

# How Microphones Work

Microphones appear in an almost endless variety of shapes, sizes, and design types, but no matter what their physical attributes, their purpose is same—to convert acoustic vibrations (in the form of air pressure) to electrical energy so it can be amplified or recorded. Most achieve this by the action of the air vibrating a diaphragm connected to something that either creates or allows a small electron flow.

There are three basic mechanical techniques that are used in building microphones for professional audio purposes, but all three types have the same three major parts:

- ▶ **A diaphragm.** The sound waves strike the diaphragm, causing it to vibrate in sympathy with the sound wave. In order to accurately reproduce high-frequency sounds, it must be as light as possible.
- ▶ **A transducer.** The mechanical vibrations of the diaphragm are converted into an electronic signal by the transducer.
- ▶ **A casing.** As well as providing mechanical support and protection for the diaphragm and transducer, the casing can also be made to help control the directional response of the microphone.

Let's take a closer look at the three types of microphones—dynamic, ribbon, and condenser.

**EDDIE KRAMER:** *To me a microphone is like a color that a painter selects from his palette. You pick the colors that you want to use.*

## The Dynamic Microphone

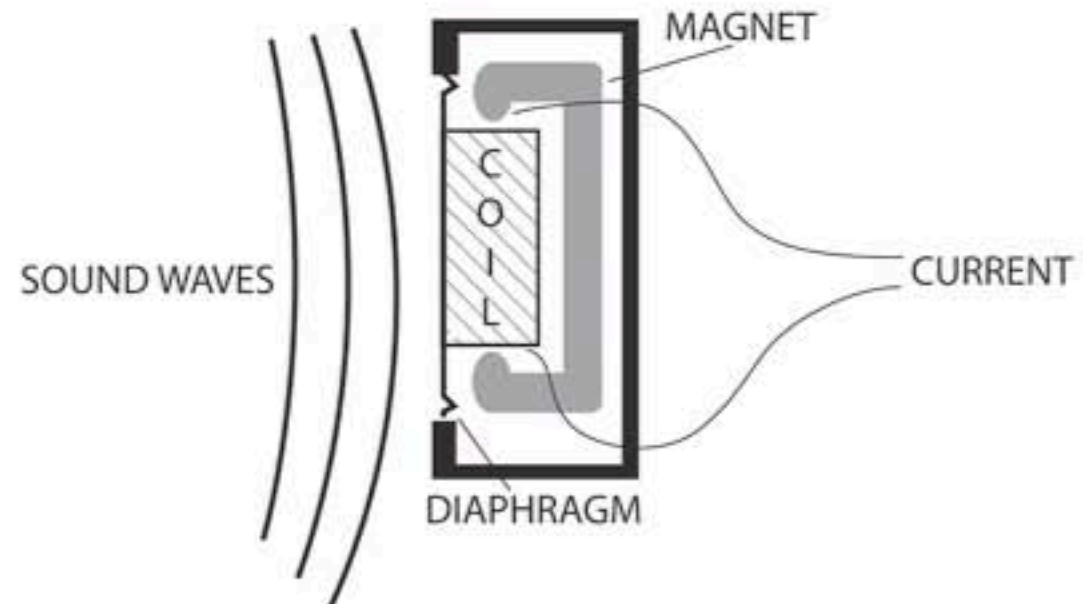
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The dynamic microphone is the workhorse of the microphone breed. Ranging from really inexpensive to moderately expensive, there's a dynamic model to fit just about any application.

### HOW IT WORKS

In a moving-coil (more commonly called *dynamic*) microphone, sound waves cause movement of a thin metallic diaphragm and an attached coil of wire that is located inside a permanent magnet. When sound waves make the diaphragm vibrate, the connected coils also vibrate in the magnetic field, causing current to flow. Since the current is produced by the motion of the diaphragm, and the amount of current is determined by the speed of that motion, this kind of microphone is known as *velocity sensitive* (see Figure 1.1).

Figure 1.1  
Dynamic mic block diagram.



The ability of the microphone to respond to transients and higher frequency signals is dependent upon how heavy the moving parts are. In this type of microphone, both the diaphragm and the coil move, so that means it's relatively heavy. As a result, the frequency response falls off above about 10 kHz.

The microphone also has a resonant frequency (a frequency or group of frequencies that is emphasized) that is typically somewhere from about 1 to 4 kHz. This resonant response is sometimes called the *presence peak* because it occurs in the frequency region that directly affects voice intelligibility. Because of this natural effect, dynamic microphones are often preferred by vocalists, especially in sound reinforcement.

These microphones tend to be expensive because they're somewhat complex to manufacture, but they're generally very robust (you can actually hammer nails with some of them, and they'll still work!) and insensitive to changes in humidity.

- ▶ **Advantages:** Robust and durable, can be relatively inexpensive, insensitive to changes in humidity, needs no external or internal power to operate, can be made fairly small.
- ▶ **Disadvantages:** Resonant peak in the frequency response, typically weak high-frequency response beyond 10 kHz or so.

## The Ribbon Microphone

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The ribbon microphone operates almost the same as the moving-coil microphone. The major difference is that the transducer is a strip of extremely thin aluminum foil, wide enough and light enough to be vibrated directly by the moving molecules of air of the sound wave, so no separate diaphragm is necessary. However, the electrical signal generated is very small compared to that of a moving-coil microphone, so an output transformer is needed to boost the signal to a usable level (see Figures 1.2 and 1.3).

Figure 1.2  
Ribbon mic block diagram.

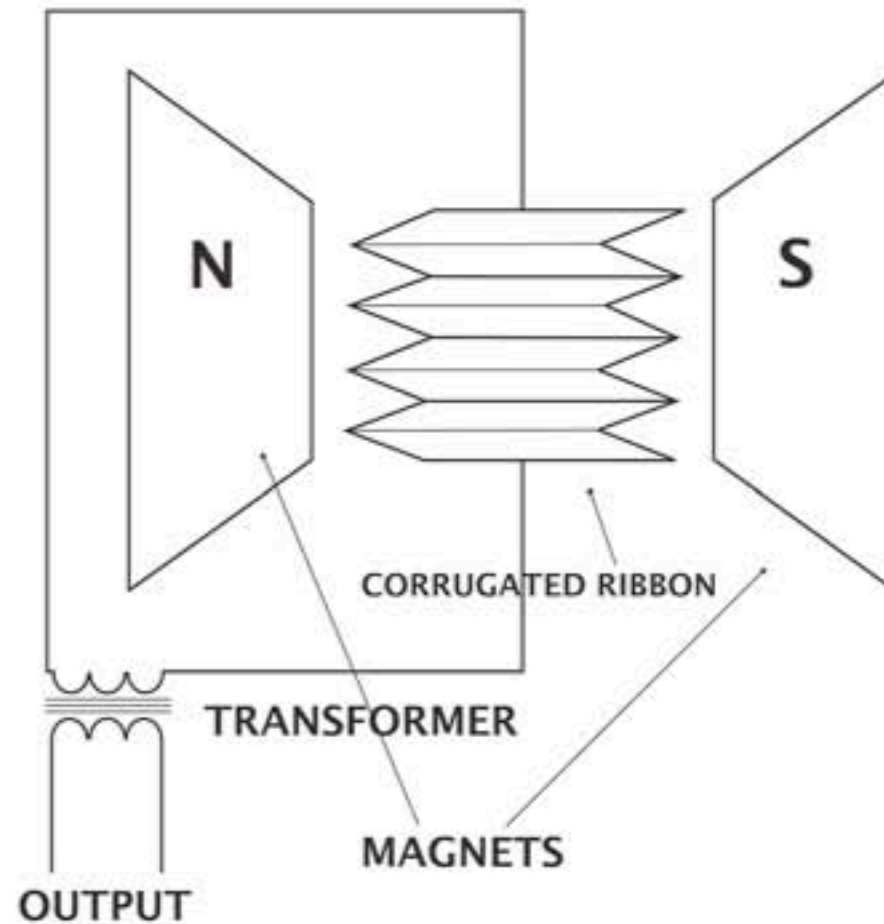


Figure 1.3  
Ribbon mic transducer. Courtesy of  
Royer Labs.



Like the dynamic microphone, the high-frequency response is governed by the mass of the moving parts. But because the diaphragm is also the transducer, the mass is usually a lot less than a dynamic type. As a result, the upper frequency response tends to reach slightly higher, to around 14 kHz. The frequency response is also generally flatter than for a moving-coil microphone.

All good studio ribbon mics provide more opportunity to EQ to taste because they “take” EQ well. Ribbon mics have their resonance peak at the bottom of their frequency range, which means that a ribbon just doesn’t add any extra high-frequency hype like condenser mics do.

- ▶ **Advantages:** Relatively flat frequency response, extended high-frequency response as compared to dynamics, needs no external or internal power to operate.
- ▶ **Disadvantages:** Fragile—requires care during operation and handling, moderately expensive.

## A Short History of Ribbon Microphones

You're going to read a lot about ribbon microphones in this book because they seem to have been rediscovered in recent years and therefore have recently returned to widespread use. So, a bit of history seems in order.

The ribbon-velocity microphone design first gained popularity in the early 1930s and remained the industry standard for many years, being widely used on recordings and broadcasts from the '30s through about the early '60s.

Ribbon-microphone development reached its pinnacle during this period. Though they were always popular with announcers and considered state of the art at the time, one of the major disadvantages of early ribbon mics was their large size, since magnetic structures and transformers of the time were bulky and inefficient. When television gained popularity in the late 1940s, their size made them intrusive on camera and difficult to maneuver, so broadcasters soon looked for a more suitable replacement.

About that time, newer breeds of condenser and dynamic microphones were developed that were a lot more compact and far more rugged. As a result, television and radio began to replace their ribbons with these new designs. Since ribbon mics were being used less and less, further development was considered unnecessary, and the ribbon soon suffered a fate similar to that of the vacuum tube when transistors hit the scene.

Although ribbon mics might have been out of favor in broadcast, recording engineers never quite gave up on the technology. While always fragile, ribbon mics still provided some of the sweetest sounds in recording, as most old-school engineers realized. As a result, vintage ribbon mics commanded extremely high prices in the used marketplace.

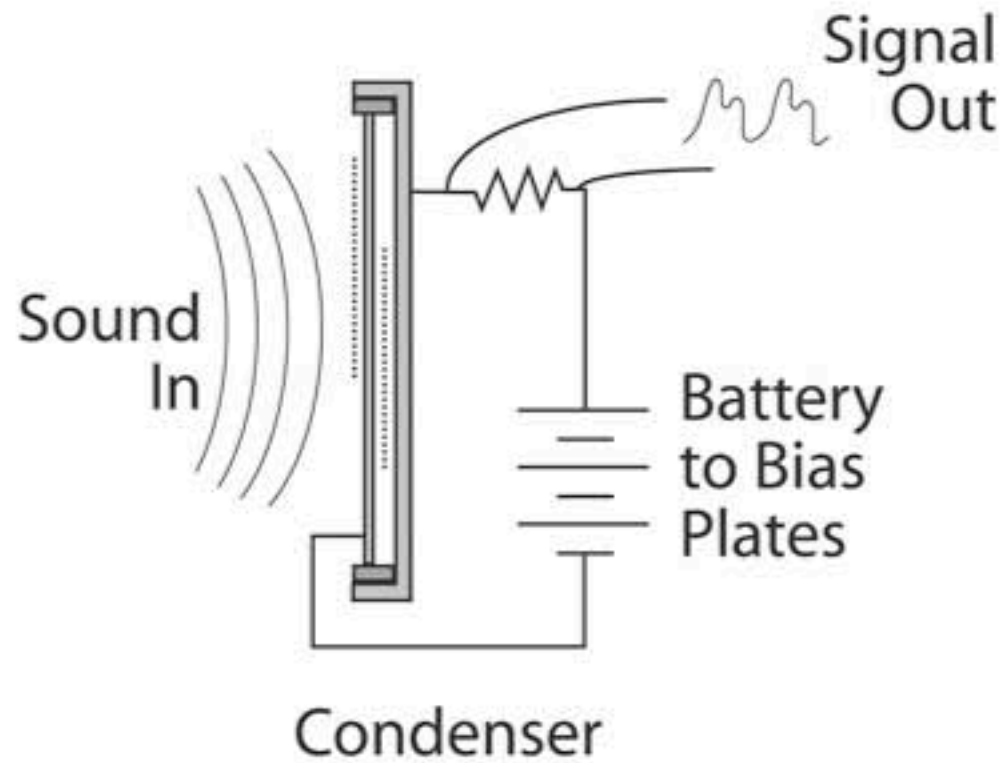
Recently, a few modern manufacturers began to not only revive the technology, but improve it as well. Companies like Coles, Beyer, Royer, and AEA now make ribbon microphones at least as good as or better than the originals, and they are a lot more robust as well. Thanks to recent developments in magnetics, electronics, and mechanical construction, modern ribbon microphones can be produced smaller and lighter, yet still maintain the sound of their vintage forbearers while achieving sensitivity levels matching those of other types of modern microphones. Their smooth frequency response and phase linearity make them ideally suited for the digital formats that dominate the industry today.

## The Condenser Microphone

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The condenser microphone has two electrically charged plates—one that can move, which acts as a diaphragm, and one that is fixed, called a *backplate*. This is, in effect, a capacitor (or *condenser*) with a positively and negatively charged electrode and an air space in between. Sound depresses the diaphragm, causing a change in the spacing between it and the backplate. This change in capacitance and distance between the diaphragm and the backplate causes a change in voltage potential that can be amplified to a usable level. To boost this small voltage, a vacuum tube or FET transistors are used as an amplifier. This is why a battery or phantom power is needed to charge the plates and also to run the preamp. Because the voltage requirements to power a vacuum tube are so high and therefore require some large and heavy components, some microphones have the power supply in a separate outboard box (see Figure 1.4).

Figure 1.4  
Condenser mic block diagram.



A condenser has an omnidirectional pickup pattern in its native state. In order to make it directional, little holes are punched in the backplate. The object of the holes is to delay the arrival of sound at the rear of the diaphragm to coincide with the same sound at the front, which then cancels the sound out. The size and position of the holes determines the frequencies that will be cancelled.

Most large-diaphragm condensers are multi-pattern microphones. This design consists of a single backplate placed between two diaphragms. By varying how much signal from each diaphragm is fed to the preamp, the microphone can have selectable patterns ranging from a tight cardioid, to a figure-eight, to full omnidirectional.

Condenser mics, however, always ring (resonate) a bit, typically in the 8- to 12-kHz range. A condenser mic's pattern of resonance is a major part of its character. Their built-in top-end response bump limits the EQ you might want to add, since a little bit of high-frequency boost can start to sound a bit "edgy" rather quickly.

### The Electret Condenser

Another less expensive type of condenser microphone is the electret condenser. An electret microphone uses a permanently polarized electret material as a diaphragm, thus avoiding the necessity for the biasing DC voltage required in a conventional condenser. Electrets can be made very small and inexpensive and are the typical microphones on portable tape recorders. Better quality electret condensers incorporate a preamplifier to match their extremely high impedance and boost the signal. One of the problems with early electret condenser microphones, however, is that the electret material loses its charge over time.

- ▶ **Advantages:** Excellent high-frequency and upper harmonic response, can have excellent low-frequency response.
- ▶ **Disadvantages:** Moderate to very expensive, requires external powering, can be relatively bulky, low-cost (and some expensive) models can suffer from poor or inconsistent frequency response, two mics of the same model may sound quite different, humidity and temperature affect performance.

### CONDENSER MIC FALLACIES

*A large-diaphragm condenser has more low end than a small-diaphragm condenser.*

This is not necessarily true. In many cases, small-diaphragm condensers reproduce the low end even better than their larger kin.

*A cardioid condenser has a better low-end response than an omni.*

Not true. In condenser mics with an omnidirectional polar response, the bass response is only limited by the electronics. So even a very small diaphragm can have a flat response down to DC.

*A large-diaphragm condenser has a flatter response than a small-diaphragm condenser.*

Not true. Large-format capsules are prone to low-frequency resonance, which means they can have trouble reproducing low frequencies at a high level. They "bottom out" by the diaphragm hitting the backplate, which is

the popping that can occur when a singer is too close to an unfiltered microphone. In order to minimize this, some microphones over-damp the capsule, making the mic sound either thin or alternatively lumpy in response, while some address this by adding a low-frequency roll-off or EQ circuitry to try to put back frequencies suppressed in the capsule.

*A small-diaphragm condenser is quieter than a large-diaphragm condenser.*

Not true. The difference in the size of the diaphragm translates into a difference in signal-to-noise ratio. The bigger diaphragm gives you more signal for a certain electrical noise level and therefore can be quieter than the small diaphragm.

*Condenser mics have consistent response from mic to mic.*

They're not as close as you might think. Despite what the specs might say, there can be vast differences in the sound between two mics of the same model, especially in the less expensive categories. This particularly applies to tube-type mics, where there are not only differences between the capsules, but also matching of the tubes. This is usually the result of not enough attention being paid to small details during design and manufacture.

So unless two mics are specifically "matched," differences in response are inevitable. The exact value of precise matching of microphones is open to much debate, however. One school of thought says that you need closely matched response for a more precise stereo soundfield, while another school thinks that the difference can actually enhance the soundfield.

### **CONDENSER OPERATIONAL HINTS**

While not particularly fragile, condenser microphones do require a bit more care than dynamic microphones. Here are some tips to keep your condenser mics sounding as good as new.

1. The most commonly seen problem with condenser microphones is dirt on the capsule, which causes the high-end response to fall off. Since a condenser is always carrying a static charge when operating, it will automatically attract small airborne particles. Add to this people singing and breathing into it, and you have your response slowly deteriorating. Because the metal film of the capsule is very thin, the layer of dirt can actually be much thicker than the original metal film and polymer support. Despite what is commonly believed, the mesh grill of the mic will not do much more than stop people or objects from touching the capsule, and the acoustic foam inside the grill has limited effect.



Cleaning a capsule is a very delicate and potentially damaging operation that is best left to a professional, so the next best thing is preventive maintenance.

- ▶ Always use a pop filter.
  - ▶ Keep your condenser microphones cased when not in use.
  - ▶ Cover the mic if it will be left set up overnight.
2. Humidity and temperature extremes can have undesirable effects on performance. When exposed to a warm or humid room after a period of very low temperature, condensation in the casing can cause unwanted noises or no signal until the unit has dried out.
  3. Don't blow into the microphone. Some diaphragms can bottom out onto the plate and stick. (Switching off the microphone and disconnecting the power supply may unstick it, though.)
  4. A condenser microphone can be overloaded, which can cause either distortion or harshness of tone. Usually this is not from the diaphragm overloading, but from the high output from the capsule overloading the built-in FET preamplifier. This is less likely in the case of a vacuum tube model since tubes naturally "soft clip" (overload in a sonically unobtrusive manner). Most internal mic preamps have a -10-dB pad switch to lower the output from the capsule. In the event that this is insufficient, the bottom-end roll-off will also reduce power from the capsule.

### PHANTOM POWER

Unlike dynamic and ribbon microphones, all condenser microphones require power of some type. Older tube condensers require an outboard power supply, while electret condensers are sometimes powered with a battery. All other condenser microphones require power from an outside source, called *phantom power*. This is a 48-volt DC power source fed by a recording console, microphone preamp, or DAW interface over the same cable that carries the audio. On most recording consoles, phantom power is switchable, since it may cause a loud pop when disconnecting a cable connected to a dynamic mic.

## Microphone Specifications

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Although hardly anyone selects a microphone on specifications, it's good to know some of the issues. The following won't delve too much into the actual electronic specs as much as the considerations they imply for your application.

**SENSITIVITY**

This is a measure of how much electrical output is produced by a given sound pressure. In other words, this tells you how loud a microphone is. Generally speaking, for the same sound pressure, ribbon microphones are the quietest, while condensers, thanks to their built-in preamplifier, are the loudest.

Where this might be a concern is in how your signal chain is responding when recording loud signals. For instance, a condenser mic on a loud source might easily overload the console, outboard microphone preamp, or DAW interface because of its inherent high output.

On the other hand, the low output of a ribbon mic placed on a quiet source might cause you to turn up that same mic preamp to such a point that electronic noise becomes an issue.

Sensitivity ratings for microphones may not be exactly comparable, because different manufacturers use different rating systems. Typically, the microphone output (in a soundfield of specified intensity) is stated in dB (decibels) compared to a reference level. Most reference levels are well above the output level of the microphone, so the resulting number (in dB) will be negative. Thus, as in Table 1.1, a ribbon microphone with a sensitivity rating of  $-38$  will provide a 16-dB hotter signal than a microphone with a sensitivity of  $-54$  dB, which will in turn provide a 6-dB hotter signal than one rated at  $-60$  dB. *Note that good sensitivity does not necessarily make a microphone "better" for an application.*

**Table 1.1 Typical Microphone Sensitivities**

Ribbon	Dynamic	Condenser
$-60$ (Beyer M160)	$-54$ (Shure SM57)	$-38$ (Neumann U 87 in omni)

**OVERLOAD CHARACTERISTICS**

Any microphone will produce distortion when it is overdriven by a loud sound level. This is caused by various factors. With a dynamic microphone, the coil may be pulled out of the magnetic field; in a condenser, the internal amplifier might clip. Sustained overdriving or extremely loud sounds can permanently distort the diaphragm, degrading performance at ordinary sound levels. In the case of a ribbon mic, the ribbon could be stretched out of shape, again causing the performance to seriously degrade. Loud sound levels are encountered more often than you might think, especially if you place the mic very close to loud instruments like a snare drum or the bell of a trumpet. In fact, in many large facilities, a microphone that has been used on a kick drum, for instance, is labeled as such and is not used on any other instrument afterward.

### **FREQUENCY RESPONSE**

Although a flat frequency response has been the main goal of microphone companies for the last three or four decades, that doesn't necessarily mean that a mic is the right one for the job. In fact, a "colored" microphone can be more desirable in some applications where the source has either too much emphasis in a frequency range or not enough. Many mics have a deliberate emphasis at certain frequencies, which makes them useful for some applications (vocals in a live onstage situation, for example). In general, though, problems in frequency response are mostly encountered with sounds originating off-axis from the mic's principal directional pattern. (See the upcoming "Polar Patterns (Directional Response)" section.)

### **NOISE**

Noise in a microphone comes in two varieties: self-noise generated by the mic itself (as in the case of condenser microphones) and handling noise.

Condenser microphones are most prone to self-noise because a pre-amplifier must be used to amplify the very small signals that are produced by the capsule. Indeed, the audio signal level must be amplified by a factor of more than a thousand, and any electrical noise produced by the microphone will also be amplified by that amount, making even slight amounts of noise intolerable. Dynamic and ribbon microphones are essentially noise-free but subject to handling noise.

Handling noise is the unwanted pickup of mechanical vibration through the body of the microphone. Many microphones intended for handheld use require very sophisticated shock mountings built inside the shell.

### **POLAR PATTERNS (DIRECTIONAL RESPONSE)**

The directional response of a microphone is the way in which the microphone responds to sounds coming from different directions around it. The directional response is determined more by the casing surrounding the microphone than by the type of transducer it uses.

The directional response of a microphone is recorded on a polar diagram. This polar diagram shows the level of signal pickup (sometimes shown in decibels) from all angles and at different frequency ranges. It should be noted that all mics respond differently at different frequencies. For example, a mic can be very directional at one frequency (usually higher frequencies) but virtually omnidirectional at another.

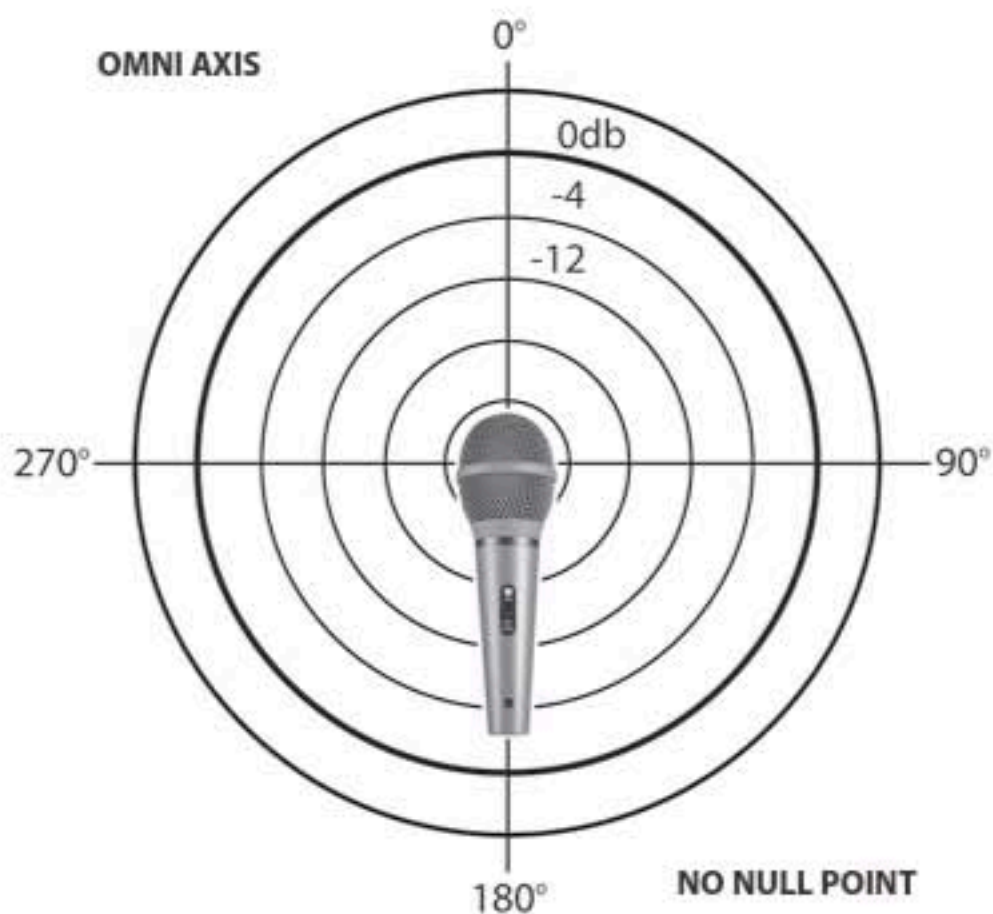
A microphone's polar response pattern can determine its usefulness in different applications, particularly multi-microphone settings where proximity of sound sources makes microphone leakage a problem.

There are four typical patterns commonly found in microphone design: omnidirectional, figure-8, cardioid, and hypercardioid.

**Omnidirectional**

An omnidirectional microphone picks up sound equally from all directions. The ideal omnidirectional response is where equal pickup occurs from all directions at all frequencies. (See Figure 1.5.)

Figure 1.5  
Omnidirectional polar pattern.



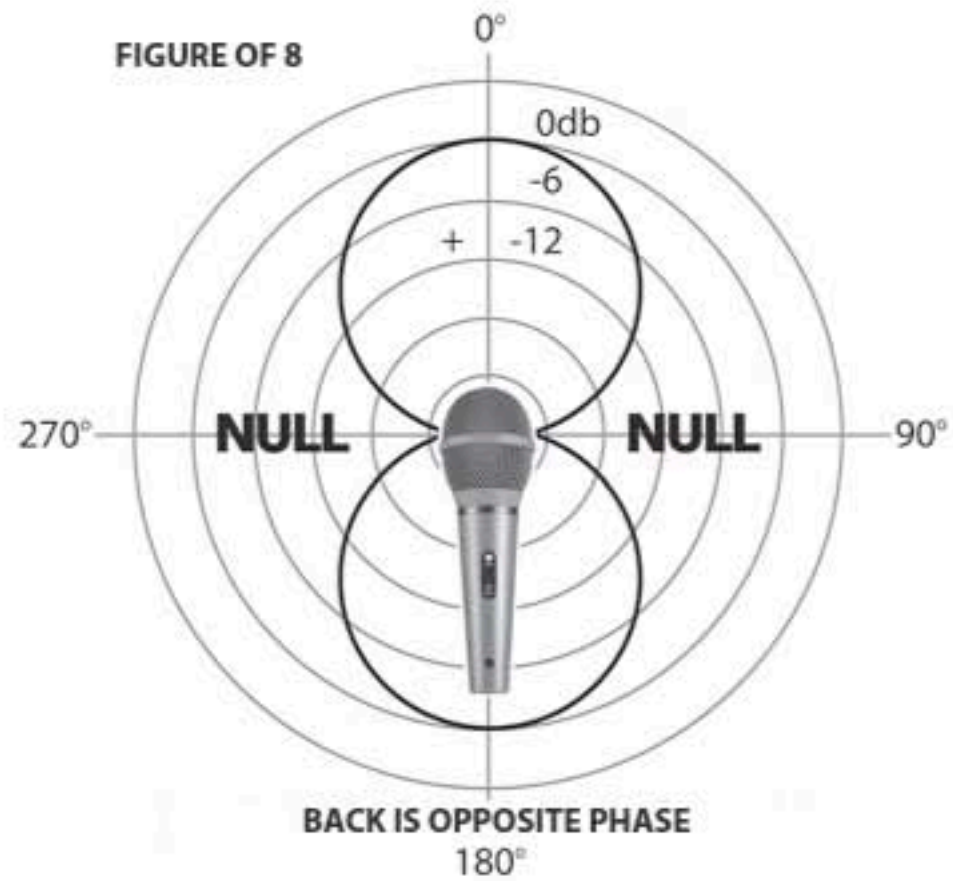
**MICHAEL BISHOP:** *I like to start with an omni before anything. Now, there are particular instances where I'll immediately go to something like a figure-8, but I'll use figure-8s and omnis more than anything.*

**Figure of Eight**

Figure of 8 (or *bidirectional* or *figure-8*) microphones pick up almost equally in the front and back, but nearly nothing on each side. It should be noted that the frequency response is usually a little better (as in brighter) on the front side of the microphone, although the level between front and rear will seem about the same.

Because the sensitivity on the sides is so low, figure-8s are often used when a high degree of rejection is required. (See Figure 1.6.)

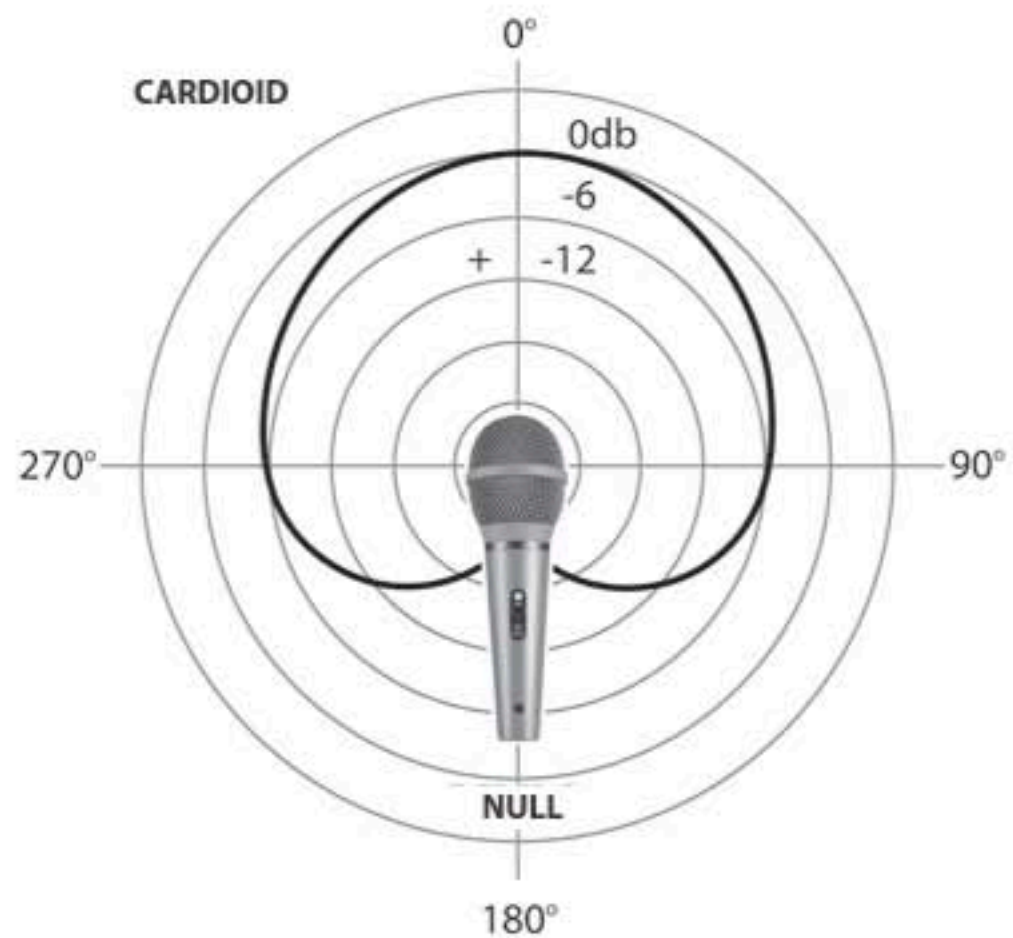
Figure 1.6  
Figure-8 polar pattern.



#### Cardioid

The cardioid microphone has strong pickup on the axis (in the front) of the microphone, but reduced pickup off-axis (to the side and to the back). This provides a more or less heart-shaped pattern, hence the name *cardioid*. (See Figure 1.7.)

Figure 1.7  
Cardioid polar pattern.



### Hypercardioid Microphones

By changing the number and size of the ports (openings) on the case, it is possible to increase the directionality of a microphone so that there is even less sensitivity to sounds on the back and sides. (See Figure 1.8.)

Figure 1.8  
Hypercardioid polar pattern.



### PROXIMITY EFFECT

A peculiarity of the pressure-gradient microphone is that it has a different frequency response in the near and far field. Cardioid and hypercardioid microphones experience low-frequency buildup the closer you get to the mic, which is known as *proximity effect*. In many cases this can be used to good effect, adding warmth and fullness to the source, but it can also make the frequency response seem out of balance if it is not taken into account.

## Specialty Microphones

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Although not used much in music recording, there are several additional types of mics available for special situations.

### SHOTGUN MICROPHONES

There are a number of applications that require an even more highly directional microphone, such as in news gathering, wildlife recording, or recording dialogue on movie and television sets. One such microphone is the shotgun (sometimes called *rifle* or *interference tube*) microphone. This consists of a long tube with slots cut in it connected to a cardioid microphone. (See Figure 1.9.)

Figure 1.9  
 Neumann KMR 82 shotgun mic.  
 Courtesy of Neumann USA.



Sound arriving from the sides enters through a number of slots in the interference tube, and those frequencies tend to cancel at the microphone. Sound entering at the end of the tube goes directly to the microphone, providing large differentiation between the source and other background noise. The tube is normally covered with a furry windshield for outdoor use.

#### **LAVALIERE**

Extremely small “tie clip” microphones are known as *lavalier* mics (sometimes just called *lavs*). They are usually electret condenser and omnidirectional and are generally designed to blend in with an article of clothing. One of the major problems with lavalieres is handling noise, which can be quite severe if an article of clothing (such as a jacket) is rubbing against it. Therefore, placement becomes crucial. (See Figure 1.10.)

Figure 1.10  
DPA 4026 lavalier mic. Courtesy  
of DPA Microphones.



### PZM

The *pressure-zone microphone*, or *PZM* or *boundary mic*, is designed to decrease the amount of echo or reverberation when recording in a large room. It accomplishes this by placing the microphone capsule very close to a flat surface. This flat surface is called the *boundary* and is why this type of microphone is also called a *boundary microphone*. Getting the microphone capsule close to the boundary cuts down on the large array of reflected sound waves hitting it from all angles. The waves that are reflected off of the closely positioned boundary are much stronger than waves that have bounced all around the room. This helps the microphone to become more sensitive, and as a result keeps the audio from sounding too reverberant.

PZM microphones, which are omnidirectional, are flat and designed to be mounted to a wall or placed on the floor or a tabletop. The bigger the boundary underneath the microphone, the better it will perform. The PZM is an omnidirectional microphone. (See Figure 1.11.)

Figure 1.11  
Crown PZM6D microphone.  
Courtesy of Crown Audio.





## WIRELESS

It's long been the dream of many performers to increase their freedom by removing the connecting cable from the microphone, and guitarists in the studio have wanted to play in the control room ever since overdubs became possible. Until recently, wireless systems weren't of sufficient quality to use in the studio, but the latest generation has begun to rival the wired versions.

A wireless system consists of three main components: an input device, a transmitter, and a receiver. The input device provides the audio signal that will be sent out by the transmitter. It may be a microphone, such as a vocalist's handheld model, or a lavalier "tie-clip" type. In wireless systems designed for use with electric guitars, the guitar itself is the input device.

The transmitter handles the conversion of the audio signal into a radio signal and broadcasts it through an antenna. The antenna may stick out from the bottom of the transmitter or it may be concealed inside. The strength of the radio signal is limited by government regulations. The distance that the signal can effectively travel ranges from 100 feet to more than 1,000 feet, depending on conditions.

Transmitters are available in two basic types. One type, called a *body-pack* or *belt-pack* transmitter, is a small box about the same size as a packet of cigarettes. The transmitter clips to the user's belt or may be worn on the body. For instrument applications, a body-pack transmitter is often clipped to a guitar strap or attached directly to an instrument, such as a trumpet or saxophone. In the case of a handheld wireless microphone, the transmitter is built into the handle of the microphone, resulting in a wireless mic that is only slightly larger than a standard wired microphone. Usually, a variety of microphone elements or *heads* are available for handheld wireless microphones. All wireless transmitters require a battery (usually a 9-volt alkaline type) to operate. (See Figure 1.12.)

Figure 1.12  
Shure UHF wireless transmitter  
and receiver. Courtesy of Shure  
Incorporated.



The job of the receiver is to pick up the radio signal broadcast by the transmitter and change it back into an audio signal. The output of the receiver is electrically identical to a standard microphone signal and can be connected to a typical microphone input in a sound system.

Wireless receivers are available in two different configurations. Single-antenna receivers utilize one receiving antenna and one tuner, similar to an FM radio. Single-antenna receivers work well in many applications but are sometimes subject to momentary interruptions or *dropouts* in the signal as the person holding or wearing the transmitter moves around the room.

Diversity receivers provide better wireless microphone performance. A diversity receiver utilizes two separate antennas spaced a short distance apart, utilizing (usually) two separate tuners. An “intelligent” circuit in the receiver automatically selects the better of the two signals, or in some cases a blend of both. Since one of the antennas will almost certainly be receiving a clean signal at any given moment, the chances of a dropout occurring are reduced.

Wireless systems operate in two different frequency spectrums: VHF and UHF. Audio performance for VHF and UHF is nearly identical, but some of the high-end (and much more expensive) UHF systems offer real improvements in audio bandwidth, transient response, and system noise floor. In terms of operational range or distance, UHF offers some advantages, especially in inhospitable RF environments. Another advantage is that broadband RF interference (compressors, elevator motors, computers, and so on) is often below UHF frequencies.

### **STEREO MICS**

Stereo microphones are essentially two microphones in a single casing or body. These are designed primarily for ease of placement, since the body is considerably smaller than two separate microphones. An added advantage is that the capsules are normally closely matched in response. The capsules usually rotate in order to give some flexibility as to the recorded sound-field. Examples are the Royer SF-12, Neumann SM 69, Shure VP88, and AKG C 24. (See Figure 1.13.)

Figure 1.13  
*Neumann SM 69 stereo mic.*  
*Courtesy of Neumann USA.*



### **PARABOLIC**

If you watch football on television, you've probably seen a parabolic microphone on the sidelines. This is usually a clear handheld dish that an operator will point out onto the field in an attempt to pick up some of the sounds of the games.

Similar to a radio telescope, a parabolic microphone is essentially an omni mic that is pointed toward the middle of a rounded (parabolic) dish. The dish provides acoustical amplification by focusing the sound on one place. If the dish itself acoustically amplifies a certain frequency range by 10 dB, it means that there's 10 dB less electronic amplification (and therefore 10 dB less noise) required within that range. The acoustical amplification increases with frequency, with the lowest frequency depending upon the diameter of the dish.

The problem with parabolic mics is that they will not respond to wavelengths longer than the diameter of the dish. This tends to make them sound unnatural for many sounds unless the dish is really huge.

Although widely used in sports broadcasting, it's not surprising that the parabolic microphone is one of the staples of the spying and espionage business as well. However, the most common use for parabolic mics in recording is to record birdcalls, since most bird chirps are only composed of high frequencies. (See Figure 1.14.)

*Figure 1.14*  
*Parabolic mic.*



## Microphone Accessories

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A number of accessories have become indispensable in modern vocal recording.

### POP FILTERS

Not to be confused with windscreens (see the following section), pop filters, either built into the mic (such as in a Shure SM58) or external, can work great or be of little value. All microphones are subject to plosives or pops. However, many engineers are fooled into thinking that a foam windscreen is all that's needed to control them when, in fact, positioning and vocal/microphone technique come more into play in the reduction of these pops. (See Figure 1.15.)

Figure 1.15  
Pop filter (Pauly Superscreen  
model 120-01). Courtesy Pauly  
Superscreens.



The problem with pop screens built into mics is that they are simply too close to the capsule. Wherever high-speed air meets an obstacle, such as a pop screen, it will generate turbulence, which takes a few inches to dissipate. If the mic capsule is within that turbulence, it will pop. Another problem with acoustic foam used within microphones is that it becomes brittle over time, and eventually tiny bits of it break off and find their way inside the capsule (which is definitely not good for the sound quality).

Spitting on a valuable mic is a really big reason to use a pop screen. Condensation from breath can stop a vintage condenser microphone in its tracks in a very short time.

External pop screens are designed to be as acoustically benign as possible, especially in the areas of transients and frequency response. That being said, they are not acoustically transparent, especially at very high frequencies. A U 87-style windscreens will knock the response at 15 kHz down about 2 to 3 dB, for instance.

Although there are many models of pop filters available commercially, it's fairly easy to build your own. Buy an embroidery hoop and some pantyhose, cut a leg of hose until you have roughly a square sheet, clamp it in the embroidery hoop, then place it between the mic and the singer.

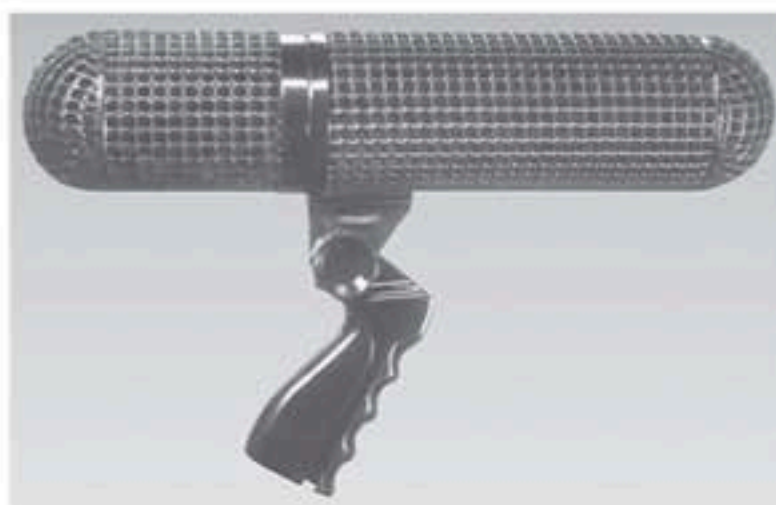
A lot of people affix pop filters to a gooseneck device that attaches to the boom stand that holds the mic. It's usually easier to mount to a pop filter on a second boom because it makes positioning less frustrating and more exact.

### WIND SCREENS

Unlike pops, wind requires a completely different strategy. Wind isn't a nice, smooth flow, but rather turbulent and random. The noise that it causes is the change in air pressure physically moving the element or ribbon in the microphone. The vibrations of wind (which is low-frequency in nature) against the element are substantially stronger than the sound vibrations. Also, the more turbulent the wind, the less you will be able to find the null in a directional mic's response.

Although acoustic foam-only may be sufficient for omni mics in gentle breezes, directional mics require more elaborate two-stage windscreens. For any amount of wind, a "blimp," which is much more effective and will kill on the order of 20 to 30 dB, is required. Companies such as Lightwave and Rycote make a variety of blimps and windscreens that are frequently used for location recording. For windscreens in general, the larger it is, the more effective it be. A spherical shape is best because it's the least affected from wind in all directions. (See Figure 1.16.)

*Figure 1.16*  
*A blimp windscreen (Lightwave Superscreen). Courtesy of Lightwave Audio.*



A trick that's been used on presidential outdoor speeches is to slip a condom (use the non-lubricated ones) over the microphone and then slip a foam windscreen over it to visually improve the arrangement. Although the frequency response will suffer, the wind noise will be attenuated.

### SHOCK MOUNTS

Shock mounts are designed to shield the microphone from picking up transmission noises that occur through the mic stand. Shock mounting is largely dependent on the mass of the microphone. Large-diaphragm mics are much more massive, and therefore present a greater inertia to mechanical noise. Small-diaphragm mics, on the other hand, have far less mass

and therefore do not present the inertia of their larger cousins. As a result, the shock mount has to be much “looser,” therefore causing the mount to be “floppier” and sometimes more difficult to position. (See Figure 1.17.)

*Figure 1.17*  
*A shock mount.*



# Basic Stereo Techniques

Even if you never intend to record an ensemble larger than a standard rock-and-roll rhythm section, a good grasp of the many techniques for stereo recording is essential and will come in handy sooner or later. Stereo miking is commonly used when recording drum kits, pianos, string sections, Leslies, and the like, and can certainly be applied to just about any recording situation.

So although we won't discuss these techniques in an orchestral music sense (where a lot of knowledge beyond the scope of this book is a necessity), this chapter will provide a basic overview of the many methods of stereo miking.

First of all, stereo miking is an improvement over mono miking because it provides:

- ▶ A sense of the soundfield from left to right
- ▶ A sense of depth or distance between each instrument
- ▶ A sense of distance of the ensemble from the listener
- ▶ A spatial sense of the acoustic environment—the ambience or hall reverberation

## Types of Stereo Miking

---

There are four general mic techniques used for stereo recording, each with a different sound and different sets of benefits and disadvantages:



- ▶ Coincident pair (including X/Y, M-S, and Blumlein)
- ▶ Spaced pair
- ▶ Near-coincident pair (the famous ORTF method)
- ▶ Baffled-omni pair or artificial head

### COINCIDENT PAIR

A coincident pair consists of two directional mics mounted so that their grilles are nearly touching, but with their diaphragms angled apart in such a way that they aim approximately toward the left and right sides of the ensemble. For example, two cardioid microphones can be mounted angled apart, their grilles one above the other. The greater the angle between microphones and the narrower the polar pattern, the wider the stereo spread.

### X/Y

While there are several variations of the coincident pair, the X/Y configuration is the easiest and mostly widely used. X/Y requires two identical directional microphones.

Unlike what you may think, the mics are not crossed in an X pattern in this configuration. In fact, the mic capsules are placed as close as possible to one another in a 90-degree angle. (See Figure 5.1.)

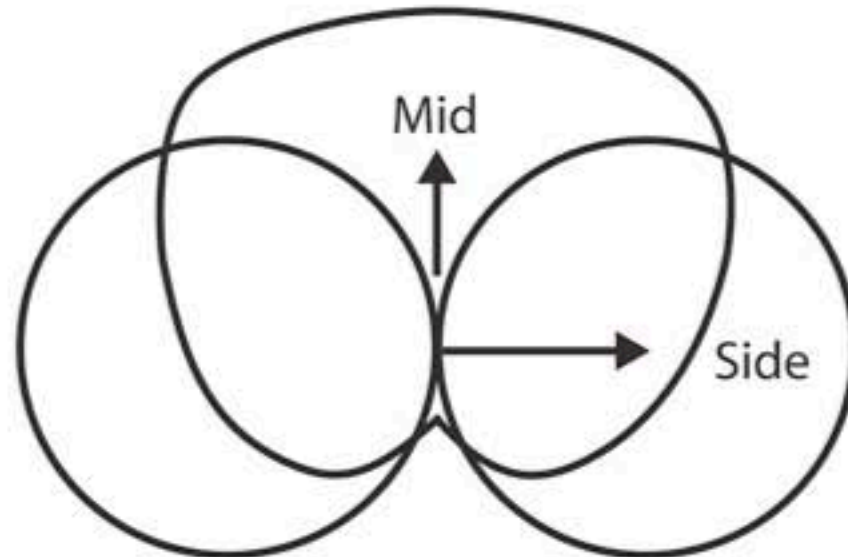
*Figure 5.1*  
*Two AKG 451s in an X/Y*  
*configuration.*



**M-S**

M-S stands for mid-side and consists again of two microphones—a directional mic (an omni can be substituted as well) pointed toward the sound source and a figure-8 mic pointed toward the sides. Once again, the mics are positioned so their capsules are as close to touching as possible. (See Figure 5.2.)

Figure 5.2  
M-S miking.



M-S is great for stereo imaging, especially when most of the sound is coming from the center of the ensemble. Because of this, it's less effective on large groups, favoring the middle voices to which the mics are closer.

M-S has no phase problems in stereo, with excellent mono compatibility, which can make it the best way to do room and ambience miking under the right circumstances. In many cases it can sound more natural than a spaced pair. If the source is extra large, sometimes using M-S alone will require too much distance to get the whole section or choir into perspective, so multiple mic locations must be used. If a narrower pickup pattern is required to attenuate hall sound, then a directional mic, such as a cardioid or even a hypercardioid, will work for the M mic. Just be aware that you may be sacrificing the bottom octaves as a result.

For best placement, walk around the room and listen to where the instrument or sound source sounds best. Note the balance of instrument to room, and the stereo image of the room as well. Once you have found a location, set up the directional mic where the middle of your head was.

Listening to either of these mics alone may sound okay, or they may even sound horribly bad. In order to make this system work, the mic's output signals need an additional "decoding" step to reproduce a faithful stereo image. The directional creates a *positive* voltage all the way around,

and the bidirectional mic is nulled to the front (of the source, the side of the mic) and creates a positive voltage from anything coming from the left and a *negative* voltage from anything coming from the right.

While you can buy an M-S decoder, you can easily emulate one with three channels on your console or DAW. On one channel, bring up the cardioid (M) forward-facing mic. Bring up the figure-8 mic (S) on two additional channels, either by muting the outputs or by patching from the "insert out" of the S channel to the "insert in" of an adjacent channel. Pan both channels to one side (like hard left), then run a tone down the first S channel, flip the phase of the second S channel, and bring up the level until the two channels cancel 100 percent.

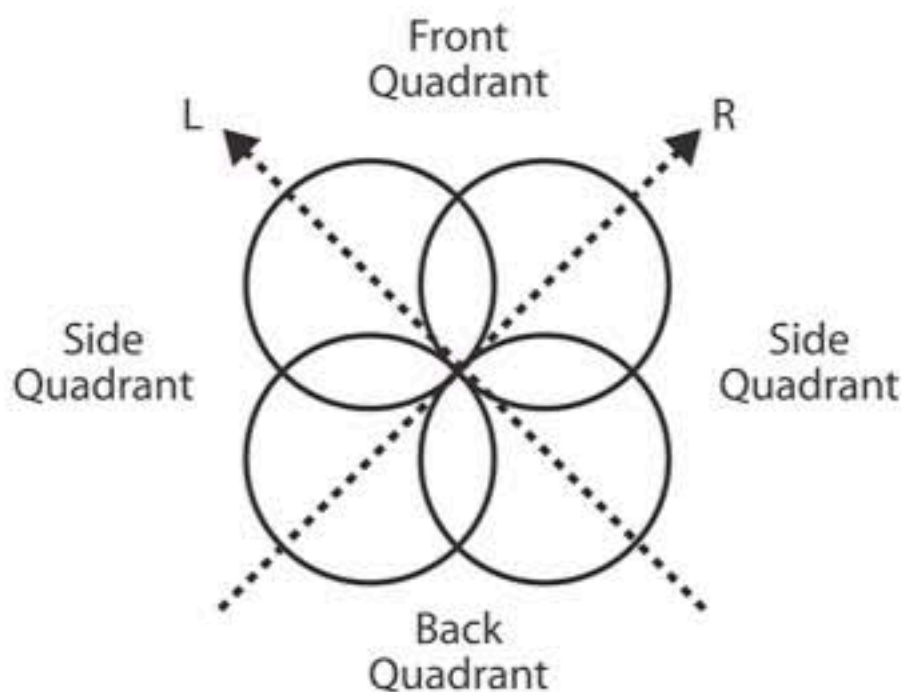
Now pan the first S channel hard left and the second S channel hard right, balance the cardioid (M) channel with your pair of S channels, and you have your matrix.

A nice additional feature of this method is that you are able to vary the amount of room sound (or change the focus) by varying the level of the bidirectional S mic.

#### **Blumlein Array**

Developed for EMI Records in 1935 by audio pioneer Alan Blumlein, the Blumlein stereo setup is a coincident stereo technique that uses two bidirectional microphones angled at 90 degrees to each other. This technique provides the best results when placed close to the sound source, since the low frequency response will decrease as the distance is increased. Blumlein stereo has a higher degree of separation between the channels than X/Y stereo, but must be used with care since it also picks up sound sources located behind the stereo pair.. (See Figures 5.3 and 5.4.)

Figure 5.3  
A Blumlein array.



*Figure 5.4  
Bruce Swedien with a Royer  
R-121 Blumlein array. Courtesy of  
Royer Labs.*



### **Stereo Mic**

Although not normally thought of as a coincident mic pair, a stereo mic uses two coincident mic capsules mounted in a single housing for convenience. Because of their close proximity to one another, this method may provide the easiest coincident mic setup. (See Figure 5.5.)

*Figure 5.5  
Royer SF-12 stereo microphone.  
Courtesy of Royer Labs.*



### Coincident-Pair Technique

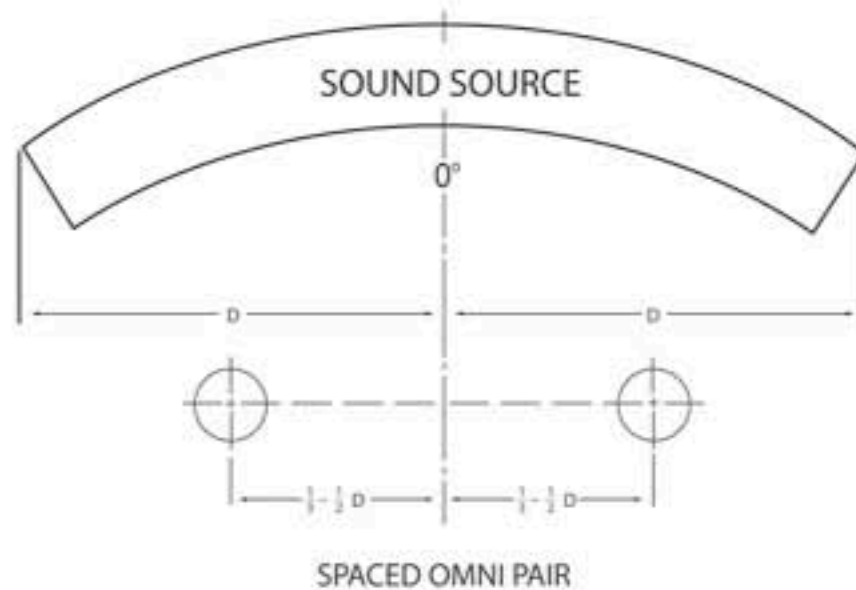
The coincident-pair technique has the following features:

- Imaging is very good.
- Stereo spread ranges from narrow to accurate.
- Signals are mono-compatible.
- Stereo is not as wide as other methods.

### SPACED PAIR

With the spaced-pair technique, two identical mics are placed several feet apart, aiming straight ahead toward the musical ensemble. The mics can have any polar pattern, but the omnidirectional pattern is the most popular for this method. The greater the spacing between mics, the greater the stereo spread. (See Figure 5.6.)

Figure 5.6  
Spaced-pair diagram.



If the spacing between mics is too far apart, the stereo separation seems exaggerated. On the other hand, if the mics are too close together, there will be an inadequate stereo spread. In addition, the mics will tend to favor the center of the ensemble because the mics are closest to the center instruments.

In an attempt to obtain a good musical balance, the mics are usually placed about 10 or 12 feet apart, but such spacing results in exaggerated separation. One solution is to place a third microphone midway between the original pair and mix its output to both channels. That way, the ensemble is recorded with a good balance, and the stereo spread is not as exaggerated.

The spaced-pair method tends to make off-center images relatively unfocused or hard to localize. In addition, combining both mics to mono sometimes causes phase cancellations of various frequencies, which may or may not be audible.

The advantage with spaced miking is a warm sense of ambience in which concert hall reverberation seems to surround the instruments and sometimes the listener. Another advantage of the spaced-mic technique is the ability to use omnidirectional microphones. An omni condenser mic has more extended low-frequency response than a unidirectional condenser mic and tends to have a smoother response and less off-axis coloration.

### Spaced-Pair Technique

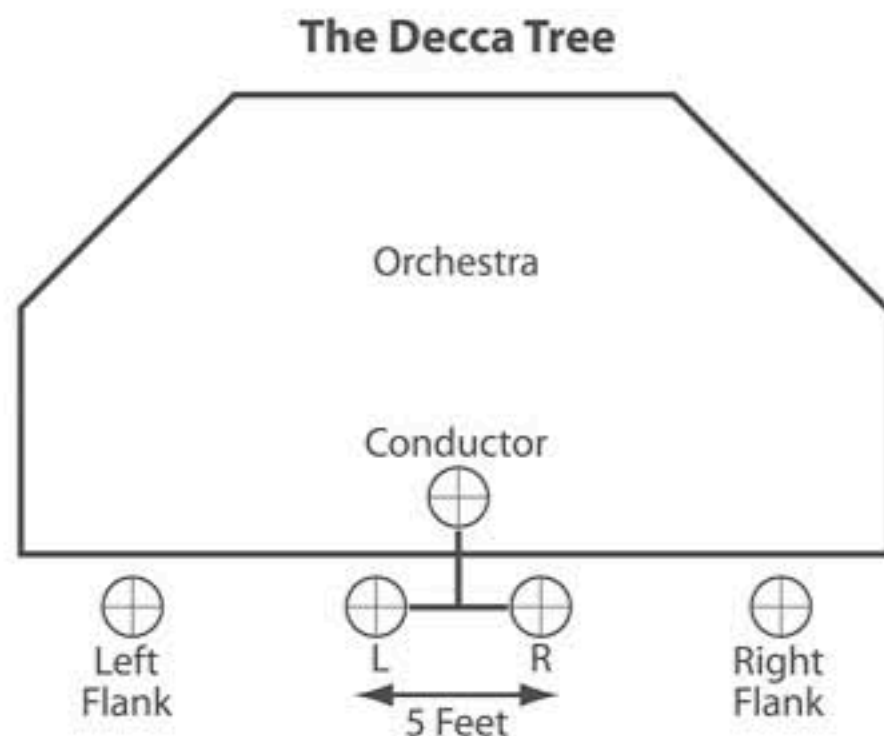
The spaced-pair technique has the following features:

- Off-center images are diffuse.
- Stereo spread tends to be exaggerated unless a third center mic is used.
- Provides a warm sense of ambience.
- Phasing problems are possible.

### THE DECCA TREE

A variation of the spaced pair is the Decca Tree, which is essentially a spaced pair with a center mic connected to a custom stand and suspended over the conductor (see Figure 5.7). Decca Records, who had a long tradition of developing experimental recording techniques, including surround sound and proprietary recording equipment, developed the Decca Tree as a compromise between the purist stereo pair and multi-mic arrays for orchestral recording by Decca engineers in 1950s. Apart from individual engineer's choice of mic, it remains unchanged to this day. It is still in use in film scoring/classical orchestral and opera recording because it produces a very spacious stereo image with good localization.

Figure 5.7  
The Decca Tree.



The classic Decca Tree setup uses three Neumann M 50s arranged in a triangle 10 to 12 feet above the conductor's position, although the spacing varies with venue and size of ensemble. The left mic is panned left, the right mic panned right, and the center mic panned to the center, with additional mics ("spot" mics or "sweeteners") used over violins (usually panned to the left), cellos (usually panned to the right), and harp/timps and soloist.

The distance between mics depends on the size of the ensemble. For orchestra, the left and right mics are 8 to 10 feet apart, with the center about 6 to 7 feet in front of the left-right axis.

It is a little-known fact that Decca has used (and maybe still uses) mics other than the M 50 on the tree. In particular, Decca has used M 49s and KM 56s on the tree, but modern substitutions include the TLM 50 or M 150, the Brauner VM1, or the DPA 4003 with APE spheres.

#### NEAR-COINCIDENT PAIR

The most common example of the near-coincident method is the ORTF system, which uses two cardioids angled 110 degrees apart and spaced 7 inches (17 cm) apart horizontally. (ORTF stands for *Office de Radiodifusión Television Française* or the *Office of French Radio and Television Broadcasting*.) This method tends to provide accurate localization; that is, instruments at the sides of the orchestra are reproduced at or very near the speakers, and instruments halfway to one side tend to be reproduced halfway to one side. ORTF provides a much greater sense of space due to time/phase differences since the capsules are as far apart as your ears. (See Figure 5.8.)

Figure 5.8  
An ORTF setup using AKG 451s.



### Near-Coincident Pair Technique

Near-coincident pair features include the following:

- Sharp imaging.
- Accurate stereo spread.
- A greater sense of air and depth than coincident methods.
- Wide image and depth of Blumlein without as much of the reverberant field.

### BAFFLED-OMNI PAIR

A baffled-omni pair, whether using two mics or a dummy head, is simply trying to emulate the way our ears are placed on our heads, and therefore the way we hear.

In this method, two omnidirectional mics are separated a few inches by a baffle between them. The baffle is a hard disk covered with absorbent foam (as in the Jecklin Disk; see Figure 5.9). Or, the baffle is a hard sphere with the mics flush-mounted on opposite sides (as in the Schoeps spherical mic; see Figure 5.10).



*Figure 5.9*  
*Jecklin Disk. Courtesy of Josephson*  
*Engineering.*



*Figure 5.10*  
*Schoeps KFM 360 spherical mic.*  
*Courtesy of Schoeps GmbH.*



With the baffled-omni pair, the level, time, and spectral differences between channels create the stereo images. The omni condenser mics used in this method have excellent low-frequency response.

Also falling into this category are dummy heads, such as the Neumann KU 100. (See Figure 5.11.)

*Figure 5.11*  
*Neumann KU 100 dummy head.*  
*Courtesy of Neumann USA.*



#### **Baffled-Omni Pair Technique**

The baffled-omni pair technique has the following features:

- Images are sharp.
- Stereo spread tends to be accurate.
- Low-frequency response is excellent.

For more information on stereo recording:

- ▶ [www.kellyindustries.com/microphones/stereo\\_miking\\_techniques.html](http://www.kellyindustries.com/microphones/stereo_miking_techniques.html)
- ▶ [www.tape.com/Bartlett\\_Articles/stereo\\_microphone\\_techniques.html](http://www.tape.com/Bartlett_Articles/stereo_microphone_techniques.html)
- ▶ [www.josephson.com](http://www.josephson.com) (click on Tech Notes and then Jecklin Disk)
- ▶ [www.csun.edu/~record/stereo](http://www.csun.edu/~record/stereo)
- ▶ [www.prosoundweb.com/studyhall/shure/mics/stereo/techniques.php](http://www.prosoundweb.com/studyhall/shure/mics/stereo/techniques.php)
- ▶ [homerecording.about.com/od/microphones101/ss/stereo\\_mics.htm](http://homerecording.about.com/od/microphones101/ss/stereo_mics.htm)
- ▶ [www.shurenates.com/issue25/article.asp?flash=true](http://www.shurenates.com/issue25/article.asp?flash=true)

# Basic Multichannel Tracking

Although it's possible to get a good sound by just randomly placing a mic in front of an instrument or vocal, it can sound so much better with a little forethought regarding the many concerns involved. And, the more mics you record with, the more complex these concerns become. This chapter will give you some insights on how to get good sounds quickly.

## Choosing the Right Mic

---

While it's safe to say that most engineers rely on experience when choosing microphones, there are some things to think about when selecting a microphone.

**MICHAEL BEINHORN:** *There's no one microphone that does every single thing.*

- ▶ **Select a microphone that complements the instrument you'll be recording.** For instance, if you have an instrument that has a very edgy top end, you wouldn't want to choose a mic that also has that quality, since those frequencies will be emphasized. Instead, choose a mic that's a bit mellower, such as a ribbon. This is one of the reasons why a ribbon mic works so well on brass, for instance.
- ▶ **Is the mic designed to be used in the free field or in the diffuse field?** *Free field* means that the sound source dominates what the mic hears. *Diffuse field* means that the reflections play a large role in what the mic hears. Mics designed for free-field use have a very flat frequency response in the high frequencies and, as a result, can sound dull when placed farther away. Diffuse-field mics have a boost in the upper frequencies that makes them sound flat when placed farther away.

- ▶ **Select a mic that won't be overloaded by the source.** You wouldn't want to put a ribbon mic or many condensers on a snare drum with a heavy-hitting drummer, for instance.
- ▶ **Choose the right polar pattern for the job.** If leakage is a consideration, then choose a mic with the proper directional capabilities for the job. If a mic is flat on-axis, it will roll off the highs when it's 90 degrees off-axis. If it's flat 90 degrees off-axis, it will have a rising high end when it's on-axis.
- ▶ **Is proximity effect an issue?** If you're close-miking, will the bass buildup from proximity be too much? If so, consider an omni.

### Microphone Considerations

1. Condensers of a given polar pattern will tend to give you more room sound than dynamics of the same polar pattern.
2. Omnis will give you lower bass extension than cardioids.
3. Large-diaphragm condensers have lower self-noise than small-diaphragm condensers.
4. Small-diaphragm condensers are generally less colored off-axis than large-diaphragm condensers.

## The Secret to Getting Good Sounds

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Contrary to what many who are starting out in recording might think, just having great equipment doesn't guarantee a great sound. While you can't really quantify how much each variable contributes to how something ultimately sounds (because each situation, even within the same project, is unique), you can generally break it down to something like this:

- ▶ The player and the instrument contribute about 50 percent to the overall sound (sometimes a little more, sometimes a little less—but always the greatest portion).
- ▶ The room contributes about 20 percent to the overall sound. (Even on close-miked instruments, the room is far more responsible for the ultimate sound than many engineers realize.)
- ▶ The mic position contributes about 20 percent to the overall sound. (Placement is really your acoustic EQ and is responsible for the instrument's blend in the track.)

- ▶ The mic choice contributes about 10 percent to the overall sound. (This is the last little bit that takes a good sound and makes it great.)

If something doesn't sound right, there are many things to change before you reach for the EQ. Try the following in this order:

1. Change the source (the instrument you are miking), if possible.
2. Change the mic placement.
3. Change the placement of the instrument in the room.
4. Change the mic.
5. Change the mic preamplifier.
6. Change the amount of compression and/or limiting (from none to a lot).
7. Change the room (the actual room you are recording in).
8. Change the player.
9. Come back and try it another day.

**AL SCHMITT:** *I don't use any EQ when I record. I use the mics for EQ. I don't even use any compression. The only time I might use a little bit of compression is maybe on the kick, but for most jazz dates I don't.*

**MACK:** *One of my big things is not to use EQ, or as little as possible, and not to add any, but find what's offensive and get rid of that as opposed to cranking other stuff to compensate.*

### SECRETS OF MIC PLACEMENT

Quickly finding a mic's optimum position is perhaps the single most useful talent an engineer can have. Bruce Sweiden says mics are the voodoo magic of recording. Steve Albini says every mic has someplace where it will sound best. Sometimes the search resembles questing for the Holy Grail. You should always trust your ears and begin by listening to the musician in your studio, find a sweet spot, and then begin your microphone placement there. If you don't like the resultant sound, then move the mic or swap it with another. EQ is the last thing you should touch.

**FRANK FILIPETTI:** *I've been doing this long enough to know that a change in microphones or position is worth a lot more than tweaking EQs. I have a tendency to think that if you start tweaking EQs too soon, you're going to miss some obvious things, so the first thing I do is get the session sounding great flat.*

**Remember:** Mics cannot effectively be placed by sight, which is a mistake that is all too easy to make (especially after reading a book like this). The best mic position cannot be predicted, it must be found.

### To Find the Sweet Spot

- To correctly place an omni microphone, cover one ear and listen with the other. Move around the player or sound source until you find a spot that sounds best.
- To place a cardioid microphone, cup your hand behind your ear and listen. Move around the player or sound source until you find a spot that sounds best.
- For a stereo pair, cup your hands behind both ears. Move around the player or sound source until you find a spot that sounds best.

Before you start swapping gear, know that the three most important factors in getting the sound you want are *mic position*, *mic position*, and *mic position*.

1. Get the instrument to make the sound you want to record first. If you can't hear it, you can't record it.

**AL SCHMITT:** *The major trick in all of this...is that you go out in the studio, stand next to the conductor, and listen to what's going on. Your job is to go in and capture exactly what he wants to hear out there. So my microphone techniques are still the same as they were 30 years ago.*

2. Use the cover-one-ear-and-listen technique described a moment ago to find the best place to start experimenting with mic position.
3. Position the mic and listen. Repeat as much as necessary.

#### Placement Considerations

The following are some issues to consider before mic placement:

- ▶ The only reasons for close-miking are to avoid leakage to other mics so that the engineer can have more flexibility in balancing the ensemble in the mix. If at all possible, give the mic some distance from the source in order to let the sound develop and be captured naturally.

**EDDIE KRAMER:** *In regard to mic techniques, what I adapted was this classical idea of recording—i.e., the distance of the microphones to the instruments should not be too close if you wanted to get anything with tremendous depth. Obviously, I used close-miking techniques as well, but it started with the concept that “distance makes depth” that Bob Auger taught me. Generally, the basic philosophy of getting the mics up in the air and getting some room sound and some air around the instrument was what we used. Then you’d fill in with the close mics.*

- ▶ Mics cannot effectively be placed by sight. The best mic position cannot be predicted, it must be found. It’s okay to start from a place that you know has worked in the past, but be prepared to experiment with the placement a bit because each instrument and situation is different.
- ▶ If the room ambience will be the majority of the sound you want to record, start with those mics that pick up the room and then add mics that act as support to the room mics.
- ▶ To overcome phase problems, consider inserting an X/Y phase scope on your stereo bus. Leave it running all the time so your eyes and ears get in sync with what is in phase and what is not. Monitoring phase this way does not guarantee good mic placement, but it does allow bad placement to be spotted more easily.
- ▶ Around 300 Hz is where the proximity effect often shows up and is why many engineers continually cut in this range. If many directional microphones are being used, they will be subject to proximity effect, and you should expect a buildup of this frequency range in the mix.

**ED CHERNEY:** *“If something is a little dark, then it might be because 2 or 300 is building up, so you dip a little of that out....”*

- ▶ A huge sound is a larger-than-life sound. One way to accomplish a larger-than-life sound is by recording a sound that is softer than the recording will most likely be played back. Ever listen to the guitar sounds on Eric Clapton’s seminal recording of “Layla?” Both he and Duane Allman used little Fender Champ and Princeton Reverb amplifiers (the Champ is 6 watts into an 8-inch speaker, and the Princeton is 12 watts into a 10-inch), but the guitar sounds are huge.

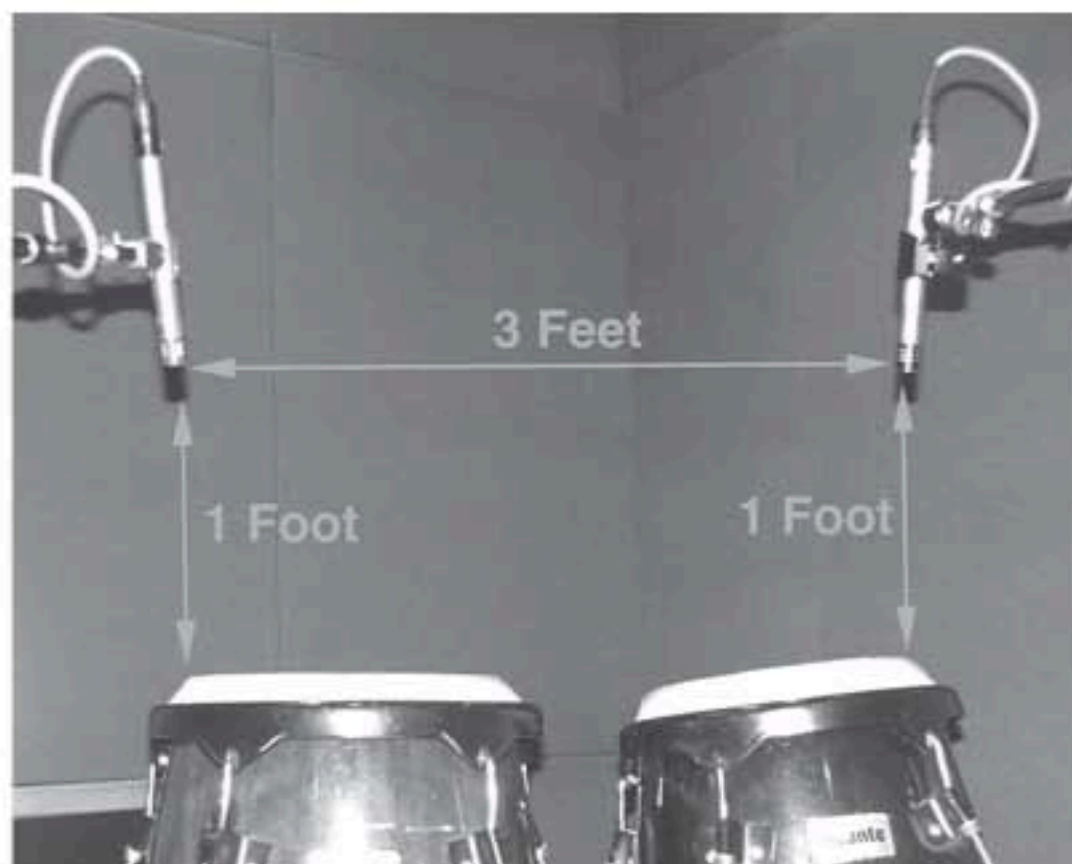
### **The 3-to-1 Principle**

The 3-to-1 principle is pretty important when considering any multi-mic setups because, if you observe the rule, you can stop any phase problems before they start. Simply put, the 3-to-1 principle states that in order to maintain phase integrity between microphones, for every unit of distance between the mic and its source, the distance between any other mics should be at least three times that distance. For instance, if a pair of microphones



were placed over the sound board of a piano at a distance of 1 foot, the separation between the two mics should be at least 3 feet. If the distance from the source were 2 feet, the distance between mics should be at least 6 feet. (See Figure 6.1.)

Figure 6.1  
The 3-to-1 principle.



This principle is not a hard-and-fast rule, but it certainly is a good guideline for eliminating phase problems. Remember, if you record something with a phase problem, no amount of EQ or processing afterward can ever make it right.

### CHECKING PHASE

Checking microphone phase should be one of the first things that an engineer does after the mics are wired up and tested. This is especially the case in a tracking session where a lot of mics will be used, since having just one mic out of phase can cause uncorrectable sonic problems that will haunt the recording forever. A session that is in phase will sound bigger and punchier, while just a single out-of-phase mic will make the entire mix sound tiny and weak.

If you're going to be absolutely thorough, there are actually two tests—one for polarity and one for phase. The polarity check is used mainly to be sure that all mics are pushing and pulling the same way and to check for miswired cables. (Yes, they're out there, especially if you build your own.) The phase check will make sure that you minimize the interference between the mics when they're placed.

# Glossary

- ADC** Analog-to-digital converter. This device converts the analog waveform into the form of digital 1s and 0s.
- AIFF** Audio Interchange File Format (also known as *Apple Interchange File Format*) is the most used audio file format in the Apple Macintosh operating system.
- API** Automated Process Incorporated; an American console manufacturer noted for their sonic qualities.
- attack** The first part of a sound. On a compressor/limiter, a control that affects how that device will respond to the attack of a sound.
- attenuation** A decrease in level.
- automation** A system that memorizes and then plays back the position of all faders, mutes, and sometimes panning and EQ on a console or in a DAW.
- bandwidth** The number of frequencies that a device will pass before the signal degrades. A human being can supposedly hear from 20 Hz to 20 kHz, so the bandwidth of the human ear is 20 Hz to 20 kHz.
- bass management** A circuit that utilizes the subwoofer in a 5.1 system to provide bass extension for the five main speakers. The bass manager steers all frequencies below 80 Hz into the subwoofer along with the LFE source signal. *See LFE.*
- bidirectional** A microphone with a figure-8 pickup pattern. *See Chapter 1.*
- binaural** A stereo recording technique using a model of a human head with microphones placed where the ears would be. This type of recording provides exceptional reproduction using headphones, but does not translate well to speakers.
- Blumlein** A stereo miking configuration utilizing two figure-8 microphones. *See Chapter 5.*
- buss** A signal pathway.

- capsule** The part of a microphone that contains the primary electronic pickup element. *See Chapter 1.*
- cardioid** A microphone that has a heart-shaped pickup pattern. *See Chapter 1.*
- chamber (reverb)** A method of creating artificial reverberation by sending a signal to a speaker in a tiled room that is picked up by several microphones placed in the room.
- chorus** A type of signal processor in which a detuned copy is mixed with the original signal, which creates a fatter sound.
- click** A metronome feed to the headphones to help the musicians play at the right tempo.
- clip** To overload and cause distortion.
- close-miking** Placing a mic close to an instrument in order to decrease the pickup of room reflections or other sound sources.
- coincident pair** A pair of the same model microphones placed with their capsules as close together as possible. *See Chapter 5.*
- color** To affect the timbral qualities of a sound.
- comb filter** A distortion produced by combining an electronic or acoustic signal with a delayed copy of itself. The result is peaks and dips introduced into the frequency response. This is what happens when a signal is flanged. *See flanging.*
- condenser** A microphone that uses two electrically charged plates (thereby creating an electronic component known as a *condenser*) as its basis of operation. *See Chapter 1.*
- contractor** The person who hires the musicians for a session. Mostly used for orchestral sessions.
- cue mix** The headphone mix sent to the musicians that differs from the one that the producer and engineer are listening to. *See Chapter 9.*
- cut** To decrease, attenuate, or make less.
- DAC** Digital-to-analog converter. This device converts digital 1s and 0s back to an analog waveform.
- DAW** Digital Audio Workstation. A computer with the appropriate hardware and software needed to digitize and edit audio.

- decay** The time it takes for a signal to fall below audibility.
- Decca Tree** A microphone arrangement used primarily for orchestral recording that uses a spaced pair with a center mic connected to a custom stand and suspended over the conductor. *See Chapter 5.*
- delay** A type of signal processor that produces distinct repeats (echoes) of a signal.
- DI** Direct Injection (also known as a direct box); an impedance-matching device that bypasses the use of a microphone and makes it possible to connect an electric instrument, such as an electric guitar, directly into a console, mixer, or recorder. *See Chapter 3.*
- diaphragm** The element of a microphone moved by sound pressure. *See Chapter 1.*
- digital domain** When a signal source is digitized, or converted into a series of electronic pulses represented by 1s and 0s, the signal is then in the digital domain.
- digitize** To record from the analog to the digital domain.
- direct** To “go direct” means to bypass a microphone and connect the guitar, bass, keyboard, and other electronic instruments directly into a recording device.
- direct box** *See DI.*
- directional** A microphone that has most of its pickup pattern in one direction. *See Chapter 1.*
- Dolby SR** Dolby Spectral Recording; a noise-reduction process used with analog tape.
- double** To play or sing a track a second time. The inconsistencies between both tracks make the part sound bigger.
- dynamic** A dynamic microphone changes acoustic energy into electrical energy by the motion of a diaphragm through a magnetic field. *See Chapter 1.*
- echo** For older engineers this is another word for reverb. For newer engineers this is another word for delay.
- edgy** A sound with an abundance of midrange frequencies.
- EMT 250** A digital reverb (the very first digital reverb, in fact) noted for its smooth sound.
- EQ** Equalizer, or to adjust the equalizers (tone controls) to affect the timbral balance of a sound.

- equalizer** A tone control that can vary in sophistication from very simple to very complex. *See parametric equalizer.*
- exciter** An outboard effects device that uses phase manipulation and harmonic distortion to produce high-frequency enhancement of a signal.
- FET** Field Effect Transistor; a solid state electronic component that has many of the same electronic qualities as a vacuum tube. Meant as a replacement for the vacuum tube, the FET has a much longer useful lifetime but lacks the sonic qualities.
- figure-8** A microphone with a pickup pattern primarily from the front and rear. *See Chapter 1.*
- FireWire** A fast computer interface that uses what's known as a *peer-to-peer* configuration, where each device has some built-in intelligence that can determine the best way for data to transfer. Usually used when more than just a few tracks are to be recorded at the same time.
- 5.1** A speaker system that uses three speakers across the front and two stereo speakers in the rear, along with a subwoofer.
- flam** A sound source played slightly off-time with another.
- flanging** The process of mixing a copy of the signal back with itself, but gradually and randomly slowing the copy down to cause the sound to "whoosh" as if it were in a wind tunnel. This was originally done by holding a finger against a tape flange (the metal part that holds the tape on the reel), hence the name.
- Fletcher-Munson Curves** A set of measurements that describe how the frequency response of the ear changes at different sound pressure levels. For instance, we generally hear very high and very low frequencies much better as the overall sound pressure level is increased.
- gobo** A portable wall used to isolate one sound source from another.
- groove** The pulse of the song and how the instruments dynamically breathe with it.
- ground** A switch on some audio devices (mostly guitar amps and direct boxes) used to decrease hum.
- hi-pass filter** An electronic device that allows the high frequencies to pass while attenuating the low frequencies. Used to eliminate low-frequency artifacts such as hum and rumble.

- impedance** The electronic measurement of the total electronic resistance to an audio signal.
- input buffer** A temporary memory location that collects the bursts of digital information and sends it smoothly to the CPU in the computer.
- I/O** The input/output of a device.
- latency** Latency is a measure of the time it takes (in milliseconds) for your audio signal to pass through your system during the recording process. This delay is caused by the time it takes for your computer to receive, understand, process, and send the signal back to your outputs.
- lavaliere** A small microphone (sometimes called a *tie tac* or *lav*) favored by broadcasters because of its unobtrusiveness. *See Chapter 1.*
- layer** To make a larger, more complex sound picture by adding additional tracks via overdubbing.
- leakage** Acoustic spill from a sound source other than the one intended for pickup.
- Leslie** A speaker cabinet, usually used with a Hammond organ, that features rotating high- and low-frequency speakers.
- LFE** Low Frequency Effects channel. This is a special channel of 5- to 120-Hz information primarily intended for special effects, such as explosions in movies. The LFE has an additional 10 dB of headroom in order to accommodate the required level.
- line level** The normal operating signal level of most professional audio gear. The output of a microphone is boosted to line level by a preamplifier.
- makeup gain** A control on a compressor/limiter that applies additional gain to the signal. This is required since the signal is automatically decreased when the compressor is working. Makeup gain “makes up” the gain and brings it back to where it was prior to being compressed.
- Marshall cabinet** The most widely used guitar speaker cabinet. It contains four 12-inch speakers and is manufactured by Jim Marshall Amplifiers.
- modulate** The process of adding a control voltage to a signal source in order to change its character. For example, modulating a short slap delay with a 0.5-Hz signal will produce chorusing. *See chorus.*
- M-S** Mid/Side; a stereo microphone technique utilizing a directional and a figure-8 microphone. *See Chapter 5.*

- mute** An on/off switch. To mute something would mean to turn it off.
- near-field** The listening area where there is more direct than reflected sound.
- Neve** An English console manufacturer noted for its sonic qualities.
- non-coincident pair** A stereo miking technique in which two microphones are placed apart from one another at the approximate distance of your ears. *See Chapter 5.*
- null** The point on the microphone pickup pattern where the pickup sensitivity is at its lowest.
- off-axis** A sound source away from the primary pickup point of a microphone.
- omnidirectional** A microphone that picks up sound equally from any direction. *See Chapter 1.*
- on-axis** A sound source aimed at the primary pickup point of a microphone.
- ORTF** Office de Radiodiffusion Télévision Française; a stereo-miking technique developed by the Office of French Radio and Television Broadcasting using two cardioid mics angled 110 degrees apart and spaced 7 inches (17 cm) apart horizontally. *See Chapter 5.*
- overalls** In orchestral recording, the primary microphone arrangement (such as a Decca Tree).
- overheads** The microphones placed above the head of a drummer used to pick up either the entire kit or just the cymbals. *See Chapters 7 and 8.*
- pad** An electronic circuit that attenuates the signal (usually either 10 or 20 dB) in order to avoid overload.
- parametric equalizer** A tone control in which the gain, frequency, and bandwidth are all variable.
- phantom image** In a stereo system, if the signal is of equal strength in the left and right channels, the resultant sound appears to come from in between them. This is a phantom image.
- phase meter** A dedicated meter that displays the relative phase of a stereo signal. *See Chapter 6.*
- phase shift** The process during which some frequencies (usually those below 100 Hz) are slowed down ever so slightly as they pass through a device. This is usually exaggerated by excessive use of equalization and is highly undesirable.

- plate (reverb)** A method to create artificial reverberation using a large steel plate with a speaker and several transducers connected to it.
- plug-in** An add-on to a computer application that adds functionality to it. EQ, modulation, and reverb are examples of DAW plug-ins.
- pop filter** A piece of acoustic foam, placed either internally near the diaphragm or externally over the mic, designed to reduce plosives, or “pops.” *See Chapter 1.*
- preamplifier** An electronic circuit that boosts the tiny output of a microphone to a level more easily used by the other electronic devices in the studio.
- predelay** A variable length of time before the onset of reverberation. Predelay is often used to separate the source from the reverberation so the source can be heard more clearly.
- presence** Accentuated upper midrange frequencies (anywhere from 5 to 10 kHz).
- proximity effect** The inherent low-frequency boost that occurs with a directional microphone as the signal source gets closer to it. *See Chapter 1.*
- Pultec** An equalizer sold during the '50s and '60s by Western Electric that is highly prized today for its smooth sound.
- pumping** When the level of a mix increases and then decreases noticeably. Pumping is caused by the improper setting of the attack and release times on a compressor.
- punchy** A description for a quality of sound that infers good reproduction of dynamics with a strong impact. Sometimes means emphasis in the 200-Hz and 5-kHz areas.
- PZM** Pressure-zone microphone. *See Chapter 1.*
- Q** Bandwidth of a filter or equalizer.
- range** On a gate or expander, a control that adjusts the amount of attenuation that will occur to the signal when the gate is closed.
- ratio** A parameter control on a compressor/limiter that determines how much compression or limiting will occur when the signal exceeds threshold.
- recall** A system that memorizes the position of all pots and switches on a console. The engineer must still physically reset the pots and switches back to their previous positions as indicated on a video monitor.



- reference level** This is the sound pressure level at which a sound system is aligned.
- release** The last part of a sound. On a compressor/limiter, a control that affects how that device will respond to the release of a sound.
- resonant frequency** A particular frequency or band of frequencies that are accentuated, usually due to some extraneous acoustic, electronic, or mechanical factor.
- return** Inputs on a recording console especially dedicated for effects devices such as reverbs and delays. The return inputs are usually not as sophisticated as normal channel inputs on a console.
- reverb** A type of signal processor that reproduces the spatial sound of an environment (such as the sound of a closet or a locker room or the inside of an oil tanker).
- Rhodes** An electronic piano designed by Harold Rhodes and marketed by Fender in the '60s and '70s.
- ribbon** A microphone that utilizes a thin aluminum ribbon as the main pickup element. *See Chapter 1.*
- roll-off** Usually another word for hi-pass filter, although it can refer to a low-pass filter as well.
- rotor** The high-frequency rotating speaker of a Leslie tone cabinet.
- scope** Short for *oscilloscope*, an electronic measurement device that produces a picture of the audio waveform.
- sibilance** A rise in the frequency response in a vocal where there's an excessive amount of 5 kHz, resulting in the "S" sounds being overemphasized.
- soundfield** The direct listening area.
- SoundField** A B-format microphone for recording stereo or 5.1. *See Chapter 11.*
- spaced pair** A stereo-miking technique in which the microphones are placed several feet apart. *See Chapter 5.*
- SPL** Sound pressure level.
- spot mic** A microphone used during orchestral recording to boost the level of an instrument or soloist.

- stems** Mixes that have their major elements broken out separately for individual adjustment at a later time.
- sub** Short for *subwoofer*.
- subwoofer** A low-frequency speaker with a frequency response from about 25 Hz to 120 Hz.
- sweetener** Another name for a spot mic.
- synchronization** When two devices—usually storage devices such as tape machines, DAWs, or sequencers—are locked together with respect to time.
- talkback** The communication link between the control room and the cue mix in the musicians' headphones allowing the producer or engineer to speak with the musicians.
- threshold** The point at which an effect takes place. On a compressor/limiter, for instance, the Threshold control adjusts the point at which compression will take place.
- timbre** Tonal color.
- transformer** An electronic component that either matches or changes the impedance. Transformers are large, heavy, and expensive but are in part responsible for the desirable sound in vintage audio gear. *See Chapter 3.*
- trim** A control that sets the gain of a device, usually referred to on a microphone preamplifier. *See Chapter 3.*
- tube** Short for vacuum tube; an electronic component used as the primary amplification device in most vintage audio gear. Equipment utilizing vacuum tubes runs hot, is heavy, and has a short life, but has a desirable sound.
- tunnel** A makeshift extension mounted to a bass drum used to isolate a microphone placed away from the drum head. *See Chapter 8.*
- USB 2.0** A computer interface used by many DAW I/O boxes that uses a *master-slave* configuration that adds system overhead, which results in slower data flow.
- windscreen** A device placed over a microphone to attenuate the noise caused by wind interference.
- X/Y** A stereo-miking technique in which the microphone capsules are mounted as closely as possible while crossing at 90 degrees. *See Chapter 5.*