style and taste. Mic pres have also tapped into the growing market of those systems that are based around a digital audio workstation, which doesn’t need a console or mixer but does require a quality pre (or set of pres) for plugging mic signals directly into the soundcard.

**Phantom Power**

Most modern professional condenser mics don’t require internal batteries, external battery packs, or individual AC power supplies in order to operate. Instead, they are designed to be powered directly from the console through the use of a **phantom power** supply. Phantom power works by supplying a positive DC supply voltage of +48V through both conductors (pins 2 and 3) of a balanced mic line to the condenser capsule and impedance preamp. This voltage is equally distributed through identical value resistors, so that no differential exists between the two leads. The −48V side of the circuit is supplied to the capsule and preamp through the audio cable’s grounding wire (pin 1). Since the audio is only affected by potential differences between pins 2 and 3, the carefully matched +48V potential at these leads is therefore not electrically “visible” to the input stage of a balanced mic preamp. Instead, only the alternating audio signal that’s being simultaneously carried along the two audio leads is detected (Figure 4.27).

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

**Microphone Techniques**

Each microphone has a distinctive sound character that’s based on its specific type and design. A large number of types and models can be used for a variety of applications, and it’s up to
the engineer to choose the right one for the job. At this point, I’d like to point out two particular styles for choosing the types and models of microphones that you might have in your own studio or production toolbox. These can basically be placed into the categories of:

* Choosing a limited range of mics that are well suited for a wide range of applications
* Amassing a collection of mics that are commonly perceived as being individually suited for a particular instrument or situation

The first approach not only is ideal for the project studio and those who are just starting out and are on a limited budget but is also common practice among seasoned professionals who swear by a limited collection of their favorite mics (often chosen in stereo pairs). These dynamic and/or condenser mics can be used both in the project studio and in the professional studio (it never hurts to have your own favorites around when traveling to different environments). The second approach (I often refer to it as the “Alan Sides Approach”) is better suited to the professional studio (and to personal collectors) who actually have a need to amass their own “dream collection” and offer it to their clients. In the end, both approaches have their merits . . . Indeed, it’s usually wise to keep an open mind and choose the range and types of mics that best fit your needs, budget, and personal style.

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

**Pickup Characteristics as a Function of Working Distance**

In studio and sound-stage recording, four fundamental styles of microphone placement are directly related to the working distance of a microphone from its sound source. These important pickup styles (which are described in the following sections) include:

* Distant miking
* Close miking
* Accent miking
* Ambient miking

**Distant Microphone Placement**

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

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

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

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

- Temporarily correct for bad or excessive room reflections by using absorptive and/or offset reflective panels.
- Place the mic closer to its source and add artificial ambience.

If a distant mic is used to pick up a portion of the room sound, placing it at a random height can result in a hollow sound due to phase cancellations that occur between the direct sound and delayed sounds that are reflected off the floor and other nearby surfaces (Figure 4.29). If these
delayed reflections arrive at the mic at a time that’s equal to one-half a wavelength (or an odd multiple thereof), the reflected signal will be 180° out of phase with the direct sound. This could produce dips in the signal’s pickup response that would color the signal. Since the reflected sound is at a lower level than the direct sound (as a result of traveling farther and losing energy as it bounces off a surface), the cancellation is usually only partially complete.

Although raising the mic will have the effect of reducing reflections (due to the increased distances that the reflected sound must travel), moving the mic close to the floor will conversely reduce the path length and raise the range in which the frequency cancellation occurs. In practice, a height of 1/8 to 1/16 inch will raise the cancellation above 10 kHz. One such microphone design type, known as a boundary microphone (Figures 4.30 and 4.31), places an electret-condenser or condenser diaphragm well within these low height restrictions. For this reason, this mic type can be a good choice for use as an overall distant pickup, when the mics need to be out of sight (i.e., placed on a floor, wall, or large boundary).

**Close Microphone Placement**

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

- It creates a tight, present sound quality.
- It effectively excludes the acoustic environment.
Because sound diminishes with the square of its distance from the sound source, a sound that originates 3 inches from the pickup will be much higher in level than one that originates 6 feet from the mic (Figure 4.32). Therefore, whenever close miking is used, only the desired on-axis sound will be recorded—while extraneous, distant sounds (for all practical purposes) won’t be picked up. In effect, the distant pickup will be masked by the closer sounds and/or will be reduced to a relative level that’s well below the main pickup.

Whenever an instrument’s mic also picks up the sound of a nearby instrument, a condition known as leakage occurs (Figure 4.33). Since this “leaked” signal will be picked up by both its intended mic and a nearby mic (or mics), it’s easy to see how the signals could be combined together within the mixdown process. When this happens, level and phase cancellations often make it more difficult to have control over individual tracks in mixdown without affecting the level and sound character of other tracks. As a result, try to avoid this condition during the recording process, whenever possible.
To avoid the problems of leakage, any or all of the following methods can be tried:

- Place the mics closer to their respective instruments (Figure 4.34a).
- Place an acoustic barrier (known as a flat, gobo, or divider) between the two instruments (Figure 4.34b).
- Use directional mics.
- Spread the instruments farther apart.

Whenever individual instruments are being miked close (or semi-close), it’s generally wise to follow the 3:1 distance rule. This principle states that, in order to reduce leakage and maintain
phase integrity, for every unit of distance between a mic and its source a nearby mic (or mics) should be separated by at least three times that distance (Figure 4.35). It should be noted that some err on the side of caution and avoid leakage even further by following a 5:1 distance rule. As always, experience will be your best teacher. Although the close miking of a sound source offers several advantages, a mic should be placed only as close to the source as is necessary, not as close as possible. Miking too close can color the recorded tone quality of a source.

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

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

Do-It-Yourself Tutorial: Close Miking

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

Accent Microphone Placement

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

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

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

**Do-It-Yourself Tutorial: Ambient Miking**

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

**Stereo Miking Techniques**

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

The four fundamental stereo miking techniques are

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

**Spaced Pair**

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

**X/Y**

*X/Y stereo miking* is an intensity-dependent system that uses only the cue of amplitude to discriminate direction. With the X/Y coincident-pair technique, two directional microphones of the same type, manufacture, and model are placed with their grills as close together as possible (without touching) and facing at angles to each other (generally between 90 and 135°). The midpoint between the two mics is pointed toward the source, and the mic outputs are equally panned left and right. Even though the two mics are placed together, the stereo imaging is excellent—often better than that of a spaced pair. In addition, due to their close proximity, no appreciable phase problems arise. Most commonly, X/Y pickups use mics...
that have a cardioid polar pattern (Figure 4.38a), although the Blumlein technique is being increasingly used. This technique (which is named after the unheralded inventor, Alan Dower Blumlein) uses two crossed bidirectional mics, which are offset by 90° to each other, and often yields excellent ambient results (Figure 4.38b). Stereo microphones that contain two diaphragms in the same case housing are also available on the used and new market. These mics will either be fixed (generally in a 90° or switchable X/Y pattern) or designed so that the top diaphragm can be rotated by 180° (allowing for the adjustment of various coincident X/Y angles.

M/S

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

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

**Decca Tree**

Although not as commonly used as the above techniques, the *Decca tree* is a time-tested, classical miking technique that uses both time and amplitude cues in order to create a coherent stereo image. Attributed originally to Decca engineers Roy Wallace and Arthur Haddy in 1954, the Decca tree (Figure 4.41) consists of three omnidirectional mics (originally, Neumann M50 mics were used). In this arrangement, a left and right mic pair are placed 3 feet apart, while a third mic is placed 1.5 feet out in front and panned in the center of the stereo field. Still favored by many in orchestral situations as a main pickup pair, the Decca tree is most commonly placed on a tall boom, above and behind the conductor. According to lore, when Haddy first saw the array, he remarked: “It looks like a bloody Christmas tree!” The name stuck.
With the advent of 5.1 surround sound production, it’s certainly possible to make use of a surround console or DAW to place sources that have been recorded in either mono or stereo into a surround image field. Under certain situations, it’s also possible to consider using multiple-pickup surround miking techniques in order to capture the actual acoustic environment and then translate that into a surround mix. Just as the number of techniques and personal styles increases when miking in stereo compared to mono... the number of placement and technique choices will likewise increase when miking a source in surround.
Although guidelines have been and will continue to be set, both placement and mixing styles are definitely an art and not a science.

**Ambient Surround Mics**

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

**Surround Decca Tree**

One of the most logical techniques for capturing an ensemble or instrument uses five mics and makes use of several variations on the Decca tree. For example, Ron Streicher has developed an ingenious and simple system of adding two rear-facing mics to the existing three-mic Decca tree system (although he has modified the system by placing a coincident M/S mic in the forward-center position), as shown in Figure 4.42a. Another simpler approach is to place five cardioid mics in a circle, such that the center channel faces toward the source, thereby creating a simple setup that can be routed L–C–R–RL–RR (Figure 4.42b).

One last approach (which doesn’t actually fall under the Decca tree category) involves the use of four cardioid mics that are spaced at 90° angles, representing L–R–RL–RR, with the on-axis point being placed 45° between the L and R mics (Figure 4.43). This configuration can be easily made by mounting the mics on two stereo bars that are offset by 90°.

**Recording Direct**

As an alternative, the signal of an electric or electronic instrument (guitar, keyboard, and so on) can be directly “injected” into a console, recorder or DAW without the use of a microphone. This option can produce a cleaner, more present sound by bypassing the distorted components of a head/amp combination. It also reduces leakage into other mics by eliminating room sounds. In the project or recording studio, the direct injection (DI) box (Figure 4.44) serves to interface an instrument with an analog output signal to a console or recorder in the following ways:

- It reduces an instrument’s line-level output to mic level for direct insertion into the console’s mic input jack.
- It changes an instrument’s unbalanced, high-source impedance line to a balanced, low-source impedance signal that’s needed by the console’s input stage.
- It electrically isolates audio signal paths between the instrument and mic/line preamp stages (thereby reducing the potential for ground-loop hum and buzzes).
Figure 4.42. Surround Sound Decca examples: (a) Ron Streicher’s Surround Sound Decca tree (courtesy of Ron Streicher, author of The New Stereo Soundbook and proprietor of Pacific Audio-Visual Enterprises, Pasadena, CA); (b) placing five cardioid microphones in a circular pattern (with the center microphone facing toward the source) to create a modified, mini-surround Decca tree.
Most commonly, the instrument’s output is plugged directly into the DI box (where it’s stepped down in level and impedance), and the box’s output is then fed into the mic pre of a console or DAW. If a “dirtier” sound is desired, certain boxes will allow high-level input signals to be taken directly from the amp’s speaker output jack. It’s also not uncommon for an engineer, producer, and/or artist to combine the punchy, full sound of a mic with the present crispness of a direct sound. These signals can then be combined onto a single tape track or recorded to separate tracks (thereby giving more flexibility in the mixdown stage). The ambient image can be “opened up” even further by mixing a semi-distant or distant mic (or stereo pair) with the direct (and even with the close miked amp) signal. This ambient pickup can be either mixed into a stereo field or at the rear of a surround field to fill out the sound.
Microphone Placement Techniques

The following sections are meant to be used as a general guide to mic placement for various acoustic and popular instruments. It’s important to keep in mind that these are only guidelines. Several general application and characteristic notes are detailed in Table 4.1, and descriptions of several popular mics are outlined in the “Microphone Selection” section to help give insights into mic placements and techniques that might work best in a particular application.

As a general rule, choosing the best mic for an instrument or vocal will ultimately depend upon the basic character of the sound that you’re searching for. For example, a dynamic mic will often yield a “rugged” or “punchy” character (which is often further accentuated by a close proximity bass boost that’s associated with most directional mics). A ribbon mic will often yield a mellow sound that ranges from being open and clear to slightly “croony”... depending on the type and distances involved. Condenser mics are often characterized as having a clear, present, and full-range sound that varies with mic design, grill options, and capsule size.

Before jumping into this section, I’d like to again take time to point out the “Good Rule” to anyone who wants to be a better engineer, producer, and/or musician:

\[
\text{Good musician} + \text{good acoustics} + \text{good mike} + \text{good placement} = \text{good sound}
\]

<table>
<thead>
<tr>
<th>Needed Application</th>
<th>Required Microphone Choice and/or Characteristic</th>
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<tbody>
<tr>
<td>Natural, smooth tone quality</td>
<td>Flat frequency response</td>
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<tr>
<td>Bright, present tone quality</td>
<td>Rising frequency response</td>
</tr>
<tr>
<td>Extended lows</td>
<td>Dynamic or condenser with extended low-frequency response</td>
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<tr>
<td>Extended highs (detailed sound)</td>
<td>Condenser</td>
</tr>
<tr>
<td>Increased “edge” or mid-range detail</td>
<td>Dynamic</td>
</tr>
<tr>
<td>Extra ruggedness</td>
<td>Dynamic or modern ribbon/condenser</td>
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<tr>
<td>Boosted bass at close working distances</td>
<td>Directional microphone</td>
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<tr>
<td>Flat bass response up close</td>
<td>Omnidirectional microphone, or omnidirectional microphone at close working distances</td>
</tr>
<tr>
<td>Reduced leakage, feedback,</td>
<td>Directional microphone, or omnidirectional microphone</td>
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<tr>
<td>and room acoustics</td>
<td></td>
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<tr>
<td>Enhanced pickup of room acoustics</td>
<td>Place microphone or stereo pair at greater working distances</td>
</tr>
<tr>
<td>Reduced handling noise</td>
<td>Omnidirectional, vocal microphone, or directional microphone with shock mount</td>
</tr>
<tr>
<td>Reduced breath popping</td>
<td>Omnidirectional or directional microphone with pop filter</td>
</tr>
<tr>
<td>Distortion-free pickup of very loud</td>
<td>Dynamic or condenser with high maximum SPL rating</td>
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<tr>
<td>sounds</td>
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<tr>
<td>Noise-free pickup of quiet sounds</td>
<td>Condenser with low self-noise and high sensitivity</td>
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As a rule, starting with an experienced, rehearsed, and ready musician who has a quality instrument that’s well tuned is the best insurance toward getting the best possible sound. Let’s think about this for a moment. Say that we have a live rhythm session that involves drums, piano, bass guitar, and scratch vocals. All of the players are the best around, except for the drummer, who is new to the studio process. Unfortunately, you’ve just signed on to teach the drummer the ropes of proper drum tuning. The session will go far less smoothly than it otherwise would, as you’ll have to take the extra time to work with the player to tune the drums and set up the mics in order to get the best possible sound. Once you’re rolling, it’ll then be up to you or the producer to pull a professional performance out of someone who’s new to the field.

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

Brass Instruments

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

Trumpet

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

Trombones come in a number of sizes; however, the most commonly used “bone” is the tenor that has a fundamental note range spanning from E2 to C5 (82–520 Hz) and produces a series of complex overtones that range from 5 kHz (when played medium loud) to 10 kHz (when overblown). The trombone’s polar pattern is nearly as tight as the trumpet’s: Frequencies below 400 Hz are distributed evenly, whereas its dispersion angle increases to 45° from the bell at 2 kHz and above. The trombone most often appears in jazz and classical music. The *Mass in C Minor* by Mozart, for example, has parts for soprano, alto, tenor, and bass trombones. This style obviously lends itself to the spacious blending that can be achieved by distant pickups within a large hall or studio. On the other hand, jazz music often calls for closer miking distances. At 2 to 12 inches, for example, the trombonist should play slightly to the side of the mic to reduce the chance of overload and wind blasts. In the miking of a trombone section, a single mic might be placed between two players, acoustically combining them onto a single channel and/or track.

**Tuba**

The bass and double-bass tubas are the lowest pitched of the brass/wind instruments. Although the bass tuba’s range is actually a fifth higher than the double bass, it’s still possible to obtain
a low fundamental of B (29 Hz). A tuba’s overtone structure is limited—with a top response ranging from 1.5 to 2 kHz. The lower frequencies (around 75 Hz) are evenly dispersed; however, as frequencies rise, their distribution angles reduce. Under normal conditions, this class of instruments isn’t miked at close distances. A working range of 2 feet or more, slightly off-axis to the bell, will generally yield the best results.

**French Horn**

The fundamental tones of the French horn range from B1 to B5 (65–700 Hz). Its “oo” formant gives it a round, broad quality that can be found at about 340 Hz, with other frequencies falling between 750 Hz and 3.5 kHz. French horn players often place their hands inside the bell to mute the sound and promote a formant at about 3 kHz. A French horn player or section is traditionally placed at the rear of an ensemble, just in front of a rear, reflective stage wall. This wall serves to reflect the sound back toward the listener’s position (which tends to create a fuller, more defined sound). An effective pickup of this instrument can be achieved by placing an omni- or bidirectional pickup between the rear, reflecting wall and the instrument bells, thereby receiving both the direct and reflected sound. Alternatively, the pickups can be placed in front of the players, thereby receiving only the sound that’s being reflected from the rear wall.

**Guitar**

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

**Acoustic Guitar**

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

**Miking Near the Sound Hole**

The sound hole (located at the front face of a guitar) serves as a bass port, which resonates at lower frequencies (around 80–100 Hz). Placing a mic too close to the front of this port
might result in a boomy and unnatural sound; however, miking close to the sound hole is often popular on stage or around high acoustic levels because the guitar’s output is highest at this position. To achieve a more natural pickup under these conditions, the microphone’s output can be rolled off at the lower frequencies (5–10 dB at 100 Hz).

Surround Guitar Miking

An effective way to translate an acoustic guitar to the wide stage of surround (if a big, full sound is what you’re after) is to record the guitar using X/Y or spaced techniques stereo (panned front L and R) . . . and pan the guitar’s electric pickup (or contact pickup, if it doesn’t have one) to the rear center of the surround field. Extra ambient surround mics can be used in an all-acoustic session.

The Electric Guitar

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

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

Recording Direct

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

Figure 4.47. Miking an electric guitar cabinet directly in front of and off-center to the cone.
The Electric Bass Guitar

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

Keyboard Instruments

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

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

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

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

Figure 4.50 shows a number of possible microphone positions that are acceptable for recording a grand piano. Although several mic positions are illustrated, it’s important to keep in mind that these are only guidelines from which to begin. Your own personal sound can be achieved through mic choice and experimentation with mic placement. The following list explains the numbered miking positions shown in Figure 4.49:

- **Position 1.** The mic is attached to the partially or entirely open lid of the piano. The most appropriate choice for this pickup is the boundary mic, which can be permanently attached or temporarily taped to the lid. This method uses the lid as a collective reflector and provides excellent pickup under restrictive conditions (such as on stage and during a live video shoot).
- **Position 2.** Two mics are placed in a spaced stereo configuration at a working distance of 6 inches to 1 inch. One mic is positioned over the low strings and one is placed over the high strings.
- **Position 3.** A single mic or coincident stereo pair is placed just inside the piano between the soundboard and its fully or partially open lid.
- **Position 4.** A single mic or stereo coincident pair is placed outside the piano, facing into the open lid (this is most appropriate for solo or accent miking).
- **Position 5.** A spaced stereo pair is placed outside the lid, facing into the instrument.
- **Position 6.** A single mic or stereo coincident pair is placed just over the piano hammers at a working distance of 4 to 8 inches to give a driving pop or rock sound.

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

**Separation**

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

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

**Upright Piano**

You would expect the techniques for this seemingly harmless piano type to be similar to those for its bigger brother. This is partially true. However, since this instrument was designed for home enjoyment and not performance, the mic techniques are often very different. Since it’s
often more difficult to achieve a respectable tone quality when using an upright, you might want to try the following methods (Figure 4.50):

- **Miking over the top.** Place two mics in a spaced fashion just over and in front of the piano’s open top—with one over the bass strings and one over the high strings. If isolation isn’t a factor, remove or open the front face that covers the strings in order to reduce reflections and, therefore, the instrument’s characteristic “boxy” quality. Also, to reduce resonances you might want to angle the piano out and away from any walls.

- **Miking the kickboard area.** For a more natural sound, remove the kickboard at the lower front part of the piano to expose the strings. Place a stereo spaced pair over the strings (one each at a working distance of about 8 inches over the bass and high strings). If only one mic is used, place it over the high-end strings. Be aware, though, that this placement can pick up excessive foot-pedal noise.

- **Miking the upper soundboard area.** In order to reduce excessive hammer attack, place a microphone pair at about 8 inches from the soundboard, above both the bass and high strings. In order to reduce muddiness, the soundboard should be facing into the room or be moved away from nearby walls.

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**Electronic Keyboard Instruments**

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

**Percussion**

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

*Figure 4.51. A Leslie speaker cabinet creates a unique vibrato effect by using a set of rotating speaker baffles that spin on a horizontal axis: (a) miking the rotating speakers of a Leslie cabinet; (b) modern portable rotary amp with built-in microphones and three XLR outputs. (Courtesy of Motion Sound, www.motion-sound.com.)*
Drum Set

The standard drum kit (Figure 4.52) is often at the foundation of modern music, as it provides the "heartbeat" of a basic rhythm track; consequently, a proper drum sound is extremely important to the outcome of most music projects. Generally, the drum kit is composed of the kick drum, snare drum, high-toms, low-tom (one or more), hi-hat, and a variety of cymbals. Since a full kit is a series of interrelated and closely spaced percussion instruments, it often takes real skill to translate the proper spatial and tonal balance into a project. The larger-than-life driving sound of the acoustic rock drum set that we've all become familiar with is the result of an expert balance among playing techniques, proper tuning, and mic placement. Should any of these variables fall short, the search for that "perfect sound" could prove to be a long and hard one. As a general rule, a poorly tuned drum will sound just as out-of-tune through a good mic as it will through a bad one; therefore, it's important to be sure that the individual drum components sound good to the ears, before attempting to place the mics.

Miking the Drum Set

After the drum set has been optimized for the best sound, the mics can be placed into their pickup positions (Figure 4.53). Because each part of the drum set is so different in sound and function, it's often best to treat each grouping as an individual instrument. In its most basic form, the best place to start when miking a drum set is to start with the fundamental "groups." These include placing a mic on the kick (1) and on the snare drum (4). At an absolute minimum, the entire drum set can be adequately picked up using only four mics by adding two overhead pickups... either spaced (3) or coincident (4). In fact, this "bare bones"
placement was (and continues to be) commonly used on many classic jazz recordings. If more tracks are available (or required), additional mics can be placed on the various toms, hi-hat, and even individual cymbals.

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

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

- The drums could be placed in their own room, isolated from other instruments.
- To achieve a bigger sound, the drums could be placed in the large studio room while the other instruments are placed in smaller iso-rooms or are recorded direct.
- To reduce leakage, the drums could be placed in the studio, while being enclosed by 4-foot (or higher) divider flats (Figure 4.54).

### Kick Drum

The kick drum adds a low-energy drive or “punch” to a rhythm groove. This drum has the capability to produce low frequencies at high sound-pressure levels, so it’s necessary to use a mic that can both handle and faithfully reproduce these signals. Often the best choice for the job is a large-diaphragm dynamic mic. Since proximity effect (bass boost) occurs when using...
a directional mic at close working distances and because the drum’s harmonics vary over its large surface area, even a minor change in placement can have a profound effect on the pickup’s overall sound. Moving the mic closer to the head (Figure 4.55) can add a degree of warmth and fullness, while moving it farther back often emphasizes the high-frequency “click.” Placing the mic closer to the beater emphasizes the hard “thud” sound, whereas an off-center mic captures more of the drum’s characteristic skin tone. A dull and loose kick sound can be tightened to produce a sharper, more defined transient sound, by placing a blanket or other damping material inside the drum shell firmly against the beater head. Cutting back on the kick’s equalization at 300 to 600 Hz can help reduce the dull “cardboard” sound, whereas boosting from 2.5 to 5 kHz adds a sharper attack, “click,” or “snap.” It’s also often a good idea to have a can of WD-40® or other light oil handy in case squeaks from some of the moving parts (most often the kick pedal) gets picked up by the mics.
Snare Drum

Commonly, a snare mic is aimed just inside the top rim of the snare drum at a distance of about 1 inch (Figure 4.56). The mic should be angled for the best possible separation from other drums and cymbals. Its rejection angle should be aimed at either the hi-hat or rack-toms (depending on leakage difficulties). Usually, the mic’s polar response is cardioid, although bidirectional and super-cardioid responses may offer a tighter pickup angle. With certain musical styles (such as jazz), you might want a crisp or “bright” snare sound. This can be achieved by placing an additional mic on the snare drum’s bottom head and then combining the two mics onto a single track. As the bottom snare head is 180° out of phase with the top, it’s generally a wise idea to reverse the bottom mic’s phase polarity. When playing in styles where the snare springs are turned off, it’s also a good idea to keep your ears open for snare rattles and buzzes that can easily leak into the snare mic (as well as other mics). The continued ringing of an “open” snare note (or any other drum type, for that matter) can be dampened in several ways. Dampening rings, which can be purchased at music stores, are used to reduce the ring and to deepen the instrument’s tone. If there are no dampening rings around, the tone can be dampened by taping a billfold or similar-sized folded paper towel to the top of a drumhead, a few inches off its edge.

Overheads

Overhead mics are generally used to pick up the high-frequency transients of cymbals with crisp, accurate detail while also providing an overall blend of the entire drum kit. Because of the cymbals’ transient nature, a condenser mic is often chosen for its accurate high-end response. Overhead mic placement can be very subjective and personal. One type of placement is the spaced pair, whereby two mics are suspended above the left and right sides of the kit. These mics are equally distributed about the L–R cymbal clusters so as to pick up their respective instrument components in a balanced fashion (Figure 4.57a). Another placement method is to suspend the mics closely together in a coincident fashion (Figure 4.57b).
often yields an excellent stereo overhead image with a minimum of the phase cancellations that might otherwise result when using spaced mics. Again, it’s important to remember that there are no rules for getting a good sound. If only one overhead mic is available, place it at a central point over the drums. If you’re using a number of pickups to close mic individual components of a kit, there might be times when you won’t need overheads at all (the leakage spillover just might be enough to do the trick).

**Rack-Toms**

The upper rack-toms can be miked either individually (Figure 4.58) or by placing a single mic between the two at a short distance (Figure 4.59). When miked individually, a “dead” sound can be achieved by placing the mic close to the drum’s top head (about 1 inch above and 1 to 2 inches in from the outer rim). A sound that’s more “live” can be achieved by increasing the height above the head to about 3 to 6 inches. If isolation or feedback is a consideration, a hypercardioid pickup pattern can be chosen. Another way to reduce leakage and to get a deep,
driving tone (with less attack) is to remove the tom’s bottom head and place the mic inside, 1 to 6 inches away from the top head.

**Floor-Tom**

Floor-toms can be miked similarly to the rack-toms (Figure 4.60). The mic can be placed 2 to 3 inches above the top and to the side of the head, or it can be placed inside 1 to 6 inches from the head. Again, a single mic can be placed above and between the two floor-toms, or each can have its own mic pickup (which often yields a greater degree of control over panning and tonal color).

**Hi-Hat**

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

- Placing the mic above the top cymbal will help pick up the nuances of sharp stick attacks.
- The open and closing motion of the hi-hat will often produce rushes of air; consequently, when miking the hat’s edge, angle the mic slightly above or below the point where the cymbals meet.
- If only one mic is available (or desired), both the snare and hi-hat can be picked up simultaneously by carefully placing the mic between the two, facing away from the rack-toms as much as possible. Alternatively, a figure-8 mic can be placed between the two with the null axis facing toward the cymbals and the kick.

Tuned Percussion Instruments

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

Congas and Hand Drums

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

Xylophone, Vibraphone, and Marimba

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

Stringed Instruments

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

Violin and Viola

The frequency range of the violin runs from 200 Hz to 10 kHz. For this reason, a good mic that displays a relatively flat frequency response should be used. The violin’s fundamental range is from G3 to E6 (200–1300 Hz), and it is particularly important to use a mic that’s flat around the formant frequencies of 300 Hz, 1 kHz, and 1200 Hz. The fundamental range of the viola is tuned a fifth lower and contains fewer harmonic overtones. In most situations, the violin or viola’s mic should be placed within 45° of the instrument’s front face. The distance will depend on the particular style of music and the room’s acoustic condition. Miking at a greater distance will generally yield a mellow, well-rounded tone, whereas a closer position might yield a scratchy, more nasal quality...the choice will depend on the instrument’s tone quality. The recommended miking distance for a solo instrument is between 3 and 8 feet, over and slightly in front of the player (Figure 4.61). Under studio conditions, a closer mic distance of between 2 and 3 feet is recommended. For a fiddle or jazz/rock playing style, the mic can be placed at a close working distance of 6 inches or less, as the increased overtones help the instrument to cut through an ensemble. Under PA (public address) applications, distant working conditions are likely to produce feedback (since less amplification is needed).
In this situation, an electric pickup, contact, or clip-type microphone can be attached to the instrument’s body or tailpiece.

**Cello**

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

**Double Bass**

The double bass is one of the orchestra’s lowest-pitched instruments. The fundamentals of the four-string type reach down to E1 (41 Hz) and up to around middle C (260 Hz). The overtone spectrum generally reaches upward to 7 kHz, with an overall angle of high-frequency dispersion being ±15° from the player’s line of sight. Once again, a mic can be aimed at the “f” holes at a distance of between 6 inches and 1-1/2 feet.

**Voice**

From a shout to a whisper, the human voice is a talented and versatile sound source that displays a dynamic and timbral range that’s matched by few other instruments. The male bass voice can ideally extend from E2 to D4 (82–293 Hz) with sibilant harmonics extending to 12 kHz. The upper soprano voice can range upward to 1050 Hz with harmonics that also climb to 12 kHz.
When choosing a mic and its proper placement, it’s important to step back for a moment and remember that the most important “device” in the signal chain is the vocalist. Let’s assume that the engineer/producer hasn’t made the classic mistake of waiting until the last minute (when the project goes over budget and/or into overtime) to record the vocals . . . . Good, now the vocalist can relax and concentrate on a memorable performance. Next step is to concentrate on the vocalist’s “creature comforts”: How are the lighting and temperature settings? Is the vocalist thirsty? Once done, you can go about the task of choosing your mic and its placement to best capture the performance.

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

- **Excessive dynamic range.** This can be solved either by mic technique (physically moving away from the mic during louder passages) or by inserting a compressor into the signal path. Some vocalists have dynamics that range from whispers to normal volumes to practically screaming . . . all in a single passage. If you optimize your recording levels during a moderate-volume passage and the singer begins to belt out the lines, then the levels will become too “hot” and will distort. Conversely, if you set your recording levels for the loudest passage, the moderate volumes will be buried in the music. The solution to this dilemma is to place a compressor in the mic’s signal path. The compressor automatically “rides” the signal’s gain and reduces excessively loud passages to a level that the system can effectively handle. (See Chapter 12 for more information about compression and devices that alter dynamic range.)

- **Sibilance.** This occurs when sounds such as “f,” “s,” and “sh” are overly accentuated. This often is a result of tape saturation and distortion at high levels or slow tape speeds. Sibilance can be reduced by inserting a frequency-selective compressor (known as a de-esser) into the chain or through the use of moderate equalization.

- **Popping.** Explosive popping “p” and “b” sounds result when turbulent air blasts from the mouth strike the mic diaphragm. This problem can be avoided or reduced by placing a pop filter over the mic, by placing a mesh windscreen between the mic and the vocalist, or by using an omnidirectional mic (which is less sensitive to popping).

- **Excessive bass boost due to proximity effect.** This bass buildup often occurs when a directional mic is used at close working ranges. It can be reduced or compensated for by increasing the working distance between the source and the mic, by using an omnidirectional mic (which doesn’t display a proximity bass build up), or through the use of equalization.

**Woodwind Instruments**

The flute, clarinet, oboe, saxophone, and bassoon combine to make up the woodwind class of instruments. Not all modern woodwinds are made of wood nor do they produce sound in the same way. For example, a flute’s sound is generated by blowing across a hole in a tube, while other woodwinds produce sound by causing a reed to vibrate the air within a tube.
Opening or covering finger holes along the sides of the instrument controls the pitch of a woodwind by changing the length of the tube and, therefore, the length of the vibrating air column. It’s a common misunderstanding that the natural sound of a woodwind instrument radiates entirely from its bell or mouthpiece. In reality, a large part of its sound often emanates from the fingerholes that span the instrument’s entire length.

**Clarinet**

The clarinet comes in two pitches: the B clarinet, with a lower limit of D3 (147 Hz), and the A clarinet, with a lower limit of C3 (139 Hz). The highest fundamental is around G6 (1570 Hz), whereas notes an octave above middle C contain frequencies of up to 1500 Hz when played softly. This spectrum can range upward to 12 kHz when played loudly. The sound of this reeded woodwind radiates almost exclusively from the finger holes at frequencies between 800 Hz and 3 kHz; however, as the pitch rises, more of the sound emanates from the bell. Often, the best mic placement occurs when the pickup is aimed toward the lower finger holes at a distance of 6 inches to 1 foot (Figure 4.62).

**Flute**

The flute’s fundamental range extends from B3 to about C7 (247–2100 Hz). For medium loud tones, the upper overtone limit ranges between 3 and 6 kHz. Commonly, the instrument’s sound radiates along the player’s line of sight for frequencies up to 3 kHz. Above this frequency, however, the radiated direction often moves outward 90° to the player’s right. When miking a flute, placement depends on the type of music being played and the room’s overall acoustics. When recording classical flute, the mic can be placed on-axis and slightly above the player at a distance of between 3 and 8 feet. When dealing with modern musical styles,
the distance often ranges from 6 inches to 2 feet. In both circumstances, the microphone should be positioned at a point 1/3 to 1/2 the distance from the instrument’s mouthpiece to its footpiece. In this way, the instrument’s overall sound and tone quality can be picked up with equal intensity (Figure 4.63). Placing the mic directly in front of the mouthpiece will increase the level (thereby reducing feedback and leakage); however, the full overall body sound won’t be picked up and breath noise will be accentuated. If mobility is important, an integrated contact pickup can be used or a clip mic can be secured near the instrument’s mouthpiece.

**Saxophone**

Saxophones vary greatly in size and shape. The most popular sax for rock and jazz is the S-curved B-flat tenor sax, whose fundamentals span from B2 to F5 (117–725 Hz), and the E-flat alto, which spans from C3 to G5 (140–784 Hz). Also within this family are the straight-tubed soprano and sopranino, as well as the S-shaped baritone and bass saxophones. The harmonic content of these instruments ranges up to 8 kHz and can be extended by breath noises up to 13 kHz. As with other woodwinds, the mic should be placed roughly in the middle of the instrument at the desired distance and pointed slightly toward the bell (Figure 4.64). Keypad noises are considered to be a part of the instrument’s sound; however, even these can be reduced or eliminated by aiming the microphone closer to the bell’s outer rim.

**Harmonica**

Harmonicas come in all shapes, sizes, and keys and are divided into two basic types: the diatonic and the chromatic. Their pitch is determined purely by the length, width, and thickness of the various vibrating metal reeds. The “harp” player’s habit of forming his or her hands around the instrument is a way to mold the tone by forming a resonant cavity. The tone can be deepened and a special “wahing” effect can be produced by opening and closing a cavity that’s formed by the palms; consequently, many harmonica players carry their preferred microphones with them (Figure 4.65) rather than being stuck in front of an unfamiliar mic and stand.
Figure 4.64. Typical microphone positions for the saxophone: (a) standard placement; (b) typical “clip-on” placement.

Figure 4.65. The Shure 520DX “Green Bullet” microphone, a preferred harmonica pickup for many musicians. (Courtesy of Shure Brothers, Inc., www.shure.com.)
Microphone Selection

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

Shure SM-57

The SM-57 (Figure 4.66) is widely used by engineers, artists, touring sound companies, etc., for instrumental and remote recording applications. The SM-57’s mid-range presence peak and good low-frequency response make it useful for use with vocals, snare drums, toms, kick drums, electric guitars, and keyboards.

Specifications

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

*Figure 4.66. Shure SM-57 dynamic microphone. (Courtesy of Shure Brothers, Inc., www.shure.com.)*
Audix D2

This dynamic hypercardioid drum and instrument microphone (Figure 4.67) has a warm, contoured response for added bottom and punch on drums, instruments, and brass. It features a VLM noise rejection capsule and compact design for easy placement and is milled from a solid block of aluminum.

Specifications

- **Transducer type:** moving-coil dynamic
- **Polar response:** hypercardioid
- **Frequency response:** 44 to 18,000 Hz
- **Maximum SPL rating:** 144 dB

The Ball

So named for its unique shape and inimitable Blue styling, the Ball (Figure 4.68) is a phantom-powered dynamic. Although dynamic mics don’t require an external supply to operate, the Ball incorporates a phantom-powered proprietary active balancing circuit in its output stage to maintain a constant, pure-resistive, 50-ohm load across the useable frequency spectrum, which has a dramatic effects on the transducer’s acoustic balance, phase coherence, noise specification, and overall sound.

*Figure 4.67. Audix D2 dynamic microphone.*
(Courtesy of Audix Corporation, www.audixusa.com.)
Specifications

- **Operating principal:** dynamic transducer with active “class A” phantom-powered solid-state circuitry
- **Polar pattern:** cardioid
- **Frequency response:** 35Hz to 16kHz
- **Sensitivity:** 3.5 mV/Pa at 1kHz (1 Pa = 94 dB SPL)
- **Output impedance:** 50 Ω
- **Recommended load impedance:** 2 kΩ
- **Maximum SPL:** 162 dB SPL (2 kΩ load at 1% THD)
- **Output noise:** 17 dB “A” weighted
- **Power requirement:** 48 V DC phantom power
- **Current draw:** 2.5 mA

**AKG D112**

Large-diaphragm cardioid dynamic mics, such as the AKG D112 (Figure 4.69), are often used for picking up kick drums, bass guitar cabinets, and other low-frequency, high-output sources.

**Specifications**

- **Transducer type:** moving-coil dynamic
- **Polar response:** cardioid
Beyerdynamic M-160

The Beyer M-160 ribbon microphone (Figure 4.70) is capable of handling high sound-pressure levels without sustaining damage while providing the transparency that often is inherent in ribbon mics. Its hypercardioid response yields a wide-frequency response/low-feedback characteristic for both studio and stage.

Figure 4.69. AKG D112 dynamic microphone. (Courtesy of AKG Acoustics, Inc., www.akg.com.)

- Frequency response: 30 to 17,000 Hz
- Sensitivity: $-54$ dB $\pm 3$ dB re. 1 V/microbar

Figure 4.70. Beyerdynamic M-160 ribbon microphone. (Courtesy of Beyerdynamic, www.beyerdynamic.com.)
Specifications

- Transducer type: Ribbon dynamic
- Polar response: Hypercardioid
- Frequency response: 40–18,000 Hz
- Sensitivity: 52 dB (0 dB = 1 mW/Pa)
- Equivalent noise rating: −145 dB
- Output impedance: 200 Ω

Royer Labs R-121

The R-121 is a ribbon mic with a figure-8 pattern (Figure 4.71). Its sensitivity is roughly equal to that of a good dynamic mic, and it exhibits a warm, realistic tone and flat frequency response. Made using advanced materials and cutting-edge construction techniques, its response is flat and well balanced; low end is deep and full without getting boomy, mids are well defined and realistic, and the high end response is sweet and natural sounding.

Specifications

- Acoustic operating principle: electrodynamic pressure gradient
- Polar pattern: figure-8
- Generating element: 2.5-micron aluminum ribbon
- Frequency response: 30 to 15,000 Hz ±3 dB

Figure 4.71. Royer Labs R-121 ribbon microphone. (Courtesy of Royer Labs, www.royerlabs.com.)
• Sensitivity: $-54$ dBV re. 1 V/Pa ±1 dB
• Output impedance: 300 Ω at 1 K (nominal); 200 Ω optional
• Maximum SPL: >135 dB
• Output connector: male XLR three-pin (pin 2 hot)
• Finish: Burnished satin nickel; matte black chrome optional

**AEA R84**

The AEA R84 ribbon mic (Figure 4.72) is a cost-effective, general-purpose large geometry ribbon (LGR) mic that’s ideally suited for solo and accent work. The R84 comes standard with a bidirectional polar response and a black/bright chrome “radio” finish.

**Neumann KM 180 Series**

The 180 Series consists of three compact miniature microphones (Figure 4.73): the KM 183 omnidirectional and KM 185 hypercardioid microphones and the successful KM 184 cardioid microphone. All 180 Series microphones are available with either a matte black or nickel finish and come in a folding box with a windshield and two stand mounts that permit connection to the microphone body or the XLR-connector.

**Specifications**

• *Transducer type:* condenser
• *Polar response:* cardioid (183), cardioid (184), and hypercardioid (185)

*Figure 4.72. AEA R84 ribbon microphone. (Courtesy of Audio Engineering Associates, www.wesdooley.com.)*
Figure 4.73. Neumann KM 180 Series condenser microphones. (Courtesy of Georg Neumann GMBH, www.neumann.com.)

- Frequency response: 20 Hz to 20 kHz
- Sensitivity: 12/15/10 mV/Pa
- Output impedance: 50 W
- Equivalent noise level: 16/16/18 dB

AKG C3000B

The AKG C3000B (Figure 4.74) is a low-cost, large-diaphragm condenser mic. Its design incorporates a bass rolloff switch, a –10-dB pad, and a highly effective internal windscreen.

Figure 4.74. The AKG C3000 B condenser microphone. (Courtesy of AKG Acoustics, Inc., www.akg.com.)
The mic’s dual-diaphragm capsule design is floated in an elastic suspension for improved rejection of mechanical noise.

**Specifications**

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

**Marshal MXL 2001**

This low-cost, large-capsule, gold-diaphragm condenser (Figure 4.75) has all the fullness and warmth characteristic of the classic large-capsule mics. It has a realistic, natural tone that makes it a good, all-around mic for both vocal and instrumental recording.

**Specifications**

- **Transducer type:** condenser
- **Polar response:** cardioid
- **Frequency response:** 30 to 20 kHz
- **Sensitivity:** 15 mV/Pa
- **Output impedance:** 200 W
- **Equivalent noise level:** −20 dB−A

*Figure 4.75. The Marshal MXL 2001 condenser microphone. (Courtesy of Marshall Electronics, Inc., www.mxlmicro.com.)*
Neumann Solution-D Digital Mic

The system of the Solution-D digital microphone (Figure 4.76) consists of three components: the digital microphone, D-01; the digital microphone interface (DMI)-2; and the remote control software (RCS), which permits operation and thus remote control of the microphone. Signal and data transmission of the microphone conform to the AES 42-2001 Standard, which identifies the transmission of output signals, power supply for the microphones, and remote control of all typical microphone functions and parameters.

- **Acoustical transducer:** new double-diaphragm capsule K 07 large diaphragm, diameter 30 mm with protected internal electrodes
- **Sensitivity:** 12 mV/Pa at 1 kHz, 0 dB gain
- **Equivalent SPL:** CCIR 468-3, 18 dB; DIN/IEC 651, 7 dB-A
- **S/N ratio:** CCIR 468-3, 76 dB; DIN/IEC 651, 87 dB
- **Weight:** 700 g (approx.)
- **Diameter:** 63.5 mm
- **Length:** 185 mm
- **Interface:** AES 42-2001
- **Dynamic range:** 130 dB (complete system including capsule); 140 dB (ADC input shorted)
- **Internal resolution:** 28 bit
- **Sampling rates:** 48 kHz/96 kHz; alternatively, 44.1 kHz/88.2 kHz
- **Remote controllable functions:** Polar pattern (15 patterns), omni...cardioid...figure-8
  - Low cut, flat; 40 Hz; 80 Hz; 160 Hz
  - Pre-attenuation, 0 dB, −6 dB, −12 dB, 18 dB
  - Gain, 0–63 dB in 1-dB steps, clickless

*Figure 4.76. Neumann Solution-D digital microphone. (Courtesy of Georg Neumann GMBH, www.neumann.com.)*
Switch functions: Soft muting, phase reverse, signal light

Synchronization: AES 42, mode 2 (default); AES 42, mode 1 (see DMI)

Signals: Blue and red LED (switchable via control software and Auxiliary User Port)

Output: XLR 3 M, AES 42-2001

Data format: transmission of audio and status data from the microphone, phantom powering and remote control data to the microphone

Studio Projects LSD2 Stereo Mic

The Studio Projects LSD2 stereo mic (Figure 4.77) is comprised of two separate dual-membrane solid-state microphones contained within a single housing. Its capsules are mounted in close proximity on a vertical axis—the upper capsule assembly having the ability to rotate 270° horizontally, relative to the lower capsule. Two three-way switches control the polar response, high-pass filtering, and −10 dB pad for each capsule. The switches on the front of the body correspond to the lower fixed capsule, while the switches 180° opposite on the back of the mic control the rotating upper capsule. It is the combination of capsule articulation and independent pattern switching that allows a user of the LSD2 to achieve all manner of coincident pair stereophonic recording techniques. Due to the close proximity of the capsules, there is no phase cancellation resulting from the time delay between the two signals.

Specifications

- **Type**: Stereo condenser microphone with vertically coincident 1.06-inch (27-millimeter) dual diaphragms
- **Polar pattern**: Cardioid, omnidirectional, figure 8
- **Frequency response**: 30~20,000 Hz
- **Sensitivity**: 12 mV/Pa = −38 dB (0 dB = 1 V/Pa)
- **Output impedance**: <200 Ω
- **Load impedance**: >1200 Ω
- **Maximum SPL**: 139/146 dB SPL for 1% THD @ 1000 Hz (0 dB/ −10 dB pad, 0 dB SPL = 0.00002 Pa)

Figure 4.77. Studio Projects LSD2 stereo microphone. (Courtesy of Studio Projects USA, www.studioprojectusa.com.)
- **Noise**: Line, 28 dB; A weighted, 18 dB-A
- **Signal-to-noise ratio**: 76 dB
- **Power requirement**: 24 to 52.5 V phantom power
- **Current consumption**: 2.5 mA
- **Circuit**: transformerless circuit featuring extremely low self-noise and large dynamic range
- **Connector**: Gold-plated 7-pin XLR
- **Size**: 2.1-inch (53.34-millimeter) diameter
- **Low Cut**: 6 dB/octave at 150 Hz
- **Pad**: −10dB