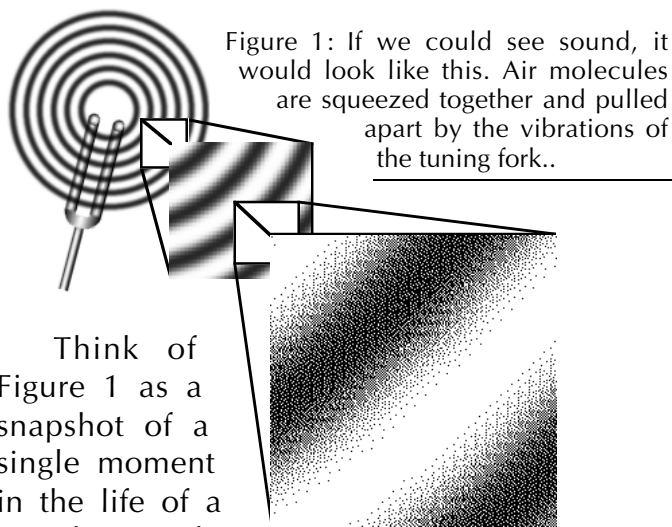


From chapter 2 Music-sicle?

“Architecture is frozen music.” When Friedrich von Schelling wrote that, he was being poetic. The scientific truth is if there were such a thing as frozen music, or any kind of sound, it would be pressure. This is usually air pressure, but it can also be pressure in water, wood, or anything else that conducts sound. It’s not precisely the kind of pressure you feel when deadlines approach, but the pressure that compresses molecules together. Sound travels through the medium of molecules bumping into each other. They can be air molecules, iron molecules, helium molecules, or just about any other substance.

If the pressure keeps changing repeatedly within certain speed ranges, you’ve got sound. Something—a drum skin, human vocal cords, the cone of a loudspeaker, or anything else that makes noise—starts vibrating back and forth. As its surface moves towards us, it squeezes air molecules together. As it moves away from us, it creates a very slight vacuum that pulls the molecules apart.

If we could freeze sound and see the individual molecules, they’d look like Figure 1.



cules are represented as tiny black dots . . . and as we enlarge sections of the picture, we can see individual ones.

The life of a sound

The tuning fork vibrates back and forth. When its surface moves towards the air molecules next to it, it squeezes them together. Those compressed molecules push against the ones a little farther from the tuning fork, and that squeezes the farther molecules together. The farther molecules now push against ones even farther, and so on: as the squeezing spreads out to successive layers of molecules, the pressure spreads out.

Air molecules, like everything else in the physical universe, take time to move from one place to another. So even while the pressure is spreading outward, the tuning fork—which is vibrating back and forth—may start moving back in the other direction. The air molecules next to the fork rush back in to fill the space where it was, pulling them a little further apart than normal. This very slight vacuum—engineers call it rarefaction—pulls on the next layer of molecules a little farther from the tuning fork, spacing them apart. And the process repeats to successive layers.

Everybody feels the pressure

As you can imagine, pent-up molecules try to push away from what’s squeezing them together. They don’t necessarily push in a straight line from the sound source, so the sound spreads in all directions. Because of this basic principle:

- You can point a light, but you can’t point a sound.
 - You can aim a lens to avoid something out-of-frame, but you can’t effectively aim a microphone to “miss” a distracting sound.
- ✂ ✂ ✂

From Chapter 4 **Setting the standard**

⌘ ⌘ ⌘ In 1940, the phone company and radio networks agreed on standard levels for long-distance broadcast and telephone lines: zero decibels would match 1/1000 watt (one milliwatt); any audio power from the tiniest microphone signal to the biggest amplifier output could be described as a ratio in dBm (decibels referred to that milliwatt). Wattage is a function of voltage and impedance, but telephone lines always had 600 ohms impedance, at that impedance, 0 dBm—or exactly one milliwatt—equals .775 volts. If you're not sure why something with a zero has any voltage at all, go back and read about decibels in Chapter 2.

The phone company developed meters calibrated to read zero at .775 volts, and added specific timing characteristics to make them respond the way we hear. They called them Volume Unit, or VU, meters. Broadcasters used them to measure program loudness and verify they were sending the phone company the right level. But they quickly discovered that the meters' internal rectifiers distorted the sound, something that didn't matter with phone calls. They reduced the distortion by isolating the meter with a resistor. But the resistor also reduced the voltage by 4 dB. Rather than design completely new VU meters, broadcasters decided that in their world the standard line level would be +4 dBm, or 1.228 volts across 600 ohms. When a VU meter reads zero, that's the voltage it's seeing.

- Professional sound equipment is often rated "+4 dBm line level" and operates at a nominal signal level of approximately 1.25 volts.

Of course, Ohm's Law insists that +4 dBm—a level in milliwatts—can equal a precise voltage only at one impedance. So two other standards evolved. In the United States, the

National Association of Broadcasters took just the voltage part of the phone company standard and called it 0 dBu—the "u" stands for unterminated: an open circuit with no impedance.

- Modern pro gear may be specified as "+4 dBu" instead of "+4 dBm." This also is a nominal level of about 1.25 volts.
- Unless you're dealing with transformers or very long transmission lines—both of which seldom appear in a modern video studio—dBm and dBu are equivalent.

In Europe, the IEC said exactly 1 volt would be a more rational reference: they called it 0 dBV. It turned out that tubed hi-fi equipment worked best with an input voltage around 10 dB below this reference, about a third of a volt. This is also sometimes called the IHF standard because it was subsequently adopted by the Institute of High Fidelity in the United States.

- Hi-fi, prosumer, and computer multimedia equipment is often rated "-10 dBV line level" and operates at a nominal signal level of roughly 0.3 volts.

So we have two different standards—three if your equipment is sensitive to impedance—to describe the nominal input voltage of a circuit. Since they're all based on the same rules of physics and math, they're easily related (Table 1).

Level in dBV	Voltage	Level in dBu (dBm across 600 ohms)
+6	2.0	+8.2
+4	1.6	+6.2
+1.78	1.228	+4 pro standard
+0	1	+2.2
-2.2	0.775	0
-6	0.5	-3.8
-8.2	0.388	-6
-10 consumer standard	0.316	-7.8
-20	0.1	-17.8

Table 1: Voltage compared to dBV and dBu



From Chapter 7 About microphones

✂ ✂ ✂ A camera's lens defines a precise rectangle. Point it in the right direction, and you can frame a shot that includes only what you want the viewer to see. Everything else is out of camera range.

With a lens, we could take a basic two-shot (Figure 1) and zoom in on just the horn-playing trickster. His bear friend would be completely gone (Figure 2).

Figure 1: A basic 2-shot



Figure 2: A longer lens lets us focus on just one character

But a mic doesn't work that way. When we talk about a mic's pickup pattern, we're merely indicating in which direction the mic is slightly more sensitive. It actually continues to hear things from all around the room, and there is no such thing as "just out of microphone range". Figure 3 is a visual analogy of what a good shotgun mic would hear, focused on just one subject.

Three visual aspects of Figure 3 translate to what you'll hear with real-world directional microphones:

- A directional mic will make sounds coming from the front seem slightly closer. That's why our horn player is slightly larger, compared to the bear, in figure 3.



Figure 3: Unfortunately, the mic isn't as discriminating. Even the most directional shotgun won't exclude the bear.

- It does this by lowering the volume of sounds from other directions. That's why the rest of the scene is darker.
- A directional mic will change the timbre of sound coming from other directions, adding a coloration that emphasizes some frequencies and lowers others. Note how the bear's colors are subtly changed.

The more directional a mic is, the more you hear these effects. However, better-quality mics will have less of the off-axis coloration.

Actually, Figure 3 is a simplification. What that directional mic really hears is more like Figure 4. . . except it would continue beyond the photo, picking up me, my camera, and things behind me.

Try this

Take your best directional microphone, camera, and good headphones. Turn on a radio, at conversational volume, in an average room of your house or office. Stand three feet away from a radio, and listen to it through the mic.

Now turn the mic 180°, so its back is pointing to the radio. Listen again. . . and you'll still hear the radio. It'll just be a little bit softer and not sound as good.

Try the same thing at 6' and the difference will be even less. ✂ ✂ ✂

From Chapter 8 **Boom Mic Technique**

✂✂✂ It takes a human model to demonstrate additional details, so we'll use Chris O'Donnell (not the model-turned-actor, but one of the best boom operators in the Boston area). Note, in Figure 7, how Chris keeps his inner arm (left, in this photo) straight and rigid. It's not directly overhead, because he doesn't need the height, but vertical enough that his bones provide most of the support. The outside arm is at the end of the boom, slightly bent so it can move easily. This gives control without being tiring.



Figure 7: Chris shows how to hold a boom properly, overhead. . .

Chris is tall, so for some shots he can hold the boom at chest level (Figure 8). This position is less tiring than overhead. In fact, many boom operators bring an apple box or step ladder to get some extra height, no matter how tall they are. But the strategy is still the same: he keeps the inside arm rigid, bearing most of the weight straight down; his outside arm does the steering.



Figure 8: . . . and at chest level.

Check the position of the hands in Figure 8 as well. The inner hand is under the boom rather than wrapped around it, and acts as a pivot. The outer

hand is wrapped around the top of the boom,

because the other hand's pivoting action makes this end push up.

In both photos, you can see how Chris holds the cable tightly at the end of the boom instead of letting it flop. Cable management is important, because a section of wire hitting the pole will make noise. That's also why there are a couple of loops at his belt: he can move the boom suddenly, and the loops prevent cable from dropping or rubbing against the floor.

The wire is looped around the pole in these photos, a common practice outside of L. A. Some boom designs let you run wires inside the pole, which seems to be the preferred Hollywood method. It doesn't make any difference which you choose, so long as the cable is kept tight and can't swing or bounce when you move the boom. If you're using inside wiring, wrap a couple of turns of cable around your finger where it exits the boom, so you can keep tension on the wire to stop it from swinging around inside.

You also have to secure the cable at the mic end, both to keep it taught and to provide a strain relief that protects the connector. Figure 9 shows the end of Chris' boom, with a couple of hairballs doing that job. Common practice, here in New England, is to mount the Ponytailers with "balls in back", facing the rear end of the mic. This way, you can glance at the entrance to the windscreen and see which way the mic, inside, is pointed. (You can also use tape to secure the cable.) If the cable is inside the pole, add a small loop of wire at the mic end; it protects the mic and connector if something snags the wire between takes.

Two More Tips

Booming requires strength and agility. You can build up the arm muscles necessary with pull-ups and overhead weight presses. You can strengthen your hands and fingers—just as important—with squeeze balls.



From Chapter 9 Balancing Adapters

✂ ✂ ✂ The transformer adapter is either a small metal box that screws into the camera's tripod socket (and has its own socket on the bottom), or a belt-pack that can be worn or taped to a tripod leg, with a cord that plugs into camera's mic input jack. It contains transformers to balance the mic inputs, and individual volume controls. Most also have switches to combine the mics to a single channel, and a line-level input. At least one model also includes a phantom power supply, which provides 48 volts for the microphone from a 9-volt battery. The adapters let you use balanced cables and microphones, which are virtually immune to wiring noise. Non-powered ones cost between \$170 - \$225; phantom power adds about \$50. Figure 1 shows the BeachTek DX6A, an under-camera model with phantom.

Camera mounting screw

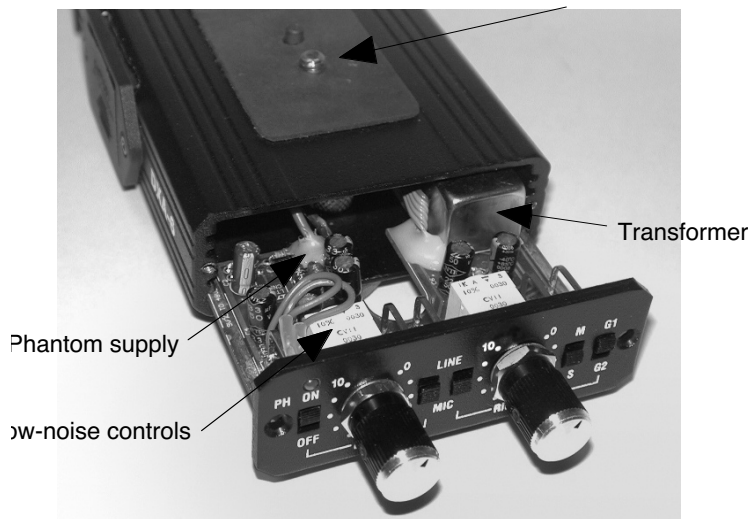


Figure 1: A typical under-camera transformer adapter, with added circuitry for phantom power.

There are two philosophies used in building these adapters. Most of the BeachTek units are designed for the footprints and grounding schemes of specific line of cameras. Studio 1 Productions' adapters have switches to adjust their grounding, and can be used with any camera or MiniDisc recorder. Either approach can give you good sound. The BeachTeks are a little bit easier to use, but might not work if you upgrade cameras. (Their DX6A has switchable

grounding and is more flexible.) While older Studio 1s are designed for under-camera use, their more recent units are designed as belt-packs with long flexible cords.

Adapter caveats

Some belt-pack adapters aren't balanced, and just contain a volume control, DC-blocking capacitor, and XLR jack. This defeats the noise-immunity of balanced wiring. You may be able to get by with unbalanced wiring for short cable runs, but for best results, look for the word "transformer" in the specifications. ✂ ✂ ✂

Unbalanced adapters

✂ ✂ ✂ You can get low-cost XLR-to-mini cables, designed for use with prosumer cameras, at video supply houses. These contain capacitors to block the DC voltage, which otherwise could cause distortion in some microphones. But they defeat the mic's balanced wiring, which can increase noise pickup through the cable. So use these adapters only if the mic cable will be kept short, or if you're feeding the camera from a nearby mixer mic-level outputs. If you're using a mixer with line-level outputs, use one of the attenuators described later.

If you're willing to do a little soldering, you can make a cable adapter for a few dollars' worth of parts. You'll need a female XLR connector for the mic (Radio Shack #274-011), a mini-plug to match your camera (#274-284), and a few feet of 2-conductor shielded wire (#278-514). The secret is a small electrolytic capacitor (like the 220 microfarad, 16 volt #272-956) to stop the recorder's polarizing voltage from causing distortion in the mic. The capacitor rating isn't critical—anything over about 100 μf and 6 volts should be adequate—and you'll find smaller units at electronics suppliers. The Shack's capacitor is small enough to squeeze inside the XLR shell, though you may have to break off a piece of the cable clamp to make it fit (Figure 2). ✂ ✂ ✂

From Chapter 14 Editing Music

✂✂✂ Once you've learned to count accurately with the music, the editing part is simple. All you have to do is match the numbers. Start by loading track 24 of the book's CD into your editing system. This is one of a number of pieces written specifically for this book by Doug Wood, president of the Omnimusic Library,¹ and it's designed to be typical of the kind of straight corporate or documentary theme you'd find in most libraries. It counts in 4.

If you're using a desktop NLE, follow the steps below. If you've got a different kind of system, the procedure will be only slightly different. Read along through these instructions, and we'll cover the differences later.

- Open the music in a clip window, start playing it, and count along. You might want to play it a few times before proceeding to get used to the tempo.
- Start tapping the marking button very lightly on each count. When you get to each "one," tap hard enough to actually make a mark. Depending on your system, you might then have to switch to a different button to mark the next "one" without erasing the first. When you're done, you should have a bunch of marks looking something like the gray flags in Figure 9.

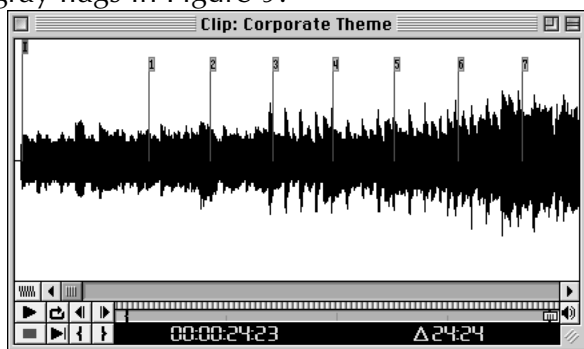


Figure 9: Tap hard enough to make a mark each time you count the number 1.

¹¹ Protected by copyright. You can license similar pieces, and a lot more, from Omnimusic (www.omnimusic.com).

Move the clip to a timeline, and you're ready to edit. If you want to shorten the piece, cut out the space between two marks and butt the pieces together. In Figure 10, we cut from Marker 2 to Marker 4, pulling out about three seconds. (Because of the way chords flow in most music, it usually sounds better to cut from an even number to another even one, or from an odd number to another odd one.) If we wanted to extend the music, we could have taken that stretch between Marker 2 and Marker 4 and laid it into the timeline twice.

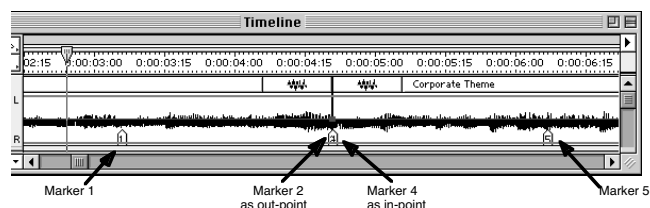


Figure 10: Shortening a cue by cutting from one numbered mark to another

This is a very short edit—there's only about three seconds between the markers—but the technique works for edits of any length. If you need to edit longer stretches of music, make the marks farther apart. . . say, every fourth "one".

Editing off the marks

As you listen through the sample corporate piece on the CD, notice how the horn melody begins around 12 seconds in. It actually starts on a "4," one beat ahead of the downbeat. But you'll still use those markers you made to cut the song accurately, and start that melody anywhere you want.

- Move the entire clip to one track of the timeline. Play through it, and notice the marker number just after you'd like the melody to start. For the sake of this example, we'll say it's Marker 3.
- Go back to the clip, and notice which marker occurs just after the melody's actual start. If your clip looks like our Figure 9, it'll be Marker 7. Slide the clip onto another track, lining up Marker 7 against Marker 3 (as in Figure 11). ✂✂✂



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