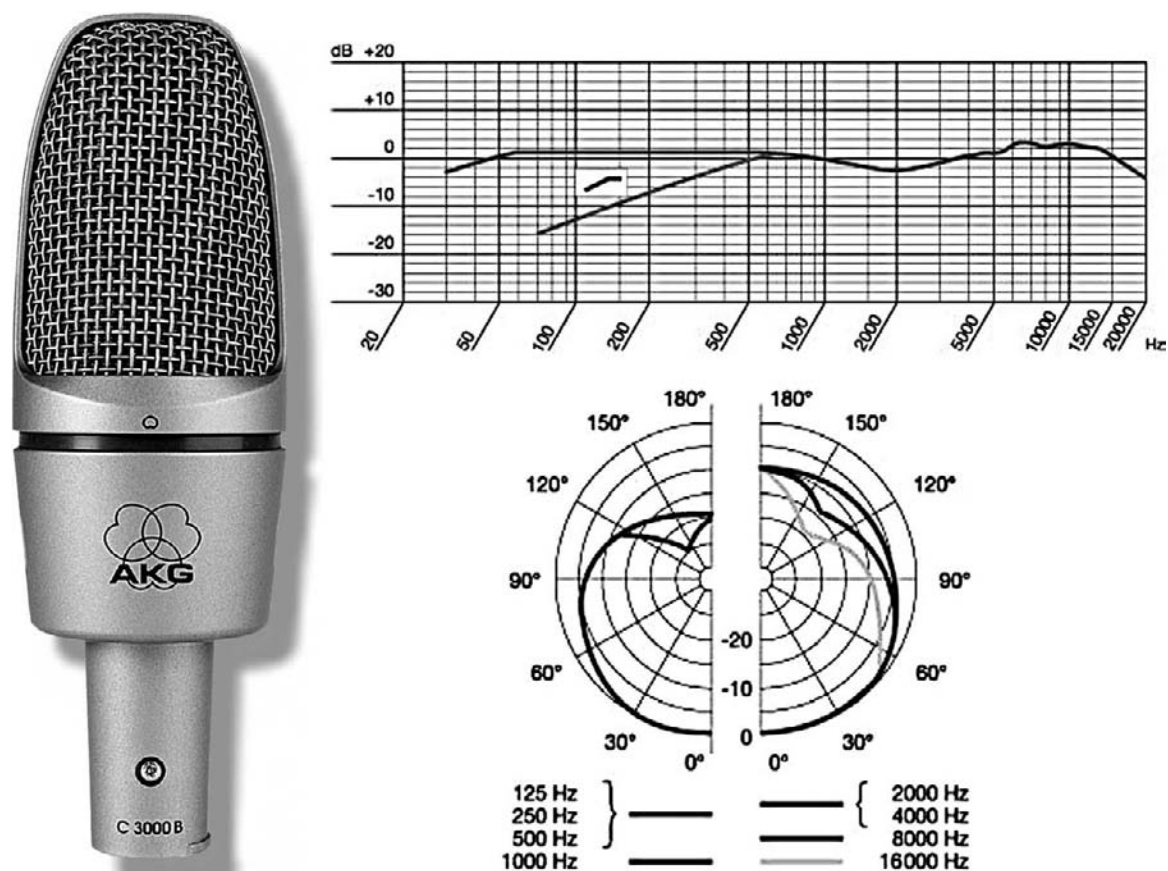


FIGURE 4.70
The AKG C3000 B condenser microphone. (Courtesy of AKG Acoustics, Inc., www.ake.com.)

FIGURE 4.71
The Marshal MXL V67i condenser microphone.
(Courtesy of Marshall Electronics, Inc., www.mxlms.com.)





FIGURE 4.72
Telefunken M216
stereo microphone.
(Courtesy of
Telefunken USA,
www.telefunkenusa.com.)

- *Power requirements:* Phantom power $48\text{ V} \pm 4\text{ V}$
- *Current consumption:* $<3.0\text{ mA}$.

Telefunken M216 stereo mic

The Telefunken M216 matrix design (Figure 4.72) features a matched pair of 1-inch dual-sided capsules, a new, old-stock NOS ECC81 vacuum tube, matched custom-wound transformers, and a custom power supply with stereo matrix and Z40 encoding/decoding settings. Each channel offers nine different polar patterns. Cardioid, omni and figure-8 capabilities and numerous in-between patterns result in an extensive range of stereo imaging possibilities.

Specifications

- *Type:* selectable pattern stereo condenser microphone
- *Frequency range:* 20 to 20 kHz
- *Polar pattern:* cardioid, omni, figure-8 and six in-between patterns
- *Sensitivity:* 14 mV/Pa
- *Impedance:* 200 Ω
- *Signal-to-noise ratio:* 125 dB
- *Equivalent noise level:* 20 dB
- *Power requirements:* Ela M916 with quadraphonic outputs.