RECORD NOW VOL. 1

A Quick-Start Guide

Choosing & Using Microphones

Presented by:

M-AUDIO

Contents

Chapter 1

Microphone Design

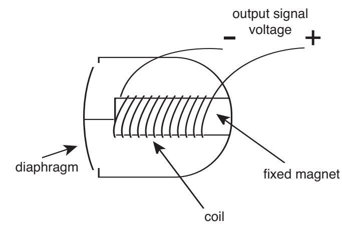
While all microphones are designed for the common purpose of converting variations in sound pressure to electronic signals, different technologies have their benefits depending upon the application. This chapter examines the merits of different design types, capsule sizes, polar patterns, electronics and more.

Microphone Types

The three main types of microphones in common use today are dynamic, ribbon and condenser. Each has unique attributes appropriate for different applications.

Dynamic microphones

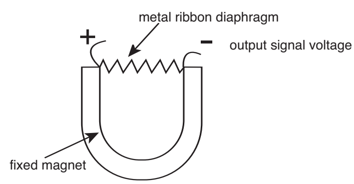
The *dynamic* or *moving-coil* microphone is the easiest to understand. It is the classic technology taught in grade school as the inverse of the common speaker. A plastic or metal diaphragm is attached to a copper coil that is, in turn, suspended in a magnetic field. Sound pressure waves hitting the diaphragm cause it to move, and with it, the coil within the magnetic field. The resulting magnetic fluctuations translate to electrical fluctuations generally corresponding to the physical fluctuations of the original sound wave.



In dynamic mics, sound pressure moving the diaphragm causes the attached voice coil to interact with a magnetic field to produce an electric signal

Due to the requirement of attaching the coil directly to the diaphragm, dynamic diaphragms are thicker and, therefore, less sensitive than the ribbon and condenser microphones discussed shortly. These same design considerations also give the ability to take the greatest amount of sound pressure before distorting, as well as the greatest amount of physical abuse. Dynamics are also the easiest and least expensive to make. Dynamics also to tend to color the sound in the range of about 5k to 10k, and start to sound thinner when more than about a foot away from the source.

For these reasons, dynamic mics are most often found in the average stage situation. After all, live performance environments are much more likely to subject mics to torture such as high volume, sweat, the elements, shock and being dropped. In the studio, dynamic mics are most often used to close-mic drums due to the possibility of wayward drum sticks. Large-diaphragm dynamics are often used on kick drums due to high sound pressure levels and low-frequency content.



In ribbon mics, sound waves cause a thin metal ribbon to move within a magnetic field to produce a current

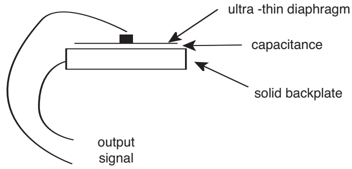
Ribbon microphones

Ribbon mics are another form of dynamic microphone distinct from the moving-coil variety. A very thin metal ribbon suspended between the poles of a powerful magnet moves in response to sound waves, thus cutting through the magnetic field and inducing a flow of electrons. The resulting low-voltage output is typically fed to a step-up transformer and sent down the mic cable. The extreme thinness of the ribbon makes this type of mic the most sensitive, especially at very low sound levels. They are most often used in close-miking situations and, because they are also the most fragile and costly mic design, ribbons are typically reserved for very controlled conditions.

Like moving-coil dynamics, ribbon mics color the sound in a way that is often employed to warm up brassy sounds. (Ribbons are a great choice for recording sax, for example.) They also tend to have very low output, thereby requiring more electronic gain a factor that necessitates high-quality preamp electronics in order to avoid noise.

Condenser microphones

Condenser mics are the most common for studio use. A thin electrically conductive diaphragm is suspended over a back plate, forming a delicate flexible capacitor. When sound waves excite the diaphragm, the distance between the diaphragm and back plate changes and with it the capacitance. This capacitance change, in turn, produces a voltage change. Associated circuitry converts these variations in voltage to a signal that is sent to the preamp. The power required by this design is serviced by the 48-volt phantom power commonly found on preamps and mixer inputs.



In condenser mics, sound waves hitting the diaphragm change the capacitance in the field between the charged diaphragm and backplate

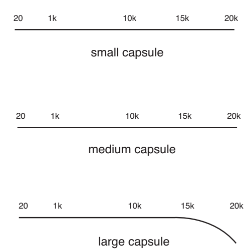
The diaphragms of condenser microphones are made of extremely thin metal or metalized plastic similar in thickness to kitchen plastic wrap. Their thinness makes condenser mics very accurate in frequency response and extremely sensitive to transients, such as the initial crack of a drum being struck. In addition to imparting the least sonic coloration of any microphone design, the sensitivity of condensers extends much further from the source than other mics, thus allowing greater flexibility. This greater sensitivity also provides the engineer with the option of picking up more the room ambience a factor that can add a great deal of realism to a recording.

Condensers are more delicate than moving-coil dynamics, yet much more resilient than ribbons. Due to sensitivity to low-frequency handling noise and the delicacy of the diaphragm, condensers are invariably used in conjunction with a shock mount, and often with the addition of a pop filter. The sonic characteristics of condensers and the need for TLC make them more ideally suited for studio recording. That is not to say that condensers can t be used for some tasks on stage just that the environment should be controlled, such as in a professional show where cables are secured, mics are shock-mounted against vibration, and the stage is restricted to professional personnel.

Since condenser construction technology is much more labor-intensive and sophisticated compared to that of dynamics, good quality condensers tend to cost comparatively more money. Condensers are excellent choices for miking vocals, acoustic guitar, piano, orchestral instruments, saxophone, percussion and sound effects. As condensers are the predominant type of microphone for studio use, this guide will focus on condenser applications.

Capsule Size

The capsule incorporates the all-important diaphragm assembly that translates sound pressure into an electric signal. Condenser capsules come in three basic sizes small, medium and large. Generally speaking, frequency response is a function of diaphragm size. Consider what happens with speakers of different size. As woofers get larger, they become more efficient at producing low frequencies and less efficient at producing high frequencies. In general, the same is true as the diameters of diaphragms increase (with some caveats we ll cover in a minute).



Without intervention, microphones tend to be less linear as the diaphragm size increases

Signal-to-noise ratio of the microphone as a whole generally owes in part to diaphragm size. The more surface area that a diaphragm has, the greater its potential sensitivity to sound pressure and the stronger the output signal. As a result, large diaphragms inherently exhibit much better signa-to-noise ratios than do small ones.

Small Capsules

Small capsules are typically those with diaphragm diameters of less than about 1/2 . Categorically, they are extremely accurate through the audible range of 20Hz to 20kHz. Their poor signal-to-noise ratio, however, requires tricks with electronics and relegates small capsules to being most useful for measurement rather than recording.

Medium Capsules

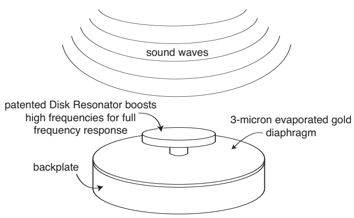
Medium capsules have diaphragms that are approximately 1/2 inch to 3/4 inch in diameter. Given the right design and manufacturing, they typically exhibit flat frequency response from about 20 to 18k. Their diaphragms are also large enough to deliver signal-to-noise ratios acceptable for professional use.

Large Capsules

Large capsules have diaphragms measuring 3/4 inch to one inch or even greater. Since larger diaphragms yield better signal-to-noise ratios and greater sensitivity without having to induce additional gain stages, bigger is typically considered better. Large capsules also tend to produce greater low frequency detail a quality that can t be measured so much as heard. Large capsules exhibit a proximity effect (most predominantly in the cardioid polar pattern), meaning that they tend to sound more boomy as they get closer to the source. Large diaphragm M-Audio mics include the Solaris, Luna and Nova.

The Diaphragm

The *diaphragm* is a critical component because it is responsible for responding directly to sound waves. The sensitivity of a mic is partially related to the thinness of its diaphragm. (Recall that the comparatively thin diaphragm of a condenser is largely what makes this type of mic much more linear and sensitive to detail than a dynamic moving-coil mic.)



Originally, condenser diaphragms were made from very thin, light metal such as nickel. As technology evolved, it became possible to use synthetic materials such as mylar in order to create tissue-thin membranes. Since condenser diaphragms need to conduct electricity, these synthetic materials have a thin layer of gold applied to them the thinner, the better. Most modern condenser diaphragms are 6 to 12 microns in thickness. (A human hair is 40 microns in diameter.) The M-Audio large capsule mic, the Solaris, employs a special ultra-thin 3-micron, highly resilient mylar diaphragm. This delivers a degree of sensitivity unparalleled in the industry. (Physics dictates that we employ 6-micron diaphragms in our Luna and Nova models.)

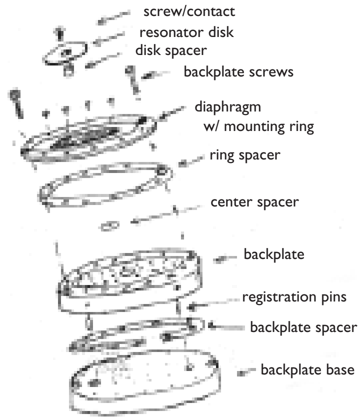
In the old days, manufacturers would apply the gold to the diaphragm using a process known as sputtering. They would place the diaphragm substrate in a vacuum jar, atomize the gold, and then blow the gold onto one side of the material. Today s vacuum chambers are far superior, allowing us to use a refined technique where we place our ultra-thin mylar film in a complete vacuum and evaporate the gold in such a way that it adheres uniformly to the mylar. The result is a diaphragm that is we feel is the most sensitive in the industry.

Condenser diaphragms can be extremely sensitive to humidity and temperature changes. In order to minimize that, we temper our diaphragms by baking them for specific times at specific temperatures in order to insure maximum stability and performance.

The Backplate

In a condenser mic, the diaphragm is suspended over a *backplate* that carries one half of the electrical charge that results in the capacitance. The backplates of the best classic condenser microphones were made of solid brass. In an effort to cut costs, most modern manufacturers make the backplate out of injection-molded plastic and metalize them in some way. Critical listeners invariably prefer the sound of solid brass. Needless to say, we use solid brass backplates in all M-Audio mics.

The spacing between the diaphragm and backplate is critical. In order to avoid problems with barometric pressure, the spacer ring has a break in order to allow air to move freely between these two components. We precision drill approximately 100 extremely fine holes in the backplate, some going all the way through and some only going partially through. This combination further allows the appropriate amount of damping for the diaphragm. We then lap the surface in order to ensure that it is completely flat. This operation requires such precision that we measure the results not with a ruler, but with reflected light.



The major components of a large M-Audio condenser capsule are a solid brass backplate and an ultra-thin evaporated gold diaphragm

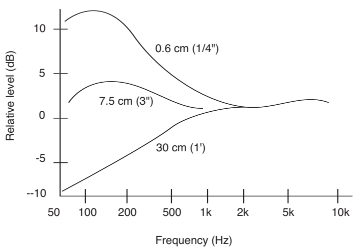
This level of precision is only possible due to modern computer-controlled manufacturing techniques. The important distinction is that these operations are programmed and supervised by human technicians at every step. All-in-all, there are several hundred precision operations that go into making each of our solid-brass capsules. That s more than the number involved the crafting of the average Martin guitar and we re talking about something the size of a 50-cent piece.

Patterns

The term *polar pattern* is used to describe the response of a microphone to sound sources from various directions. Each type of polar pattern has its own place and usage in the recording process. Note that the classic polar pattern definitions apply most accurately when sounds hit the microphone on axis that is to say, approaching perpendicular to the planar surface of the diaphragm. In general, microphones tend to become more directional in focus as frequencies increase. Put another way, capsules are generally less sensitive to high frequencies off axis. This phenomenon is typically less significant in medium capsules than in large capsules.

Cardioid pattern

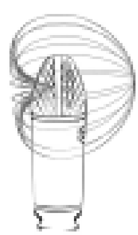
The *cardioid* is the most common polar pattern found in microphones. The name derives from this pattern s resemblance to a heart shape. Cardioids are unidirectional, meaning that they pick up sound primarily from the front of the capsule. The back of the capsule rejects sound, allowing the engineer to isolate the signal source from other performance elements or background noise. More noticeable in larger capsule designs, cardioid patterns typically exhibit a *proximity effect* a boost in low-mid frequencies as the proximity between the source and mic increases. Proximity effect is also more prominent with both larger capsules and lower frequencies.



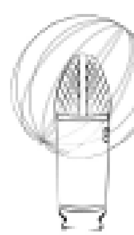
The proximity effect causes increased output in the low-mids as distances between the mic and source increase

Omni pattern

As the name implies, the *omni-directional,* or *omni* pattern, picks up sounds equally well from all directions. Omni is used to capture room resonance along with the source, thereby yielding a more open sound compared to the more focused quality of cardioid. Omni is great for vocal groups, Foley sound effects, and realistic acoustic instruments assuming that acoustic space of the recording environment is desirable.



Cardoid patterns are most sensitive on the side of the capsule



Omni patterns are sensitive to sound from all directions

Omni also exhibits significantly less proximity effect than cardioids. One result is that omnis are somewhat less sensitive to the movements of an animated vocalist. Another is that omnis tend to have less need for EQ. As mentioned earlier, while omnis pick up 360 degrees of sound, they tend to be more directional as frequencies increase especially in larger capsules.

Figure 8 or bidirectional pattern

The *figure 8* or *bidirectional* pattern is equally sensitive on the two opposing faces of the microphone, yet rejects sound from the sides. This pattern does exhibit the proximity effect found in cardioid patterns.

The figure 8 is excellent for capturing a duet or face-to-face interviews with a single mic. The —40dB side rejection spec also makes it great for isolating an instrument like a snare from the rest of the drum kit. Figure 8 is also one of the key components of M/S (mid-side) miking an advanced stereo recording technique we ll look at little later.

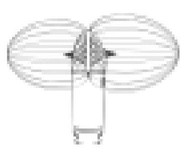
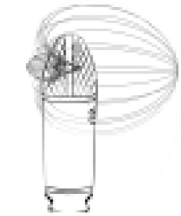


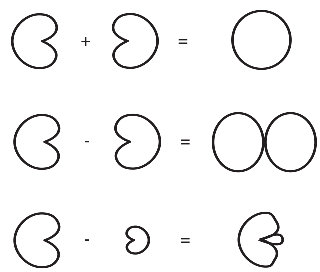
Figure 8 patterns are sensitive on opposing sides and exhibit strong rejection at 90 degrees off axis



The super-cardioid pattern has an even greater focus of sensitivity than cardioid

Super-cardioid pattern

The *super-cardioid* pattern exhibits an even narrower area of sensitivity than the classic cardioid and is used for very sonically focused recording. Super-cardioid is great for zeroing in on that perfect sweet spot for instruments such as piano or drum. This pattern is also ideal for live recording sessions where isolation is important, including minimizing bleed between a vocalist and their own instrument.

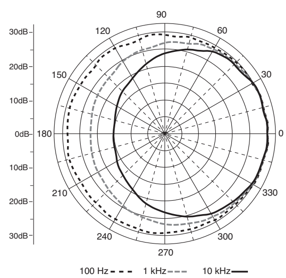


In multi-pattern microphones, two cardioids combine in different ways to create other patterns

Single pattern vs. multi-pattern mics

The most inexpensive way to make a microphone is with a single fixed pattern. Cardioids have openings in the backs of the capsules that produce the physics of a unidirectional pattern. This is an inherently fixed pattern design. An omni-directional pickup pattern can be achieved by sealing the back of the capsule, resulting in another fixed pattern. Super-cardioids employ yet a different design. In most cases, different back-end electronics are required for each pattern, thus making it difficult to make interchangeable capsules.

The secret to building a single mic with multiple pickup patterns is placing two cardioids back-to-back in combination with various electronic tricks. An omni-directional pattern occurs as the result of wiring two back-to-back cardioids in phase with each other. Similarly, those same two opposing cardioids wired out of phase yield a figure 8 or bi-directional pattern*. Tweaks to the polarity and output level result in a super-cardioid pattern. While the presence of two high-quality diaphragm/backplate assemblies increases the cost, this solution provides the best polar pattern performance and is still significantly less expensive than buying multiple microphones in order to have a choice of patterns at your disposal.



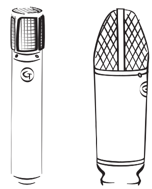
All microphones are less sensitive to high frequencies off axis (omni example shown)

This approach to capsule design can be seen in the M-Audio Solaris. The Solaris employs an opposing pair of the same diaphragm/backplate assemblies, thus allowing for the selection of multiple patterns via switches on the body of the mics.

**Tip: Note that the out-of-phase wiring of the two sides of a figure 8 capsule can play tricks on the uninitiated. One side will sound strange to a vocalist or speaker who is simultaneously monitoring the mic* *signal through headphones. That's because one side of the mic is in phase with the performer (and therefore reinforcing their perception of their own sound) while the other side is not. Addressing the in-phase side while monitoring produces optimal monitoring results.*

Top Address vs. Side Address Designs

The orientation of the diaphragm within the head of the microphone determines if the microphone is addressed from the top or the side. While not an absolute rule, medium diaphragms are typically top-address while large diaphragms are usually side-address. As you might surmise from the previous discussion about design considerations in attaining various polar patterns, top-address mics typically have single pattern (at least without physically changing the capsule) while side-address mics lend themselves to the possibility of back-to-back capsules for switchable patterns. Note that on side-address mics, the side with the logo is usually the primary or cardioid side.



Side address and top address microphone designs

Polar patterns aside, the practicality of side-address versus top-address designs also has to do with logistics. Top-address microphones can usually fit into tighter spots than can side-address mics (between drums, for example). This is yet another reason why pro engineers keep a variety of mics in their arsenal.

Microphone Electronics

As we ve seen, the microphone capsule is responsible for translating sound waves into electrical signals. The other important part of the microphone is the head amp that conditions the sound coming from the capsule so that it can be transmitted through a length of cable to an external preamp or console.

Part of a head amp s job is impedance conversion. (See A Word About Impedance for more information on impedance.) The average line-matching transformer found in dynamic or ribbon microphones has to convert on the order of several thousand ohms down to around 200 ohms (or from half an ohm up to about 200 ohms). The condenser microphone presents a challenge of a different magnitude converting a signal in the range of two billion ohms down to 200 ohms. This incredible leap is beyond the scope of most output transformers, requiring the addition of a specialized amplifier.

A Word About Impedance

Impedance essentially describes the resistance in a circuit. Water flowing through a pipe is a good analogy to electrons flowing through a wire. Let s say you ve got a pump designed to put 100 pounds of pressure into an eight-inch pipe. If you double the size of the pipe to 16 inches, you get half the pressure. While the pressure is now only 50 lbs, there is no damage to the system. Halving the size of the pipe, on the other hand, yields twice the pressure that the system was designed for. As a result, back-pressure affects the pump, further reducing its efficiency and increasing the potential of an explosion.

In terms of audio electronics, the pipe scenario is analagous to inputting the output from a 100-watt amp into 8-ohm speakers. While using 16-ohm speakers is safe (though it reduces output power), switching to 4-ohm speakers will almost certainly blow up the amp. That s why guitar amps designed to run into different speaker ratings often have output transformers with 4-, 8- and 16-ohm taps which appropriately condition the output signal.

Guitar pickups and most dynamic mics are considered to be high impedance, meaning that they exhibit an impedance of many thousands of ohms. Low-impedance signals are generally around 200 ohms or less. While the high-impedance signals typically exhibit greater voltage, they can only be run through about 20 feet of cable before they begin to lose high frequencies (or require additonal amplification in order not to). Low-impedance signals can typically be run much further without detriment.

An output transformer and/or amplifier serves as a sort of translator and, in audio, we expect that translation to be excellent in order to maintain frequency response, dynamic range, and signal-to-noise ratio. Just as a professional language translator costs more than someone who just took a few years of foreign language in high-school, pro-quality output transformers and amplifiers cost more than garden-variety ones. (A single transformer like those used in each channel of pro consoles and outboard preamps can cost more than a complete inexpensive multi-channel mixer.) Because the quality of this formidable translation is so critical in a professional-quality microphone, all M-Audio mics employ high-quality Class A electronics in the head amp.

Tubes vs. solid state

The head amp can employ either tube electronics or less expensive solid state electronics. Before we can effectively compare these two technologies, it is important to understand some fundamental concepts. There are three main ways to measure how accurately an electronic circuit passes sound frequency response, total harmonic distortion (THD), and dynamic distortion. Frequency response is the simplest to understand. We re simply talking about whether any highs or lows are rolled off, or if any frequencies are cut or boosted to exhibit a non-linear frequency response. Both tube and solid state electronics can be made without significant deficiencies in frequency response.

Regarding THD, all electronics induce some kind of harmonic distortion, i.e. harmonics that are not present in the original source. The nature of the harmonic distortion has more to do with the associated circuitry than with tubes versus solid state. *Class A* circuitry (where all amplifying components handle the entire signal waveform) tends to produce lower-order harmonics. On the other hand, *Class B* (where the positive and negative parts of the waveform are amplified by two separate devices) tend to produce higher-order harmonics. For this reason, Class A strikes most people as sounding warmer. (All M-Audio mics employ Class A circuitry.)

That brings us to the third, more mysterious element called dynamic distortion something that the industry didn t even have the technology to measure until quite recently. Dynamic distortion refers to the accuracy or transparency over time, particularly critical regarding the transient at the very beginning of a sound. Take the recording of a finger snap, for example. You can roll off the highs and lows and/or introduce a good amount of distortion, yet still perceive the sound as a snapping finger. Change the dynamic, however, and that snap can quickly lose its characteristic snap. In general, accuracy in reproducing dynamics can make the difference between something sounding full and three-dimensional or flat and two-dimensional.

Ironically, the discussion comes down to measuring things that don t matter and not measuring things that do. Tubes measure greater in THD than solid state. While one can measure the difference between .01 percent THD and .001 percent THD, it s practically impossible to hear that difference. On the other hand, while it s difficult to measure dynamic distortion you can definitely hear it. Solid state electronics exhibit many orders of magnitude more dynamic distortion than tubes. This is a major reason why tube mics make recordings sound truer to life.

Tube electronics

Tubes cost more money to manufacture than comparable solid state electronic components. In fact, the music industry is one of the few places where tubes have value in the face of more modern electronics.

It s a known fact that the average tube exhibits more inherent noise than solid state electronics. In general, the smaller the tube, the better. Larger tubes have a greater propensity for being microphonic, i.e. generating noise from mechanical movement of the internal parts. They also use higher voltages that result in more heat and subsequently more noise. Most manufacturers tube mics employ larger 12-volt tubes like the 12AX7 an older tube design that is noisier when used in microphone design.

TIP: One of the first things to be aware of is that not all products advertised as being tube mics employ tubes in the main signal path. Some popular low-cost mics utilize less expensive solid-state circuitry, putting a tube in the side-chain. (You can literally cut the tube out of the circuit on some models and the mic will *still work.) The theory there is that the tube is used as a sort of processor to "warm" up the sound. The reality is that these are still solid state mics masquerading as tube mics as cheaply as possible.*

Because of the physics behind tube operation, tube mics have classically been subject to certain physical restriction on the length of the cable between the microphone and power supply. As a result, tube mics are normally restricted to cable lengths of about 15 feet. This has sometimes required the use of solid state mics in scenarios such as drum overheads, remote recording or orchestral recording.

Solid state electronics

Solid state microphones cost significantly less to manufacture than tube mics. As such, they are found in the less expensive condenser mics on the market. (As stated earlier, some manufacturers put low-quality tubes in their solid state mics like an effects circuit in order to advertise products as being tube mics.)

In most solid state condensers, the key components are a series of op amps. All M-Audio mics employ FETs (field effect transistors) instead. Logic says that op amps should be preferable because they have lower measured amounts of THD. As discussed previously, while that difference in THD specs is measurable it is not audible in well-executed microphone applications. Op amps, however, can have much more dynamic distortion than FETs something you can hear. Moreover, many designs use multiple op amps to do the job of one FET. The difference is so profound that many people think that our solid state mics sound like most manufacturers tube mics.

The Myth of Tube Warmth

There is a common myth that tubes are warmer sounding. It certainly can be said that cranking up a tube amp will make an electric guitar sound warm, fat or distorted. That scenario, however, is one in which distortion is desirable. On the other hand, distortion is the enemy of the engineer who is attempting to record a sound source faithfully and realistically. Here, you want accuracy and transparency rather than any coloration that might be described subjectively with a word like warmth. Fortunately, there are many types of tubes and related circuitry that result in comparatively transparent sound.

It has also been said that tubes warm up digital recordings. This implies that there is something inherently deficient in digital recording. While some purists will always make a case for analog over digital, the fact is that a vast number of todays pro recordings are made with digital technology such as M-Audio s 24-bit/96k Delta cards, USB and FireWire solutions.

Digital recording significantly increased the dynamic range, allowing us to better hear the dynamics of recorded material. As a result, people were quick to label digital recording as cold, when using solid state mics. When using a tube mic, everything suddenly sounded warmer by comparison. In actuality, digital recording simply gave us the means of hearing differences we didn t hear before (such as how tube output is dynamically truer than solid state).

Manufacturing Standards

There are quite a number of condenser microphones to choose from on the market today. Many look professional on the outside and, indeed, most will give you acceptable sound. However, the fact is that most companies engineer for profit. This guide was designed to help you think about what s inside those shiny cases and much of that comes down to manufacturing standards.

The story behind affordable matched pairs for stereo-miking

One of the factors that make a significant difference between amateur and professional recordings is the use of stereo miking techniques. Pro engineers have long relied on matched pairs of microphones to attain optimal results from stereo recording methods. Why a matched pair? You wouldn t consider monitoring with a mismatched pair of speakers, right? Similarly, you want the left and right mics hearing exactly the same way in order to achieve a balanced sound.

From a technical perspective, the two mics need to be as identical as possible in frequency response. A flat frequency response implies that there is no deviation in the output level versus the input level at any and all frequencies across the audible spectrum. While a flat frequency response is theoretically ideal, it is rarely achieved completely in any audio component. For example, a mic might exhibit a 1dB boost at 1kHz and start rolling off 3dB per octave at 14kHz. A perfectly matched pair would exhibit the same exact characteristics in both mics. Here again, such an exacting match is rare. Therefore manufacturers each establish their own window of acceptable deviation that they will certify as a being a matched pair there is no industry standard. (Please note that we are actually talking about two different variables that are subject to interpretation and little disclosure the deviation between two matched microphones of the same model, as well as their deviation from the given manufacturer s standard reference mic for that model.)

Even the most famous of classic microphones have exhibited disparities in frequency response of 6dB of more from unit to unit. In such circumstances, manufacturers must search through a batch of mics to select a pair that is relatively close in response on the order of 2dB up or down for a total window of about 4dB. It is often necessary to place a special order (and pay surcharge as large as 20 percent of normal cost) for such matched pairs. This is not the case with M-Audio microphones. In order to pass inspection, all mics in our line must be within +/-1dB of not only each other, but of our golden reference mic for that model the one we won t sell for any price.

Higher standards

M-Audio is able to offer incredibly high quality and tight tolerances at affordable prices for several reasons. The first is that highly skilled technicians use the latest computer-controlled equipment for manufacturing and testing.

The reality of today s marketplace is that most companies manufacture their products offshore in order to be profitable. Many microphones on the market today are made in China or other countries where labor is less expensive even the ones that say that they are made elsewhere. At M-Audio, manufacturing is a hybrid operation. The designs all start in the USA, as do the manufacturing of all critical path elements like transformers, capacitors, resistors and diaphragm material. We then complete the machining and assembly in our own facility in Shanghai. In this way we attain the best of both worlds quality and affordable pricing.

While we re on the subject of standards, let s talk about the frequency response graphs that are often included with microphones. These graphs illustrate the deviation between input and output across the frequency spectrum. The ideal is to have as flat a line as possible indicating as little deviation as possible. Such graphs can be misleading because the industry has no universally accepted measurement standards that factor in distance from the mic, volume, angle relative to axes, and so forth. Moreover, there is no standard for rendering these graphs. Major deviations apparent on a graph calibrated vertically at +/-10dB look much more like a flat line if displayed on a graph calibrated at +/-100dB. So in a world where everybody draws nice looking graphs because they feel they must in order to be competitive, we simply decline to play the game until such time that standards exist that level the playing field. As stated earlier, all M-Audio mics are manufactured to within +/-1dB of each other and our golden reference standard. We re confident that your ears will tell you everything else you need to know.

Chapter 2

Caring for Microphones

High-quality condenser mics like the M-Audio line represent an investment. A few basic tips will help ensure a lifetime of excellent performance.

Shock Protection

As you now know, condenser mics are constructed with extremely thin diaphragms and very high tolerances. As such, condensers should be protected from abuse, especially physical shock. (M-Audio capsules are rubber-mounted internally, but the need for caution still applies.) Keep condenser mics away from situations in which they might be physically abused. Unlike a dynamic microphone, condensers should always be mounted on a stand rather than hand-held (or swung around on the end of a mic cord by a vocalist exhibiting showmanship). Similarly, wayward drum sticks, guitar necks, violin bows and the like are not friends of condensers. As indicated earlier, condensers should only be used live in controlled situations where the stage is protected from the elements and is the exclusive domain of professionals. Take great care to avoid dropping a consender mic or knocking over a mic stand holding one we recommend duct-taping cables to the floor in order to avoid tripping over them.

A soft mount (also know as a shock mount) one that suspends the mic in an elastic we b i s usually desirable because the mount absorbs vibrations from the floor, passing trucks or airplanes, and any modest inadvertent physical shock. While hard mounts provide no such shock absorption, they are sometimes useful in tight situations or when exact placement is required (such as in an X-Y stereo miking configuration).

Pop Filters and Windscreens

When pronouncing p , t and b sounds, vocalists often project extra energy toward the microphone. A common result of this extra energy is unwanted pops in the sound, as well as the expulsion of saliva a form of moisture detrimental to a condenser mic. For these combined reasons, a pop filter is highly recommended when recording vocals with condenser microphones. Typically a thin mesh stretched over a circular frame, the pop filter is mounted between the vocalist and the mic capsule. (In a pinch, you can even construct a pop filter with a hanger and pantyhose.)

Windscreens, as the name implies, are sometimes used in outdoor recordings in order to reduce wind noise and particulate matter striking the diaphragm. Windscreens typically consist of a thickness of foam custom designed to fit over the capsule. Windscreens can reduce both low and high frequency response, so they are typically not used as substitutes for pop filters.

Temperature and Humidity

The thin diaphragms and tight tolerances of condenser microphones make them susceptible to temperature and humidity extremes. Never use condenser microphones when there is risk of water damage (such as rain). Avoid high humidity situations such as seaside climates lacking air conditioning.* The operating temperature of most condensers is 50…F to 95…F. If a condenser has been outside in a colder environment (such as transporting it in winter), allow the mic to slowly acclimate to room temperature before applying power in order to avoid condensation on the capsule. Similarly, be careful not to leave condenser mics to bake in the trunk of your car on a hot, sunny day. These same precautions apply to tube power supplies as well.

**Tip: Even the best condenser will start producing a crackling noise if inadvertently exposed to too much humidity. In this event, an old trick is to place it near the heat of a light bulb for about half an hour.*

Cleaning and Storage

Always store a condenser microphone in its case when not in use. Particulate matter such as dust can attach itself to the diaphragm and cause degradation of performance over time. In most cases, wiping the metal exterior of a microphone down with a dry or slightly damp rag will be sufficient to remove dust, dirt, fingerprints and the like. In the event that further cleaning is necessary, spray a non-abrasive household cleaner such as Fantastik or Formula 409 onto a rag and wipe the metal exterior with the rag. NEVER spray directly onto the microphone as it may damage the capsule. NEVER attempt to clean the inside of a microphone. If performance degrades, contact M-Audio for factory repair.

Chapter 3

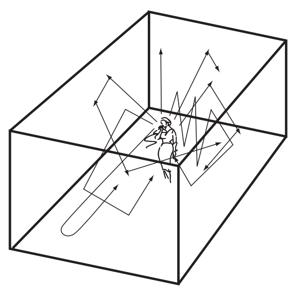
Basic Miking Concepts

Microphone placement is an area in which art meets science. Microphone choice and placement is somewhat subjective, much in the same way that choosing a guitar and amp is a matter of personal preference. Furthermore, each situation brings a difference confluence of performer, sound space, recording equipment and creative forces. The question is not one of using the right or wrong mic or technique, but simply one of what works best in each unique situation. Nonetheless, it s good to know the rules in order to break them with the greatest success. Here, then, are some generalizations to consider. Note that since condenser mics are used in the vast majority of studio situations, all of the following application tips apply to condenser mics.

All recording spaces have a unique ambient quality that determines how sound from the source will be reflected. Those reflections are candidates for being picked up in the microphone(s) along with the direct sound from the source. The choice of microphone, pattern and placement depends in part on the balance you wish to strike between the sound source and the ambient characteristic of the recording space. Another critical consideration is isolation from other sound sources. In many ways, it all comes down to envisioning the sonic focus you want the mic to have.

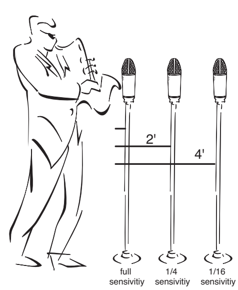
Close-Miking vs. Distance-Miking Techniques

In general, *close-miking* techniques (where the microphone is very close to the sound source) are used in conjunction with a cardioid or super-cardioid to focus the pickup pattern on the source while simultaneously avoiding any significant influence from the surrounding space. Close-miking with cardioids (or super-cardioids for extreme situations) is also very useful in isolating the sound source from other performers. Note also that the closer the mic is to the source, the more prevalent with be the performance by-products such as breath, fret noise, snare rattles and piano hammers. With close-miking, the illusion of space is likely to be added electronically in post-production via reverb and/or other forms of time-delay devices.



Microphones potentially receive reflections from the room and other objects as well as sounds emanating directly from the source

Placing any mic at a greater distance from the source will add more of the room reflections. *Distance-miking* refers to microphone placement intended to incorporate at least some room reflections. An omni pattern really opens up the recording to incorporate the full ambience of a room. Regardless of the pattern, a proper balance must be found in order to maintain the presence of the source while incorporating surrounding ambience. When enough mics are available, engineers often employ both close- and distance-miking techniques simultaneously in order to control the balance of direct and room sound.



Sound power falls off exponentially with distance according to the law of inverse squares

The farther the microphone is placed from the source, the less sensitive it is to the sound emanating from that source. This falloff is not linear. Microphone sensitivity exhibits the *law of inverse squares* i.e. sound power reaching the microphone varies inversely as the square of the distance from the source. For example, the typical mic is exposed to only one-quarter the sound power at twice the distance from the source. (You can think of this as the aural equivalent of the exponential falloff in light as you get further away from a light bulb.)

Recall also that large-diaphragm cardioid microphones exhibit a proximity effect where the low-mid frequencies increase as the distance between the source and mic decreases. (The proximity effect is not a big issue with omni patterns or medium-sized capsules of any pattern.) With large diaphragms, then, the placement of the mic affects volume, room ambiance and tonality.

Large Capsules vs. Medium Capsules

You can achieve excellent results in most situations using our large-capsule mics.

As a rule of thumb, the large-capsule mics like ours will have more sensitivity in the low end than the medium capsules simply because the diaphragms are larger. As previously mentioned, they also exhibit more proximity effect in cardioid patterns (which can be a plus or a minus depending on the circumstances). Further, they take up more physical space so they are less adaptable in tight situations. Conversely, medium capsules tend to exhibit flatter frequency response regardless of distance and are more flexible when space is a consideration. You ll eventually want to have both large- and medium-capsule models in your mic locker.

Dealing with Unwanted Low-Frequencies

Extraneous low-frequency content such as that induced by passing trucks or standing waves in the room can present a problem during recording. Low frequencies are harder to compensate for with acoustic treatment than are higher frequencies. Most condenser mics have a switch that introduces a high-pass filter rolling off low frequencies starting at around 75Hz. This feature should be used with care, since sound sources such as the male voice have content in this range. On the other hand, low frequency roll-off can sometimes be used intentionally, like in a situation where you want to reduce the boomy quality of an acoustic guitar. It is best to induce as little electronic circuitry as possible. Use critical listening to determine if low-frequency roll-off is truly beneficial.

The Mic Preamp

Before the low-level signal from a mic can be used in the recording and mixing process, it must be run through a preamp in order to boost the gain. Therefore, most pro recording engineers will tell you that next most critical piece of gear after the microphone is the mic preamp.
Even the best microphone inputs on an affordable mixing board, sound card or all-in-one recorder don t hold a candle to a dedicated mic preamp. Pro studios routinely pay thousands of dollars per channel for dedicated outboard preamps. While that s not realistic for most project studios and home recordists, it is indicative of the fact that good quality mic preamps are an important thing to consider in your studio budget. If you re looking for a good preamp at budget prices, check out M-Audio s DMP3. And if you re interested in a high-end preamp that won t break the bank, see the inset which follows, containing information about our revolutionary TAMPA preamp featuring Temporal Harmonic Alignment.

Use of Processing During Recording

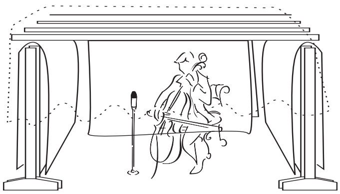
Engineers have varying opinions about the amount of processing to use during the actual recording process. Part of it comes down to how much processing gear you have at your disposal for the subsequent mix session. The predominant wisdom is to process as little as possible at any stage period. That s a major reason why having a good mic is essential to high-quality recordings. The more EQ and compression you have to apply, the further you get away from a natural sound.

Nonetheless, many engineers like to record with basic EQ and compression for two reasons to overcome major deficiencies in the sound, and to achieve the hottest possible levels before clipping for the highest signal-to-noise ratios. In other words, processing during recording is typically used to ensure that the signal is technically optimal. On the other hand, processing more specialized effects such as reverb, chorus and delay are usually reserved for the mixing process, to yield maximum flexibility in these more subjective and creative areas.

The Recording Environment

Professional studios often have several different acoustic spaces available from small, relatively dead isolation booth to cavernous rooms with natural reflections and long delay times. Home recordists have fewer options, yet experimenting with recording in different rooms may yield interesting results. Large rooms and tall ceilings will give a more open sound than small rooms and low ceilings. The amount of furniture and reflectivity of various surfaces is also an influence. A carpeted floor, for example, has a damping effect as opposed to the reflectivity of a wood or tile floor.

There are many times when it is beneficial to create methods of isolating the microphones or otherwise controlling the room acoustics. Such scenarios include having a poor sounding room, having an open mic in the same room as recording gear exhibiting fan noise, or recording multiple performers simultaneously. In cases like these, consider solutions such as applying acoustic treatment to the room, creating a temporary isolation booth by hanging or tenting blankets, or building movable partitions. Moving blankets, egg-crate foam and carpet are



It is often beneficial to devise methods of controlling room acoustics such as constructing a tent using blankets

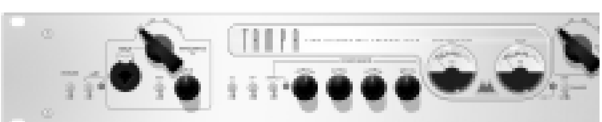
The Revolutionary New TAMPA Preamp

Reflecting back on our discussion about tube versus solid state electronics, most highly revered mic preamps are based on tube technology. Unfortunately tubes are part of what typically drives the price of preamps into thousands of dollars. That s why our design team set out to find out just why tubes sound so good, and devise a way to land that sound at solid state prices. The result is far beyond tube modeling; it s a whole new technology we call Temporal Harmonic Alignment.

Natural sound sources such as strings, drum heads and vocal chords share a characteristic temporal or phase relationship of harmonics to the fundamental when vibrating. Not coincidentally, our ears exhibit the same qualities. Electronic circuitry induces distortion in the form of additional harmonics that do not exhibit that relationship. Tubes strike the ear as having such a warm sound because the added harmonics have the same temporal relationship as natural mechanisms although predominantly in the midrange. This results in a sweet spot that makes things like vocals and guitars sound especially pleasing. We designed TAMPA technology to produce that same phase relationship found in both tubes and nature. And unlike tubes, TAMPA s sweet spot spans the full spectrum of your sound.

TAMPA also includes a dual optical servo compressor that alone is worth the price of admission. Three fundamental problems plague engineers in designing compressors distortion, noise and accuracy. The VCA technology used in inexpensive compressors exhibits less than professional specs on all counts. Simple optical servo technology is much more quiet and accurate, yet has its own issues with distortion. The dual optical servo technology we use in TAMPA yields low noise, consistent accuracy and low distortion and it comes built into a killer preamp.

TAMPA s entire signal path is designed to yield maximum fidelity without compromise, including discrete Class A circuitry throughout. You also get tons of other professional features like an impedance selector for optimizing vintage mics, and a massive 30dB of headroom. Audition a TAMPA for yourself and you ll see what all the fuss is about.



common acoustic damping materials for home studio use. Music stands can also be reflective something you can compensate for by simply draping towels over them.

While a reasonable amount of absorption is often desirable for isolation, too much damping can create an anechoic space that literally sucks the life out of a recording. In more permanent project studios, consider creating a flexible acoustic environment. One solution is a series of gobos or movable panels with a reflective surface on one side and an absorptive surface on the other. These can then be moved and placed as desired for a given project. Another solution is to create reflective walls with movable absorptive drapes in front of them.

Finally, don t overlook the acoustic resources you have available. Many a vocal track has been recorded by running a mic into a tile or marble bathroom. (People like singing in the shower for good reason the sonic reflectivity can make even mediocre voices sound great!) Recording engineers have frequently placed speakers and mics in concrete stair wells to transform the concrete acoustics into reverb chambers. The drum track for Led Zeppelin s classic When the Levy Breaks was so incredibly ambient because John Bonham s drums were set up in the stairwell of a stone castle. Similarly, some classic Jimmy Page tracks were realized by placing the guitar amp in a fireplace and miking the top of the chimney. Again, the only real rule is to use what works for the track.

Phasing Issues with Multiple Microphones

The use of two microphones can introduce problems owing to phase discrepancies between the mics and that potential increases with the number of microphones in concurrent use. In essence, phasing problems occur when a sound reaches different mics at different times. Telltale signs are different notes from the same source sounding at different volumes, or bass response that is overly strong or overly shallow.

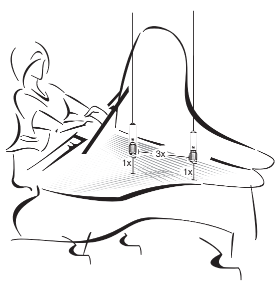
Here are a few tips in minimizing phasing problems when using multiple mics:

Move the mics. The first line of defense is to just get into the studio with headphones on and move one or more of the problem mics until the phasing issue is resolved.

Check the cables. If a cable is accidentally wired out of phase, it can cancel out the signal from a neighboring mic. Make certain that the mic cables are wired with continuity (i.e. pin 1 on one end goes to pin 1 on the other end, and so forth).

Apply the 3:1 Rule. If possible, microphones should be three times further away from each other than from the source. As an example, microphones placed 5 inches away from a sound source should be at least 15 inches apart from each other. (This does not apply to the coincident stereo miking techniques we ll discuss shortly.)

Minimize the number of microphones in concurrent use. The more open mics you have, the greater the potential for phasing issues. While it might be tempting to put a separate mic on each component of a drum kit, for example, the tradeoff is the amount of time it might take to eliminate phasing complexities. Less can be more in situations where you have difficulty getting phasing under control.



Placing two microphones three times the distance from each other as they are to the sound source can eliminate phasing problems

Separate the sound sources. With the exception of stereo recording, the general idea behind using multiple microphones is to isolate the sound sources. Phasing issues provide another reason to isolate the sources. Solutions include simply spreading the mics apart, putting them in separate rooms or isolation areas, or using baffles, gobos and the like to provide additional separation. In the case of two mics on the same instrument, it is sometimes beneficial to devise a baffle that goes between the mics.

Minimize reflective surfaces. Hard surfaces like wood floors, smooth walls, windows and mirrors are a common culprit in phase issues because they reflect sound back into the microphone. If things sound odd, try moving the performer and/or mic. Also experiment with damping those reflections with blankets, towels, baffles and the like.

Avoid boxing in mics. Microphones typically need a little breathing room in order to avoid reflection. Omnis placed in a corner, for example, often sound like they re, well, in a corner! Similarly, placing the back of a cardioid too close to a surface or corner can sonically block the rear ports, thereby distorting the effective polar pattern of the mic. Also, exercise care when using baffles and gobos because these mechanisms do not completely absorb sound and can actually cause reflections when placed too close to the mic.

Chapter 4

Stereo Miking Techniques

The use of stereo miking techniques utilizing matched pairs can make all the difference between mediocre and outstanding recordings. After all, we listen to the world around us in stereo via matched pairs of ears. Stereo miking can be used in applications ranging from individual instruments to small ensembles to full orchestras and other concert events. In this section, we ll cover some of the proven stereo miking techniques that have been used on countless professional recordings. (For the purposes of this guide, stereo miking techniques are a subset of multi-microphone techniques specifically aimed at accurately capturing a sound source with a left-right balance similar to the way our ears perceive a sound source.)

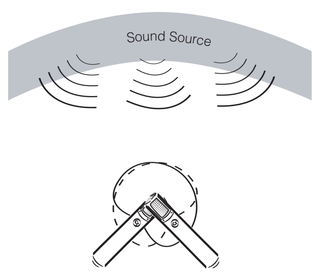
Several factors must be considered in determining the best stereo miking technique for your specific application. Although results vary with different polar patterns, it is common to use distance from the source to determine the amount of room reflection versus direct source signal desired. Physical restrictions in distance or position may also come into play, such as the need to maintain clear lines of sight from audience to stage. It is also advisable to consider mono compatibility, especially if the resulting material will wind up on radio or television.

The following stereo miking techniques fall into two basic categories coincident and spaced. *Coincident* techniques rely on the microphones being placed in extremely close proximity to one another, while *spaced* techniques place them further apart. While the coincident methods are considered to be very accurate, some listeners find them to be too accurate. Common criticisms are that the stereo field is too narrow or confined to the speakers on playback. (You can sometimes compensate for this by moving the coincident mics slightly apart from each other in order to introduce a time delay between sides.)

Spaced techniques are considered less accurate, yet more spacious sounding. In effect, widening the space between mics widens the virtual placement of our ears. As with everything surrounding microphones and their techniques, these considerations are subject to interpretation and experimentation. In fact, it is not uncommon to find engineers employing techniques from both categories simultaneously. In such a case, the coincident pairs provide a well-defined primary signal, while the space pairs are placed to capture the reflected sound that provides extra control over ambience.

X-Y

The *X-Y* miking technique employs a matched pair of microphones overlapping as much as the mic bodies allow. As pictured, place a pair of cardioid mics as close to each other as possible with the capsules at an angle to each other. The mic on the left captures the right signal and vice versa. While 90 degrees is the most common angle between the capsules, the working range is approximately 60 to 135 degrees. The wider the angle, the wider the perceived stereo field will be. In general, the distance from the sound source combined with the intended stereo spread (the width of a stage, for example) will determine the appropriate angle.

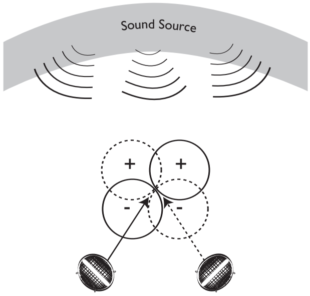


X-Y miking employs a matched pair of coincident cardioids

The use of cardioid patterns means that the X-Y configuration as a whole rejects signals from the rear. (You can also experiment with super-cardioid patterns to provide more isolation between left and right sonic imagery.) This rear rejection has several advantages. The configuration can be moved further away from a stage to preserve sight lines. The reduced sonic clutter is also of benefit when converting the stereo recording to monaural. Increasing the distance between the coincident mic pair and the sound source decreases stereo separation and captures more room reflections. In general, the X-Y technique using cardioids yields an accurate stereo image exhibiting minimal acoustic reflections, although the separation is not as significant as some other stereo miking techniques.

Blumlein

Named after British stereo pioneer Alan Blumlein, the *Blumlein* technique takes advantage of the polar patterns inherent in figure 8 (bidirectional) mics. Recall that figure 8 patterns pick up equally well on two sides while exhibiting strong rejection at 90 degrees off axis to those sides. In the Blumlein technique, two figure 8 patterns are oriented 90 degrees from each other with the positive sides facing the left and right sides of the sound source. Due to the inherent side rejection, the area of greatest sensitivity of one mic is the area of least sensitivity of the companion mic. While the patterns overlap in the center, the signal from each is 3dB down and, when combined, pick up a uniform center signal.

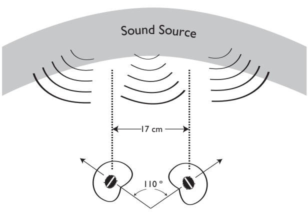


The Blumlein arrangement relies on a matched pair of coincident figure-8 patterns

The Blumlein arrangement yields very good stereo separation. Due to the fact that figure 8 s are equally sensitive on the back lobes, this configuration also picks up significant room reflections. There are drawbacks to this technique, however. The fact that the back of the left mic is also picking up reflections from the right rear of the room makes for poor mono compatibility. Further, reverberant sounds coming from the sides of the acoustic space can enter the positive lobe of one mic and the negative lobe of the other, thus causing the impression of poor localization and/or hollow effects that can be disturbing. As a result, Blumlein is best used in situations where the sound source, acoustic space and mic placement are optimal. Since this is a rarity, other stereo techniques offering superior control are more frequently used.

ORTF

Developed by the French national broadcasting agency, Office de Radio T l vision Fran aise, the *ORTF* technique is intended to emulate the placement of ears in the average adult human head. Two cardioid capsules are placed 17cm (about 6 - 3/4 inches) apart at a 110 degree angle to one another. ORTF can produce the wide imagery and depth common to the Blumlein technique, however the use of cardioids means that the configuration captures much less reverberant reflection.



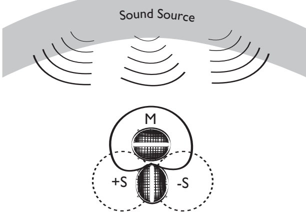
The ORTF technique positions a matched pair of mics in a configuration similar to that of human ears

The specified distance for ORTF makes wavelengths below about 500 Hz effectively phase coherent. The time delays or phase incoherence above that frequency typically contribute to a sense of stereo separation, along with the perception of a pleasing open or airy quality. ORTF also exhibits adequate monophonic compatibility. Similar experiments by the Dutch broadcasting counterpart Nederlandsche Omroep Stichting yielded the NOS technique where a pair of cardioids are placed 30cm apart at a 90 degree angle.

Mid-Side

The *Mid-Side* technique utilizes special processing to capture very precise stereo imagery with excellent mono applicability. A mid microphone (typically a cardioid) faces the center of the sound source and captures the primary sound. A figure 8 (the side) is placed along the same vertical axis with its lobes facing right and left, thereby picking up the extreme left and right information due to the side rejection inherent in the figure 8 pattern.

This configuration does not constitute stereo until the signals are processed through an M-S encoder matrix such as the M-Audio Octane Preamp. The encoder adds the mid and side signals together to create one side of the stereo signal, and subtracts the side signal from the mid signal to create the other. The result is a very accurate translation of the stereo listening field. The presence of an M-S balance control in the encoder also allows the engineer to control the ratio of mid signal to side signal, and therefore the perceived width of the stereo field.

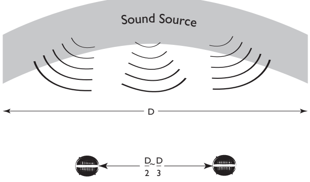


The Mid-Side technique electronically derives a stereo signal from a center mic coincident with a figure 8

Note that Mid-Side is the only stereo miking technique that does not rely explicitly on a matched pair of microphones. However, high quality microphones are imperative for overall sonic integrity, as well as to ensure well-balanced capsules within the figure 8 mic. Note also that Mid-Side offers a great deal of flexibility because the mid does not have to be a cardioid. If more audience noise or reflections from the back of the room are desired, an omni could be used as the mid mic to great effect. The Mid-Side technique also offers excellent mono compatibility because the recombination of the two out-of-phase side signals cancels them out to leave only the mid or center signal. This process simultaneously minimizes side reflections that can yield confusion in a mono conversion.

Spaced Omni

The *spaced omni* technique is often used for recording orchestras. It employs a matched pair of omni mics typically positioned four to eight feet in front of the sound source. The mics are normally at the same height as the performers, although raising them to 10 feet or more in the air can increase perceived ambience. The distance between the mics should be approximately 1/3 to 1/2 the width of the sound stage. While spaced omni provides excellent depth and stereo image, the center of the field can tend to be less distinct. In situations where there is too much unwanted background noise or the mics must be placed further away due to logistics, experiment with using carefully placed cardioids or supercardioids with this spaced mic technique.

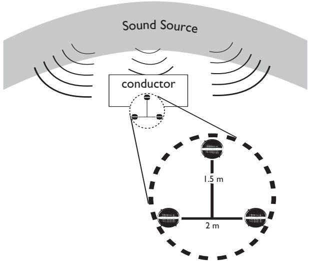


The spaced omni technique places omnis at a distance of 1/3 to 1/2 of the sound stage width from each other

As pointed out earlier, spaced miking techniques are not as technically accurate as some correlated miking techniques. They can be susceptible to phase anomalies owing to reflections entering the mics from various surfaces in the recording environment although some people actually find this pleasing. Many engineers consider spaced mic techniques best for recording relatively uncorrelated sounds such as a pipe organ or outdoor ambience. Spaced techniques are also useful in creating the surround channels for surround sound.

Decca Tree

Staff engineers at Decca Records (now Thorn-EMI) developed a technique known as the *Decca Tree* in England in the 1950s. This method and numerous variations are still very popular today in the recording of film scores.



The Decca Tree technique is very popular in film scoring

A T-shaped fixture houses a microphone classically an omni at each of its three ends. The two mics at either end of the cross-arm are positioned approximately two meters (approx. 79 inches) apart, while the central microphone is 1.5 meters (59 inches) away at the bottom of the T. This structure is then mounted about eight to ten feet in the air and positioned so that the central mic is just behind the conductor s head. The mics are tilted down at about 30 degrees and fanned out to cover the physical spread of the orchestra. Another pair of mics is often placed further back in the hall on either side of the orchestra in order to capture room reflections in the ambient space.

Decca Tree is favored in the film industry because it provides a spacious sound along with good stereo imagery that works well with processes like Dolby and surround sound. There is also the advantage of a discrete center mic for both monaural and center channel use. Variations abound, including the substitution of other polar patterns, spreading or narrowing the distance between the mics, and aiming the left and right mics at specific orchestra sections to be featured.

CHAPTER 5

Specific Miking Applications

Now that we ve covered some basics, it s time to look at some time-honored guidelines for common recording situations. While experimentation is definitely encouraged, these techniques will get you in the ballpark and, more importantly, provide additional understanding about microphones and placement techniques so that you can find what works in any given situation.

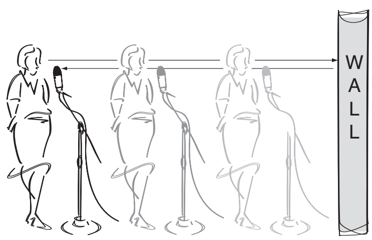
Vocals

Vocals are perhaps the most difficult subject to mic. Each vocalist is different and there can be a tremendous amount of dynamic range within a single performance. Vocalists also tend to move when they sing, providing yet another challenge.

Angling the mic downward can reduce unwanted vocal energy from reaching the diaphragm

A large diaphragm capsule is traditionally desired on vocal tracks. Large diaphragms are generally better equipped to accommodate a vocalist s potentially high dynamic levels. The proximity effect tends to add fullness to the voice, as well. That same proximity effect can be overwhelming when used on a performer that already has an extremely deep voice. In this event, a medium capsule can be more appropriate because of the reduced proximity effect.

Tube mics and preamps are highly recommended for vocals, as vocals tend to be the featured element.



In order to avoid primary sonic reflection, vocalists and mics should not be positioned directly in front of a hard surface

Cardioids are typically used when close-miking a vocalist, especially when the acoustic space is not necessarily something you want featured in the track. On the other hand, omni can yield excellent results when you do want to feature the room s natural ambience. In the case of recording multiple vocalists, there may not be enough resources for separate mics or tracks. For a duet, placing the performers on either side of a figure-8 capsule works well. For background vocalists or an entire singing group, place the singers in a semicircle around a cardioid. Position the individual vocalists closer to or further from the mic in order to achieve the desired balance in their levels.

In all cases, the distance between the vocalist and the microphone will determine how present or intimate the sound is, as well how much reflected sound is picked up. Note also that the law of inverse squares dictates that slight movements on the part of the singer will have much less effect on the mic output level if he or she is not eating the mic. A good starting distance is 12 to 18 inches away from the vocalist.

A pop filter is often used to reduce vocal plosives

Vocal mics are usually placed at the same level as the performer s mouth. Raising the mic produces a more nasal sound, while lowering it yields a more chesty sound. Avoid extremes, as they tend to stretch or constrict the subject s throat enemies of a good vocal performance. You can also experiment with angling the mic down at the performer s mouth in order to avoid projecting the energy of the breath directly into the microphone. Also, make certain that the vocalist s headphones are the closed-cup variety, as open-cup designs tend to leak sound that the microphone will capture.

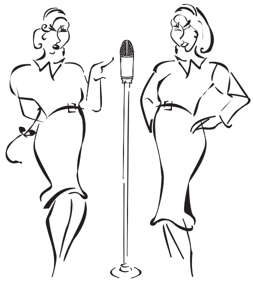
As previously mentioned, a pop filter is highly recommended in order to soften plosives and to protect the condenser diaphragm from saliva. Pop filters can also help in controlling a singer s proximity to the mic. (If a vocalist has a tendency to get too close to the mic, place the pop filter as far away from the mic as you want the vocalist to be, then have them sing directly into the filter.)

Pop filters do not help in controlling sibilance, commonly found in the hissing portion of the letter s. You can try moving the mic lower and/or farther away to reduce sibilance, as well as encourage the performer to turn their head slightly during those passages to avoid projecting the sibilance directly into the mic. If these approaches are not effective, you may need to process the vocal with a de-esser a special form of compressor that targets sibilant frequencies.

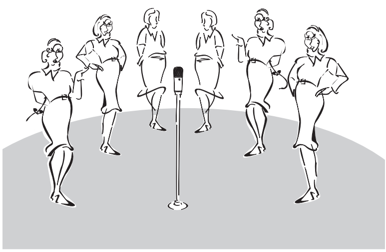
Most engineers do use some compression when tracking vocals. A little compression goes a long way, and too much can squash a vocal. A few dB of compression is typically all that is necessary in order to prevent peaking while getting the maximum signal to the recorder. EQ adjustments should only be aimed at getting an optimal signal, rather than trying to perfect final tonality (something that should be reserved for the mix).

Be aware that a hard surface directly in front of the vocalist can create a strong sonic reflection that colors the vocal sound. If this is not desired, move the vocalist further back from that surface in order for the reflection to fall off more before reaching the mic.

Our ears are very highly tuned to the human voice, so even casual listeners are much more critical of recorded vocals than of any other instrument. Once you think you have a good miked sound for vocals, it s always a good idea to compare it directly with the sound of the vocalist just singing in the room in order to get a reality check.



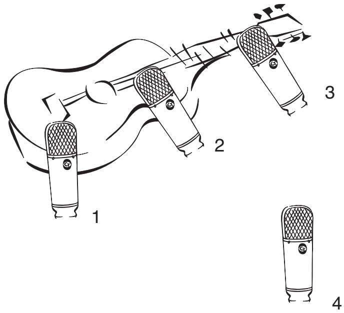
A figure-8 pattern can accommodate two vocalists facing each other in a duet



Positioning for multiple singers around a cardioid

Acoustic Guitar

There are a variety of popular mic placements for acoustic guitar. The one that might seem the most logical miking the sound hole is actually the least desirable in most situations. The result is very boomy, especially when combined with the proximity effect of a large diaphragm. With the exception of the audience perspective technique, place the mic(s) nine to 12 inches away from the focal area and experiment from there. In close-miking scenarios, encourage the guitarist to remain stationary in order to minimize fluctuations in the sound. Hard mounts are preferable to ensure exact placement, unless a soft



Different microphone positions bring out different qualities of an acoustic guitar mount is needed to minimize vibrations from tapping feet and the like.

In most cases, cardioids are the best choice for miking acoustic guitar. While you can certainly attain good results with a large diaphragm mic, medium diaphragms are often preferred in order reduce proximity effect.

1. Bridge. Miking the bridge yields a very sweet sound. Angle the mic either away from the sound hole or tilted up from underneath. The difficulty with this position is that the performer s hand can obstruct the pickup pattern, yielding a fluctuation in sound. Working with the performer to find a way to capture the bridge sound unobstructed can be well worth the effort.

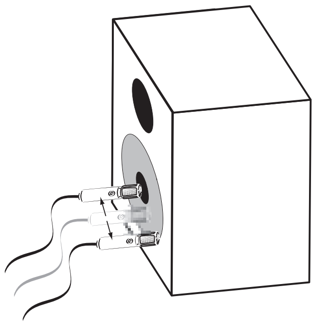
2. Neck joint. The most popular position is where the neck joins with the body. This placement yields a nice balance of warm body and bright neck, including natural fret and finger noises. Deploying a matched pair with one mic at the joint and the other at the bridge can result in one of the most satisfying acoustic guitar sounds. Panning the results hard left and right can yield a dramatic stereo sound.

3. Nut. Miking the nut usually results in a bright jangly sound. Additionally miking the bridge and panning the two mics hard left and right produces an extremely wide stereo effect.

4. Audience perspective. Close-miking techniques are often a bit too in your face for solo classical guitar. In that event, try placing the mic three to four feet in front of the performer and level with the sound hole in order to capture more room sound. To create more of a concert ambience, deploy a matched pair left and right at even greater distances. You can also attain the best of both worlds by using multiple mics to combine the aforementioned close-miking techniques with this distance-miking technique.

Electric Guitar

The most common way to mic a guitar amp is to place a cardioid four to six inches in front of the speaker. If it sounds as if the microphone or preamp are overloading, try engaging the pad switch on the condenser typically providing a 10 to 15dB reduction in gain. Proximity effect applies here, so closer positions yield more low mids. Experiment with moving the mic from the center to the outer edge of the speaker. The center typically has a more edgy quality, while the outer portions are mellower. One time-honored trick is to don headphones and move the mic until the hiss sounds minimal and you ll probably like the sound at that spot.



Close-miking the center and edge of the guitar amp speaker yields different tonalities

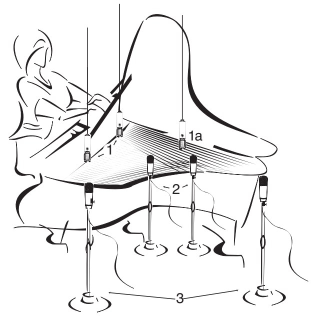
As with most miking situations, moving the mic further away brings in more of the room sound. Consider miking the cabinet in stereo at various distances. One popular technique is to use an inexpensive dynamic mic tight on the speaker and an M-Audio condenser mic further back, panning the two in stereo.

Grand Piano

The key to getting a great recording of a grand piano is to start with a great piano in a great recording environment. That said, every instrument and situation is different and the same piano can also sound different at the hands of different performers. Each situation has to be evaluated differently.

The extremely wide frequency range of the piano combined with the broad throw makes stereo miking with one or more matched pairs the de facto standard. Large diaphragms such as the M-Audio Solaris and Luna are excellent choices. The exact placement has a great deal to do with the type of tonality you are desiring. In most cases, the lid should be all the way open. When isolation is required, many engineers will adjust the lid to the halfway height, then enclose the opening and mics with a moving blanket.

1. Pop/rock. The closer the mics are to the hammers, the more percussive the resulting sound will be. For this reason, pop/rock recordings where the piano needs to cut through other tracks are often achieved by close-miking the hammers. Start by placing a matched pair of cardioid mics face down about six inches above the hammers and about one-third from each respective end. Be careful not to place them so close to the hammers vertically that the pickup area is restricted to only part of the required range of notes. Experiment with the distance between mics until you get the desired balance of wide stereo imagery and even response across the entire range. You may also need to angle the mics slightly away from each other in order to improve separation. Experiment with distance from the hammers to achieve the desired balance of percussive attack and full body. You can also try moving the bass mic further down the harp away from the hammers to achieve a bigger, warmer sound (see position 1a in the diagram).



The placement of stereo mics on a piano depends in part upon musical style

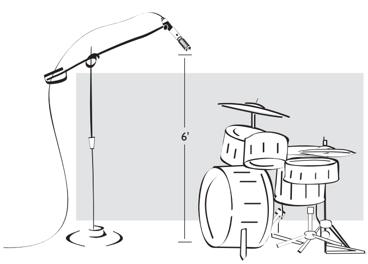
2. Pop ballad/jazz. The jazz and pop ballad genres typically require the fuller, sound attained by positioning the mics further away from the hammers. Place a matched pair of cardioids at a 45 degree angle in the deepest part of the curved side of the instrument, with one mic facing the performer and the other facing across the harp. Start with a separation of three to six inches, decreasing the angle if you move the mics further apart. It s also perfectly valid to try an X-Y or other coincident configuration at this position. The vertical position should be about halfway between the strings and the open lid.

3. Classical or solo piano. Classical and solo piano often benefits from distance miking that includes more of the ambience of a concert setting. Experiment with placing the mics four to eight feet from the curved side of the piano and at heights from five to twelve feet. Factors leading to the establishment of optimal positioning include the size and model of the piano, the acoustic space and amount of that acoustic space you wish to capture. The closer the mics are to each other, the more intimate the sound. Conversely, moving them further apart makes the piano sound larger. As always, cardioids will focus the sound more on the instrument, while omni will really open up the room (assuming that it has desirable acoustics.)

Some engineers like adding another microphone under the piano facing up to the soundboard to capture the warm, mellow sound of the resonating wood. If you have enough condenser mics at your disposal, try a combination of close-miked stereo pair on top, a single mic on the underside, and a matched pair for distance-miking to capture the sound of the hall.
The prescribed use of multiple microphones on a grand piano leaves any of these scenarios extremely susceptible to phase problems. Be sure to check for phase anomalies and mono compatibility in any of these stereo miking positions.

Drums

There are as many ways to mic drum kits as there are drummers and engineers. Professional recordings are invariably made with enough mics to achieve separation and control over each of the key elements. This luxury is not always available, so we ll explore some distance-miking options as well as close-miking techniques. In all cases, care should go into things like tuning the drums, choice of sticks and other topics that are beyond the scope of this discussion.



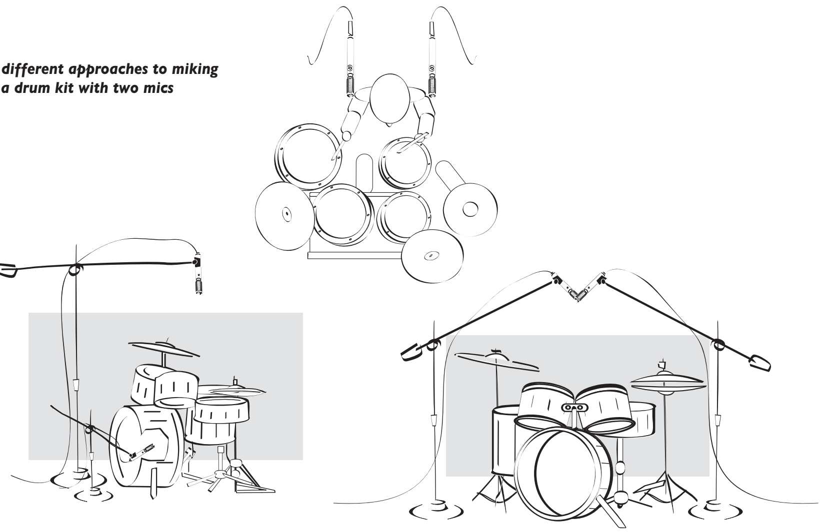
With only a single mic available, experiment to get the best overall balance of all the elements of the kit

Distance-Miking Drums. The more mics you have, the greater the control you have over the balance, stereo placement, EQ and effects for each component of the kit. We ll work our way up from a single mic through multiple mics.

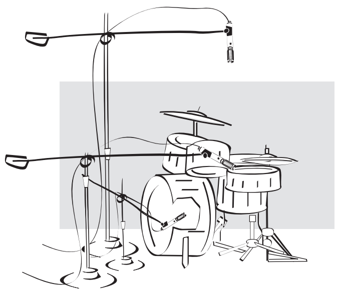
• *Single Mic*. Using a single mic, the best you can attain is some sense of balance between the individual drums, along with the amount of room reflection versus pure drums. Using a boom, try angling a cardioid mic toward the kit at about 6 feet off the ground and about one foot in front or behind the kit. Alternately, place the mic four feet above the center of the kit. If you desire more room sound, try pointing the mic directly at the kit at a distance of about eight feet and auditioning both cardioid and omni patterns.

• *Two Mics*. Several techniques are available using two mics. If you have a matched pair, try an X-Y pattern about three feet directly above the kit. Alternately, place the matched mics level with the drummer s ears and facing forward about four to eight inches on either side of his/her head. (The drummer s head actually acts as a form of baffle in this technique.) If you do not have a matched pair, try placing the one with the larger diaphragm inside the kick drum and the other on a boom about two feet over the rest of the kit.

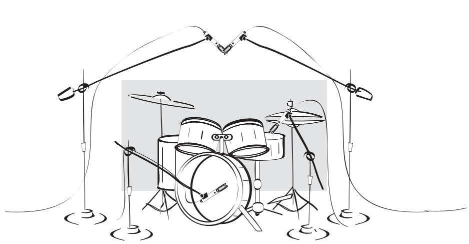
Three different approaches to miking a drum kit with two mics



• *Three Mics*. Where only three mics are available, two main techniques are favored. The first is a hybrid of the aforementioned dual-mic techniques where one mic is placed inside the kick and the other two form an X-Y several feet above the kit. The other is to mic the kick and snare separately along with one overhead. (See individual close-miking techniques for more information.)



When only three mics are available for drums, try featuring the kick and snare, capturing the rest with a single overhead



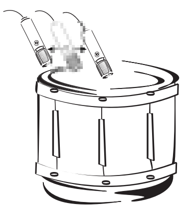
Four mics provide for featuring the kick and snare, while capturing the rest of the kit with a stereo X-Y configuration

• *Four Mics*. The use of four mics begins to open the possibilities for professional results. Place individual mics on the kick and snare, then use a matched pair in X-Y configuration for stereo overheads. (See the information on individual close-miking techniques for more information.)

Close-Miking Drums. Much of todays music focuses so strongly on the kick, snare and hi-hat that being able to mic these components of a drum individually is fairly critical. At a minimum, you also need a pair of overheads to catch everything else in stereo. In a perfect world, you have enough mics and channels to mic each element of the kit individually with the exception of the cymbals (hi-hat excluded) being captured by the stereo overheads. Cardioid or super-cardioid is ideal in most cases to isolate the elements of the kit from one another.

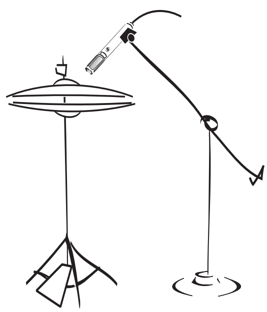
Placement of the mic in relation to the beater head determines much of the tonal quality

• *Kick Drum*. If the front head is on the kick drum and there is no hole in which to insert a mic, simply place the mic close to the front head. Placing the mic inside the kick drum provides more flexibility. Placement near where the beater strikes the head produces a tighter, punchier sound, while moving further out makes the sound larger and deeper. It is common practice to experiment with various methods of padding inside of the kick in order to increase punch and reduce boom. Use as large a diaphragm as possible. (In fact, some of the best kick drum mics are oversized dynamics such as the EV RE20.) You may need to switch on the mic s built-in pad if the sound pressure is overloading the electronics of either the mic or the preamp.



The edge of a drum provides more tone, while the center provides more attack

• *Snare*. Snare drums are one of the few places where dynamic mics are routinely used in the studio. One of the reasons for this is that the snare mic is the most likely to be hit by errant drum sticks. That said, you can certainly use a condenser, especially in more subtle applications such as those involving brushes. Standard practice is to angle the mic down toward the drum at about two inches from the rim. Moving the mic further in provides more attack and less body something that is true for most drums. Snares are sometimes dampened with duct tape, wallets and other mechanisms if they are too lively. Some engineers routinely place a condenser under the snare drum as a second mic in order to capture the sizzle of the snares themselves.



Miking the edge of a hi-hat yields more sizzle, while moving it further in results in more of the bell sound

• *Hi-hat*. Most engineers place a mid-sized cardioid condenser facing down at the outer edge of the hi-hat. This position tends to produce more of the sound of the stick striking the cymbal, where moving it further inward captures more of the quality of the bell. In either event, orienting the cardioid diaphragm downward helps to reject bleed from a neighboring overhead cymbal.

• *Toms*. As with the snare, tom-toms are often the domain of dynamic mics like the Sennheiser 421 because of the possibility of being hit by drum sticks. Here again, condensers are perfectly valid in controlled situations. Miking the toms individually provides the flexibility of balancing, panning and EQing them separately in the mix.

Mid-sized cardioids in X-Y configuration over the kit is the most common method of overhead drum miking

• *Overheads*. A matched pair of condensers is the generally preferred method of overhead miking. Mid-sized capsules are used more typically than large capsules in this application due to the smooth high frequency response.

However, you can experiment with large capsule mic s like the M-Audio Luna, which is noted for it s performance in the higher frequencies.If using a pair, they can either be used in an X-Y coincident fashion or spaced several feet apart over the left and right portions of the kit. In both cases, experiment with a height of anywhere between two and five feet above the kit, depending upon the room. In general, high ceilings are helpful when miking drum overheads because there is more room for the sound to breathe before being reflected back.

• *Room mics*. If the drums are in a sizable room, you can attain avery large drum sound by placing a stereo pair of omnis out in the room that can be mixed in with the individual mics. Adding compression can make the sound appear to be even bigger.

Chapter 6

The M-Audio Family of Microphones

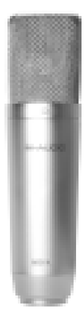
At this point, you ve learned a good deal about how microphones work, how they are built and how they are typically applied. Along the way, we ve occasionally referred to some of the specific models in the Groove Tubes family of mics. Here s a brief overview of the entire M-Audio lineup:

All M-Audio mics share a great deal in common. Here s a brief review in the context of some of the mic construction features discussed in first part of this guide.

¥ 20Hz—20kHz frequency response (+/- 1dB) in FET mics
¥ industry-thinnest, ultra-sensitive diaphragms
¥ precision-crafted solid brass backplate
¥ industry-lowest dynamic distortion
¥ FET rather than op amps for improved dynamic distortion
¥ Class A head amp electronics
¥ Precision manufacturing and quality control

Model:	Solaris
Diaphragm:	1.1" x 3 microns
Electronics:	Class A FET
Patterns:	Cardioid, Omni & Figure 8
Address:	Top
Applications:	Excellent on stringed instruments, overhead cymbals and vocalists who already have a deep voice. Great for close-miking anything without proximity effect. Good results for most other applications, including stereo miking.

Model:	Luna
Diaphragm:	1.1" x 6 microns
Electronics:	Class A FET
Patterns:	Cardioid
Address:	Top
Applications:	The ultimate acoustic guitar mic. Exceptional on stringed instruments, overhead cymbals and vocalists. Excellent for close-miking anything without proximity effect. Very good results for most other applications.



Model:	Nova
Diaphragm:	1.1" x 6 microns
Electronics:	Class A FET
Patterns:	Cardioid
Address:	Side
Applications:	Excellent on vocals and grand piano. A good all-around instrument mic. Excellent sensitivity and pattern flexibility for duets, stereo and ensemble recording.

Appendix A

Troubleshooting Tips

Please consult the following before calling tech support (or even a friend).

Problem: No audio from mic

¥ Phantom power not on. Turn on phantom power on your mic preamp or mixer channel.
¥Gain is too low Make certain the pad is not turned on, both on the mic and preamp/mixer. Turn up input gain and/or channel trim.
¥ Mic cable isn t connected.

Problem: Audible buzz

¥ Bad mic cable. Replace cable.
¥ Mic cable crossing a power cable. Route audio cables so that they are not near power cables or power supplies. When they must cross, make them cross at a 90 degree angle for minimal overlap.
¥ Mic preamp gain is too high. Check the signal path to ensure that high preamp gain is not compensating for attenuation in the subsequent audio path.

Problem: Sound is muffled

¥Addressing wrong side of the mic. Address the top on top-address mics address the side with the M-Audio logo, on side-address models.
¥ Mic is too close to the source. Increase distance between mic and source.

Problem: Low level from mic

¥ Mic cable may be damaged. Try a different mic cable.

Appendix B

Contact Information

M-Audio, formerly known as Midiman, is a leading provider of digital audio and MIDI solutions for today s electronic musicians and audio professionals. Founded in 1988, M-Audio now has independent offices in the US, Canada, UK, Germany, France, and Japan. M-Audio s ability to parlay advanced technology into affordable products has led the company to win some of the international audio community s highest praise and awards including being named the industry s fastest growing company for the past two years running by *Music Trades* magazine.

M-Audio s mission of Refining the Studio delivers new levels of control, virtualization and mobility that transform the way computer-centric musicians compose, perform and live their creative lives. In addition to manufacturing its own product lines like the popular Delta audio cards, Studiophile reference monitors and USB Keystation MIDI controllers, M-Audio also wholly owns Evolution Electronics LTD and distributes other best-of-class products such as Propellerhead s Reason and Ableton Live. In 2002, the company successfully launched a product line that brings professional-quality audio to the consumer electronics market.

For more information you may visit our Web site, contact us by phone or visit an authorized M-Audio dealer.

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