CHAPTER 10

Sound Recording Systems

This chapter is about sound and the audio recording equipment used for both film and video. Many of the principles that apply to one type of system are relevant to others. See Chapter 11 for discussion of the sound recordist's role and recording techniques.

SOUND

What we hear as sound is a series of pressure waves produced by vibration. A violin, for example, works by vibrating air rapidly back and forth. When you pluck the string, it makes the body of the violin vibrate—when it moves one way, it *compresses* the air (pushes it) in that direction; when it moves the other way, that pressure is temporarily reduced. Sound waves travel through the air and cause your eardrum to *oscillate* (move back and forth) in response to the sound. Like ocean waves breaking on a beach, sound waves alternately press forward and recede back.

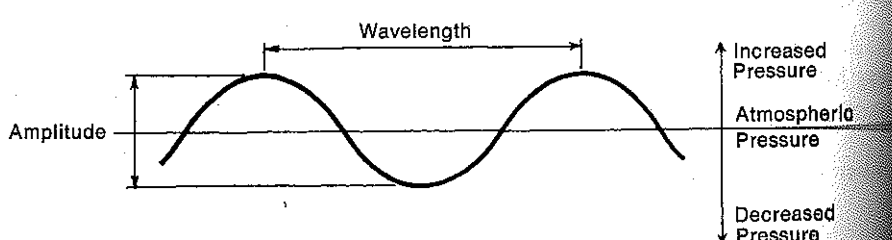


Fig. 10-1. Graph of a simple sound wave at an instant in time. The height of the wave (or *amplitude*) corresponds to loudness. The distance from one peak to the next (the *wavelength*) corresponds to the sound's frequency. (Carol Keller)

Loudness

The *loudness* or *volume* of a sound results from the amount of pressure produced by the sound wave (the *sound pressure level* or *SPL*). Loudness is measured in *decibels* (*dB*), which are used to compare the relative loudness of two sounds. The softest audible sounds occur at the *threshold of hearing*. The volume of normal conversation is about 65 dB above threshold, thus its sound level is said to be 65 dB. The *threshold of pain* is at about 130 dB, equivalent to the noise of a jet passing within 100 feet.

When we work with recording systems, sound level is expressed in dB units that reflect the electrical voltage (see below). With audio in a recording system, if we increase a sound's level by 6 dB, it will sound about *twice* as loud. For more details, see Setting the Recording Level, p. 414.

Dynamic Range

For any passage of sound—be it music, speech, or noise—the difference in volume between the quietest point and the loudest is called the *dynamic range*. The dynamic range of a symphony orchestra is about 80 dB, which represents the difference in volume between the full group playing fortissimo and a single violin playing very softly. Actually, the dynamic range of the orchestra is somewhat lessened by the shuffling and coughing of the audience, which may be louder than the quiet solo violin.

Dynamic range is a term used in evaluating audio systems. The human ear has a dynamic range of 130 dB between the thresholds of hearing and pain. High-quality analog tape recorders have a dynamic range of 70 dB or higher between the loudest sounds that can be recorded without audible distortion and the low-volume sound of the tape noise. Tape noise, or hiss, is always present in analog recordings and, like the shuffling sounds of the symphony audience, it determines the lower limit of the dynamic range (the *noise floor*). The dynamic range of a tape recorder is sometimes called the *signal-to-noise* (*s/n*) *ratio*. The "signal" is the sound we want to record; the "noise" may be tape hiss or system noise from the amplifiers and circuits in the recorder and the microphone.

With digital recorders, tape noise is virtually eliminated (though there can still be other types of noise). Digital recorders usually have much better dynamic range—up to about 100 to 120 dB or more.

Frequency

Musical notes are pitches; the modern piano can produce pitches from a low A to a high C. The lower notes are the *bass*, the higher ones the *treble*. What is perceived as pitch is determined by the *frequency* of the sound wave. Frequency is a measure of how frequently the waves of sound pressure strike the ear—that is, how many cycles of pressure increase/decrease occur in a given length of time. The higher the frequency, the higher the pitch. Frequency was formerly measured in cycles per second; now the same unit is called a *hertz* (*Hz*). Musical notes are standardized according to their frequency. Orchestras usually tune up to concert A, which is 440 Hz. Doubling this or any frequency produces a tone one *octave* higher.

The male speaking voice occupies a range of frequencies from about 100 to 8,000 Hz (8 kHz). The female speaking voice is slightly higher, ranging from about 180 to 10,000 Hz. The ear can sense low frequencies down to about 20 Hz, but these sounds are very rumbly and are felt throughout the body. At the other extreme, sounds above 20,000 Hz are audible to dogs and bats, but seldom humans. For more on working with voice in postproduction, see Frequency Range and EQ, p. 651.

When sound volume is low, the ear is much more sensitive to midrange frequencies (2,000 to 4,000 Hz) than to low or high frequencies. Thus, a low-frequency sound seems quieter than a middle-frequency sound if they have the same sound pressure. Some audio equipment has a "loudness" control that increases the low bass when the volume is down to compensate for this deficiency. When sound volume is high, the ear responds much more evenly to all frequencies of sound; low, middle, and fairly high frequencies of the same sound pressure all seem equally loud.

Tone Quality and Harmonics

All naturally occurring sounds are made up of a mixture of waves at various frequencies. A violin string vibrates at a basic frequency (called the *fundamental*), as well as at various multiples of this frequency. These other frequencies are called *harmonics* or *overtones*, and they are usually quieter than the fundamental. With tones that sound "musical," the frequencies of the harmonics are simple multiples of the fundamental. Most other sounds, such as a speaking voice or a door slam, have no discernible pitch; their harmonics are more complexly distributed. The relative strengths of the harmonics determine tone quality or timbre. When a man and a woman both sing the same note, their voices are distinguishable because the man's voice usually emphasizes lower harmonics than the woman's. Pinching your nose while talking changes, among other things, the balance of harmonics and produces a "nasal" tone quality.

Frequency Response

Frequency response is used to describe how an audio system responds to various frequencies of sound. As noted above, at low volume the ear favors middle-frequency sounds, and at high volume its frequency response is more even or flat. A good audio recorder is capable of providing a fairly *flat* frequency response throughout the frequency range of human hearing.

Because all sounds incorporate a spread of frequencies, if you change the frequency response of your equipment by increasing or decreasing the response to low, middle, or high frequencies, you can change the character of the sounds. The bass and treble controls on a radio do this to some extent; most people like to turn the bass up in dance music to make the rhythm, carried by low-frequency instruments such as the bass guitar and bass drum, seem more powerful. *Equalizers* (see Fig. 16-15) are often used to alter the frequencies of sounds during recording or after. With an equalizer, you could boost low frequencies to make, say, a truck engine sound deep, rumbly, or menacing, or you could boost the high frequencies of a piano to make its sound "brighter." If we diminish high frequencies without changing the bass, the effect is like putting cotton in your ears: The sound is muddy and dull.

Telephones have a fairly limited frequency response, which is centered on the middle frequencies needed to understand speech. In movies, the sound of someone

talking through a phone can be simulated with an equalizer by cutting the low and high frequencies and boosting the midrange.

HOW AUDIO IS RECORDED

This section is about how analog and digital recording works—the fundamental process of capturing and storing sound. Though most recording is done digitally these days, it's important to understand analog because it's still very much a part of the process.

ANALOG AUDIO RECORDING

In simplest terms, the idea of analog recording is to convert sound energy in the air to magnetic energy, which can be stored on tape. When the tape is played back, the process is reversed to reproduce sound (see Fig. 10-2).

The *microphone* responds to sound waves by producing electrical waves that have essentially the same character in terms of frequency and amplitude. Most modern microphones employ an extremely light *diaphragm* that can move with the slightest variations in sound pressure. Mics vary in the way the moving diaphragm generates electricity. The common *dynamic microphone* that comes with some home audio recorders is also called a *moving-coil* microphone because it has a very light coil of wire attached to the diaphragm. When the diaphragm moves back and forth, the coil moves past a magnet and creates an alternating electric current that flows through the wires in the coil. Thus, sound pressure is translated into electric pressure, or *voltage*.

This voltage travels from the microphone to a *mic preamp (preamplifier)*, which increases its strength, and may supply the mic with power. Then it goes to the magnetic *recording head*. A recording head is an electromagnet, not unlike the ones used in metal scrap yards or that kids sometimes play with. When electricity passes through the head it generates a magnetic field. The head is a C-shaped piece of metal with wire coiled around it. On its front is an extremely narrow opening called the *gap*. The head completes a flow of energy: Advancing and receding sound waves become electrical waves, which finally result in a magnetic field that is oriented first in one direction and then in the opposite.

Magnetic tape is made up of a thick support material or *base* and a thin emulsion that stores the information. Tape emulsion is called *oxide* and contains small particles of iron. Each piece of iron is a miniature bar magnet with distinct north and south poles. When a particle of iron passes the gap in the recording head, the magnetic polarity of the particle aligns itself with the magnetic field at the head. When the tape moves on, it maintains that alignment. Since the magnetic field is always alternating back and forth, any given stretch of tape contains groups of particles that alternate in their alignment. The orientation of the particles corresponds to the original sound in this simplified way: The louder the sound, the more particles will be forced to line up the same way; the higher the frequency, the closer together the alternating groups will be.

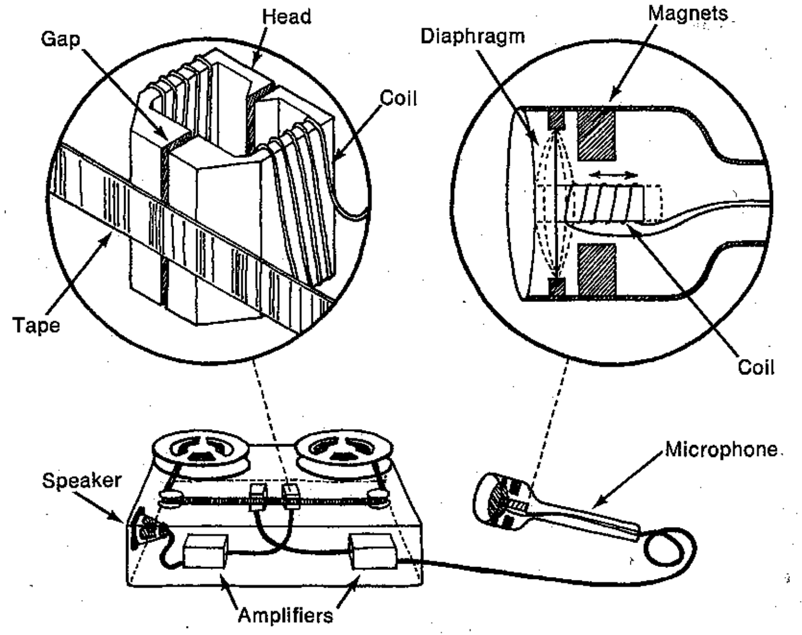


Fig. 10-2. The magnetic recording process. The playback head of the tape recorder is shown in a cutaway view (the width of the gap is exaggerated for clarity). The microphone cutaway shows the components of a dynamic microphone. (Carol Keller)

When you play back the tape, it is passed over the same head, or a similar playback head. Now the magnetic field stored in the iron particles creates an electric current in the wires coiled around the head. This signal is amplified and sent to a loudspeaker, which acts like a moving-coil microphone in reverse. Instead of the microphone's diaphragm, the speaker employs a paper cone that is connected to the coil. When current passes through the coil, it moves the cone, which in turn pushes the air to produce sound pressure waves. If you stand in front of a large bass speaker, you can both hear and *feel* the sound waves generated by the paper moving back and forth.

DIGITAL AUDIO RECORDING

Before reading further, see The Basic Idea, p. 208.

The way digital audio is recorded is similar in many respects to the analog audio process described above. First of all, microphones and speakers are analog, so sound is captured and reproduced using the same equipment regardless of the recording format (the same microphone could feed an analog or digital recorder). The difference is in the way digital recorders process and store the sound.

With analog recording, sound is converted to a voltage; the voltage is converted to a magnetic field, which is then stored on tape. In digital audio recording, we start the same way: Sound is converted to a voltage. Then the *analog-to-digital* (*A/D*) con-

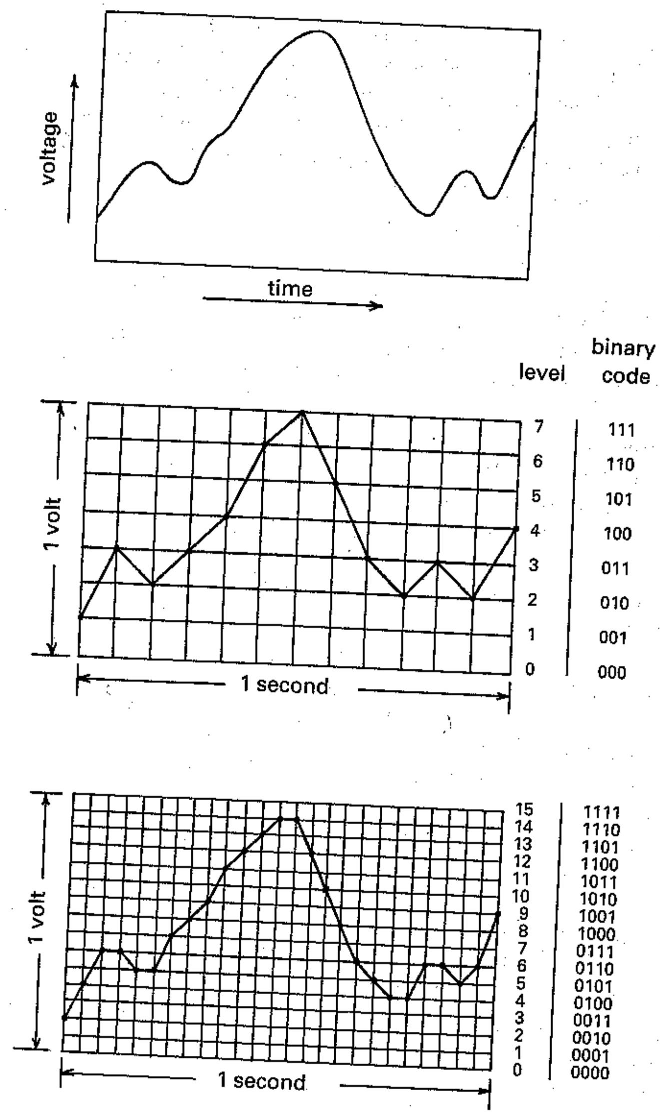


Fig. 10-3. Digital recording. (top) The original analog signal. Note that the voltage level changes continuously over time. (middle) To record the signal digitally, we take *samples* (measurements) at regular time intervals (the vertical lines). The level of the signal at each sample is measured according to a level scale (the horizontal lines). The total number of units in the scale is determined by the *number of bits* in the system; a three-bit system is pictured here. Samples can only be measured in *one-unit increments*. A measurement that falls between, say, level two and level three must be rounded down to two or up to three. (bottom) By taking more samples in the same period of time (higher *sample rate*) and using more bits per sample, we can make a higher-resolution recording. Pictured here is a four-bit system with twice the sampling rate as the middle graph. Now the scale has finer gradations, allowing us to measure the signal more precisely. Note that this curve better approximates the shape of the original analog signal. (Robert Brun)

verter processes the sound by repeatedly measuring the voltage level (*sampling* it) and converting those measurements to numbers (*quantizing*). This two-step process is the heart of digital recording. The quality of the recording depends how *often* we sample the voltage and how *accurately* we measure each sample.

Once we have the sound expressed as digital data we can then store it on tape, hard drive, or in other media. The basic concepts of sampling and quantizing are quite similar for both video and audio recording. If you understand one, it can help you make sense of the other.

Sample Rate

The first part of the digitizing process is to take a series of samples or measurements of the sound level. Take a look at Fig. 14-21, which shows an audio waveform (a visual representation of a sound). You can see that it's constantly changing (oscillating). High-frequency signals change very fast and low-frequency sounds change more slowly. To accurately measure the level of high-frequency sounds we need to take samples more frequently than for low sounds.

As a simplified example, imagine you're making popcorn and want to count how fast the kernels are popping. When you start heating the popcorn, kernels pop only every few seconds, so you can take your time counting them (low frequency). But as the corn heats up, many kernels start popping every second, so you have to count much faster to get an accurate count.

High frequencies are of particular concern because without them, a recording may have poor quality and sound muddy or dull.[1] A Swede named Henry Nyquist proved that the sampling rate has to be at least *twice* the maximum frequency we hope to capture. Because humans can perceive sounds up to about 20,000 Hz (20 kHz), a digital audio recorder needs to sample at least 40,000 times a second (40 kHz) to capture that range of frequencies.

Different digital audio recorders use different sample rates. Too low a sample rate results in *aliasing*, with poor high-frequency reproduction. The higher the sample rate, the better the frequency response and quality. Increasing the sample rate also increases the amount of data that needs to be stored.

CDs have a sample rate of 44.1 kHz (this rate is sometimes referred to as "CD quality" and is often used for music). Many video cameras and recorders use 48 kHz, which is a standard professional sample rate. Very high-quality recorders used for high-end production and music recording may operate at 96 kHz or even 192 kHz. Some DV cameras can be operated at 32 kHz—a low sample rate that compromises quality.

Bit Depth or Precision

Sample rate is an expression of how often we measure the audio signal. *Bit depth* or *precision* refers to how accurately we measure each sample.[2]

To get the idea of bit depth, consider this simple example. Say you had to measure people's height with a stick. The stick is one foot long and you can only record

1. If a video recording lacks high frequencies, fine detail in the picture may be lost, making it appear unsharp.

2. Bit depth is also called *bit length* and *word length*.

the height in one-stick increments. So, you could measure a six-foot-tall man very accurately (six sticks). But when you measure a woman who's five feet, six inches tall, you either have to record her height as five sticks or six sticks—either way, you're off by half a foot.

Now, imagine that we do the same thing with a shorter stick that's only six inches long. We can still measure the man's height precisely (twelve sticks). And when we measure the woman, now we can be just as accurate (eleven sticks).

Digital systems use a measurement scale to record the voltage of each audio sample. The scale has a number of levels. In an 8-bit system there are 256 levels.[3] Each level is the equivalent to our "sticks." The quietest sound could be given the level 1 and the loudest would be given level 255. But there's no such thing as a fraction of a level. If the signal level fell midway between 125 and 126, it would be rounded down to 125 or up to 126. Either way, that would introduce an error that could degrade the sound. In the digital audio world, that's called a *quantizing error*; it's a form of noise.

For greater precision we could use more bits. In a 16-bit system (which is fairly typical in professional video cameras) there are 65,536 levels. Now we can measure different voltage levels much more accurately and reproduce sounds more precisely. The more bits used for each sample, the higher the quality and the lower the noise. However, increasing bit depth, like raising the sample rate, increases the amount of audio data to be processed and stored.

Some high-quality recording systems use 20 bits or 24 bits or more. Even though you might not be able to hear the difference between 16 bits and 24 bits, when digital audio gets processed during postproduction, errors get multiplied, so more precision keeps the sound cleaner in the end.[4] Often, 16-bit recordings are converted to 24 bits for mixing.

Keep in mind that using more bits doesn't mean recording *louder* sounds. As you can see in Fig. 10-3, the one-volt maximum signal is divided into eight levels in the middle graph and sixteen levels in the lower graph. The maximum level is the same in both. However, an interesting thing happens at the *bottom* of the scale where the very quietest sounds are recorded. Sounds that are lower than the first level will disappear entirely from the recording (they will be recorded as zero). This is the *noise floor*. But if we use more bits, the first level is now lower, and we may be able to catch quiet sounds that would have been too low to register before. This reduces noise and increases dynamic range.[5]

Resolution and Sound Quality

Together, the sample rate and bit depth contribute to the *resolution* of a digital audio recording. Low-resolution recordings may sound "gritty" or overly crisp ("cold"). When the recording can't capture the subtleties of the original sound, artifacts result, which may be disturbing to the ear. High-resolution recordings

3. Digital systems use binary (base 2) numbers, and each digit is a *bit*. Eight bits is 2^8, which is 256.

4. When the sound level is very low—near the noise floor—there is a noticeable difference between 16-bit and 24-bit recording.

5. Systems that use *dither* actually allow you to hear sounds *below* the lowest level. Dither is a special form of low-level random noise. Generally, dither should be used when converting from a high bit depth to a lower one (such as converting 24-bit to 16-bit).

sound more faithful to the original sound source in terms of frequency response, dynamic range, and lack of noise. In the 1950s, vinyl LP records were considered "hi-fi" (high fidelity). Today, our standards for fidelity are a whole lot higher.

Recording and Transmitting Digital Audio

Once audio has been digitized, we can record it on hard drives, memory cards, or tape. With digital audio or video, it's very easy to move data around because you only need ones and zeros to represent any value.[6] A number can be transmitted from one place to another by sending a series of electrical pulses: send an *on* pulse for the ones, send an *off* pulse for the zeros.

This is why digital recordings on tape are so much more "robust" than analog recordings. To record an analog signal, you have to capture and store in the tape *tiny* variations in magnetic levels. With digital, the signal has only two possible levels: completely on or completely off (representing ones or zeros). The tape has a much simpler and cruder job: record either "full on" or "full off."

Tape noise is a problem with analog recordings because very quiet sounds may be lost in the noise of the tape. Digital systems generally don't have a problem with tape noise because even quiet sounds are recorded with pulses that are either full on or full off, so there's no confusion about what is signal and what is tape noise (however, as we've seen, there are still other types of noise in digital).

Digital tape recorders use rotating heads as in a video camcorder (see Fig. 3-8) instead of the fixed (stationary) heads used in analog tape recorders.

including chromium dioxide, or CrO_2, and "metal" formulations may offer lower noise and better frequency response. Special tape formulations are more often used for analog cassette recorders than for ¼-inch recording in which tape noise is less of a problem. With ¼-inch recording today, the choices in tape stock are very limited.

A common analog tape problem is *print-through*, which occurs when the sound on one layer of tape becomes imprinted on the next layer out on the reel, causing a slight echo that can be heard during quiet moments before or after loud sounds. Print-through is mostly noticeable on material recorded at quiet locations where there is no background sound to mask the echo. It can be minimized by storing the tape tail-out on the reel, by keeping the tape in cool locations, and by rewinding it every now and then.

Noise Reduction

Many analog tape recorders and video camcorders are equipped with *noise reduction* units such as one of the Dolby systems. There have been several different versions of Dolby noise reduction, including A, B, C, and SR (Spectral Recording). Dolby C works by boosting the level of mid- and high-frequency sounds as they are recorded and then diminishing their level in playback, leaving the sound signal normally balanced. Tape hiss is also a high-frequency sound; however, since it is inherent in the tape, it is not affected by the Dolby during the recording. It is diminished, though, when the Dolby reduces the high frequencies in playback.

If a recorder has a noise reduction unit, you should generally use it. Tapes recorded with noise reduction *must* be played back with it, on a machine that employs the same system. Always indicate on the tape box whether a recording was made with noise reduction and which system. Tapes made without noise reduction should not be played back with it or the sound quality will be degraded (for example, with Dolby C, high frequencies may be lost).

There are several Dolby multichannel and digital audio systems (see Chapter 16).

THE MICROPHONE

Microphone Types

There are a few basic types of microphones used for film and video production. *Condenser microphones* are used extensively. They are often quite sensitive and some are expensive. Condenser mics use a capacitor circuit to generate electricity from sound, and they need power supplied to them to work. Power may come from batteries in the microphone case, on the mic cable, or in the recorder itself. *Electret condenser* mics employ a permanently charged electret capacitor. They can be made very cheaply and may require no power supply.

Dynamic or *moving-coil microphones* are typically used by musical performers, amateur recordists, and many professionals. They are simpler and less sensitive than condensers, but usually quite rugged and resistant to handling noise and they require no batteries or special power supply.

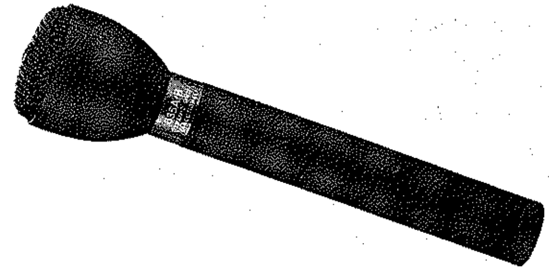


Fig. 10-16. Electro-Voice 635 omnidirectional dynamic microphone. Simple and durable. (Electro-Voice)

Directionality

Every mic has a particular *pickup pattern*—that is, the configuration of directions in space in which it is sensitive to sound.

Omnidirectional or *omni* microphones respond equally to sounds coming from any direction.

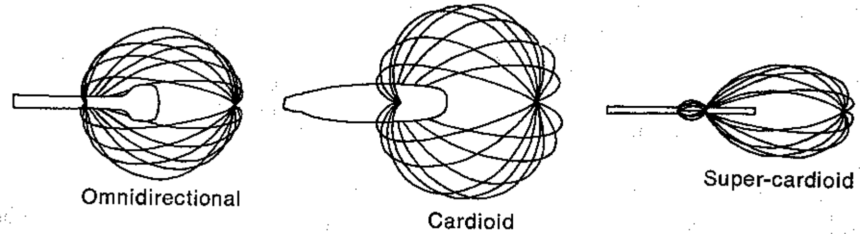


Fig. 10-17. Representations of the directional sensitivity of omni, cardioid and super-cardioid microphones (not drawn to the same scale). These indicate each mic's response to sound coming from different directions. Imagine the omni mic at the center of a spherical area of sensitivity; the diaphragm of the cardioid mic is at about the position of the stem in a pattern that is roughly tomato-shaped. Though the lobes of sensitivity are pictured with a definite border, in fact sensitivity diminishes gradually with distance. (Carol Keller)

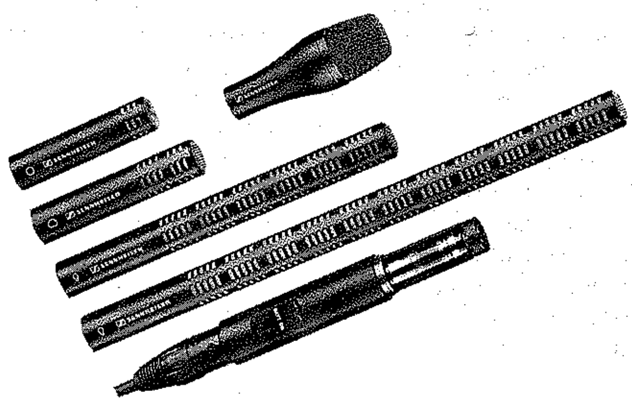


Fig. 10-18. Sennheiser K6 modular condenser microphone system. The powering module is at bottom (works with internal batteries or phantom power). Interchangeable mic heads range in directionality from super-cardioid to omni. (Sennheiser Electronic Corp.)

Cardioid mics are most sensitive to sounds coming from the front, less sensitive to sounds coming from the side, and least sensitive to those coming from behind. The name derives from the pickup pattern, which is heart-shaped when viewed from above. *Hyper-cardioid* microphones (sometimes called *in-line*, *short-tube shotgun*, or *mini shotgun*) are even less sensitive to sounds coming from the side and behind. *Super-cardioid* microphones (*long-tube shotgun* or *shotgun*) are extremely insensitive to any sounds not coming from directly ahead. However, some hyper- and super-cardioid mics have a certain amount of sensitivity to sound emanating from directly behind as well. Because the names for these microphone types are not entirely standardized (one company's "hyper-cardioid" is another's "super-cardioid"), be careful when you select a microphone.

Bidirectional mics have a figure-eight pickup pattern with equal sensitivity on either side; these mics may be used in a studio placed between two people talking to each other.

Boundary microphones (sometimes called *PZM* or *Pressure Zone Microphones*) are mounted very close to a flat plate or other flat surface and may have a hemispherical pickup pattern. These are sometimes used for recording a group of people when the mic can't be close to each speaker, or when recording music (mounted on a piano, for instance).

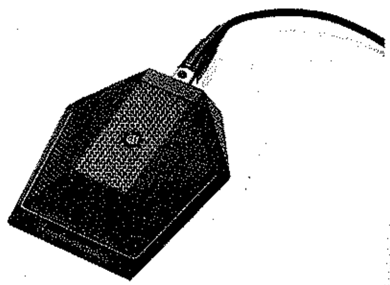


Fig. 10-19. Mounted on a flat surface, a boundary mic uses the surface to boost its audio response. (Audio-Technica, U.S., Inc.)

Manufacturers print *polar diagrams*—graphs that indicate exactly where a microphone is sensitive and in which directions it favors certain frequencies. It's important to know the pickup pattern of the mic you are using. For example, many people are unaware of the rear lobe of sensitivity in some hyper- and super-cardioid mics, which results in unnecessarily noisy recordings (see Fig. 10-20).

Hyper- and super-cardioid microphones achieve their directionality by means of an interference tube. The tube works by making sound waves coming from the sides or back of the mic strike the front and back of the diaphragm simultaneously so that they cancel themselves out. In general, the longer the tube is, the more directional the mic will be. For proper operation, don't cover the holes in the tube with your hand or tape. Usually, the more directional a microphone is, the more sensitive it will be to wind noise (see Windscreens and Microphone Mounts, p. 395).

Contrary to popular belief, most hyper- and super-cardioid mics are not more sensitive than cardioid mics to sounds coming from directly ahead; they are *not* like zoom lenses; they don't "magnify" sound.[10] However, directional mics do exclude more of the competing background sound, so that they can produce a good recording at a greater distance from the sound source—as recordists say, the "working distance" is greater.

10. *Parabolic mics*, which look like small satellite dishes (sometimes used at sporting events), actually *do* have a magnifying (amplifying) effect.

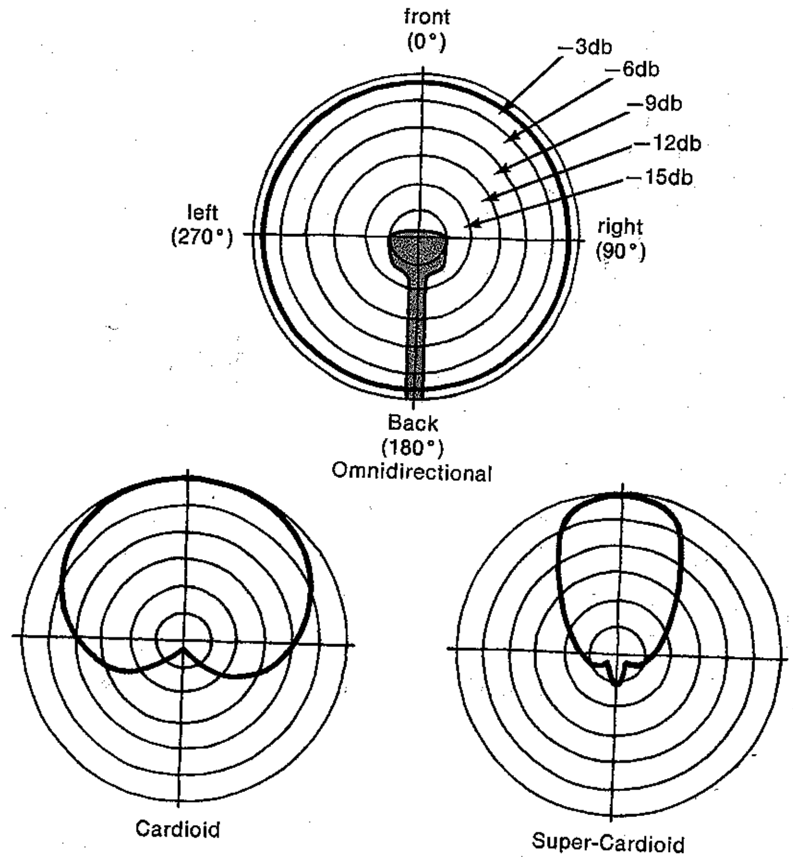


Fig. 10-20. Polar diagrams indicating the sensitivity of omnidirectional, cardioid, and super-cardioid microphones. Imagine each diagram as a cross section of the mic's sensitivity, with the microphone lying along the vertical axis (as the omni mic is here). The microphone's diaphragm would be positioned at the center of the graph. (Carol Keller)

One disadvantage of highly directional mics is that you may encounter situations in which it's hard to capture important sounds within the narrow lobe of sensitivity. A classic case is trying to record a two-person conversation with a super-cardioid mic: When the mic is pointed at one person, who is then *on-axis*, the other person will be *off-axis*, his voice sounding muffled and distant. Panning a long microphone back and forth is an imperfect solution if the conversation is unpredictable. In such cases, it may be better to move far enough away so that both speakers are approximately on-axis. Unfortunately, the best recordings are made when the microphone is close to the sound source.

Microphone Sound Quality

Microphones vary in their frequency response. Some mics emphasize the bass or low frequencies, others the treble or high frequencies.

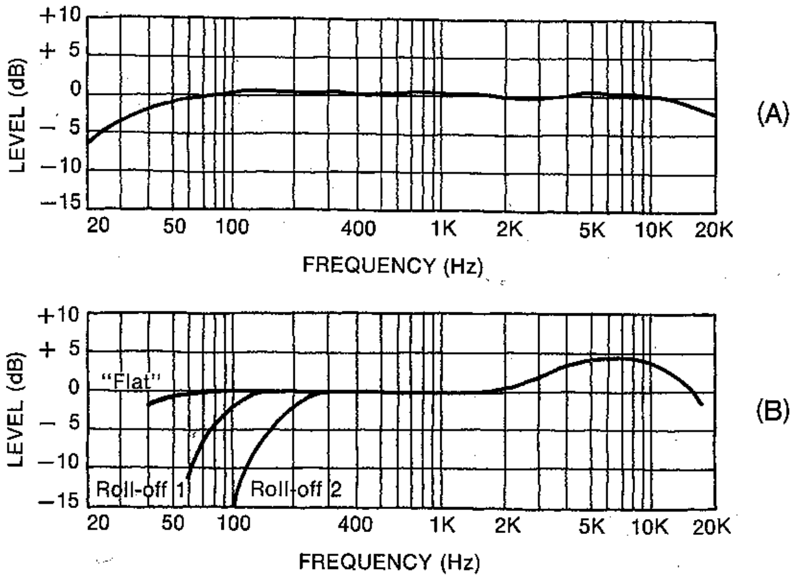


Fig. 10-21. (A) The relatively flat frequency response of a good-quality audio recorder. Where the graph drops below the 0 dB line indicates diminished response. (B) A microphone frequency response curve. This mic is more sensitive to high frequencies. The three parts of the curve at left represent increasing amounts of bass roll-off controlled by a built-in, three-position switch (see p. 429). (Carol Keller)

The frequency response of a microphone or recorder is shown on a *frequency response graph* that indicates which frequencies are favored by the equipment. Favored frequencies are those that are reproduced louder than others. An "ideal" frequency response curve for a recorder is flat, indicating that all frequencies are treated equally. Many mics emphasize high-frequency sounds more than midrange or bass frequencies. Recordists may choose mics that favor middle to high frequencies to add clarity and *presence* (the sensation of being close to the sound source) to speech. Some mics have a "speech" switch that increases the midrange ("speech bump"). In some situations, a mic that emphasizes lower frequencies may be preferred. For example, a male vocalist or narrator might like the sound coloration of a bassy mic to bring out a fuller sound. A large-diaphragm mic, like the Neumann U-89, can be used for a "warm" sound.

Sometimes recordists deliberately *roll-off* (suppress) low frequencies, especially in windy situations (see Chapter 11).

When you purchase a mic, check the frequency response graph published by the manufacturer. An extremely uneven or limited response (high frequencies should not drop off significantly before about 10,000 Hz or more) is some cause for concern.

One of the things that distinguishes high-quality mics is low "self-noise"—the mic itself is quiet and doesn't add hiss to the recording. The better the recorder, the more likely you are to notice mic noise.

The microphones that come with recorders and cameras are often not great and may need to be replaced. Set up an *A/B test* where you can switch from one mic to another while recording. However, you may find that you prefer the sound of the less expensive of two mics. An A/B test is especially important if you need two matched microphones for multiple mic recording (see Chapter 11).

Windscreens and Microphone Mounts

The sound of the wind blowing across a microphone does not in the least resemble the gentle rustle of wind through trees or the moan of wind blowing by a house. What you hear instead are pops, rumble, and crackle. When recording, don't let wind strike a microphone (particularly highly directional mics) without a *windscreen.* A windscreen blocks air from moving across the mic.

A minimal windscreen is a hollowed-out ball or tube of *acoustifoam*—a foam rubber–like material that does not muffle sound (see Fig. 9-17). This kind of windscreen is the least obtrusive and is used indoors and sometimes in very light winds outside. Its main use is to block the wind produced when the mic is in motion and to minimize the popping sound caused by someone's breathing into the mic when speaking.

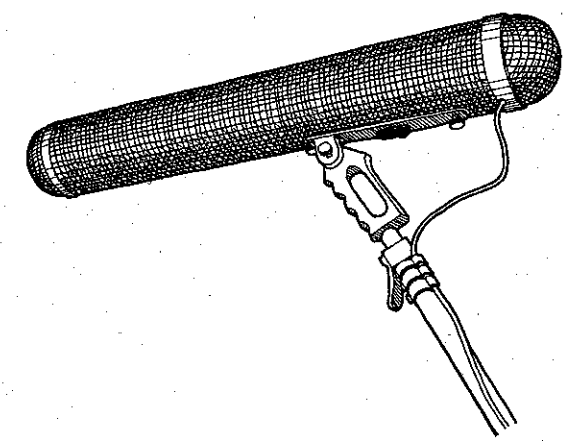


Fig. 10-22. A zeppelin windscreen for a shotgun mic, shown with a pistol grip and mounted on a microphone boom. (Carol Keller)

For breezier conditions, a more substantial windscreen is needed. Many recordists carry a soft, fuzzy windscreen with built-in microphone mount, such as the Rycote Softie (see Fig. 10-23). A Softie can also be used to cover a camera mic. For heavier wind, a windscreen called a *zeppelin* can be used.[11] Like its namesake, this is large and tubular; it completely encases the mic. In strong winds, an additional sock-like, fuzzy covering can be fitted around the zeppelin. A good windscreen should have no noticeable effect on the sound quality in still air.

When you are caught outside without an adequate windscreen, you can often use your body, the flap of your coat, or a building to shelter the mic from the wind. Hide a lavalier under clothing or put the tip of a wool glove over it (see below). Often, a bass roll-off filter helps minimize the rumble of wind noise (see Chapter 11). Omni mics may be the least susceptible to wind noise.

Besides wind noise, microphones are extremely sensitive to the sound of any

11. In the U.K., windscreens are called "wind shields," and zeppelins may be called "blimps."

Fig. 10-23. (left) Rycote rubber shock mount and pistol grip. (right) Rycote Softie windscreen works well in windy conditions. (Rycote Microphone Windshields, Ltd)

moving object, such as hands or clothing, which touches or vibrates the microphone case. *Hand noise*, or *case noise*, becomes highly amplified and can easily ruin a recording with its rumbly sound. The recordist should grip the microphone firmly and motionlessly, grasping the looped microphone cable in the same hand to prevent any movement of the cable where it plugs into the mic. Even better, use a pistol grip that has a shock mount (usually some form of elastic or flexible mounting) to isolate the mic from hand noise (see Fig. 10-23).

In many recording situations, a *fishpole* (collapsible) boom should be used to enable the recordist to stand away from the action (see Fig. 9-13). A shock mount will isolate the mic from hand or cable noise on the boom.

Lavalier or Lapel Microphones

Lavalier microphones (*lavs*) are very small mics generally intended to be clipped on the subject's clothing.[12] Also called *lapel mics*, these are often used with a wireless transmitter (see below) or may be connected by cable to the recorder. Most TV news anchors wear one or two lavs clipped on a tie or a blouse. Lavaliers are quite unobtrusive and are easy to use when there isn't a sound recordist. Lavs are often used for interviews because they can result in clear, loud voice tracks. They're useful for recording in noisy environments because they're usually positioned so close to the person speaking that they tend to exclude background sound and the reverberation of the room. However, this also results in a "close" sound, which sometimes sounds unnatural. Many professional recordists don't like using lavs and prefer the sound from a good cardioid or hyper-cardiod mic on a boom because it's more natural, more "open," and doesn't risk clothing or body noises. In some situations, recordists like to record with a lavalier on one track and a boom mic on another, to get both good voice and some background sound; avoid mixing both mics on the same track in the field.

12. Lavalier used to refer to a larger type of mic designed to be hung on a cord around the subject's neck. These are rarely used today.

Most lavaliers are omnidirectional, although hypercardioid lavs are available. Many have a flat frequency response. However, when you clip a mic on someone's shirt you may get too much bass (from being right over the chest cavity) and not enough treble (since the mic is out of line with the speaker's mouth). Some lavs have a midrange speech bump (see Microphone Sound Quality, p. 393) to compensate.

A good position for a lavalier is in the middle of the chest at the sternum (breastbone). For subjects wearing a T-shirt or a sweater, sometimes the mic is clipped on at the collar. The problem with collar placement is it may be too close to the subject's voice box and may cause sound variations if the subject turns his head away from the mic. If the subject is looking generally in one direction (perhaps for an interview), put the mic on that side of his collar.

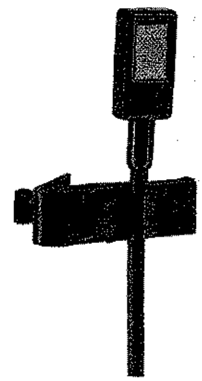


Fig. 10-24. Lavalier mic. The Countryman Isomax omnidirectional lavalier is available in a choice of three frequency responses in either matte black or white. (Fletcher Chicago/Countryman)

Often it is preferable to hide the mic; for shooting dramatic material, it's essential. Clip or tape the mic under clothing, but listen carefully for case noise caused by the cloth rubbing on the mic. Silk and synthetic fabrics are the worst for noise; cotton and wool are often fine. Small frames or cages are available to provide separation between the cloth and the mic. You can improvise with some rolled-up tape to prevent rubbing (see Fig. 10-25). Leave enough slack in the cable so that the body movements don't pull on the mic; make a small loop in the cable for strain relief and tape or clip it in place.

Sometimes you can get better sound by hiding a lav in the subject's hair or a hat. Carry some moleskin or surgical tape for taping mics to skin.

For more on using lavaliers, see Chapter 11.

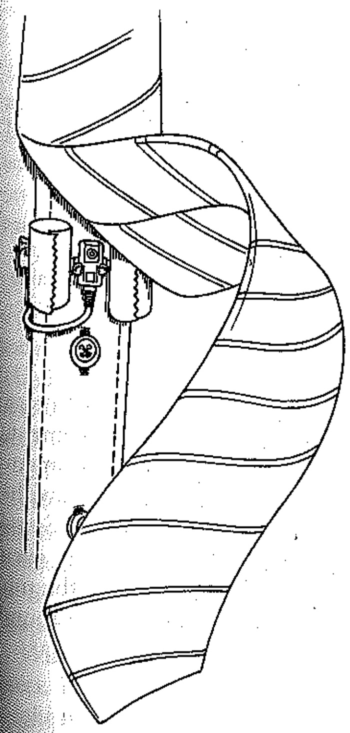


Fig. 10-25. Hiding a lavalier. To prevent cloth from rubbing on the mic, one technique is to use rolled-up gaffer's tape. Some recordists like to hide the mic under a tie's knot. Most recordists prefer not to place mics under clothing whenever possible. (Robert Brun)

Wireless (Radio) Microphones

To allow the camera and subject greater freedom of movement, a *wireless*, or *radio*, *microphone* can be used. A wireless isn't really a mic at all, just a radio transmitter and receiver. In a typical film or video shoot, a lavalier mic is clipped on the subject, plugged into the concealed transmitter that's about the size of a pack of cards. A receiver mounted on the recorder or camera picks up the signal with a short antenna. Wireless sound quality may not be as high as with *hard-wired mics* (mics connected by cables), but some wireless systems are excellent and are used regularly in professional productions. High-end systems can cost thousands, although much more affordable wireless systems costing a few hundred dollars are surprisingly good. Some systems are digital, others analog, and some combine both technologies.

Fig. 10-26. Wireless mic. Sennheiser Evolution system with bodypack transmitter, receiver, lavalier mic, XLR cable, and plug-on transmitter for handheld mics. This affordable UHF wireless system offers a wide choice of frequency bands so you can avoid radio interference. (Sennheiser Electronic Corp.)

Using a wireless opens up many possibilities for both fiction and documentary shooting. You never need to compromise camera angles for good mic placement, since the mic is always close to the subject but out of view. In unscripted documentaries, there are great advantages to letting the subject move independently, without being constantly followed by a recordist wielding a long microphone. Some people feel uncomfortable wearing a wireless, knowing that whatever they say, even in another room, can be heard. As a courtesy, show the person wearing the mic where the off switch is. Some recordists object to the way radio mics affect sound

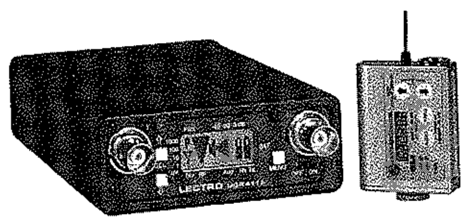


Fig. 10-27. Lectrosonics wireless receiver with super-miniature transmitter. Lectro makes some systems for the DV market, but many are very high-quality, expensive units for pros. (Lectrosonics, Inc.)

perspective: Unlike typical sound recording, when the subject turns or walks away from the camera wearing a wireless, the sound does not change.

A wireless transmitter and receiver can be used for many purposes: to connect a handheld mic, a standard boom mic, or a mic mixer to the camera or recorder; or to transmit timecode or a headphone feed on the set.

Wireless transmission is not completely reliable. Depending on the physical obstructions and competing radio transmissions in the area, wireless signals may carry up to several hundred feet or they may be blocked altogether. Many newer wireless systems operate in the UHF bands, which may have less interference than VHF bands. It's important to find a transmitting frequency that isn't being used by a local television station or taxi company (unless you want the sound of a cabbie in your movie). Newer wireless systems usually offer a choice of frequencies (and are called "frequency agile"). Talk to a recordist or dealer to find the frequencies that are likely to be interference-free in the area where you're shooting; lists of open frequencies should be available on the Web. Some radio mics are sold with a particular subset of frequencies; get the band that works best in your area.

When there's interference or loss of signal, you can get a loud noise on the track. Some systems broadcast on more than one frequency simultaneously to avoid breakup. "Diversity" radio mics use multiple antennas for the same reason. Inexpensive consumer wireless systems can get interference from many household sources. Avoid electric motors, computer monitors, and other electronic interference. Make sure the *squelch* control is properly adjusted to prevent noise when the radio signal is lost.

Always position the receiving antenna as close to the transmitter as possible. If the signal breaks up, experiment with different antenna positions. Make sure the transmitter antenna is straight. Some systems use very short stub antennas and some use longer ones. Run a long one around the belt line or up the subject's back (clip it on clothing near the shoulder and hide it under a coat or sweater; use a rubber band between the clip and the antenna for strain relief). Antennas sometimes work best if dangled away from the body.

Check and/or replace transmitter and receiver batteries every few hours.

Most professional wireless transmitters use a limiter (see p. 423) to prevent excess volume levels. Many models have a level adjustment and some have a light to indicate excess volume. With the subject speaking normally, turn the level up until the light flashes often, then turn it down a bit (for more on level adjustments, see Chapter 11).

Wireless receivers can be mounted directly on a camera with various brackets or plates. For handheld work with a small camera, this may increase the camera's weight noticeably, especially when more than one receiver is used.[13] You can also put receivers on your belt or in a shoulder bag with a wire to the camera (see Fig. 11-1). It's important to match the output level of the receiver to the audio input on the camera or recorder. Some receivers work at line level, others at mic level, and some are switchable. For more on this, see Mic and Line Level, p. 401, and Gain Structure, p. 423.

13. Zaxcom makes a wireless system that can transmit two audio channels to one receiver, lightening the load a bit.

CHAPTER 11

Sound Recording Techniques

This chapter is about methods of audio recording for video and film. See Chapter 10 for discussion of audio recording equipment.

PREPARING FOR A SHOOT

GATHERING GEAR

Lists of useful audio equipment for a video or film shoot appear on p. 322. Here are some considerations in choosing audio equipment.

Recorders and Formats

When shooting video, the choice of camcorder or VTR format is usually driven by picture needs, but it clearly has an impact on sound as well. In selecting a system, you must consider: How many audio tracks do you need? Does the camcorder allow manual adjustment of audio levels? Can you use external mics? Are there professional mic connectors (such as XLR) or will you need adaptors? If you're working with a sound recordist, will you need a mic mixer so he or she can control levels? Does the camera record audio in a highly compressed format or with a low sample rate that might compromise quality?

For film or double-system video shoots, the recordist has more freedom in choosing a recorder. This decision should be made on the basis of what features are needed, the cost, and how postproduction will be done. You must choose between digital and analog, with timecode or without. Many of the questions mentioned above apply here as well. How many tracks do you need? Can you easily input professional-type mics? If recording digitally, what format will you use? How will files be exported from the recorder for archive and postproduction? Do you have enough storage media? How will the deck be powered in the field? Do you need a shoulder case? Are the controls easily readable in daylight? Sometimes when a recorder is too small, or has badly arranged controls, it may be hard to use in high-pressure situations.

Spare batteries, fuses, and an AC power connection for the recorder should be kept on hand.

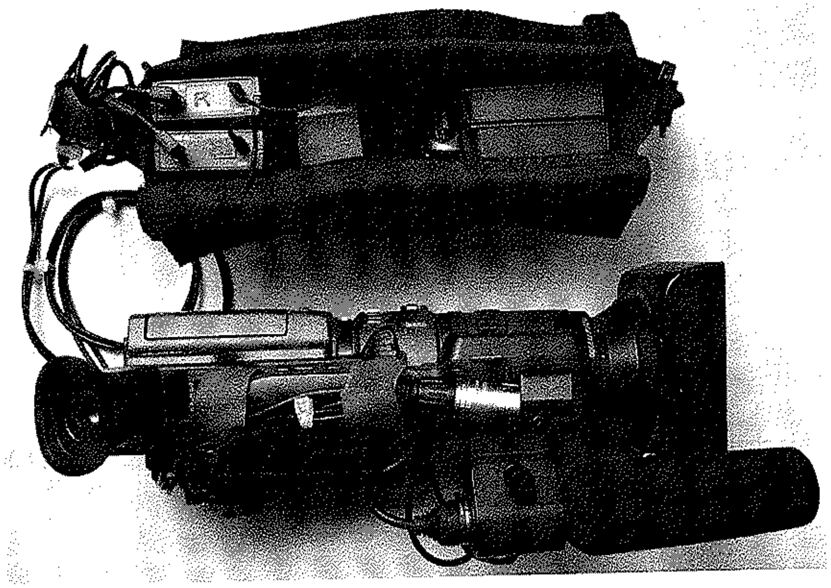


Fig. 11-1. A shooting kit. Portabrace makes a small shoulder pouch that's just the right size for spare tapes and batteries. Shown here, two wireless receivers are connected by cables to the camera, but sit in the pouch to keep the weight off the camera. A hard disk drive could be carried in a similar way.

Microphones

Unless you can carry several mics, your choice of a primary microphone is very important. Many recordists prefer a somewhat directional microphone (a hyper-cardioid, like the Sennheiser ME-66), which can exclude some background sound but is excellent for general uses. This type of mic is good for unpredictable documentary or dramatic scenes. Some recordists prefer a super-cardioid (long shotgun) mic, like the Sennheiser MKH-70. This mic allows you to stand farther back from the subject and is good for isolating voices in noisy environments, such as a train station or a parade. Some long shotgun mics are very big and awkward. In documentary work, subjects can find them intimidating. Also, a long shotgun is often too directional for recording in tight quarters (see Directionality, p. 391).

Modular microphone systems allow you to use one power supply with several heads of varying directionality (omni, cardioid and hyper-cardioid; see Fig. 10-17). This provides a great deal of flexibility. There are modular systems at both ends of the price range.

Always try to have at least one backup microphone and cable as insurance. If the second microphone is more directional or less directional than your primary mic, you'll have more flexibility. Lavalier mics (see p. 396) are very small, pack easily, and can be used for general recording in some situations.

In many shooting situations, a microphone boom is essential to allow the mic to be positioned close to the sound source while keeping the person holding the mic out of the shot. Typically, recordists carry extendible booms that can be adjusted for

each shot. Studio mic booms are mounted on a pedestal to relieve the boom person of the considerable fatigue that results from holding a boom all day. For smaller, mobile crews, borrow a stand from a lighting kit and get a bracket so you can mount the boom on it; this is particularly useful for situations such as sit-down interviews or scenes in a drama in which actors aren't moving.

In some documentary scenes, the boom may be too big or intrusive. The mic can be handheld (with or without a pistol grip shock mount) for more intimate situations. A short table stand for the mic can be handy at times.

Headphones

The choice of headphones is also important. For controlled shooting situations, it makes sense to get headphones that have good fidelity and closed ear pads that fit around the ear. These block any sounds coming directly to the recordist without having gone through the microphone first, so you can be sure of the recording without being misled by other sounds around you. Open headphones are lightweight and more comfortable. They usually rest on top of the ear (*supra-aural*). With these, you may not hear some defects that will be apparent on a better sound system. Sometimes sound leaking out of these headphones may be picked up by the mic.

One problem with headphones that fit around the ear for unstaged documentary filming, especially when a directional microphone is used, is that the recordist

Fig. 11-2. (upper left) Headphones with closed earpads. (upper right) Headphones with open earpads. (bottom) Mono earpiece shown with mono connector; stereo-to-mono adaptor (so you can hear both channels from the camcorder); and right-angle adaptor (so the plug doesn't stick out from the camcorder).

can hear only sounds he or she expects to hear; if someone speaks outside the mic's range of sensitivity, the recordist will not hear or react to it. Some documentary recordists just put the headphone over one ear so they can better relate to people and react to events.

Some camcorders have a small speaker at the operator's ear. All cameras can be used with a wired earpiece that goes in one ear. These allow you to respond to what's going on around you and are not obtrusive. They aren't great for close monitoring of the sound (it's easy to be fooled by what you hear in the *other* ear that's hearing the scene without the earpiece) but you'll hear major problems like breakup or loss of audio.

Stereo headphones should be used with a mono adaptor when you are recording only one channel of audio so that you can hear it in both ears. Some headphone outputs can be switched to mono for that purpose. A mono earpiece may need a stereo adaptor if you're recording stereo so you can hear both channels.

Other Equipment

A typical professional video package includes a field mixer (see p. 400). Get a *breakaway cable* with a *headphone return* line in it. This allows the recordist to monitor the sound coming from the camcorder (as a check that the signal is really there), and breaks apart easily so the camera and sound recordist can separate quickly.

One or more wireless microphone systems (see p. 397) will greatly increase your mobility and flexibility for both film and video shoots.

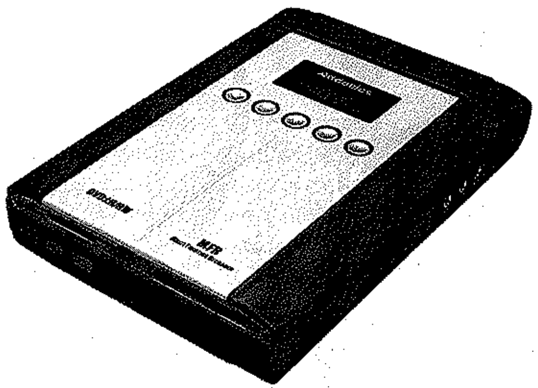


Fig. 11-3. External DVD burner. Addonics model can transfer data stored in ten different types of Flash media directly to DVD or CD, or record video to disc via a USB port. (Addonics Technologies, Inc.)

Recordist's Tools and Supplies

Having a few tools at hand can mean the difference between easily finishing a shoot and canceling it. Many repairs are very simple. After phoning and consulting a technician, even inexperienced persons can often make adjustments or isolate what needs to be repaired or replaced.

For any system:
Permanent felt-tip marker (e.g., Sharpies)
Small roll of gaffer's tape
Spare batteries for recorder and mic
Sound report sheets or log

For video recorder or DAT:
Head-cleaning cassette

For digital recorder:
Flash memory cards
CDs or DVDs for backup
External FireWire drive

For open-reel or analog recorders:
Head-cleaning fluid or isopropyl (rubbing) alcohol
Cotton swabs or head-cleaning sticks
Spare take-up reel and reel retainer nut
Single-edge razor blades (for breaking tape)

For repairs:
Swiss Army knife with scissors
Screwdriver handle and detachable blades: two sets, medium and jeweler's size
Small needle-nose pliers with wire-cutting edge
Small volt/ohm meter ("multitester")
Battery-operated soldering iron, rosin-core solder
Short length of light wire
Fuses for the recorder or camera
Head demagnetizer (for analog tape heads)

Preparing the Recorder

Equipment should be checked thoroughly before using. This is especially important if it has been transported, used by someone else, or come from the rental house. School equipment is more likely to be malfunctioning than working properly. Many recordists check their equipment whenever they arrive at a new location for shooting. The preparation you do depends a lot on the particular technology you're using. If anything in this list is unfamiliar, look elsewhere in this chapter and in Chapter 10.

1. ***Clean the heads.*** This should be done as a matter of course for analog recorders (see p. 388); for camcorders and DAT machines, it may not be necessary unless there's evidence of a problem. For hard drive and memory card recorders, this does not apply.
2. ***Check the batteries.*** If possible this should be done with the machine rolling in "record" position to see how the batteries read under load. Many rechargeable batteries will read fairly high on the meter until they are ready to give out, then the voltage drops sharply, so be prepared if the reading seems at all low. See Batteries and Power Supplies, p. 130, for more on managing batteries.
3. ***Check the settings.*** There are numerous settings; here are a few to check: Is the limiter or AGC on or off? If recording digitally, what about sample rate,

bit depth, file formats? Scan through physical switches and menu selections. Check the manual if you're unsure about any settings.

4. ***Timecode and/or pilot.*** For a timecode-capable deck, make sure that the timecode can be properly recorded and played back. On analog sync recorders, check that the pilot signal is actually being recorded on tape.
5. ***Microphones.*** Do you have cables? Windscreen? Various clips for lavs? Are the recorder inputs set correctly for mic level (and phantom power if needed)? Wireless mics should be checked for clear radio channels (without interference) and fresh batteries.
6. ***Test the audio.*** Do a test recording, which you can erase when you begin recording for real. Check the meter. Make sure you can move the level control without causing static. If not, moving the control rapidly back and forth a few times can help. Set the headphone level adjustment, if there is one. Gently move the recorder, cable, and microphone while listening through headphones to be sure there are no loose connections. This should not produce noise or static. Play back the recording and listen carefully for any defects. If the sound is muddy on an analog deck, try cleaning the heads. If the sound doesn't improve, this may indicate that the heads need demagnetizing.

If you can't get the recorder to work properly, systematically isolate various components. Try a different mic or mic cable; plug the mic into a different input; make sure the recorder is not in "pause"; check the AC/battery power switch (if there is one); try running the recorder on AC power; try cleaning the battery contacts with an eraser or grit-free abrasive, and so on.

Before going out to shoot, coil up excess cabling and fasten with Velcro cable ties (see Fig. 11-4). Make the recorder and/or mixer package as neat and compact as possible. It's much easier to concentrate on recording if you can move without getting tangled up. For handheld work, carry the recorder on the side of you that allows easiest access to the controls and the meter, and put some padding under the strap to spread the load. Wear soft-soled shoes and clothing that doesn't rustle.

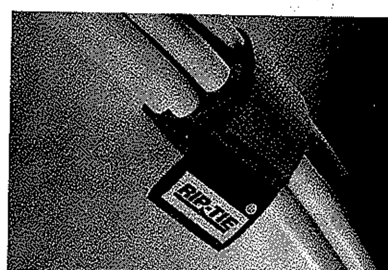


Fig. 11-4. Cable tie. This Velcro fastener can be used to keep coiled cables under control. (Fletcher Chicago/The Rip Tie Company)

Recordists on feature films often use a *sound cart* as a platform for mounting the recorder, mixer, and accessories that can be wheeled between setups (see Fig. 1-42).

THE SOUND RECORDIST'S ROLE

The sound recordist is responsible for placing the microphones (although someone else may hold them), operating the recorder or mixer, and making sure

that the quality of the recording is good. For staged (controlled) work, the recordist, who is sometimes called the *mixer*, can usually experiment with various mic positions and monitor the level of a rehearsal before shooting begins.

On video shoots, the recordist usually does not control the camcorder, but he or she must ensure that the sound actually recorded in the camera is acceptable. When possible, this means monitoring from the camera with headphones and/or doing periodic playback to check the recording.

For unstaged documentary shooting, the recordist should be alert and attentive to the action and not be glued to the meters. With time, your judgment will make you less dependent on the meter.

Recordists should be respectful of actors or subjects when placing body mics. Some recordists bring a newspaper to the set and pretty much tune out when they're not recording. Try not to "disappear" on set. Staying alert will help you and others do their jobs. On some sets, people in the sound department are treated like second-class citizens. The recordist may be forced to find mic positions only after all the lighting and blocking is done, making his job harder. He may also be blamed for airplanes flying over and the dog barking next door.

Communication

On all productions, crew members must communicate with each other. The camera operator must be able to signal to the recordist (or boom person) that he is in the frame, and the recordist must be able to indicate his need to change position. Sometimes in documentary work, only the recordist can hear whether a scene is worthy of being filmed. For all of these reasons, the filmmakers should have a set of signals with which to communicate silently. If these are hand signals, they should be sent with minimal commotion, so as not to disturb actors or documentary subjects. This requires that crew members watch each other as well as the action. The recordist, or whoever is operating the microphone, should not position himself on the camera operator's blind side (which is usually the right), and the camera operator should frequently open her other eye so that she can see the recordist while shooting. This is most important when filming improvised or unstaged action. After a crew works together for a while they begin to predict each other's needs, and eye contact precludes the need for hand signals.

If circumstances are such that a good-quality recording can't be made, the recordist should say so. If an airplane makes a take unusable, tell the director when the take is over (some directors want you to signal the problem during the take; smart directors will be looking at you as soon as they hear the plane). The director must decide whether to try to salvage a take with ADR or other postproduction cures (hence the solution for every conceivable production problem: "We'll fix it in the mix!").

Organization and Logging

It's generally part of the recordist's job to organize and label tapes or other recording media, to prepare the media to be recorded, and to keep a report or log of what's been recorded.

LABELING AND MANAGING MEDIA. Videotapes should be clearly labeled with the name of the production title, production company, tape number, date, frame rate, timecode type (drop or nondrop), timecode start, reference tone level (see below) and any special track assignments (such as LAVALIER CH. 1., BOOM MIC CH. 2).

Whenever a separate audio recorder is used, in addition to the above information, also indicate the camera roll or tape number, name of recordist and, when applicable, sample rate, bit depth, timecode frame rate (also indicating drop or nondrop), tape speed, and/or type of sync signal (for example, 60 Hz pilot). This information can additionally be spoken into the microphone and recorded.

When recording audio to file with a hard drive or Flash memory system, care must be taken in organizing file names, keeping track of drives, memory cards, or optical discs, making backups, and all the other details that will prevent the disaster of losing irreplaceable recordings due to carelessness or a corrupted file.

If recording to tape, develop a system to keep track of which tapes have been recorded and which are fresh (if you reuse tape, it is especially easy to get confused). Cassettes have little record-inhibit tabs that should be pushed in after recording to prevent accidental rerecording. Analog audio cassettes have small tabs on the edge opposite the one where you see the tape that can be punched to prevent accidental recording. These tabs can be plugged up or taped over later if you want to rerecord the tape.

When shooting film and recording on audiotape, whenever a camera roll is changed partway through the sound roll, an announcement should be made on the tape.

A reference tone should usually be recorded at the head of every videotape, audiotape, or other recording. This is discussed on p. 421.

For double-system recording, slates are generally used to sync sound and picture and to provide information on scene and take number. See Recording Double System for Film and Video, p. 434.

LOGGING. For a video shoot, someone may be assigned to keep logs of each take (see Keeping a Log, p. 363). On a double-system film or video shoot, the sound recordist often keeps a *sound report*, which is a written log of each take, noting the length, any problems, and whether the director considers the take good or bad (similar to the camera report in Fig. 9-31). On some productions, a *sound take number* is given each time you record audio (and is included on the slate). This identifies each piece of sync and wild sound. Since sound take numbers advance chronologically throughout the production (unlike scene numbers), in conjunction with the log, they aid in locating pieces of sound and picture. When recording to file using a hard drive or Flash memory recorder, the system may automatically assign a take or event number to the file name or user bits.

When recording to file, some systems can convert information in the BWF file including scene/take, time of recording, and other info to a text file such as an Avid Log Exchange file (ALE; see p. 705 for more). This can be used instead of a handwritten sound report, printed out, and/or imported into the editing system.

In unstaged documentary work, there's no time for meticulous log keeping. Instead, if a separate audio recorder is being used, the sound person should record a quick message after every shot or two, describing what was filmed, if there were problems with the slate, and so on; this can save a great deal of time in the editing room. It's a good idea to keep an informal log, listing the contents of each sound roll or group of files.

If you're recording to tape, record an announcement before sections of *wild sound* (sound recorded without picture); in some cases you may want to record wild sound to a separate tape, especially if a telecine transfer will be done. You may also want to make a verbal note when MOS shots (shots without sound) are filmed.

RECORDING TECHNIQUE

Basic Strategy

The general objective in sound recording is to place the microphone close enough to the sound source to produce a loud and clear sound track. A good track should be easily intelligible, should lack strongly competing background sounds, unpleasant echo, or distortion, and should be reasonably faithful to the tone quality of the original sound. Once a good recording is in hand, you have a great deal of freedom to alter the character of the sound later as you choose.

The ideal placement for many mics is between one and three feet in front of the person speaking, slightly above or below the level of the mouth. If the microphone is directly in line with the mouth, it may pick up popping sounds from the person breathing into it. A windscreen helps. If a directional mic is too close, it will bring out an unnatural bass tone quality. This is the *proximity effect*, which results from the particular way low frequencies interact with directional microphones. If a microphone is too far from the subject, background sound (*ambient sound*) often competes with or drowns out the speaker's voice. Also, undesirable acoustic qualities of the recording space, such as *echo* and *boominess*, become more noticeable (see Acoustics of the Recording Space, p. 425).

The microphone's position is almost always compromised by the camera's needs. It's important, however, that the sound source be solidly within the pickup pattern of the mic. Generally a boom mic will be brought in from just above the camera's frame, as close as possible without getting in the shot. The boom operator should practice during rehearsals. Be attentive to mic shadows—you may need to move away from a light to avoid them. Ask the camera operator for a frame line before the shot begins so you know how far in you can go.

Keep in mind that sound, like light, diminishes in intensity with the square of the distance (see Fig. 12-6). Thus, moving *twice* as far from the sound source diminishes the sound to *one quarter* of its previous level. If the recording level of the sound seems low, especially with respect to louder background sounds, you must get closer to the sound source and not attempt to correct the problem by turning the level way up.

Many beginners think the recordist should try to capture all sounds in a general fashion, standing back from, say, a party, a conversation, or a street scene to record all the sounds together. The result of such recordings is usually an indistinct blur. The recordist should instead select individual sounds and get close enough to record them clearly. If an overall mix of sounds is desired, it may be necessary to put together several distinct tracks later. For documentary filming in noisy conditions, you may need to get closer to the subject than you feel comfortable doing. A few experiences with scenes ruined by bad sound will help overcome shyness.

Another approach to a noisy situation is to use a lavalier mic on the subject with a wireless transmitter to get the mic in close while allowing the recordist to stand back.

Whenever you put a mic very close to the sound source it minimizes both ambient sound and the natural reverberation of sound reflecting in the recording space. In some situations, a close mic sounds artificial. For example, if the camera is filming a distant long shot and the mic is very close, the recording will lack the proper *sound perspective*. Although distorted sound perspective is found regularly in movies, you may not like it. To correct this, the recordist could move farther back, but at the risk of sacrificing clarity. Alternatively, during the sound mix you could lower the level and maybe use equalization and a little reverb to give a sense of distance to the sound (see Chapter 16). Similarly, any missing ambient sounds can be added later by mixing in additional tracks. These kinds of effects are usually handled better under the controlled conditions of sound editing than they are while making a live recording. You can always *add* background sounds, distance effects, and equalization during sound editing and mixing, but nothing can make a noisy, echoing, or weak recording sound pleasing and clear.

SETTING THE RECORDING LEVEL

One of the recordist's key jobs is controlling the volume of the recording. The loudness of sound as it passes through a camera or audio recorder is called its *level* or *gain*; this is adjusted with a control labeled *volume*, *gain*, or *level*. This control is sometimes called a *pot* (short for *potentiometer*). All prosumer and professional cameras and audio recorders have a manual control to set the recording level (some consumer cameras only have auto control). In a typical recording situation, you're controlling the level of one or more microphones.

Setting the level isn't hard. In the simplest terms, *you want to record sound as loud as you can without it being too loud.* But to know what that really means, you need to understand a few things about the nature of sound, what meters do, and the workings of your digital or analog recorder. Please see Sound, p. 368, and How Audio Is Recorded, p. 371, before reading this section.

Understanding Sound Level

Figure 11-5 shows a simplified representation of the level of a sound signal over time. Notice that the level is always changing, with peaks and valleys. The lower black line represents the *average level*. This corresponds most to how our ears perceive the *loudness* of sound.

Yet, there are peaks that rise quite a bit higher than the average. The upper black line shows the highest limit of the peaks. Peaks often come and go very quickly, so the ear may not fully perceive their loudness. However, we are very concerned with how high the peaks are because recording equipment can't tolerate peaks that are too high.

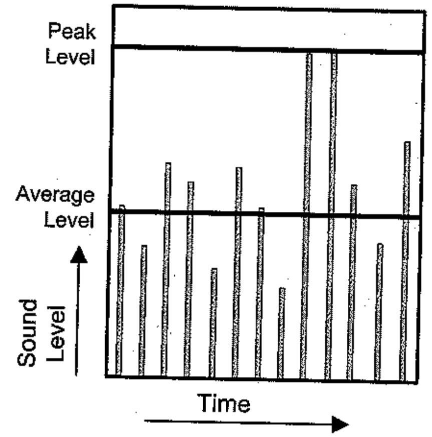


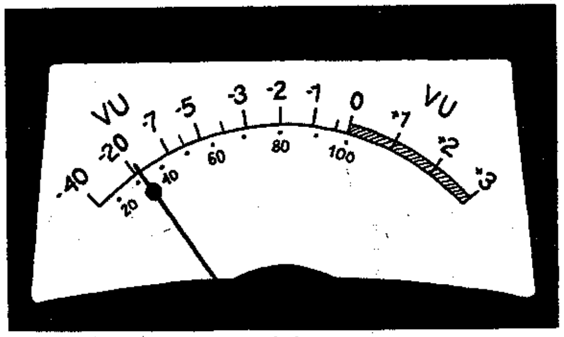
Fig. 11-5. Sound levels over time. Typical sounds contain volume peaks that are higher than the average level (how much higher depends on the sound).

The relationship of the average level to the peak level varies with the type of sound. With a "normal" male speaking voice, the peaks might be 8 to 10 dB higher than the average. With short, percussive sounds like a hammer, jangling keys, or a chirping bird, the peaks might be up to 50 dB higher than the average—a much bigger ratio.

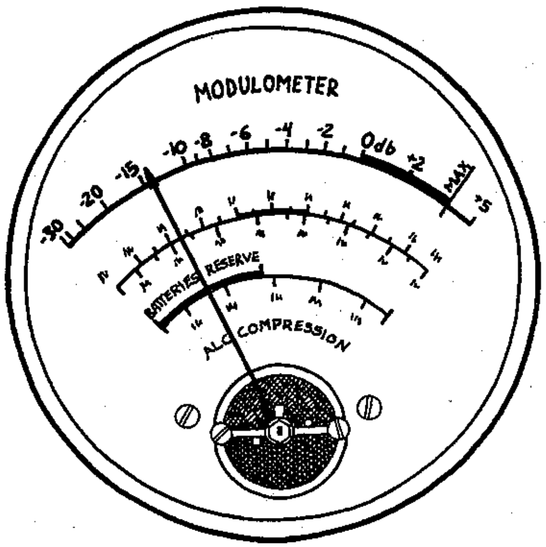
When recording sound, we need some way to measure or meter the sound level. The classic meter for analog equipment is the *VU (volume unit) meter*, which is found on both inexpensive and costly recorders (see Fig. 11-6). If a meter has a needle and is not identified otherwise, it's probably a VU meter (some VU meters use lights or LEDs instead of a needle). VU meters read the *average* sound level, so they provide the best reading of how loud things *sound*.[1] VU meters give an accurate reading of steady signals that don't have sharp peaks (like steady reference tones or a violin playing a long, slow note). However, VU meters are designed to respond relatively slowly to changes in sound level (you can think of the needle as being fairly "heavy"), which means that quick peaks, called *transients*, can pass by without deflecting the meter much (see Fig. 11-7). This is the problem with VU meters—they don't give a good reading of how high the peaks are.

Enter the *peak-reading meter*. Unlike the VU meter, the peak meter responds almost instantaneously to quick surges in volume and provides a reading of the *maximum* sound level. VU meters give too low a reading for fleeting sounds, like the aforementioned hammer or keys, because the volume peaks have passed by the time the needle is finished responding. The peak-reading meter, on the other hand, responds quickly enough to give an accurate reading of sounds of short duration. However, the drawback of peak meters is that the level on the meter may not correspond as well to perceived loudness. A recording with lots of quick peaks that shoot up high on the peak meter may *sound* the same as one that lacks those peaks and reads much lower on the peak meter.

1. It's worth noting that true VU meters are very costly, and the typical meters found on inexpensive gear only approximate a real VU response.



(A)



(B)

Fig. 11-6. (A) VU meter. (B) Modulometer on Nagra 4.2. Sound level is read on the top scale. (Carol Keller)

Peak meters come in various forms: an LED display; the *modulometer* found on a Nagra recorder. The *PPM* (*peak program meter*) widely used in Europe responds to transients faster than a VU meter but slower than a true peak reader (because distortion from super-short peaks may not be audible).

Many audio peaks come and go so fast that you can't actually see them on a meter. Peak meters may have a *peak hold* function, to let the meter linger a bit at the highest peaks so you can read them.[2] Some meters combine a VU meter with a peak indicator light, giving you the best of both worlds.

2. How quickly a meter responds to a peak, and then releases afterward is called its "ballistics." PPMs, for example, are defined by very specific ballistics, and there are a few different flavors of PPMs.

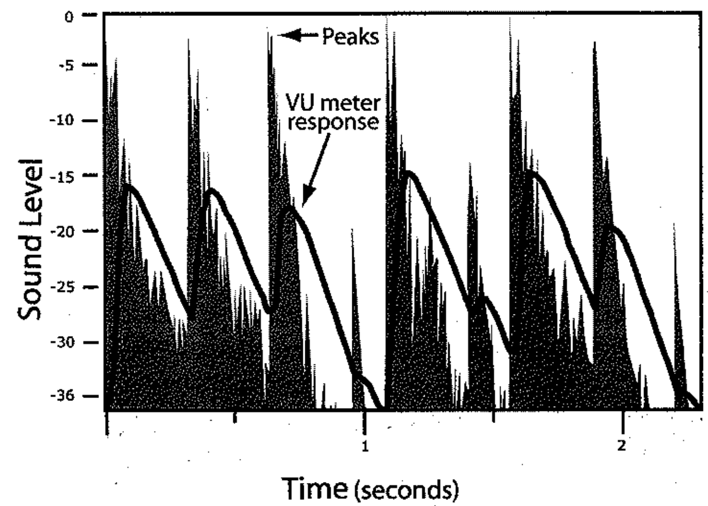


Fig. 11-7. Peak level and the VU meter. The gray area represents the sound level of a drum beat, with its short, percussive peaks. The black line is the response of a VU meter. Note that the VU meter lags behind the actual spikes in level, and doesn't rise as high or drop as low as the quick peaks and valleys. The VU meter indicates an average level that gives a good sense of how loud something sounds, but is often significantly lower than the actual peaks. (From http://en.wikipedia.org/wiki/VU_Meter)

Average and peak: Keep these two concepts in mind when you record sound. We care about the average level because that's closest to how things sound. And we care about peaks, because a signal with too much level causes recording problems (as described below). Meters can help you if used correctly, but in the end, your ears tell you more about level than a meter can.

Digital Level

The peak meters found on digital cameras or recorders typically start at 0 dB at the top of the scale and range down to a negative number like –60 or less at the bottom (see Fig. 11-8). In digital recording, 0 dB represents *all* the digital bits being used; there is nothing higher. This is *full scale*. The units on the meter are marked as "dB" and represent *dBFS* or *decibels full scale*. You could think of them as decibels *below* full scale—the number tells you how close you are to the top. You *never* want the signal to actually reach 0 dB because anything above this level will be clipped and distorted.

As a general rule, you want to record the signal as high (loud) as you can. This provides the best dynamic range and keeps the sound signal as far from the noise floor as possible. In film and video production, sound levels are always readjusted during editing and mixing, so even if you record a sound loudly in the field, you can always make it quiet later if that's called for (see Fig. 11-9).

So, the goal is to set the level in the camera or recorder relatively high, without letting the signal reach 0 dB. How best to do that depends on your equipment, the sound, and your preferences.

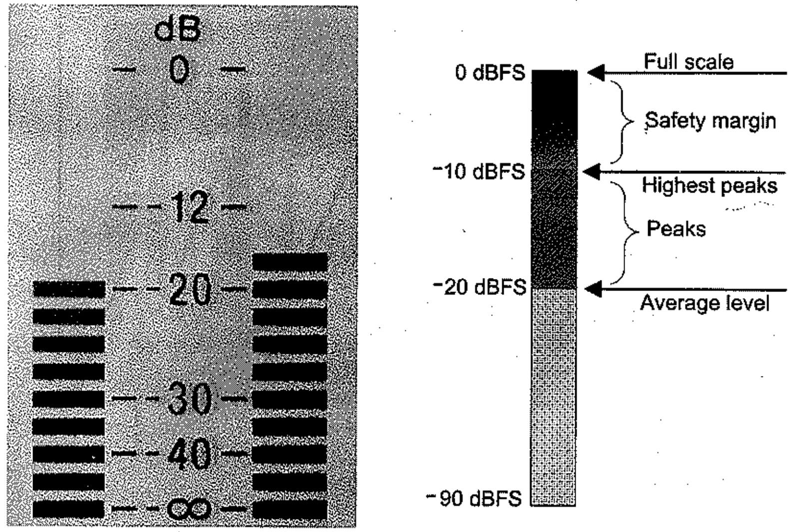


Fig. 11-8. (left) Digital peak reading meter found on Sony camcorder. The left channel reads −20 dBFS. (right) Rough guidelines for setting the level during field recording. Different situations call for different levels.

With professional recording for broadcast in the United States, the level is often set so that the average level of voices or music on the peak meter reads around −20 dBFS or so and the loudest peaks don't go above −10 dBFS. This leaves a safety margin of 10 dB before reaching the top, in case of any sudden, hotter (louder) peaks (see Fig. 11-8). If you're looking for a simple rule to follow, this should work fine in most situations.

With consumer and prosumer machines that have more noise and less dynamic range, the level is sometimes set higher, to stay farther away from the noise floor. The average level may be set to −12 dBFS, which leaves less *headroom* or safety margin in case of a loud peak.[3] You might use this level when capturing tapes into an editing system, as you can redo anything that goes too high.

Many recordists use a limiter as a matter of course with digital recordings to protect against high peaks (see p. 423).

Digital meters often have a clipping indicator or "over" light to show when peaks hit 0 dB; sometimes these light up just *before* you hit 0. Some systems *only* have a peaking indicator. In this case, turn the level up until the light comes on frequently, then reduce the level until the light stops flashing on. Some digital equipment has VU meters (see p. 420).

With any recorder, be attentive to the type of sound and how high the peaks are,

3. The idea of "safety margin" as used above is the difference between the highest peaks and the maximum level the system can record without distortion. "Headroom," however, is usually defined as the difference between the *average* signal level and the system maximum. If you're recording the average program level at −20 dBFS with peaks up to −10 dBFS, you have 20 dB of headroom, but only 10 dB of "safety margin."

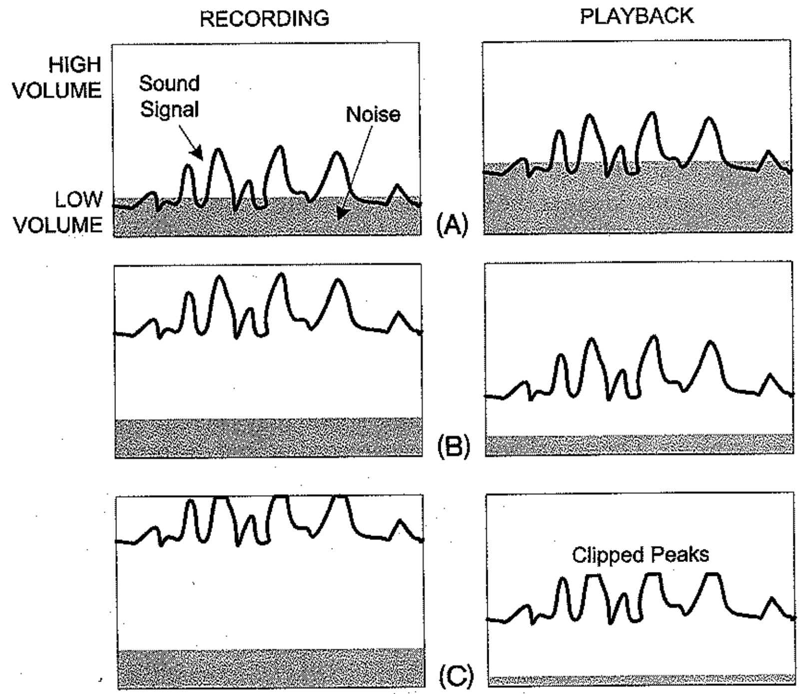


Fig. 11-9. Setting the recording level. The sound signal is what we want; the interfering noise could come from the recording system, the recording environment, or, with analog recorders, tape noise. (A) If the sound signal is recorded at a low level—near the level of the noise—when the volume is increased to a medium level in playback, the competing noise gets louder too. (B) If the signal is recorded as loudly as possible without over-recording, you can lower the volume in playback and the noise level will diminish as well. (C) If the level is set so high in recording that the signal is distorted, it will still be distorted in playback even if the level is reduced later (note the clipped peaks).

If you're filming a scene of two characters having a fight, the dialogue might go from low mumbles to loud shouts (with a few smashed dishes along the way). In situations like this in which dynamic range is high, you might want to keep the level lower than normal to protect against unexpected peaks. This is a situation in which a good limiter can help.

On the other hand, when you're recording something very even and predictable, you can nudge the level higher without risking over-recording.

If your recorder has an adjustment for headphone volume, set it at a comfortable level and leave it there. This will help you record consistent levels, and when you're in a situation in which it's difficult to watch the meter, it will help you to estimate proper recording level by the way things sound in the headphones.

Reference Tone

To help ensure that recordings are played back at the proper level, a *reference tone* (also called a *line up tone*) is recorded at the beginning of a tape or session. This helps you or others calibrate the playback level to match the level of the original recording. Many pro or prosumer recorders, mixers, and cameras have a *tone generator* or *oscillator* that can make a steady tone, typically at 1 kHz. You may have a choice of tone level, often between −12 dBFS and −20 dBFS. The idea is to use a tone level that corresponds to the average level at which you recorded (the *program level*). Though the idea is simple enough, it can get complicated because of different types of equipment and different styles of recording.

- As noted above, a standard approach for professional digital video recording in the United States is to record the average program level around −20 dBFS with peaks no higher than −10 dBFS. In this case, set the reference tone to −20 dBFS. This is a level that postproduction houses and broadcasters are used to working with and should result in fairly consistent levels. The −20 tone equates to 0 VU on a VU meter.
- When using a VU meter, tone is typically set at 0 VU. Peaks are not allowed to exceed +3 or sometimes +6 dB.
- When using a Nagra recorder, tone is typically set at −8 dB on the modulometer.
- In Europe and the U.K., −18 dBFS is often used instead of −20 dBFS as the reference level; peaks are kept under −10 dBFS.[5]
- As noted above, sometimes people record at −12 dBFS average level with prosumer cameras (and also sometimes with editing systems or in other postproduction situations with mixed tracks). In this case, use a line up tone at −12 dBFS. However, if you do so, a dub house or broadcaster may think the peaks are too high, or, if they don't bother checking and just assume the tone was recorded at −20, they may make the sound too low.

It is *very important* when recording a reference tone that you indicate in writing (and sometimes as a spoken message on the tape) what level the tone is recorded at, how high the peaks are above it, and what kind of meter you used. If you use a nonstandard tone, be sure to discuss it with anyone involved in postproduction sound or dub work.

Reference tones are just that—a reference. Just because a tone is recorded at the start of a tape or program doesn't mean the levels are correct throughout.

Some Recording Guidelines

In general, recordings sound better if the level is not changed in a noticeable way during the recording. This may mean, for example, setting the level at a compromise position between the loud and soft voices of two people talking. (Another solution might be to move the microphone closer to the soft-spoken person.) Try to anticipate surges in volume, choosing perhaps to under-record slightly an orator

5. In Europe, meter layouts vary. Reference tone may be +4 on a BBC PPM meter or 100 percent on a peak meter.

whose every sentence is greeted with loud cheers from the audience. This is another situation where a good limiter could be helpful (see below).

Often, a certain amount of level adjustment (called "riding the gain" or "riding the pot") is called for. Try to make level changes between lines of dialogue, not during them; if you have to reset during a line, adjusting the level gently and not suddenly will be less disturbing.

Try to maintain consistency in your tracks. If you're recording a scene in a drama, try to keep the characters at roughly the same level so they can be edited together without a lot of level changes.

When recording a single mic with a two-track (or more) recorder or camera, some recordists use the second track as insurance against over-recording. Record the mic on one track at full level. Also route the mic to the second track, but set the level on this track -6 dB lower (or even lower than that). This second track is used only if the first one is too hot. Be sure to write down that you've done this and record the reference tone on the second track at normal level *before* turning the level down.

When quiet sounds seem under-recorded, don't turn the level control beyond three fourths of its full range, as this will usually add system noise. Instead, get closer to the sound source. If extremely loud sounds (like live rock music) require that the level control be set at less than one fourth of its range, this too may degrade the sound signal. If this is the case, get an *attenuator*, or *line pad*, to place on the cable between the microphone and the recorder (some mics and recorders have built-in attenuators, see Fig. 10-29). The attenuator cuts down the strength of the signal.

Automatic Level Control

Many audio and video recorders are equipped with some form of automatic control of recording level. The names for the various types of automatic systems are not entirely standardized. Some types work better than others, but often the effectiveness depends more on the sophistication of the particular camera or recorder than on the type used.

AUTOMATIC GAIN CONTROL. *Automatic gain control* (*AGC*) or *automatic level control* (*ALC*) works by automatically choosing a predetermined recording level. If the sound signal coming in is too quiet, it will be boosted; and if it is too loud, the gain will be reduced. AGC requires no attention on the part of the operator, and it can be quite effective for straightforward recording in which the sound level doesn't vary too much. However, some AGC devices don't handle sudden volume changes well. Say you're recording someone in a kitchen. While he speaks, the level is fine, but when he stops, the AGC responds to the quiet by boosting the gain, bringing the sound of the refrigerator to full prominence. AGCs sometimes have a slow *release time*, and if a sudden, loud sound occurs while someone is speaking, the recording level will drop, reducing the level of the dialogue, and return to normal some moments later.

For all these reasons, AGC gets a bad rap. Indeed, some AGCs are not very good and in most situations a competent recordist setting the level manually can do much better. However, some AGCs work well and can be a real lifesaver if you're shooting alone or in a challenging situation. Test your own system and judge for yourself.

LIMITERS AND COMPRESSORS. A *limiter* is another form of automatic control that allows you to control the basic recording level; the limiter kicks in to reduce the level only if you're in danger of over-recording. Limiters often cut in fairly sharply when the sound level gets too high, protecting against sudden volume peaks (see Fig. 16-14).[6] Some limiters work so well that they're virtually "transparent" (unnoticeable); others can produce an unnatural, flattened effect in the sound.

Some wireless mics and other systems have limiters with a flashing LED indicator instead of a meter. Turn up the gain until the indicator lights up, showing that the limiter is cutting in; then decrease the level a bit until the limiter operates only on the loudest peaks.

Limiters can be very helpful for scenes in which sound levels change suddenly. They're especially useful for digital recording where too much level is a particular problem. Limiters can be a kind of insurance policy against hot peaks, but if they're relied upon too heavily (that is, if the recording level is set so high that the limiter is constantly working), the sound may lose much of its dynamic range and seem flat and without texture. Some pros use them; others avoid them.

A *compressor* is another method used in various audio systems (including some wireless mics) for dealing with excess level. Compressors reduce the level by a certain ratio (say, 2:1 or 3:1), which compresses (reduces) the dynamic range of the sound. A *compressor/expander* (sometimes called a *compander*) compresses the dynamic range during recording—to accommodate any loud volume peaks without over-recording—and then expands the dynamic range again during playback.

For more on compressors and limiters see Level and Dynamic Range, p. 648.

Gain Structure

There are many situations in which you want to use two pieces of audio gear together. For example, using a wireless mic with a camcorder. It's important to coordinate how the level is set in each, especially with analog connections. *Gain structure* or *gain-staging* refers to the process of setting the level correctly through the whole audio chain.

As a rule, you want the upstream item (say, the wireless receiver) set to output at normal full level (*unity gain*) but not so high that the sound is distorted. If you feed too low a level from the wireless to the camera, the camera will need to boost the signal, raising the noise level. If you feed a signal that's too hot, the signal will be distorted from that point forward.

In shooting, it's common to use a microphone mixer (field mixer) with a camera or audio recorder (see p. 400). Most mixers have level controls for individual channels and a master fader that controls all the tracks together. Especially when using only one channel, you generally want to avoid situations in which the channel is set low but the master is set high, which may add noise. Set the channel at a healthy level, then use the master to fine-tune the level.

6. The threshold point where a limiter cuts in and how much it limits the sound level varies among systems and is sometimes adjustable.

When using a field mixer, typically the recordist wants to set things up so he or she can control the level from the mixer, knowing that the camera will record correctly. Ideally, the field mixer should output to the camera using line level (see p. 401). Make sure any AGC is turned off in the camera. Turn on the mixer's tone generator and set it to 0 VU on the mixer's VU meter.[7] Then adjust the gain on the camera so the tone reads –20 dBFS on the camera's digital meter (but see p. 421 for more on reference tones). Now the recordist can adjust the level during the shoot using the mixer's meter and level control. You should still try to monitor the audio output from the camera with headphones, and check the camera's meter from time to time. The same concepts apply when using a mixer with a separate audio recorder.

LOCATION SOUND

Ambient Sound

Ambient sounds are the background sounds that surround any recording space. They can come from birds, traffic, waves, refrigerators, fluorescent lights, stereos, and the like. The best way to minimize their effect, when possible, is to eliminate them entirely. Don't shoot the birds, but do turn off refrigerators and air conditioners, and close windows facing out to the street.[8] When possible, locations should be chosen with ambient noise in mind. Try not to set up shoots underneath an airport flight path or by a busy highway. Sometimes you can get permits to block off a street while you are shooting; otherwise, plan to shoot at a quiet time of day. Heavy *sound blankets* (furniture packing pads) can be used to dampen noisy windows or air vents. These can be clipped and hung on grip stands to make a quieter space near the actors or subject.

Ideally, audible background sounds should remain consistent throughout a scene. Consistency is important for editing, since much condensing and rearranging of the movie's chronology are done at that time. An editor needs the freedom to juxtapose any two shots without worrying that the background tone may not match. The audience will tune out the gentle ambience of an electric fan, but will be aware of it if it pops in and out in every other shot. If you begin shooting a scene with the window closed, don't open it during the scene. In situations where you can't control some background sound (a neighbor's auto, for example), record some of the offending sound alone in case during editing you need to cover sections of the scene that lack it. In some cases, an inconsistency in the background sound will seem logical and doesn't need to be disguised.

Make every effort to turn off or lower any music that is audible at the filming location. Discontinuous music is a glaring sign that the chronology of shots has been changed. Recording music may also create copyright problems (see Chapter 19). If ambient music can't be eliminated, or if it is part of the scene you are filming, plan your editing around it when you shoot.

7. On some mixers the tone is not adjustable. Also, not all mixers have VU meters.

8. Many a crew has driven off from a location forgetting to turn the fridge back on. Recordist Frank Coakley suggests putting your car keys in it when you turn it off to remind you.

Always record thirty seconds or so of *room tone* at every location. Have everyone be quiet and stop moving while you record. Even if nothing in particular is audible at the location, every site has its distinct room tone, which is very different from the sound of pure silence or dead tape. Room tone—an expression that refers to outdoor sound as well—is used in editing to bridge gaps in the sound track, providing a consistent background.

Acoustics of the Recording Space

The size, shape, and nature of any location affects the way sound travels through it. An empty room with hard, smooth walls is acoustically *live*, reflecting sound and causing some echoing. Bathrooms are often very acoustically live; sound may reverberate in them for a second or more before dying out (try singing in the shower). A room with carpets, furniture, and irregular walls is acoustically *dead*; sound is absorbed or dispersed irregularly by the surfaces. Wide-open outdoor spaces are often extremely dead, because they lack surfaces to reflect the sound. Test the liveness of a recording space by clapping your hands once or giving a short whistle and listening to the way the sound dies out.

The acoustics of a location affect the clarity of the sound track and the loudness of camera noise. It's hard to hear clearly in an overly live room (a *boomy* location), as high frequencies are lost and rumbly low frequencies predominate. If you've ever tried to talk in a tunnel, you know what it does to the intelligibility of voices.

There are a number of ways to improve an overly reverberant location. You can use a directional mic and move closer to the sound source. A room can be deadened by closing curtains or by hanging sound blankets on stands and spreading them on the floor. Avoid positioning a microphone near a smooth wall where it will pick up both direct and reflected sound; echo may be increased or sound waves may cancel each other, weakening the microphone's response. This may also occur when mics are mounted on a short table stand over a smooth, hard surface. Avoid placing the mic in a corner or equidistant from two or more walls where reflected sound may

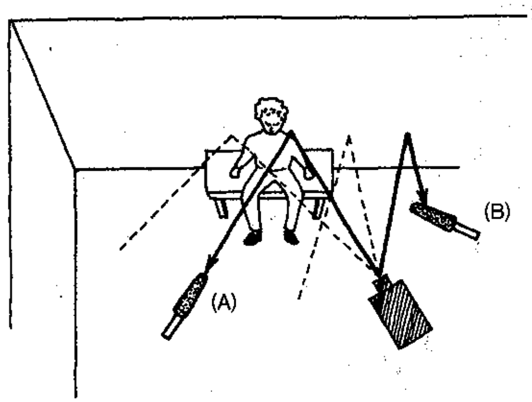


Fig. 11-10. Microphone positioning. Directional microphone positioned at (A) is pointed right at reflected noise from a film camera. Microphone positioned at (B) will reduce pickup of both direct and reflected camera noise. (For clarity in the illustration, the microphones are shown farther away from the subject than is optimal.) (Carol Keller)

cancel or echo. Sometimes boominess can be reduced by filtering out low-frequency sounds below about 150 Hz (see Bass Filters, p. 429).

If a space is too live, even a quiet camera's noise will sound loud. When you point the mic away from the camera, you often are aiming at reflected sound bouncing off a wall. When this happens, deaden the space with blankets, move closer to the subject, or use the pickup pattern of the mic to cancel out both direct and reflected camera noise.

Location Recording Problems

If you have to shoot in a noisy location, there are a number of things you can do. First, use a directional mic and get the mic as close to the source as possible. Always keep its pickup pattern in mind (see Fig. 10-20). Try not to let the subject come between the mic and a major noise source, like the street. Stand in the street to mic someone on the sidewalk; don't stand with your back to the buildings where you will pick up the sound of your subject and the street noise equally. With a supercardioid microphone, if you can point the mic upward from below (or down from above), you can minimize street level background noise, including sound reflected off buildings.

Lavalier mics are often useful for noisy settings. These are especially effective for a single subject or even sometimes when two people are near each other (even though one person is wearing the lav, you can usually still record the other person; you may want to place the mic a little lower than normal to try to even out the two voices). Lavs can also be hidden, for example, in a piece of furniture.

Shooting a group of people at a dinner table or conference table can be difficult. A boom mic is hard to manage if the conversation is unpredictable (and can be really distracting if it swings around wildly trying to catch everyone who talks; using two booms can help). Sometimes you can place a couple of omni mics or lavs in the center of the table and go wireless to the camera or recorder. Or try using a boundary (PZM) mic (see Fig. 10-19). Some lavs have a bracket so they can be taped face-down on a surface as a kind of boundary mic. Be careful about the sound of objects being put down on the table. You may need to keep the mic off the table surface, or give people a cushion—say, a placemat for cups and glasses—to soften the noise of putting things on the table.

With film cameras (and some video cameras), camera noise is a big problem. Avoid pointing the mic at the camera or its reflected sound (see Fig. 11-10). Put a film camera in a barney (see Fig. 6-15) if needed.

MUSIC, NARRATION, AND EFFECTS

Recording Music

Some suggestions for music recordings:

1. It's often best to record music in stereo (if not with more channels). See p. 431 for more on stereo mics.

2. When you record an acoustic band or orchestra, try to find a mic placement that balances the instruments nicely. Often, a stereo mic slightly behind and above the conductor's position on a high stand is used as the master mic. With many instruments, the sound will radiate out and up, so getting the mic high helps. Hanging mics is another solution. Sometimes a second mic is added to capture a vocalist or soloist. You may also want to put mics elsewhere in the space to record on separate tracks or mix with the master. When you use more than one mic, be careful to avoid phase cancellation (see p. 430).
3. When you record an individual instrument, place the mic near the point where the sound is emanating (for example, the sound hole of a guitar or the bell of a saxophone).
4. When recording amplified vocalists or instrumentalists, you often want to put your mic by the loudspeaker, not the person. When you record a person at a podium, you may get better sound by miking the person directly, but you must get the mic very close. Often with amplified speeches or musical performances you can get a line feed directly from the public address system (or a band's mixing board) to your recorder. By doing this, you avoid having to place a microphone and you usually get good-quality sound (in the case of the band's mixer, you get premixed sound from multiple microphones).
5. Generally, you want to let the musicians control the volume. Avoid using automatic level control or making sudden manual adjustments to the recording level. Find the highest level that can accommodate loud passages in the music and then try to leave the level alone. For live performances, professional recordists sometimes attend rehearsals, follow the musical score, and make slight adjustments during rests or pauses between soft and loud passages. When recording music with a digital recorder, consider using a higher bit depth and/or sample rate. With analog machines, you generally want to use the fastest tape speed you can.
6. When shooting a musical performance, plan to record fairly long takes. Unlike a lecture that can be cut into short segments and spliced together, musical performance sound must be relatively continuous. The camera should get a number of cutaways that can be used to bridge various sections or tie together takes from different performances. Shooting with more than one camera helps ensure that you will have sufficient coverage. For cutaways, get some neutral shots (such as faces) that don't show fingering or specific hand positions, which can be used anywhere.
7. If you want to use an existing recording, do a digital transfer to capture directly to your editing system. You may need to convert to a different file format using Quicktime Pro or another program. Use uncompressed sources whenever possible.
8. If you plan to use music in your movie, you should be familiar with music copyright laws (see Chapter 19).

Music Videos and Music Playback Scenes

Music videos and scenes in movies that include performing to prerecorded music present some challenges to the sound recordist. Generally, the band will have already recorded the song, and the performers will lip-sync (sing along) with it for the video. The recordist is usually responsible for having equipment to play back the song. It's a good idea to also record a scratch track on the set, which will help in syncing the footage to the song, and will capture any on-set banter or other sound that may be wanted by the director.

VIDEO CAMERAS. Shooting to music playback with video cameras is fairly straightforward. The music should be played back on a stable, speed-controlled format such as hard drive, Flash memory, DAT, or good-quality CD player. Use a camera frame rate that won't require speed changes in postproduction. In NTSC countries, 23.976 fps and 29.97 fps should work fine. In PAL countries, you can use 25 fps. The singers' mouth movements captured on video should then match up to the song in postproduction. Some video cameras may drift slightly in speed. A sync generator may be used to keep the audio playback and the camera locked together. Even without it, if shots are kept relatively short, sync drift won't be apparent in the shot.

FILM CAMERAS. Shooting with a film camera for video release in NTSC countries can be somewhat trickier. If the camera is run at 24 fps, the film will be slowed down by 0.1 percent during the telecine transfer (see Chapter 18). Compared to the master recording of the song, the singers on video will then be moving their lips 0.1 percent too slow. One way to solve this is to run the camera at 23.976 fps or 29.97 fps. Shooting at 29.97 uses 25 percent more film, and is not appropriate if you're doing music scenes for a movie that is otherwise shot at 24 fps.

Another solution is to shoot at 24 fps and *pull up* (speed up) the music playback by 0.1 percent on the set. There are several digital and analog recorders that can be made to play exactly 0.1 percent fast.[9] Or, you can create a file in a DAW that's speeded up by this amount and play that on the set. The lip-sync singing is then done to the speeded-up song, but when the picture is transferred to video, it will drop down to the speed of the original song. Always consult with the transfer house or other professionals before shooting.

Recording Narration

Narration (*voice-over*) should generally be recorded in a soundproof booth in a studio, as any background sound in the narration may cause serious problems. Any noise in the recording will be especially noticeable in the movie as the narration cuts in and out. For a low-budget production, or if you have to record on location, you can build an enclosure with sound blankets and grip stands to try to isolate the narrator. Set it up as far from windows and outside noise (such as airplanes) as possible—basements sometimes work best. You might even try a big coat closet.

Be attentive to echoes from reflected sound, especially in the small space of a recording booth. Even the stand used to hold the narration can create unwanted acoustic effects. Use a clip or small stand to hold the script. Narrators often like to use their hands when speaking, so it's better if they don't have to hold the script.

Often the mic is placed quite close to the narrator for a feeling of presence. Position the mic a few inches to the side and use a good windscreen to avoid breath popping. Listen closely while recording for breath sounds; redo takes if necessary. Many people like the warm, full sound of a large-diaphragm studio mic like a Neumann U87 or U89. For more on writing and delivering narration see p. 493.

9. An excellent resource on timecode and speed issues for music videos and many other situations is Wolf Seeburg's *Sync Sound for Film and Video* (see Bibliography).

Recording Sound Effects

Sound effects (SFX) are nonmusical, nonspeech sounds from the environment. The sounds of cars, planes, crowds, and dripping water are all considered effects. Effects usually have to be recorded individually. Don't expect to get a good recording of effects during scenes that involve dialogue. An effect may be difficult to record well either because of practicalities (positioning yourself near a jet in flight, for example) or because it doesn't sound the way audiences have come to expect it (for example, your recording of a running brook may sound more like a running shower). It can be better to purchase prerecorded effects from a sound library or mix studio (see Chapter 16) or to try to simulate an effect (crinkling cellophane to simulate "fire" sounds, for instance). There are many postproduction processes (reverb, speed changes, filtering) that can be used to create and enhance effects.

OTHER RECORDING ISSUES

Bass Filters

Many recorders, mixers, and microphones are equipped with filters that reduce the level of low-frequency sounds. These filters are variously called *bass cut*, *bass roll-off*, *high pass*, and *low-frequency attenuation* (*LFA*). Some filters cut off bass fairly sharply at some frequency, say 100 Hz. Others roll off low frequencies more gradually, often diminishing them 12 dB per octave; thus, the filter might reduce 150 Hz somewhat, 75 Hz quite a bit, and 37 Hz almost entirely.

Filtering, also called *equalization*, is done to minimize the low-frequency rumble caused by such things as wind, traffic, machinery, and microphone-handling noise. The low-frequency component of these sounds is disturbing to the listener and can distort higher-frequency sounds. If low-frequency sounds are very loud, the recording level must be kept low to avoid over-recording, and this impairs the sounds that you really care about (like voices).

Microphones and recorders may have a two- or three-position bass roll-off switch. The first position (sometimes labeled "music" or "M") provides a relatively flat frequency response with no bass filtering. The next position ("voice" or "V") provides filtering below a certain frequency. If there is a third position, it rolls off bass starting at an even higher frequency (see Fig. 10-21). Do test recordings with the filter to judge its effect. On some mics, the "M" position is optimal for most recording, and the "V" position should be used only for excessive rumble or when the mic is very close to someone speaking. Sometimes the third position removes so much of the low end that recordings sound very thin and hollow. For this reason, it's usually best to avoid using a camcorder's "wind" menu setting.

There are two schools of thought on filtering bass: One is to filter as needed in the original recording; the other is to hold off as much as possible until postproduction. The first school argues that the low frequencies will be filtered out eventually, and a better recording can be made if this is done sooner rather than later. This must be weighed against the fact that frequencies rolled off in the original recording are not replaceable later. The sound studio has better tools and is a better environment in which to judge how much bass needs to be removed.

A prudent approach is to filter bass only when excessive rumble or wind noise requires it or when trying to compensate for a microphone that is overly sensitive to low frequencies. Then filter only the minimum amount to improve the sound. If filtering is done, keep it consistent in the scene.

Multiple Microphones and Multitrack Recorders

There are many situations in which you may want to use more than one microphone. Typical examples are when recording two people who are not near each other, recording a musical group, or recording a panel discussion. Many recorders have provisions for two microphone inputs, and some machines can record from microphone and line inputs simultaneously (mics can usually be fed into the line input with the proper preamp). A mic mixer allows several mics to be fed into the recorder. Try to get microphones that are well matched in terms of tone quality. Sometimes a filter can be used on one mic to make it sound more like another.

When recording with multiple mics, be careful to avoid *phase cancellation*. This occurs when the peak of a sound wave reaches one mic slightly before or after it reaches another mic, diminishing the strength and quality of the sound signal. In phase cancellation, the diaphragm of one mic is pushed by the sound pressure while the other mic is being pulled and the two signals cancel each other out. The rule of thumb for avoiding this is that the microphones should be at least three times farther from each other than the distance from each mic to its sound source (see Fig. 11-11). Directional microphones that are angled away from each other can often be placed closer together.

Start with one mic and watch the level on the audio meter when you plug in the second mic—the strength of the sound signal should be increased, not decreased. Sometimes two microphones are wired differently, so that even if the mics are placed correctly, they cancel each other anyway. When recording in stereo, phase cancellation is not always noticeable but becomes apparent when the two mics are combined (*summed*) to one mono channel.

When using multiple mics, when someone is *not* speaking, keep his mic level down to avoid unwanted noise; this is often difficult when recording unpredictable dialogue.

If the recorder has multiple tracks, this opens up various possibilities for multiple microphones. You can record both a boom mic and a lavalier separately. You can place two mics in different positions and choose whichever sounds best later. If different subjects have their own mics and recording channels, their lines of dialogue can be separated more easily for editing and mixing purposes. Director Robert Altman used recording systems with up to forty-eight tracks with wireless mics on individual actors, each feeding a different channel. With this system, the actors can freely improvise, and everyone's lines will be recorded well, something that is almost impossible to do with only one mic and recording channel. Phase cancellation can often be avoided with multiple recording tracks, if the editor can choose the sound from one mic or another without trying to mix them together.

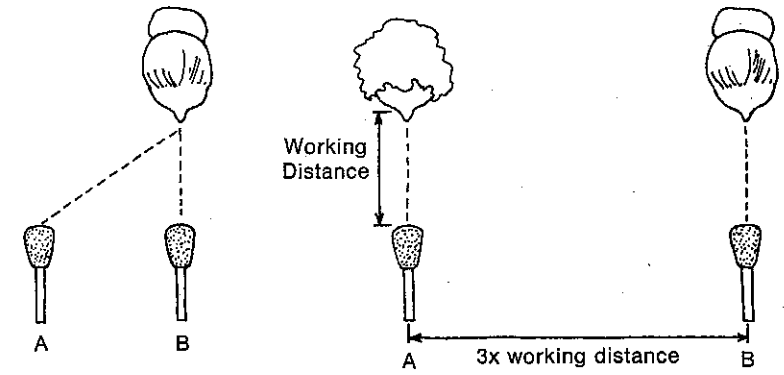


Fig. 11-11. Multiple microphones. (left) Distance from woman to microphone A is only slightly longer than distance to microphone B, leading to possible phase cancellation. (right) Separation between microphones is three times the distance from each mic to its sound source. Now the distance between microphone A and the woman at right is sufficient to minimize the chance of phase cancellation. (Carol Keller)

Having many tracks can be cumbersome for dailies screenings or for the picture editor. If you record to multiple tracks, you may need to create one or two *mixdown tracks* that combine the key dialogue tracks. Some multitrack recorders allow you to record a mixdown along with the individual source tracks.

Stereo Recording

Most video and audio recorders have at least two tracks and are capable of recording in stereo. Some camcorders are equipped with dual built-in mics that give some stereo separation between the left and right sides.

When movies are distributed to the public, in theaters, on TV, or as DVDs, they are in stereo (at least two channels) or other multichannel formats such as 5.1 channel sound (see Mix Formats, p. 654). However, it's important to make the distinction between *recording* in stereo on location and *releasing* the finished movie in stereo. For very many projects, dialogue scenes and the like are recorded in mono, even if those scenes will ultimately appear in a stereo sound track in the finished movie. A mono recording is made with one microphone (or more) on one audio track. It's very easy during the sound mix to place a "mono" sound either on the left or right side of the screen to create a stereo effect if needed.

Nevertheless, there are times when stereo recording is desired either for the entire production or for specific types of sound such as music, effects, or wide shots (perhaps to capture the sound of a horse moving from one side of the screen to the other).

There are various techniques for recording stereo. Stereo mics built into

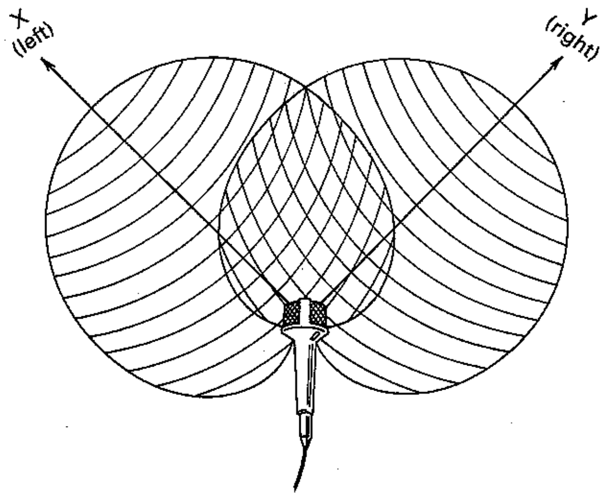


Fig. 11-12. Pickup pattern of a stereo mic in the X-Y configuration. Shown here, the two mic capsules are in one body. (Robert Brun)

camcorders usually employ the *X-Y* method (see Fig. 11-12). This uses two cardioid mics each pointed 45 degrees to the side (the two mic capsules may be in the same housing). The X-Y method is straightforward and simple, though when the two mics are separate—that is, not in the same case—mounting and controlling them must be done with care. It can sometimes be difficult to get the proper balance between the left and right side.

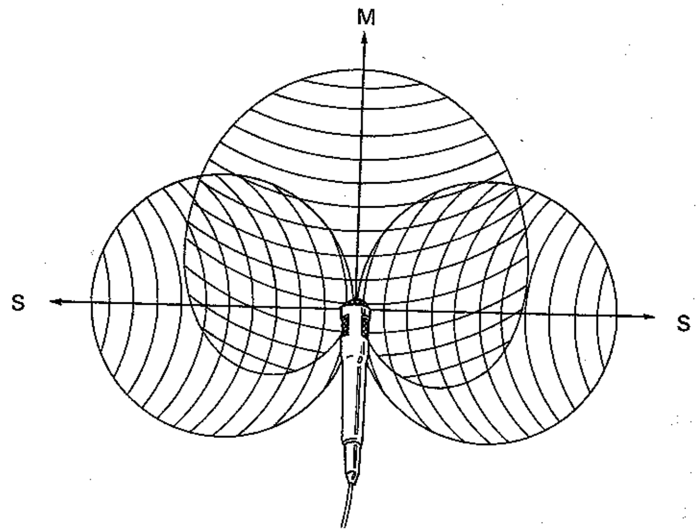


Fig. 11-13. Pickup pattern of a stereo mic in the M-S configuration. Shown here, the two mic capsules are in one body. (Robert Brun)

Some people prefer the *M-S* (*Mid-Side*) method. This also employs two mics: one with a cardioid pickup pattern and one with a figure-eight pattern (see Fig. 11-13). Mics like the Shure VP-88 have both mics built into a single housing. The cardioid mic picks up sound from the front and the figure-eight mic gets the left- and right-side image; each mic is sent to a separate track on the audio or video recorder. Later, the two tracks are "matrixed" through an M-S decoder to create the standard two-channel stereo effect.

Fig. 11-14. Shure VP88 microphone. Versatile mic that can output an M-S signal or can be switched internally to provide a stereo (X-Y) output with a choice between low, medium, or high stereo effect. Comes with a splitter cable for outputting the two channels. Use both output connectors for M-S or stereo output, or use just one in the M-S mode to record a mono signal from the cardioid capsule or the figure-eight capsule. (Shure Brothers, Inc.)

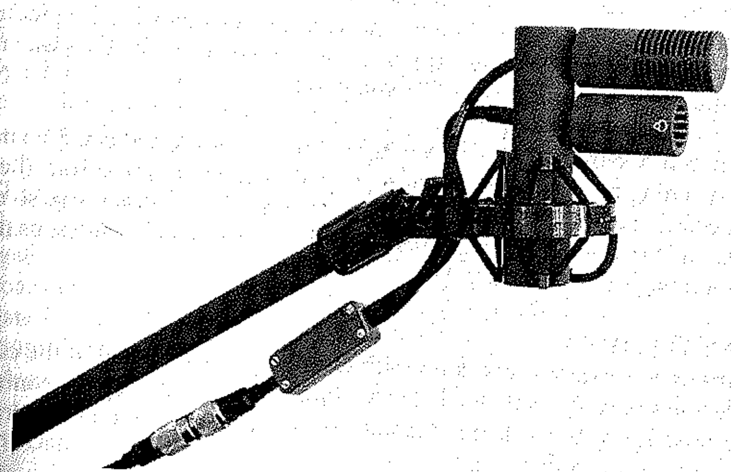


Fig. 11-15. Schoeps M-S setup showing separate cardioid and figure-eight mics that can be combined (matrixed) to create stereo. Schoeps makes very high-quality professional mics. (Posthorn Recordings, NY)

If you keep the M and S tracks separate throughout the editing process, you'll have more flexibility in the mix, because you can then adjust the relative balance of the mid, left, and right sound images, and you can isolate the mono signal from directly in front, if needed. However, by doing so, you won't be able to hear the stereo effect until after you're done editing (unless you have an M-S decoder in the editing room). Some M-S mics (such as the VP-88) allow you to output an X-Y (left-right) signal if you prefer; on some you can even choose how "wide" the stereo pattern is.

RECORDING DOUBLE SYSTEM FOR FILM AND VIDEO

Double system (or *dual system*) recording means using a separate audio recorder from the camera. All film shoots are done double system; film cameras don't record sound. Video shoots are sometimes done with a separate audio recorder in order to record higher-quality sound or more audio tracks or to provide more mobility for the recordist than might be possible by recording sound in the video camera. In this situation, it's still recommended to record audio in the video camcorder as well, to serve as a scratch track for reference.[10]

When sound and picture are recorded in separate machines, there are two basic concerns:

- Ensure that both sound and picture can be played back at the same speed or frame rate, so they don't drift relative to each other during a shot.
- Make it as easy as possible to sync the sound to the picture in postproduction—that is, to line up the sound recording with the picture so that when we see something happen on screen we hear it at the same time. Synching up is discussed on p. 573 and p. 601.

Many different techniques and technologies are used for speed control and for making the synching process easier. Today, many of them involve timecode (be sure to read Timecode, p. 203, before reading this section). Because there are so many options, this topic can get confusing; remember, in the end you'll only be using *one* of them on your shoot.

Speed Control

To record double-system sync sound, it's essential that the speed of the camera and recorder be precisely controlled. Today, film cameras with crystal-controlled motors are used for sync work (see Camera Motors, p. 235). Video cameras are usually also crystal controlled.

As for audio recorders, most recent digital machines such as hard drive, compact flash, DAT, MiniDisc, and CD recorders should have speed control that's accurate enough for sync recording. In some cases there may be slight drift over very long takes, which is usually corrected easily in editing. For precise sync, some recorders can accept a reference signal from the camera.

Speed control with analog recorders is discussed on p. 389.

PULLDOWN. In certain situations the speed of the audio needs to be adjusted after recording. Perhaps the most common occurs while shooting film in NTSC countries when video transfer and editing are planned. Film shot at 24 fps will typically be slowed to 23.976 fps in the telecine (see p. 606). This means that audio recorded in the field must also be pulled down (slowed down) by the same 0.1 percent. This can be done in a number of ways.

When working with digital audio, speed adjustment may be done by manipulating the sample rate. Unlike film or video, digital audio doesn't really have "frames"—the recording is just a constant stream of data (samples). For example, when using a 48 kHz sample rate, the recorder creates 48,000 samples each second; then, in playback, the machine makes sure 48,000 samples are played back every second to maintain constant speed.

If you're planning to slow the audio 0.1 percent, with some recorders you can set the sample rate to 48.048 kHz. This creates 480 extra samples per second when recording. However, the recording is "stamped" as having the normal 48 kHz rate, so the playback machine plays back the usual 48,000 samples per second, making the sound 0.1 percent slower. This method has the advantage of requiring no sample rate conversion or analog transfer later, either of which take time and could degrade the quality.

However, if the project will be distributed on film, this may not be the best route. Pro Tools and other DAWs have a pulldown/pullup feature for material recorded with standard sample rates. Always discuss settings with the post team before recording.

Traditional Slates

Synching involves lining up the sound and the picture. It's much easier to do if there's a distinct event that can be seen clearly in the picture and heard on the sound track. For example, the closing of a car door might suffice. A *slate* is an event in sound and picture that can be used to facilitate syncing.

The traditional slating device—called variously a *slate*, *clapper board*, *clap sticks* or, simply, *sticks*—is literally a piece of slate on which information can be chalked, with a hinged piece of wood on top that makes a sharp noise when it makes contact with the board (see Fig. 11-16). Now slates are usually plastic. Information written on the slate includes the production company, name of film, director, DP, scene and take numbers, sound take number (if any), camera and sound roll numbers, and date. A small gray card (see Fig. 8-4) or chip chart will assist in color-correcting a film workprint or video transfer.

The clapper board is usually handled by an assistant who writes and reads aloud the scene and take numbers and/or the sound take number before snapping down the hinged part of the slate at the beginning of each take. The numbers are often written on pieces of tape that can be stored on the back of the slate and quickly stuck on the front as needed.

10. This could be done as a feed from the audio recorder or just by leaving the camera mic on.

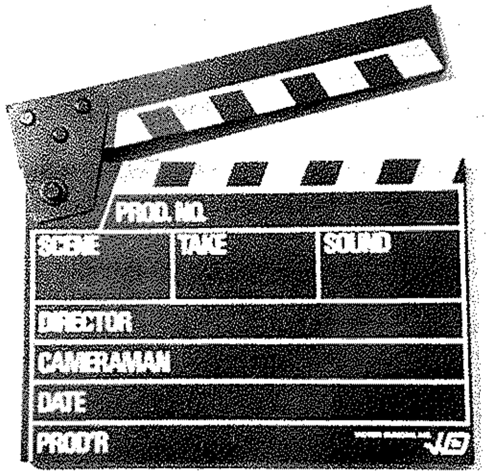


Fig. 11-16. Traditional clapper board slate. (Victor Duncan, Inc.)

Another slating device is the *slate light*, which is connected to the recorder. When its trigger is pushed, it flashes a small light (some flash consecutive numbers to identify takes) and it produces an audible beep. The slate light can be handy for documentary filming, although the light is sometimes hard to see in daylight. Slating can also be done by gently tapping the microphone once or twice or even by snapping your fingers within range of the recorder.

When you make any slate, it's imperative not to turn off the camera or recorder between the slate and the shot itself, as novices sometimes do. Make sure that the slate is clearly visible to the camera to avoid spending unnecessary time synching up. It's a good idea to say "slate" or "marker" into the mic to help the editor find the sound later.

When possible, do *head slates*, which are done at the beginning of the shot. Head slates speed the process of putting the sound and picture in sync in the telecine suite or editing room. *Tail slates*, done at the end of the shot, are sometimes preferable for unstaged documentary filming since they don't loudly announce to everyone that filming is about to begin; they may also be less disruptive for emotional, acted scenes. However, tail slates can slow down synching, especially in the telecine. Traditional clapper boards are usually held upside down to indicate a tail slate; the person slating should call out, "tail slate" or "end sticks" when doing so. If the film runs out before the last tail slate on a roll and the camera operator says "run out," you can use this as an approximate slate.

If either the camera or audio recorder miss a slate and you have to do it a second time, announce "second sticks" to alert the editor. In any situation, a gentle, quiet slate helps put actors or film subjects at ease. Generally, actors should not be rushed to begin the action immediately after the slate.

It takes longer, but it's usually not hard to put shots in sync without a slate (see Chapter 15).

OTHER SLATED INFORMATION. The term *slating* also refers to the recording of information on film or tape. MOS takes are slated, not for synchronization, but to identify the scene and take number at the head of the take ("MOS" should be written on the clapper board and the hinged bar should not be raised). It's a good idea to use a slate on all dramas—in film or video—even if not needed for synching. Each roll of film is normally slated at the head by shooting a card or clapper board with production name and company, camera roll number, date, and so on for a few seconds. Similarly, sound tapes are slated at the head with information on the roll number or the content.

Operating the Recorder

For staged work, there's a traditional protocol for beginning each take. The assistant director calls for quiet and then says "sound." The recorder is then started, and the recordist says "speed" when the machine is running smoothly. The AD then says "camera," and the camera operator calls "speed" or "rolling" when the camera has come up to speed. If slates are being used, they are done at this point; the AC reads off the scene and take numbers and calls out "marker." The director or AD then calls "action." Normally, the camera and recorder are not turned off until the director says "cut."

When doing slates, be sure the slate is shot so it's large in the frame and in focus. The numbers can be spoken aloud. Even when using a slateless system (see below), a clap stick with manual slates may still be done as a backup in case of timecode problems and for scene/take information.

When shooting film, some telecines require five to seven seconds of "preroll" before the slate or first usable audio. Consult the transfer house before you shoot to find out how much preroll, if any, they need. Make sure you let the recorder run the allotted time before the slate is done or the camera starts (and/or use prerecord to build in the preroll time; see p. 381).

In unstaged documentary work, it's important that the soundperson be ready to record at a moment's notice. If shooting appears imminent, the recorder should be put in the standby position (on some recorders this is done by pressing the "record" button, but not the "forward" button) and the recording level should be set. If the scene looks interesting, the recordist should not hesitate to roll. If the scene doesn't pan out, simply say "no shot" into the mic and stop recording. If the scene is good, the camera should roll. The first part of the scene that has no picture can usually be covered with another shot or a cutaway. There is no advantage to rolling vast amounts of sound, but often if you wait too long to start the recorder, the take will be useless (another situation where prerecord can help).

If your recorder is equipped with a separate confidence head, it's a good idea to monitor it and not the microphone directly (see p. 382). This allows you to check the recording quality and, for a tape recorder, whether you've run out of tape. Running out of tape or storage space on a Flash card is unnecessary and embarrassing. Some recordists routinely change tape or cards when the camera changes to a new roll/tape/card even if some recording time is remaining, which ensures adequate supply and means the camera crew won't have to wait for *you* to make a change later. If you don't have enough tape or media to complete a shoot, in emergencies you may be able to switch to a slower tape speed or lower sample rate, but don't forget to log this and to inform the person transferring the sound.

If shooting film that will be transferred to video for editing, it's generally a good idea to record or store all *wild sound* (sound without picture) separately so that they don't have to wade through it during the telecine session or when synching up.