

Recording and Microphones

Digital audio recording is all about 1s and 0s. The binary system is used to encode all the glorious sound fed into the computer so you can tweak, manipulate, process, mix, and edit to your heart's delight.

This chapter will take you beyond elementary recording setups in Pro Tools and focus on real-world techniques both inside and outside the software. The goal is efficiency. The faster you can get a solid sound coming through the monitors, the more time and energy you'll have for that moving performance. We will focus on the basic elements of mic placement for the most common instrumentation of a rock band, but most concepts apply to sound design, broadcast, or any type of recording.

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I/O Recording Techniques

Using Pro Tools to record a band can be a fun, creative, and rewarding experience for everybody involved. The streamlined virtual nature of the Pro Tools interface coupled with the instantaneous nature of hard disk recording is a godsend for those who want more time to create. And that's what it's all about, isn't it? In the old days of analog recording, we had to wait around for extensive patching, calibration of tape machines, setup of multiple headphone mixes, and so on. With Pro Tools, you can make all this happen in a fraction of the time. The key to all of this is preparation, preparation, preparation. We will illustrate how you can set up and manipulate the Pro Tools I/O, create lightning-fast headphone mixes, and record into your Pro Tools interface so that you have more time for the good stuff: recording the hits.

It's now time to introduce you to an imaginary band we'll use to help illustrate Pro Tools concepts throughout the book: The Condensers, a typical small rock band made up of a drum kit, a bass guitar, an electric guitar, and a vocalist.

Preconfiguring the I/O Setup

You need to be ready for the band before they even walk through the door. Here's how to preconfigure the I/O setup to accommodate the band:

1. Choose I/O from the Setup menu. Click the Default button that appears in the I/O Setup dialog.
2. Click on the Output tab at the top of the dialog to access the Output panel.
3. Double-click the default output names to highlight the field. Name your outputs as shown in Figure 1.1:
 - 1–2 MIX
 - 3–4 PHONES MIX 1
 - 5–6 PHONES MIX 2
 - 7–8 PHONES MIX 3
4. Press Enter to lock in the changes. This will provide you with a “control room” mix and three separate and distinct headphone mixes for the individual performers of The Condensers.



Note: These setup ideas are designed to work with at least an eight-channel Pro Tools interface. Users with a Digi 002, Digi 002 Rack, or any HD interface will be able to configure the I/O to resemble the examples in this chapter. Due to differences in the interfaces, there will be some variation in the dialog texts for different systems.

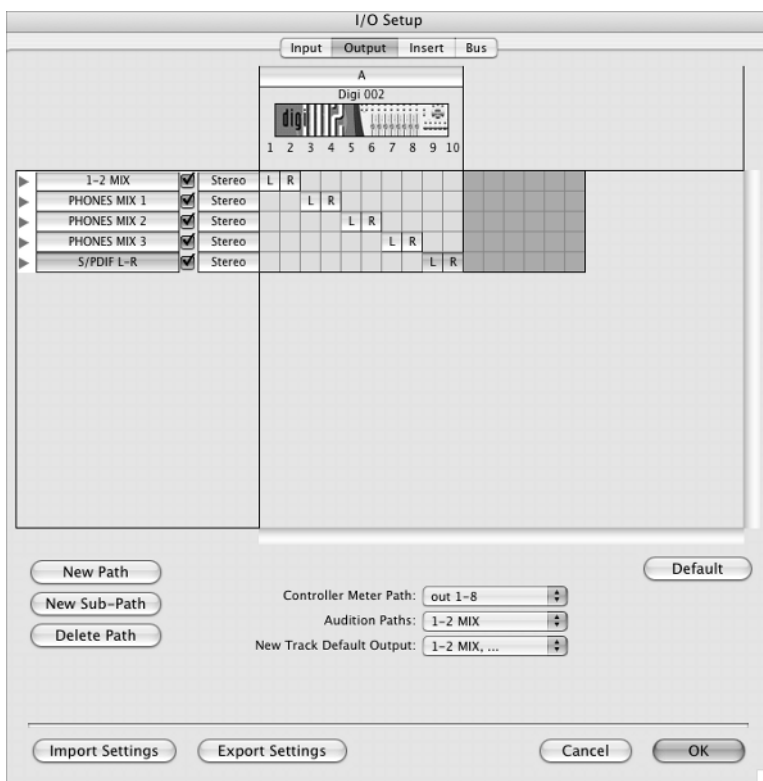


Figure 1.1 The Output tab for a 002 Rack

- Click on the Input tab (Figure 1.2) to access the Input panel and rename inputs 1–8 to Mic 1–8. If you are tracking a combination of Mic and Line inputs, you can be more specific here by naming the first four inputs Mic/Line and the second four Line, for example. Click the disclosure triangles for each channel to have access to the mono paths.
- Click on the Bus tab to access the Bus panel and assign names to the busses that follow the instrumentation. For The Condensers, the Bus panel will look like Figure 1.3.

Note: When you are done with the I/O setup, you may save it as a recallable I/O setup file. These are located by default in the Hard Drive > Applications > Digidesign > Pro Tools > IO Settings folder on your computer and may be taken with you to other sessions on your FireWire drive, iPod, or other removable media storage devices. You also have the option to click the Export Settings button. This will allow you to save your I/O settings to a specific location, such as the desktop.



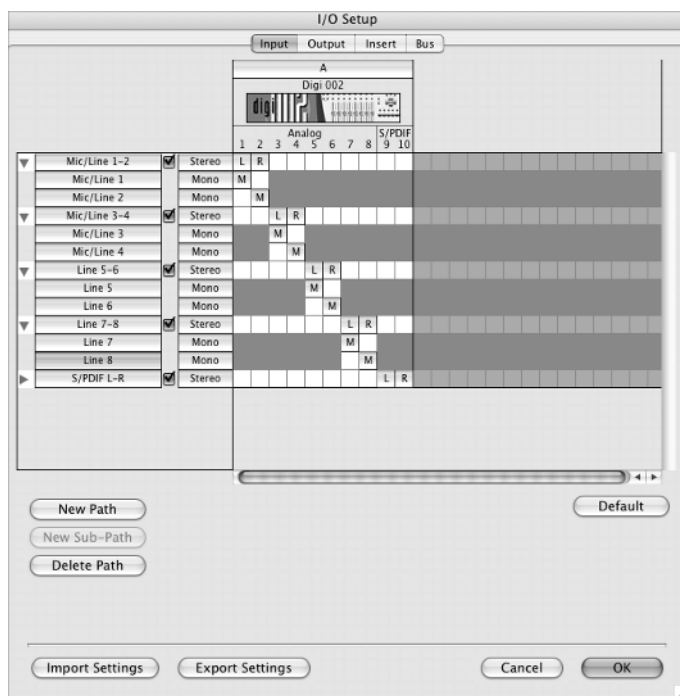


Figure 1.2 The Input tab

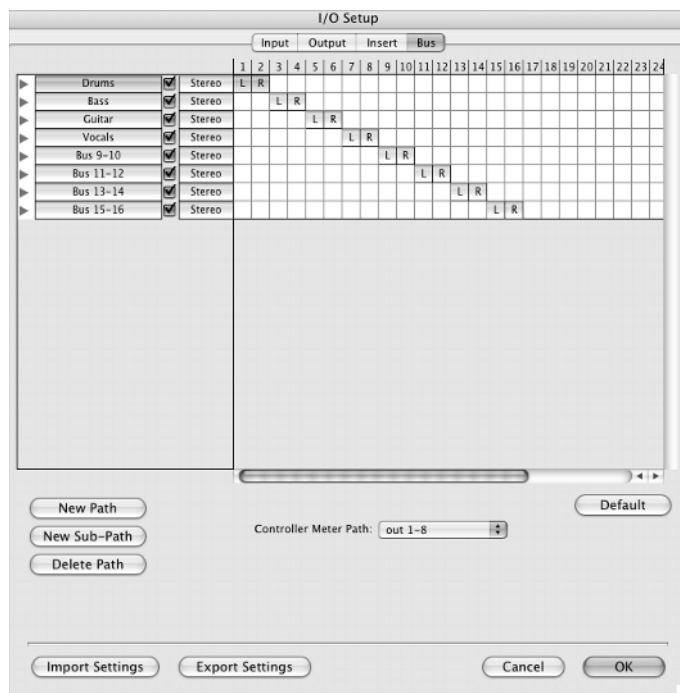


Figure 1.3 By assigning names to these busses, you will be able to bring up correctly labeled submix faders using aux channels.

Setting Up the Mixer

Now that you have the I/O in place, you can set up the mixer with appropriate channels and input and output assignments needed to track the Condensers. Here you can configure the available I/O to be labeled to accommodate the recording configuration you use for The Condensers.

The TDM mixer will look something like Figure 1.4.

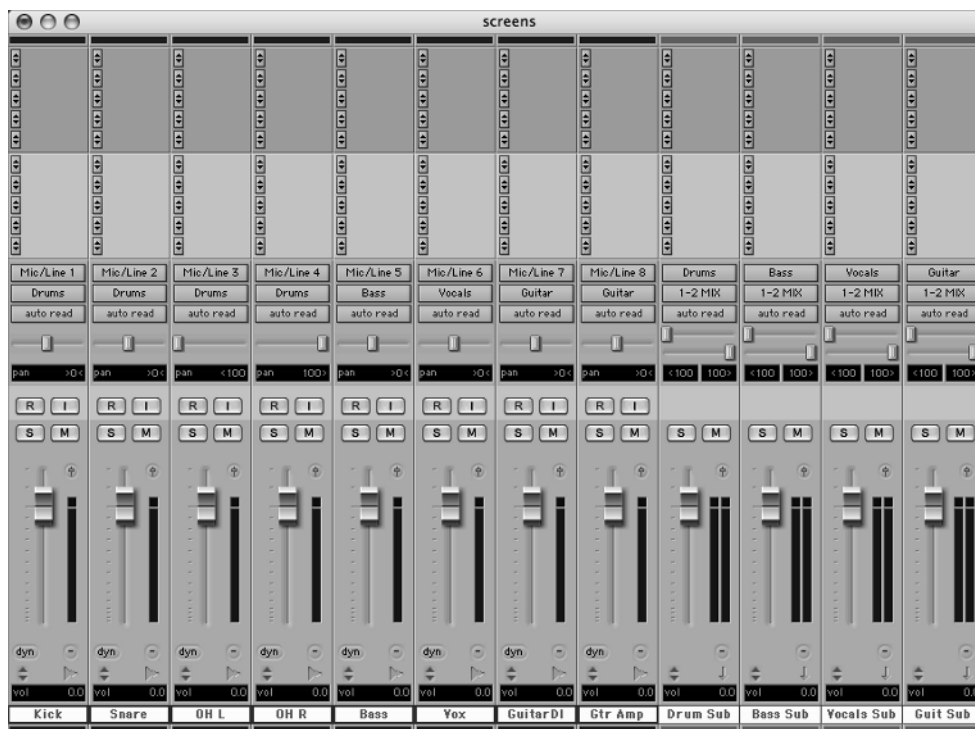


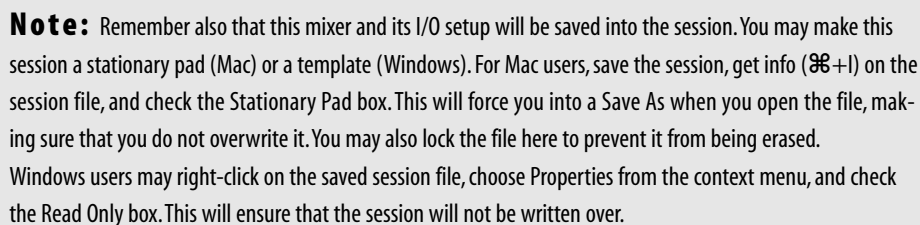
Figure 1.4 Note that the outputs are assigned to the corresponding busses you labeled in the I/O setup; this makes it easy to set up submixes across the auxiliary channels on the right.

Note: This is where the size and type of your Pro Tools hardware interface is crucial to how you proceed. TDM Pro Tools interfaces, such as the trusty old 888s or the newer HD systems, are going to have eight or more available inputs without microphone preamps. We are going to assume here that if you have a TDM system, you also have access to enough available mic pres to fill up at least eight channels of input and output (a single DIGI 96 I/O). For LE users, we will assume the use of a 002 system, using the four onboard Mic pres and the four Line inputs. The LE mixer will look like Figure 1.5.



The screenshot shows the Pro Tools 10.4.6 interface with the 'ch1' mixer. The mixer has 10 channels, each with a track name, input type, gain, pan, solo, mute, and fader. The channels are: 1 Kick, 2 Saare, 30H, 4Y0X, 5GUITARDI, 6BASS DI, 7 Drum Sub, 8 Bass Sub, 9 VocalsSub, and 10GtrSub. The interface is in a dark theme.

Figure 1.5 The LE mixer



The Headphone Mix

There's nothing a band likes more than to have a good headphone mix. A lot of engineers say that the headphone monitoring is the number-one thing to get ready in a recording situation because no matter how good the sounds being recorded are, if the band isn't comfortable, the performances will suffer.

Since you labeled three separate outputs as CUE MIX in your I/O setup, you can now use sends on each submix track to give three mixes to the band and still have your own control room "main mix" coming from the 1-2 outputs. To take full advantage of the separate headphone mixes, you are going to need either three headphone distribution amps like the ROLLS HA43 Headphone Amp or a headphone matrix amp like the Mackie HMX 56 Six-Channel Headphone Matrix Mixer and the corresponding cables.

You now have the option of putting the three headphone sends on each individual audio track for the most flexibility (see Figure 1.6).

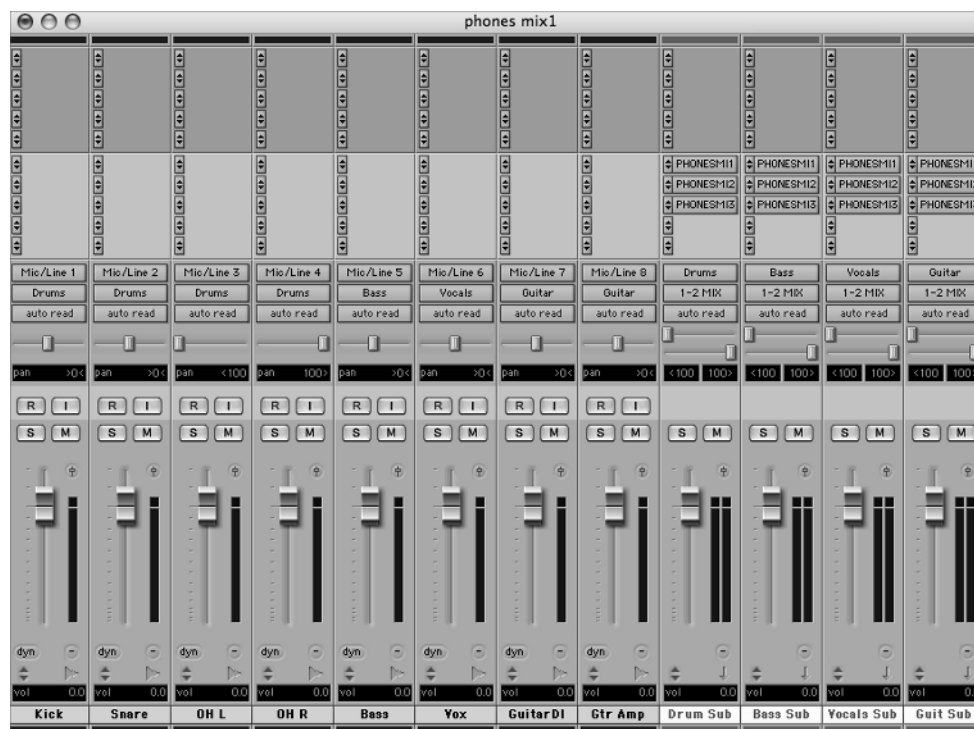


Figure 1.6 Headphone mix option 1

Or, you can put the headphone sends on the auxiliary submix channels (see Figure 1.7).

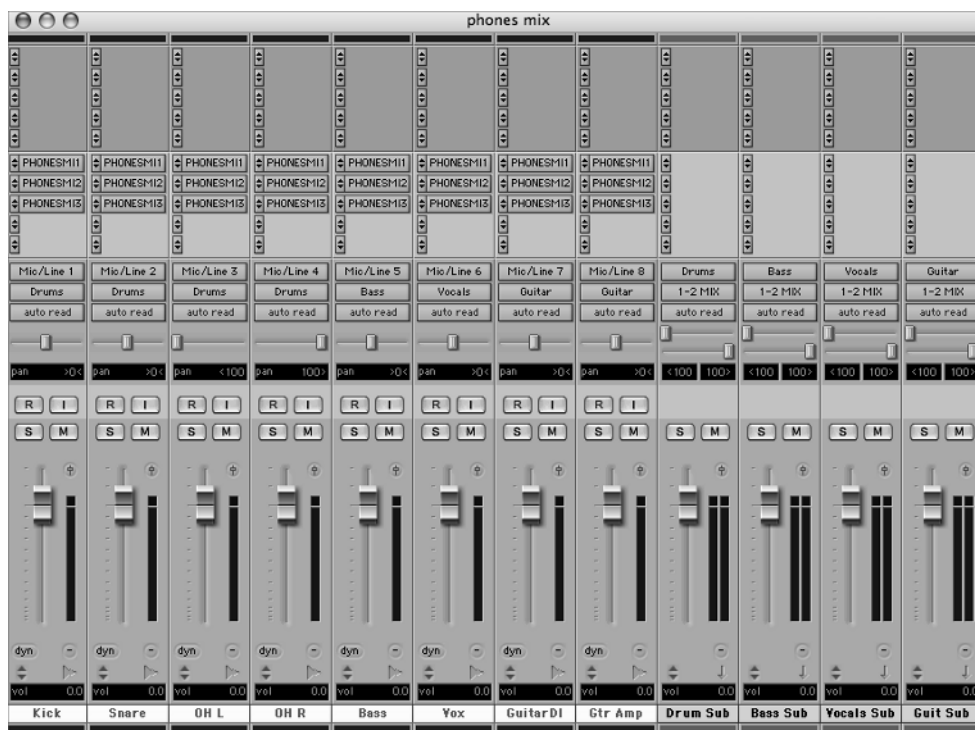


Figure 1.7 Headphone mix option 2

The outputs 3-4, 5-6, and 7-8 on the Pro Tools interface will now feed those headphone mixes. Typically, in this situation the drummer and vocalist prefer to each have their own mix, while the guitarist and bassist can agree on a third mix together.



Note: To assign and manage multiple track outputs quickly, select the tracks you wish to assign by (⌘+clicking / Ctrl+clicking) on each additional track name you want to select. Option+Shift+click / Alt+Shift+click on the output assignment and choose the bus or output you wish to send the tracks through. This will assign only the selected tracks to that bus. The same quick key works for send assignments, as in the headphone mixes in this chapter. Be careful not to only press Option/Alt here, because all tracks will be assigned regardless of selection. This is not undoable!

The Click

One of the first considerations in recording is tempo. When we think tempo, we think click track. When we think click track, we should think, “No problem, bring it on.” Tempo changes and even meter changes are a piece of cake. We need the click track not only for music’s sake but in order to put us on the grid to edit with the most flexibility and efficiency. Pro Tools is very clever in dealing with tempo, tempo changes, and tempo mapping. So if we have a bumpin’ little dance piece or some tasty electronic track, we know it will probably be a constant tempo throughout the track. Most electronic and hip-hop music in general stays with one tempo, so setting up a click for that is very simple. But in the case of multiple tempos and even meter, Pro Tools has the ability to map out these changes with musical precision.

The click track in Pro Tools has a number of different built-in sounds available to choose from (see Figure 1.8).

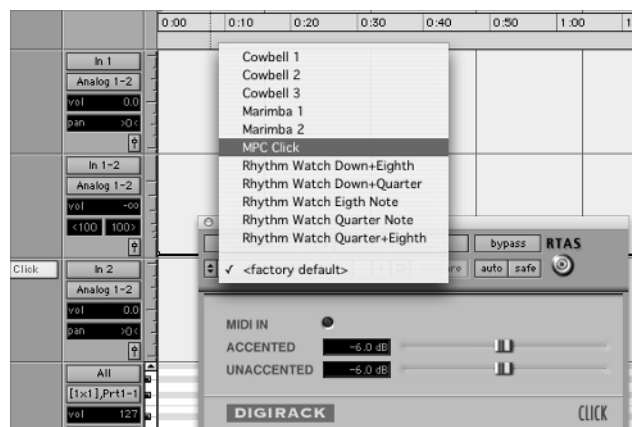


Figure 1.8 Click sounds

Better yet, perhaps, is the ability to choose from any other MIDI devices you might have, such as a nice old JV2080 or even NI’s Battery for the perfect sample of a stick click.

And now for the nuts and bolts: After selecting Grid mode (F4), the first choice to make is which tempo mode works better for the song, Manual or Conductor? If the track is geared to one tempo for the entire song, then you can use Manual Tempo mode. If the track has multiple tempo changes, you must use the Conductor mode and make tempo changes in the Tempo Ruler track. It’s easy to set up a basic click, so there can be no excuses. Create a mono aux track for the click because using an aux track saves

a voice. Make sure the Inserts view is showing in the Edit or Mix window and insert an instance of the click plug-in (see Figure 1.9).

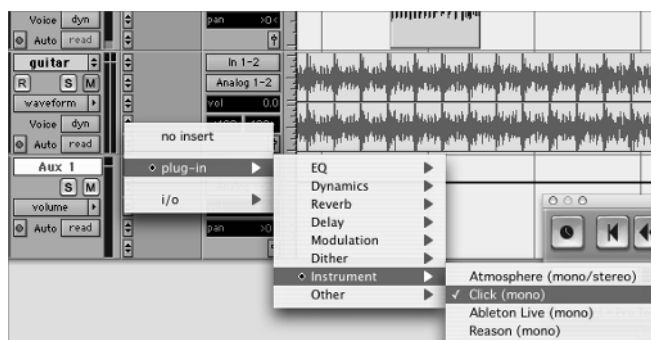


Figure 1.9 Inserting a click from the Edit window

Show the transport bar by pressing **⌘+Keypad1** / **Ctrl+Keypad1**



Note: To quickly change views in the transport bar, click the green maximize button in the upper-left corner of the transport bar. Option+click / Alt+click or **⌘+click** / **Ctrl+click** to get the rest of the views in the transport bar without going to the pull-down menu.

Pro Tools defaults to Conductor mode (Tempo Ruler enabled at 120 Beats Per Minute, or BPM). To change to Manual Tempo mode, click the Tempo Ruler Enable button (the button that looks like Arthur Fiedler) and deselect it (see Figure 1.10).



Figure 1.10 Manual Tempo mode in the fully expanded transport bar

Now you are in Manual mode. At this point you can manually enter the BPM into the counter window if you know the tempo. Manual Tempo mode also acts as a track mute so the tempo track ignores any tempo changes you have created. In the BPM window on the transport bar, click and select the current tempo, and you can either mouse up and down, type in the tempo, or move the slider bar horizontally to increase or decrease the BPM.

If you need to use Tap Tempo to find the BPM, make sure you are in Manual Tempo mode and click in to the BPM window. You can use your computer keyboard to tap tempo. Use the T key on your keyboard and tap in up to eight taps. Pro Tools averages the taps and calculates the tempo. If you would rather use a MIDI keyboard to tap in the tempo, then choose **Setups > Preferences > MIDI > General** and make sure

the Use MIDI to Tap Tempo preference is checked. On your MIDI keyboard, tap the tempo on any key up to eight times to get your tempo. Having the singer or guitarist play for you to get the tempo is a age-old rock and roll tradition.

If there are multiple tempos in the session, then highlight the Tempo Ruler Enable button on the transport bar (see Figure 1.11).



Figure 1.11 Conductor mode

Take your cursor to the Tempo ruler at the top of the Edit window. In order to insert tempo changes, you need to bring up the Tempo Change prompt box and tell it what to do. Click the Add Tempo Change button in the upper-left portion of the Edit window, which is the + symbol next to the word *Tempo*.



You can also press Control+click / Start+click in the Tempo ruler to bring up the Tempo Change box (see Figure 1.12).



Figure 1.12 Tempo Change box

In the Tempo Change prompt box, you can enter the new tempo and give it a location on the timeline. The old what and where. Very easy. Every time a different tempo is needed, call up the Tempo Change box and tell it the new BPM and location; Pro Tools's grid and click will follow right along. With the click sound selected

and the tempo mapped out, you then need to turn on the click by enabling the Metronome Click button on the transport bar.



Note: With everything set up, it's easy to toggle the click on and off by pressing 7 on the numeric keypad.

Now that you have some pointers for setting up Pro Tools, let's move to the studio and get things set up to record.



Note: Microphone manufacturers chosen in the following section are by no means a comprehensive list. We chose them because they are some of the more commonly found microphones in home and commercial recording studios.

Recording Drums: Less Is More and More Is Less

You can approach recording drums in two ways: the natural open room sound, which we like to think exists on the Led Zeppelin end of the spectrum, and the tight closed and controlled sound, which belongs on the Fleetwood Mac side of the spectrum. Everything else falls somewhere in between. The following sections outline each of these methods and how it is possible to capture those wild transient sound waves that drums produce with accuracy and confidence.

Drums require at the very minimum two microphones and at the maximum up to eight or nine microphones of all different types, so it is necessary to beg, borrow, or steal to have all those well-matched preamps and microphones ready to go and patched into your available Pro Tools inputs. Go back to the section “Preconfiguring the I/O Setup” at the beginning of this chapter for some ideas on preproduction.

The next point is not to be taken lightly. More important than any recording technique or microphone choice is the instruments you are starting with. If you have a poorly tuned, junky-sounding drum kit, there is absolutely no way you are going to get it to translate well into a recording and you'll probably end up hating yourself and losing friends along the way if you try to make it work. In addition, the player's technique

is something to take into account. Drumming for the recording studio requires a whole different skill set than playing drums live at a show. All the intricacies in feel, dynamics, and technique are going to show up indiscriminately when you hit Record, so be prepared to deal with them.

The Roomy Drum Sound

Now let's get down to brass tacks. The big room sound always starts with the overhead microphones. If you can get the whole kit to sound balanced in the overheads, most of your work is done and you can blend in other more closely mic-ed sources like salt and pepper to taste. Use your best cardioid condenser microphone here, but if you are going with a two-microphone stereo overhead setup, make sure you have a pair of the same type microphone. Small-diaphragm condensers like Neumann KM 184s, Schoepps 221s, or AKG 451s do the trick here, but you could get a huge sound out of a pair of Neumann U67s, AKG 414s, or even some Audio Technica 4050s. Phase correlation is everything here, so you may try them in an XY correlation, with the capsules pointed 90° at each other on a boom stand about a foot above the drummer's head. Make sure the drummer doesn't harm your precious microphone with his sticks though! A spaced pair works well here, with the two microphones facing downward and several feet apart from one another. In this case, it helps to monitor both microphones in mono, with their tracks panned center to check for phase anomalies. Keep the 3:1 rule in mind (see the sidebar "Phase"). If you are getting too much cymbal, you can try putting the microphone lower, about 3 or 4 feet in front of or behind the drum set.

Once the balance is perfect and the drums are sounding good, you can microphone up the kick and snare. The most popular microphone for the snare is the Shure SM57. Its ability to handle the fast transients and high sound pressure levels are unrivaled, so most people don't stray from it. But hey, no one is stopping you from a little experimentation—definitely see how other dynamic microphones sound. The biggest sound variation will come from placement of the snare microphone, so start with the microphone over the rim of the drum angled 45 to 60° downward pointing at the center of the drum. All snares are different, and each player hits differently, so you may need to experiment with the angle of the microphone to get the optimum sound.

Mic-ing the kick drum can be tough. The low frequencies take distance to fully develop, so at close range they are sometimes hard to grab ahold of. Start with a dynamic microphone that can handle high sound pressure levels, like an AKG D-112, Sennheiser 421, Beyerdynamic TGX150, or Shure Beta 52. If the kick drum has a hole cut in it (this is recommended), place the microphone halfway in the hole pointed directly at the beater on the front head. The distance is crucial and variable, so experiment a bit until you get the optimum mix of attack and low end. The brighter attack

will come from the beater hitting the front head and the low-end sounds will develop inside the drum.

That's it for the minimal style. This setup will yield the biggest, roomiest sounding results. Speaking of big and roomy, don't forget the room microphone! In addition to this setup, it is always nice to find a spot in the room where you can put a satellite condenser microphone. It is tricky to find this spot because if you are too close, it will not add much, but if you are too far, it may sound too reverby. This microphone should be recorded onto a separate track and can be later crushed to smithereens by a compressor to get that huge Bonham sound!

The Tighter, More-Controlled Drum Sound

Ahem, tighter please.... If it is control you are after, you've come to the right place. After all, Pro Tools offers us many options with editing, Sound Replacer, looping, Strip Silence, Beat Detective, and so on. (See Chapter 3, "Editing: Slip, Shuffle, and Spot Your Way Home," for more editing ideas.) Almost all of these tools require the drum recordings to be clean and the sounds to have good isolation from each other.

To get a tighter sound by close mic-ing, you can start with the basic setup for the big roomy sound we mentioned earlier but add more microphones to the mix.

The snare can sometimes benefit from a microphone underneath as well. This can pick up the metal snares for more bright character. Just about any microphone will work for this, and the phase will usually have to be inverted 180 degrees with a plug-in inserted on this microphone track (see Figure 1.13).



Figure 1.13 The phase flip symbol on Pro Tools plug-ins

The toms can be mic-ed 2 or 3 inches from the top heads near the rim, pointed toward the center with a dynamic microphone. Sennheiser 421s are popular choices here, as are SM57s. Make sure the capsule is angled in a way to cancel out cymbal noise here since you are going for isolation.

If the hi-hat isn't cutting through the way you want it, you can mic it angled away from the snare using a small-diaphragm condenser microphone like the AKG 451 or Neumann KM 184. It may be a good idea to filter out lower frequencies with an equalizer, or EQ either during or after tracking for this microphone.

Since you are tracking digitally and hard drive space is so cheap, if you've got the microphone and inputs, there can be no harm in tracking drums with this more-complex setup. As you will see in the Chapter 3, there is no limit to what is possible with isolated drum sounds in Pro Tools. After all, it is much easier to remove unnecessary tracks than to wish nonexistent tracks into being!

Dealing with Phase

If you are recording with microphones, it isn't long before you have to deal with the concept of phase. This is critical with drums, but it applies for all recording. Although it seems intimidating, it is actually a simple concept and with the right methods and some tricks in Pro Tools, easy to deal with.

Phase Technique 1: The 3:1 Rule

The 3:1 rule is a handy guideline that can help with phase relationships. To use the rule, the distance between two mics should be at least three times the distance between each mic and the sound source. This will help with keeping phase cancellations to a minimum, making sure that the sounds hit each microphone at the same part of their cycles.

In practice, you've got each mic 7 inches from an acoustic guitar. To use the 3:1 rule, you would spread the two mics at least 21 inches apart from each other.

Note: It is not a bad idea for an engineer to keep a tape measure handy when tracking live instruments.



Phase Technique 2: Listen While Placing Mics

Trust your ears. A good technique when mic-ing something with multiple mics is to listen through Pro Tools as you or your assistant moves the mics around. If you are listening to the mics mixed together, you will hear the difference when one of the mics is moved. Remember, the mics share a relationship with each other and the sound should be judged as a better or worse situation. You are looking for a best-possible relationship, one in which the mics all sound right together. With a little practice, you will know when you have hit the sweet spot and you will learn what it sounds like when mics are not sharing a good phase relationship. In a bad phase relationship, the sound has a weird EQed sound, or it's thin or filtered.

Phase Technique 3: Invert

If your waveforms are near 180° out of phase, you may select a region and invert by using the Audio Suite plug-in named Invert. This will effectively flip the phase on the entire region and invert the waveform (see Figure 1.14).

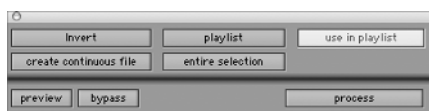


Figure 1.14 Invert Audio Suite plug-in

Phase Technique 4: Waveform Alignment

Once the information is already recorded, take a look at the waveforms in detail in Pro Tools. It is possible to nudge the waveform forward or backward slightly to put one source in better phase with another. You can change your nudge values by opening the green nudge pull-down menu.

A good choice here is 1 millisecond since we are planning to move the region very subtly. With the region selected, use the plus and minus keys on the numeric keypad to nudge the region forward until its waveform is inline with the other region (see Figure 1.15).

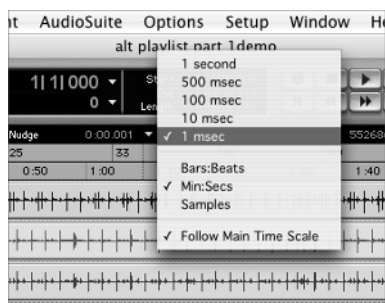


Figure 1.15 The nudge pull-down menu

Figure 1.16 shows two kick drum tracks a little out of phase due to the difference in their waveforms.

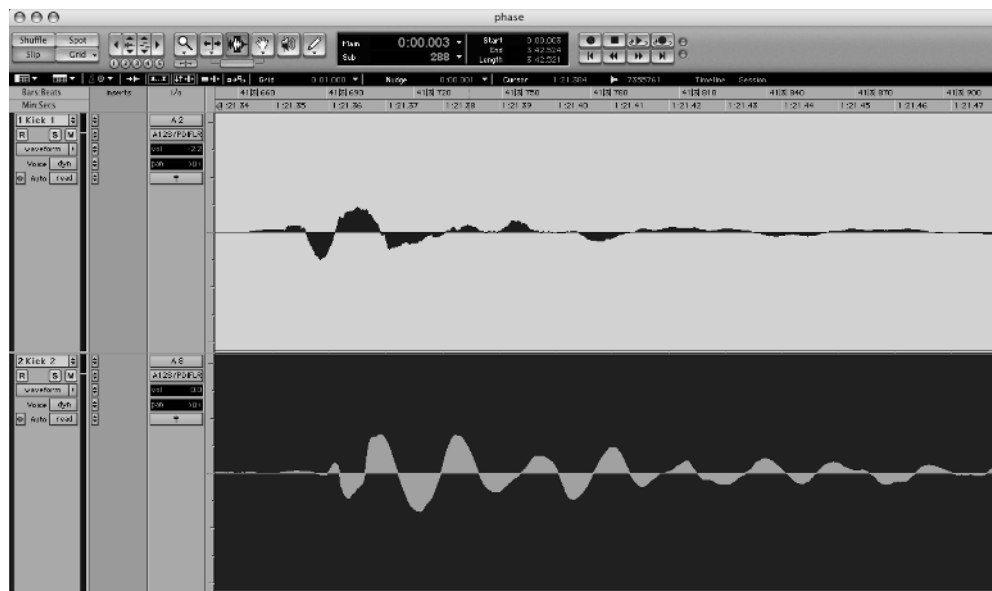


Figure 1.16 Out-of-phase kicks

Figure 1.17 shows the kick drum sounds after being nudged into a better phase relationship.

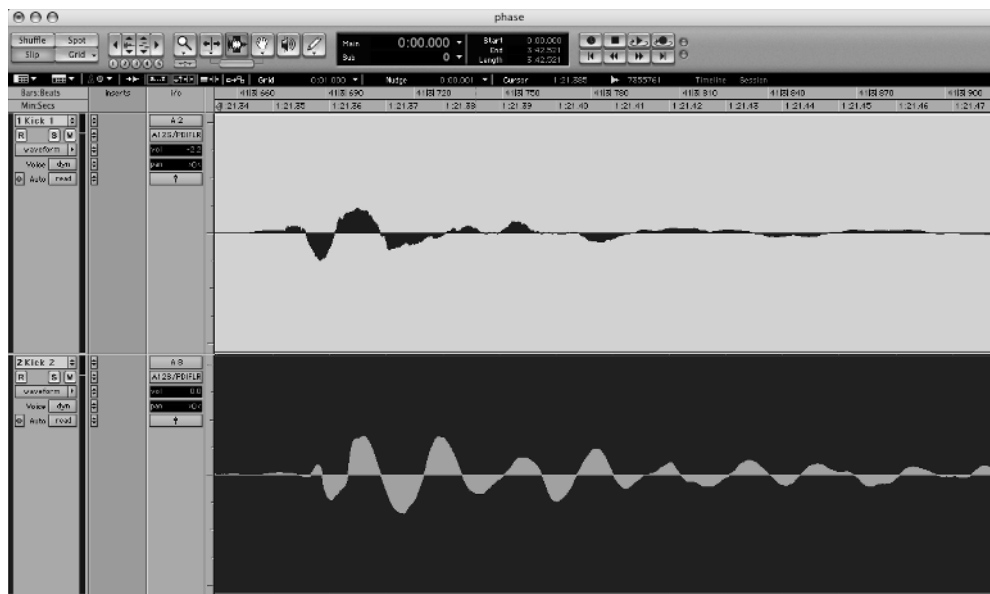


Figure 1.17 In-phase kicks

Phase

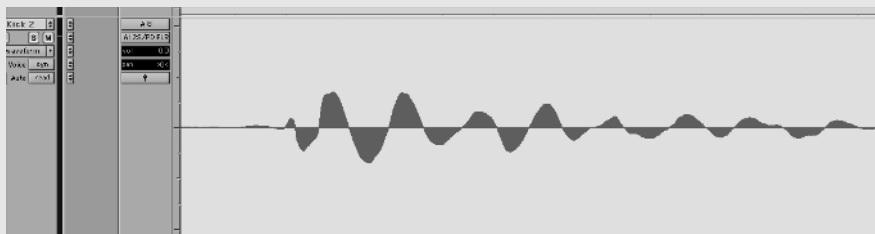
What does it mean to be “in phase” or “out of phase” when recording? Simply, when a single source is being recorded by more than one microphone, the sound hits the microphones at different times. Because sound travels through the air in repetitive cycles, the microphones are receiving the signal at different points in the cycles of the sound waves. When you mix the two signals together in playback, the two sounds may add or subtract from each other, causing certain aspects of the original sound to be either missing or emphasized.

Take the example of the drummer’s kick drum from our favorite band, The Condensers. When the beater is struck, air molecules are pushed into vibrations, which you measure in sound waves. These sound waves always travel at a constant speed, about 761 MPH, or the speed of sound. Although all sound travels at this speed, the number of times sound waves repeat their cycles is what gives different sounds their unique pitch. In the case of the kick drum, a lot of the sound being produced will have a lower pitch. You can measure the pitch of the sound made by how many times the vibrations complete their cycles in a second. This unit of measurement is called *frequency* (how frequently the waves repeat). In the audio world, you measure frequency as cycles

Continues

Phase (Continued)

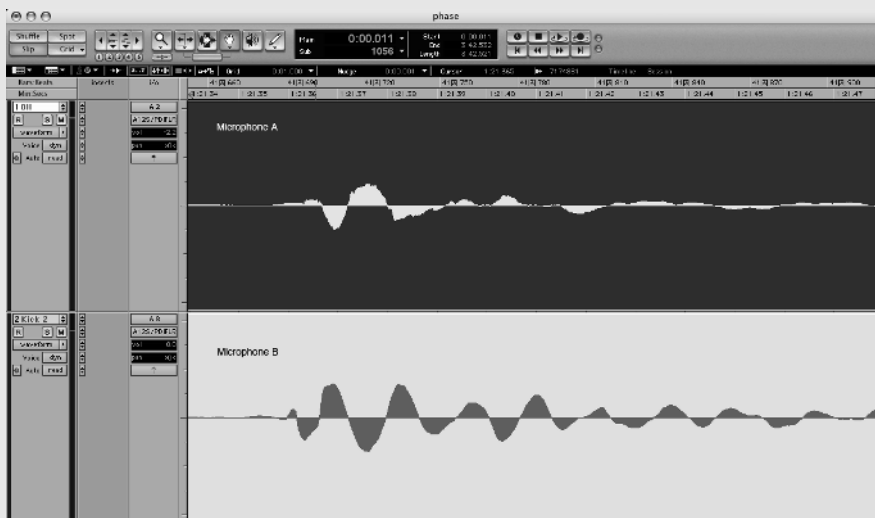
per second, or Hz. So the frequency of the sound waves determines the pitch of the sound. When you graph this out, or look at it very zoomed in on a Pro Tools track, you can see this waveform in detail.



Pro Tools's graphic waveform draws are extremely accurate and very useful when dealing with phase!

The waveform in the region called Microphone B is a graphical picture of the sound waves coming from a kick drum. When the sound hits microphone B, which happens to be inside the kick drum itself, microphone B records this waveform into the timeline.

When the same sound from the kick drum hits microphone A, which is an overhead microphone, it's actually slightly later because the microphone is farther away from the actual drum and it took longer for the sound to reach it.



Continues

Phase (Continued)

The phase issue occurs when you play both tracks together. Notice how microphone B's peak occurs at the same time as the valley of microphone A's waveform? These two conflicting signals will partially cancel each other out, resulting in a very un-phat kick drum sound. Not usually the goal!

So with this example, you can see that every time you use more than one mic to record, there will always be a phase *relationship* between all the mics in the room. Mics are rarely completely or 180° out of phase or in phase but rather slightly more in or out from each other. Don't let this scare you! There are some tried-and-true techniques to help you deal with this. Some engineers do not view phase as a problem but rather a bounty. It is our approach in this book to embrace phase relationships and use them to our sonic advantage. Keep in mind that Pro Tools always offers the advantage to slightly nudge one waveform forward or backward in time to help with the phase relationships. See the section "Dealing with Phase" for more on this.

Recording the Electric Guitar

In comparison to some other instruments, recording the electric guitar is a relatively simple, fulfilling, and zesty enterprise. In terms of different guitar tones, the guitar player, the effects pedals, and the amp settings are going to be as important as microphone placement and technique. But there are a few tricks of the trade to get that ungodly crunch or the wistfully expressive chime of the electric guitar.

The first myth to dispel with amp recording is that the size and loudness of the amp has anything to do with how it will translate when recorded. It's not the size that matters; it's how you use it. In fact, if anything, the bigger, louder amps will sound smaller in comparison to the smaller ones. Jimmy Page used a tiny 8-inch speaker, 12-watt Fender Champ to record some of his most raunchy guitar solos. That said, if it is overdrive you are going for, the amp gain has to be at a certain level for the amp to start achieving that lovely natural overdrive.

Start simple, with one cardioid microphone placed as close to the grill of the amp as possible, pointed directly at the outer portion of the best-sounding speaker in the cabinet (if there is more than one). The more you angle the microphone toward the center of the cone, the brighter the tone will get. Favorite microphones for this are the Shure SM57, Audio Technica 3035, Sennheiser 421, and Beyerdynamic TGX50 (yes, it is a kick drum microphone, but try it; you'll be pleasantly surprised).

If you want a more complex sound, add a room microphone about 5 to 8 feet in front of the amp. This technique works well in good-sounding rooms or even wood hallways. Condenser microphones work well for this, but be mindful of phase issues that will arise. Move the room microphone while listening to both microphones and also abide by the 3:1 rule and you should do fine.

Condenser, Dynamic, and Ribbon, Oh My ...

One of the most frequently asked questions we hear is, What is the difference between a condenser and a dynamic mic? And then, What is a ribbon mic? And then, of course, What does 48v mean and what is this phantom power stuff? All good questions indeed. Remember, each category of microphone (condenser, dynamic, ribbon, piezoelectric, etc.) refers to the physical method in which it converts sound waves into electrical current or voltage. Let's get to it and look at the three most common mic types.

Condenser

Condenser mics are the standard for most studio applications that require maximum clarity, subtlety, and detail in the recording—vocals, acoustic instruments, string sections, ambient room sound, and stereo overheads on a drum kit just to name a very few. Here's why we like them: They have a flatter frequency response than dynamics, and they are more sensitive and can pick up the subtle nuances of a performance more effectively, and because they are powered, they have a higher output than dynamic mics. And come on; with the proximity effect, every singer can at least try to sound like Bryan Ferry! Be careful though; these mics are easily overloaded. Monitor your preamps or board carefully to avoid clipping and distortion.

Here's the breakdown: Condensers need to be powered to operate. To power a condenser microphone, you can use an internal power source such as a battery or you can use externally supplied phantom power (see below). Physically, the condenser mic is a capacitor, converting acoustical energy to electrical energy using two plates containing voltage between them. The thinner outer plate acts as the diaphragm. The diaphragm vibrates when it is struck by sound waves, and as it moves back and forth it changes capacitance. This is then routed to the microphone wires as AC voltage and then out of the microphone via a three-pin XLR cable. It should be noted that the market for condenser mics has changed considerably in the past five years. One of the downsides of condensers in the past might have been the high asking price, but in this new millennium you can find condenser quality for a dynamic price. There have never been as many quality microphones available for such an inexpensive entry point. Competition is good. Some of the more popular condenser microphones include the Neumann U87, the TLM 103, the AKG 414, and the Audio Technica 4050.

Phantom Power

Phantom power is a method of carrying a DC current through audio cables to power condenser microphones as well as other gear. Condenser mics, being capacitors, need to be powered to operate. In the beginning, condensers had their own power supply and were bulky and cumbersome. Phantom power freed condensers from needing to supply their own juice. Condenser mics can get their power from an internal battery as well. Many people see the old 48v on the side of a

Continues

Condenser, Dynamic, and Ribbon, Oh My ... *(Continued)*

mixing board and never really know what it all means—one of the great mysteries of recording. The wiring for the audio traveling through a microphone is AC current, but to operate the microphone needs DC current too. Phantom power lets you feed both with one cable.

Dynamic

The venerable old dynamic microphone is the workhorse of the industry. More people have SM57s than Neumann M49s; that's for sure. Dynamics are valued for their price, durability, and ability to handle very LOUD music. Many types of dynamic microphones are unidirectional, allowing the microphone to focus in a certain area and reject any sounds from behind. On stage, the dynamics are more useful than condensers or ribbons because they don't distort or feed back as easily and require no power. Dynamics are more durable than condensers in general; toss an SM57 against a wall and you get a dented but functioning SM57. Toss a Telefunken U47 against the wall and you get a very broken, very expensive microphone and then probably lynched by every engineer and singer that has ever used one. Dynamic mics work on the principle of having sound waves move a diaphragm that is connected to a coil (moving coil). As the diaphragm moves, the coil connected to it moves across a magnet, creating a current in the coil, which is carried away from the microphone along the internal wires to the cable. Some of the more common uses are toms, snare, kick, guitar amps, and horns. Some of the more tried and tested dynamics are Shure SM57 and SM58, Sennheiser 421, and AKG D-112.

Ribbon

Well, if you know about ribbon mics, then you can wipe up the drool from the page now. How about the RCA 77A, one of the most famous microphones of all time? Developed in the late '20s, ribbon microphones need no external power supply and deliver some of the most sought-after sound ever recorded. You find fewer and fewer of the vintage ribbons these days for a number of reasons, but newly manufactured ribbons are on the rebound. The color of sound they can capture is almost an opiate to some engineers. Talk about them and their eyes glaze over. They are very fragile microphones and can be ruined in an instant if mishandled. This is how they work: A metallic ribbon is placed in a magnetic field, and as sound waves vibrate the ribbon, it creates a very small electric current. Because of this, the microphone has a very low output and needs a clean and powerful preamp to crank it up to suitable levels. Most ribbon microphones are very sensitive at the front and back (0° and 180°) but dead on the sides (90° and 270°) and can be as warm and lush as an afternoon on a tropical beach. Now if we can just get a gig recording wave sounds (sound waves) all day, we're good! Some vintage examples of ribbon microphones include the Coles 4038 and the Shure SM33 as well as the RCA dynasty of ribbon microphones that inspired a generation of broadcasting and recording. The newer class of ribbon microphones includes Royer Labs R-121 and Beyerdynamic M-160.

A word about ribbons: Because of their mellow high end, ribbon microphones work especially well in capturing a classic, warm, vintage tone on clean guitars. There's nothing like a Fender amp with some tremolo and spring reverb going into a Coles ribbon microphone. Sweetness.

Capturing the Acoustic Guitar

How could something as innocent and pure as a simple wooden acoustic guitar cause so much strife when you try to record it? The acoustic guitar can be a very elusive instrument to capture. The reasons are many. For such a simple device, it produces quite a complex sound, one full of rich overtones, dynamic transients, and a full frequency range of full low end and brittle treble. Getting the acoustic guitar to translate in a recording exactly how it sounds in the room it is being played in pushes recording tools and technology to the limit of their capabilities. There is a method to the madness, however, and the following techniques will get you closer to achieving that captivating acoustic guitar recording into your Pro Tools session.

Line 'em up! When you know you are going to track an acoustic guitar, the first thing to do is gather as many of them as you can. Even though DJ turntables outsell guitars to young American teens, you know you have a few friends who have one lying around. You will be surprised at how a certain unlikely cheap guitar will sound more appropriate than a very expensive Martin for the song you are recording. Another surprising fact is that often the large-bodied D-28-style instruments will actually sound tiny in comparison to the little parlor-style acoustic half its size, which sounds huge when mic-ed up.

Be aware of the room you are recording in. The sound of the acoustic guitar has a lot of live energy that will react to the surface of the room, so the same guitar will sound much different in a small deadened room than it will in a large live wood one.

The next thing to ask yourself is, What role is the acoustic going to play? Is it the intimate central element of a sensitive folk song, or is it playing a strummed support role along with electric guitars, keyboards, drums, and bass?

For that intimate sound, start simple. Place one cardioid condenser microphone 6 to 10 inches in front of the sound hole pointed directly at the 12th fret or where the body meets the neck. The Neumann KM 184 and the Schoepps 221 are good small-diaphragm microphones to try, while it is possible also to get outstanding results with large-diaphragm condensers like the Audio Technica 4040, the Neumann U87, and the AKG 414. The slightest movement here will mean a lot, so if you angle the microphone more toward the sound hole, you'll get more bass, and if you angle it more toward the fingerboard, you'll get more string and fret noise. Listen carefully to the performer playing the part they intend to record as you angle the microphone. It also helps to get on your knees and use your ear to locate the sweet spots around that general area.

For a wider, bigger sound, you have two options. The first option is to add another “over the shoulder” microphone. This microphone should be similar in type to the ones mentioned previously. It can be placed about 1 foot above the right shoulder of the (right-handed) performer, and the capsule should be angled downward at the body of the guitar. Be careful not to point the capsule directly at the sound hole or the sound may get too boomy. This microphone should be able to pick up the guitar’s body, while the other microphone can play a part in capturing more intricacies in the picking. Since you are using two microphones, heed the laws and rules of proper phase correlation as outlined in the sidebar on phase. Once the part is recorded, panning the two tracks a little apart from each other opens up the sound of the guitar in the mix (see Figure 1.18).

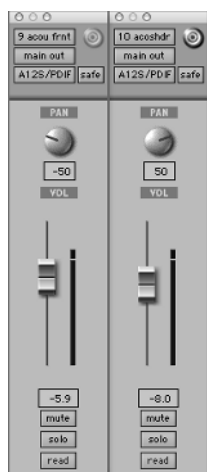


Figure 1.18 Two acoustic tracks

The other option for getting a sound more appropriate for strummed support tracks is to go stereo. This requires two identical small-diaphragm microphones. AKG 451s or Neumann KM 184s do the trick well. The capsules of the microphones should be angled 90° from each other, and the point where their angles converge and should be pointed at the 12th fret about 7 to 12 inches away.

Getting That Bass Guitar

Big bottoms anyone? The idea here is to capture the low end in a way that has a deep bass response but still retains its definition in the mix. Surprisingly, the upper mid frequencies are almost as important as the lower fundamentals in making this happen.

Many engineers are split about whether it is preferable to record electric bass through a direct insert box, or DI, directly or to mic the bass amplifier’s cabinet. We say if you have the channels, do both. Both sounds have characteristics all their own that can be mixed together at a later time to get the perfect sound. Pro Tools’s accurate waveform editing makes it easy to align both signals in phase, and hard drive space is cheap as ever, so go ahead and record both signals!

Polar Patterns

We're switching polar patterns, and we're not talking global warming!

The polar pattern on a microphone describes the physical area that is sensitive to approaching sound waves. So even though sound waves may surround the entire microphone, by selecting different patterns, you can determine which areas of the microphone will be sensitive to the waves. This allows an engineer to either focus a microphone's sensitivity or not, depending on the situation.

Unidirectional

When you need to isolate the microphone on the old Marshall stack as much as possible, a unidirectional microphone pattern is used to get the most of the amp and the least of the sound around it. Its typical polar pattern (cardioid) allows sound to enter from the front and, to a lesser extent, from the sides but offers maximum rejection to sound from the rear. The unidirectional microphone is also subject to the proximity effect. The proximity effect exaggerates the lower frequencies when a sound source is less than a foot away from the sensitive area of the microphone. This can be a cool trick to bring out bass response, but be careful how close you get! There are two other types of cardioids: supercardioid and hypercardioid. Both are similar to the cardioid's pattern but offer further refinements. Supercardioid has a more narrow focus than cardioid (115°), which helps in isolating a recording but has a bit of pickup in the rear. hypercardioid, with the tightest focus (105°), has the best side rejection from a unidirectional mic so it can really eliminate bleed, feedback, and background noise but opens up to more rear pickup. Hypercardioid is the typical microphone type built into a video camera because of these qualities.

Omnidirectional

Now say perhaps you want to record a cello in a church and the church sounds like Abbey Road Studios; then you might choose an omnidirectional microphone so you get the instrument and the room sound as well. An omnidirectional microphone records in a 360° field and has no proximity effect.

Bidirectional or Figure 8

If you need to record an interview with two people and have one microphone, then a bidirectional mic is the ideal choice since it records from the front and back (0° and 180°) equally well but rejects sounds from the sides (90° and 270°).

With a basic knowledge of these patterns, you can be much more effective getting the right sound by design, not by accident!

A good bass tone always starts with the player, the gear, and the bass. Nothing stands in the way of a well-intonated instrument, a dynamically consistent player, and a solid-sounding amp. However, if you've got those things at your disposal, you are going to want to capture them correctly.

When mic-ing the cabinet, it is best to use a large-diaphragm cardioid polar pattern microphone, either dynamic or condenser. This type of microphone will react best to the characteristics of the bass, and it is possible to take advantage of the coveted low-end response by utilizing the proximity effect that occurs in this polar pattern. Good dynamic microphone choices are the Sennheiser 421, Electro Voice RE-20, and Beyerdynamic TGX50. Suitable condenser mics are the Audio Technica 4060 or 4050, AKG 414, or for unlimited budgets, a FET or tube Neumann U47.

Place the microphone 6 to 10 inches from the best-sounding speaker cone in the cabinet. Experiment with the mic placement relative to the cone. The closer the capsule is to the center of the speaker cone, the more bright the sound will be, while moving it to face the outer edges of the cone will make for a warmer, darker sound.

A DI is a box that converts the line output of the bass, which is a high impedance signal into the low impedance signal that your Pro Tools interface likes to see (or hear). Like mic pres, all DIs are not created equal and will have a large impact on the sound you get. Popular DIs on the market are the Sans Amp, Countryman, and Yamaha. It is important to note here that many amplifiers have extensive tone controls and a built-in line-ready output. This can be a good way of getting a direct signal too.

Going for the Lead Vocal

Think about your favorite songs. Put yourself in that time and place. What is it you remember the most? That special voice, the incredible harmony, and the chorus you can't get out of your head? Robert Plant at the end of "Stairway to Heaven"? Vocals are a main part of the story in modern music, but tracking a good-sounding vocal is and always has been a challenge.

It's common to use a large-diaphragm condenser mic with a cardioid pattern to track most vocals when the dynamic range is not excessive. This can get a great sound but you can overload the mic if the singer is really pushing it.

Take the mic and find the spot in the room with the least amount of sonic debris (listen). Place the singer's mouth about an open hand's length away from the mic, about 8 to 12 inches, and position the mic above the mouth slightly. This is a good starting point for something with some dynamics in it. Not full on screaming, but good old rock and roll. If you want to get a more intimate breathy sound, position the mic about 3 to 6 inches away from the singer's mouth and about an inch or two above the mouth to keep out unwanted pops and such. The cardioid mic suffers from the proximity effect, so be sure to utilize this to the fullest.

Let's say you have some LOUD vocals to record. Time for plan B. You can back off the mic to about 1 to 3 feet away from the singer's mouth and about 8 to 10 inches above the mouth. Or you can go with plan C, which is the trusty SM57 or Sennheiser 421. These mics will deal with your problem. It's a different tone to be sure, but who says it's not a good one? Remember to get close to the microphone and watch where you point it because it is very directional.

Now the preamp comes into play. Preamps are cool. Use a tube preamp on the front end of a digital system to give your sound that analog color. Use a solid-state preamp for more clarity and detail (see the sidebar "Preamplifiers"). A good preamp by itself will not save a bad recording, but it will make a good recording sound better. It is a vital part of the recording chain. Monitor these levels carefully and don't overload the signal. The preamp feeds the compressor.

Depending on who's singing or rapping, you can bring in some compression to the chain. A little compression on the way in goes a long way, but often it is necessary. A good starting point is a 2:1 ratio on your compressor. This means for every db going in above the threshold, there is 1/2 db coming out. A ratio up to 5:1 is typical for vocals, and a ratio above 10:1 becomes noticeable as an effect. The higher ratios act as limiters. Less is more is the rule of thumb. If you overcompress as you record, you can't fix it later; besides, the cool compressors in Pro Tools are half of the fun! The compressor feeds Pro Tools.

When audio comes into a Pro Tools system, you need to control the level from the devices in your mic chain. Be sure to monitor all of your devices and watch for overloads. If you overload your mic chain anywhere, the distortion will pass through everything after that. Think of these things when you record:

- Mic choice
- Placement
- Preamp
- Compression
- EQ
- Levels

Other Vocal-Recording Considerations

Every recording technique involves some background knowledge in order to fully utilize the concepts. Here is a handful of things to consider when planning your vocal recordings.

Preamplifiers

Why is a preamp necessary? Microphones are analog devices. When you record with them into the digital world that is Pro Tools, you take this analog information and convert it into a digital signal. One of the things your Pro Tools hardware interface does is this analog-to-digital (or A-to-D) conversion. The analog information microphones provide comes in the form of electrical AC current (see the sidebar “Condenser, Dynamic, and Ribbon, Oh My . . .”). When this current comes down the cable of the microphone, its signal isn’t optimized for the converter. In essence, it is too low. This is where the mic pre comes in: The preamp amplifies the signal to the optimum level for the A-to-D converter. To make a long story short, the mic pre comes between the microphone and the Pro Tools A-to-D converter, and we use the gain control on the mic pre to set the level coming in to Pro Tools.

Are all preamps created equal? No. If you’ve snooped around the audio marketplace at all, you will be aware of hundreds of different preamps out there, all shapes and sizes, all different colors and flavors. Remember, unless it is built into the interface, like on the 002 or Mbox, higher-end Pro Tools systems require an external preamp. Of course, this doesn’t mean that you have to use the built-in preamps, either.

So what makes them better or worse? For starters, different types of preamps actually sound different. In addition, some preamps may actually work better or worse with different microphones. Many engineers are constantly on the hunt for that elusive “match made in heaven”—the perfect combination of mic and preamp.

So what makes them different? For one, you get what you pay for. The price is going to have everything to do with the quality and ultimately the sound of the preamp. Generally, the more expensive preamps are made to more consistent specifications and with higher-quality components. Within these guidelines, however, there are differences, the two main ones being tube or solid state.

Tube preamps are often held in a mythical light because they were all that was available in the “golden age” of recording. Certainly there is something to that classic tube preamp sound. This is due mainly to the fact that when tubes respond to fast transient information, they tend to “round off” the edges of sudden sharp sounds, like drums and other percussive sounds. In addition, the “color” of the tube preamp that many describe as “saturated” or “warm” is actually due to the tube circuit’s natural distortion. Many modern manufacturers are putting tubes back in their products to emulate this classic sound. One of the more famous examples of a vintage tube preamp is Bill Putnam’s 610, while a popular modern tube preamp is the Avalon VT-737SP.

Solid state preamps can react much more quickly to transient information and ultimately can be more objective in their sound. A vintage example of a solid state preamp is the API 512, while a popular modern one found in studios is the Focusrite Red series preamp.

The Mic

Well, this is a loaded topic. Who do you infuriate first? Just kidding, but really people do have strong opinions about microphones. Every microphone has its day, but where and when it does is the question. Since we are talking about recording vocals, we will start with condenser microphones.

Condenser microphones are the most common microphones used to track lead vocals in a studio setting. On stage, you would use a less sensitive dynamic mic, but in the studio where everything can be controlled, fragile microphones can be treated like royalty. Condenser microphones offer more detail and clarity than dynamic microphones, so when you want to record the most accurate vocal performance, a condenser is a good choice.

The ribbon microphone is highly sought after for broadcast and vocal work. It has a classic sound that recorded a generation of artists. Although vintage ribbons are expensive, rare, and delicate, there is a growing trend to manufacture ribbon microphones again. A number of companies are putting out a quality product.

While it's not the party line, we like using dynamic microphones to track vocals if we need a unique sound or something sounding a little vintage. These mics can take a real load before they break up, so think screamer! All it takes is one punk experience to teach you about protecting your condenser and ribbon diaphragms from Sid "wannabe" Vicious.

The Room

Before we talk gear, we need to talk room. Try to treat the room with some nonreflective material, and at the very least, deaden the area behind the singer a bit. Try not to be too close to a wall and not in the center of the room because of the standing waves. Off of the middle a bit is a good start. Always remember to be interactive with the microphone. Move it around and listen to what the room sounds like. Test, probe, and question at all times!

The Mood

The last thought on vocal recording would be the environment. As a producer, it is very important to do what it takes to put your artist in a position to succeed. You need to be part baby-sitter, part therapist, part mind reader, and part bad cop to get the job done. Candles, incense, insults, jokes—you do it all to set the mood.

Overdubbing and Punching without Fear

One of the most difficult and dangerous (yet exhilarating) aspects of analog recording was the old punch-in. Punch out too soon and you don't get the entire take. Punch out too late and, oh boy, you've got problems. Maybe you just erased part of the only good vocal take your singer had in him. Worse than that, maybe you just ruined a client's track and now the free studio time starts. No undo! We have all done this, but it still

hurts. You will never ruin a track with automated overdubbing in Pro Tools! So if you have a vocalist who just gave a great vocal performance except for one phrase, no problem. To start with, you have to configure Pro Tools's monitoring system to best fit your vocalist's needs.

Selection-Based Punching

There are two modes in Pro Tools for record monitoring: Auto Input Monitoring, which is the default mode, and Input Only Monitoring. You can toggle between the two from the Track pull-down menu or by using Option+K / Alt+K (see Figure 1.19).



Figure 1.19 Here is where to change your input monitoring mode on LE systems

Auto Input Monitoring mode is the most common mode used when overdubbing. Auto Input Monitoring mode configures Pro Tools to play back the audio on the track up until the punch-in point and then switches to monitor the input source. After punching out, Pro Tools switches back to monitor the audio already on the track. So you hear the track, then you hear the microphone, and then you hear the track again. This entire process is selection-based, as are most things on a computer. Select the audio you need to replace and a little extra in the beginning and end of your selection. Performers are notorious for changing their phrasing and the extra selection will help accommodate for this. Don't be afraid to come out of grid mode to slip mode (F2) to make your selection. (See "The Many Modes of Pro Tools" in Chapter 3 for more on Pro Tools modes.) This method is foolproof, and even in the worst-case scenario, there is always everyone's best friend, Mr. Undo.

Since it is difficult for even a seasoned musician to punch in with no lead-in, you need to set up the pre-roll. Pre-roll can be enabled on the transport bar or by pressing ⌘+K / Ctrl+K (see Figure 1.20).



Figure 1.20 Pre-roll enabled

Once pre-roll is turned on, you need to give it a count. How much time does your vocalist need to feel comfortable with the vibe of the track before coming in? Everyone is different in this, so try 1 bar and go from there. To enter the countoff, click in the Pre-Roll Amount window in the transport bar and type in the desired amount.



Note: Always press Return or Enter to lock in changes when you manually enter numeric data in Pro Tools.

The Main Counter Selection determines the kind of counting for the pre- and post-roll. This can be Bars and Beats, Minutes and Seconds, and Samples in LE and Timecode and Feet & Frames in TDM. At this point, you are ready to go. Press 3 on your numeric keypad to record, and let it rip. Automatic, every time.



Note: Use Input Only Monitoring mode when it is not desirable for the performer to hear playback of their previous takes. This is good for those situations in which the performer focuses on the previous takes instead of concentrating on the take at hand. This allows you to make your punch selection without informing the performer exactly what is being punched.

QuickPunch Mode

At first glance, QuickPunch mode seems like a throwback to the analog style of punching in and out. It is enabled from the Options menu, or by Control+clicking / Start+clicking the record button in the transport window until a little *P* appears inside the red button (see Figure 1.21).



Figure 1.21 QuickPunch mode

With QuickPunch enabled, you get to manually click the record button or press F12 when you want to record onto a record-enabled track. This happens in real time during playback just as on a tape deck. This is handy, but there are a couple of hidden features lurking in QuickPunch mode that are incredibly valuable to your recording chops:

- QuickPunch is the only way you can punch in and out of a track more than once without stopping playback. So if you have three words interspersed through-

out a singing verse that need to be replaced, you can let the track play and grab those three spots without having to stop and locate each one separately.

- During QuickPunch, Pro Tools is secretly recording the whole time when the track is playing back, regardless of when you manually punch in and out. This means that if you missed the in or out point, you can trim the handles of the regions and have access to all of the performance—a real lifesaver!

Note: Use QuickPunch mode when you are doing an ordinary selection-based punch-in with a pre-roll and post-roll enabled. This way, if your performer or voice-over talent jumps the gun or holds out a note longer than your punch-in selection, you will have access to the material that would have been chopped off in a normal selection-based punch.



Loop Record Techniques

If you're the type of Pro Tools user that wears multiple hats in the studio (engineer, musician, producer), you'll find the Loop Record feature very useful. Sometimes when recording yourself, it is easy to get frustrated with having to simultaneously start and stop recording, reset the playback cursor, and set up for an additional take. With Pro Tools's Loop Record, you can set up a loop record selection that will record again and again without having to stop after each take. Loop Record is also especially useful for helping musicians, voice-over talent, and foley artists nail difficult passages by repeating them over and over without any downtime. As if that wasn't cool enough, after loop recording, you can make a composite of all of these different performances that may have never been possible to achieve in one take. Because Pro Tools keeps accurate track of all the subsequent takes, it is easy to audition and recall each take and edit them together.

Making a Selection

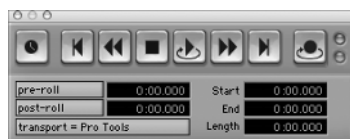
On the record-enabled target track, make a selection for the appropriate amount of time. This could be the length of a verse, or in the case of Figure 1.22, it is an eight-bar guitar solo. It is important here to make a selection a bit longer than the passage because any pre-roll setting will be active only on the first pass. To make sure the performer knows where they are relative to the song, it is sometimes necessary to include some extra space before the first note of the intended recording start time. In this example, the selection ended up being 10 bars total to give 2 bars pre-roll at the beginning of each looped take.



Figure 1.22 Loop Record selection

Enabling Loop Record

You can turn on Loop Record in the Options menu, by Control+clicking / Start+clicking on the record button in either the dedicated transport panel in the Edit window or the floating pop-up transport bar two times so that the red record button has a loop around it.



Record

To begin recording, press \mathbb{C} +spacebar / Ctrl+ spacebar, Keypad3, or F12. Note that if you are recording yourself, you can always let this first pass go to get situated behind your instrument or microphone. When the recording gets to the end of the selection, it will start over at the beginning again and repeat infinitely until you hit stop.

Note: In Macintosh OS X, $\text{⌘}+\text{spacebar}$ is configured to open Spotlight and the F12 key launches Dashboard by default. If you wish to use these key combinations to begin recording in Pro Tools, you must first disable them in the Macintosh System Preferences.



Edit

Once you have recorded multiple takes you will notice a new *whole file* named after your track in boldface type in the Regions list (right column in the Edit window). This whole file is actually all of the takes you just recorded combined in one long file. Below this you will see a list of takes with numbered suffixes after each take. These are separate versions of all the takes you just recorded! In other words, these are *region definitions* that point to each take separately in the larger whole file. (All of these files are highlighted in Figure 1.23.)



Figure 1.23 The guitar solo was looped seven times, as indicated by the regions numbered 01 through 07 after the name **gtr solo_01**. The **gtr solo_01** in boldface type is all of the takes combined into a single whole file.

After you finish loop recording, the last take (region named gtr solo_01-07) remains in the track's playlist in the Edit window. Now go to the Pro Tools > Preferences > Edit tab to set a preference that will allow you to listen to and compare all of the takes quickly and efficiently on the same track without having to move anything around (see Figure 1.24).

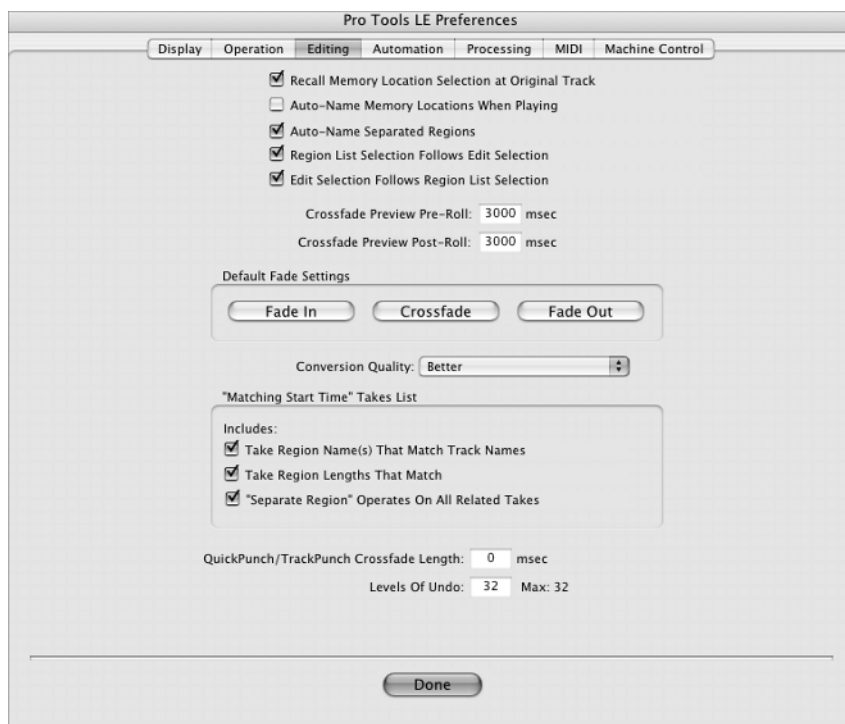


Figure 1.24 “Matching Start Time” Takes List preferences

Because all of the loop-recorded takes share the same track name and are subsequently the same length, checking the “Matching Start Time” Takes List options titled Take Region Names(s) That Match Track Names and Take Region Lengths That Match will make Pro Tools think of them as all belonging together.

Select the region of the last take (gtr solo_01-07 in Figure 1.26) by double-clicking on the region with the Selector tool. ⌘+Click / Ctrl+click on the region with the Selector tool and you will get a pull-down menu showing all related takes. This is a list of all of the loop-recorded takes right in the Edit window. You can select and listen to all of the looped performances right from here. Make sure you use the Selector tool and the entire region is selected for this to work (see Figure 1.25).



Figure 1.26 Here the region is separated four times, with sections of four different takes used.

Once the layers are cut, you can mix and match different parts of the take to arrive at a final composite. This could be something that is not even physically possible to play in one take. Is this cheating? Ask producer/musician Brian Eno, who did this back in 1972 on *Another Green World* by splicing tapes of many different guitar performances. This type of composite editing is common in the history of multitrack recording, but with Pro Tools it just takes a whole lot less time!

Clean Up

You will notice that this kind of editing makes many different numbered versions of regions in your Regions list. It may be useful here to select all of your final composite in the Edit window, choose Edit > Consolidate, and make a new whole file out of your comped performance. Once this is done, you may want to select Unused Audio Except Whole Files from the Regions list pop-up menu and get rid of all the unused regions you created while making your composite. You can read more about this type of cleanup process in Chapter 3.

Alternate Playlist Recording Techniques

Using alternate playlists is one of the most powerful and versatile techniques in Pro Tools. While the concept of recording multiple takes of an instrument or vocal performance and then “comping” or combining the takes into one cohesive performance has been around for decades in the recording world, never has it been so fast, painless, and easy as it is with Pro Tools’s alternate playlists. The audio regions and the placement and order of those audio regions on a track in Pro Tools is deemed that track’s playlist. Pro Tools lets you store and save an infinite number of these playlists within a track as alternate playlists. This means that every track can contain additional recording takes or other arrangements, ready to be called up instantaneously and edited from. This technique can help save valuable screen real estate, time, and energy while recording. It’s all about efficiency here, so you have more time to be creative.

It’s the moment you’ve been waiting for: time to record the lead vocal. Depending on the singer, vocals recording can be tricky. Some singers (even very popular ones) require many takes to get their part correct. Others need a few takes to warm up. Using alternate playlists, there is no reason why you shouldn’t be recording all of the takes. Singers sometimes hit their best performances right away, while other times they hit a peak halfway through the session and everything goes downhill from there. A lot of the time it is hard to judge these performance nuances in the midst of a tracking session. By using alternate playlists, you can go back to previous takes and compare and contrast. You also have the option of selecting the best parts of each take and combining them right then and there during the tracking session.

Record with Alternate Playlists

The Alternate Playlists menu is in the small pull-down menu identified with an up/down arrow to the right of the track name. If it is the first time you open this in a session, you will see New, Duplicate, and the name of the track you are in (see Figure 1.27).

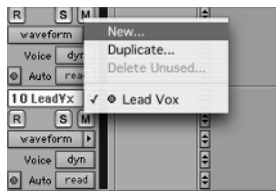


Figure 1.27 The alternate playlists menu

After the first take has been recorded, you can choose New from this list and Pro Tools will prompt you to use your track name_01 or allow you to rename the new playlist you are creating. You may find it useful to use the names Pro Tools gives you since they are created numerically, but you can rename it if you wish. When you click

OK, the track will appear empty with your new track name. Don't worry. Your previous take is not gone; it is still available from the same pull-down menu, but you now have a new, clean playlist to record an additional take. You may do this as many times as you wish, and although each track will contain more media on your hard drive, all the different takes are contained neatly within the one track in the Edit window. There is also no additional strain on system resources to have as many of these as you want hiding inside your track.

Edit on the Fly with Alternate Playlists

Once you have a few playlists in the track, your Alternate Playlist menu will look like the one in Figure 1.28.

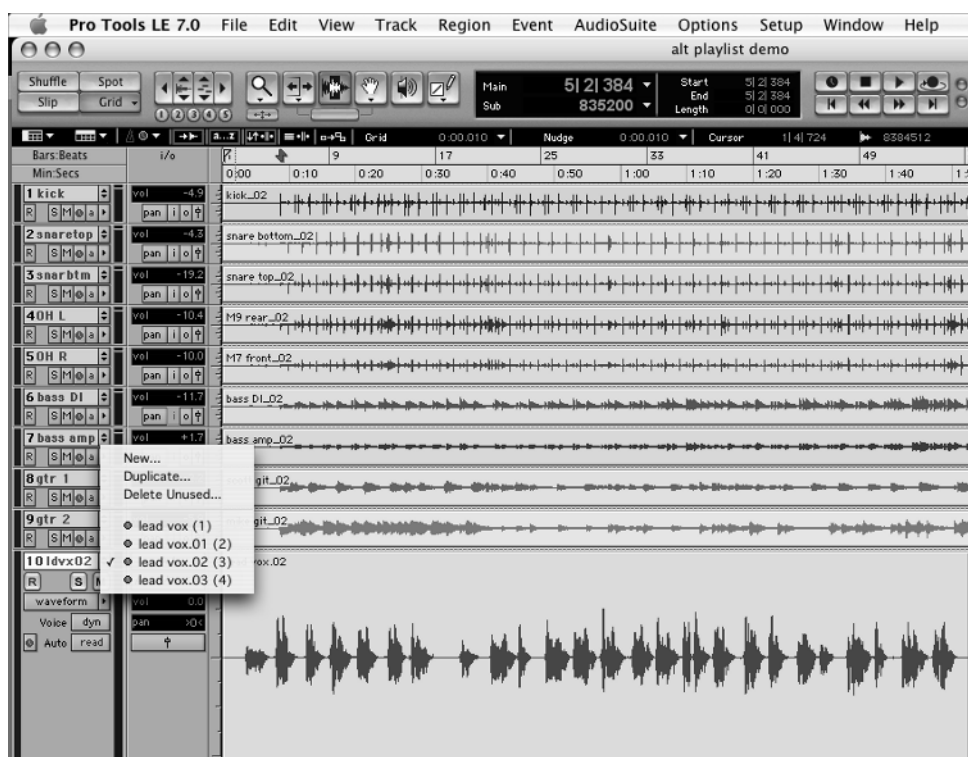


Figure 1.28 After several playlists have been added

You can now pick and choose between all of these performances instantly by calling up each take and listening one by one, or you can make a master take or a comped track (see the sidebar "Comping"). A good technique here is to make a new playlist titled comp or master and put the most solid performance in the track. Then,

using simple copy (⌘+C / Ctrl+C) and paste (⌘+V / Ctrl+V) editing, you can assemble a track with all the good parts. Pro Tools makes this easy because it keeps your selection in the same spot while you call up different playlists. Here's how:

1. Find your best performance; in this case it is titled lead_vox.02-01.
2. Open the Alternate Playlist menu and choose Duplicate.
3. Name the new track Vocal Master.
4. Select a part of a phrase from another take that you want to use; in this example, it is the last two words of a verse in the lead_vox.03-01 track (see Figure 1.29).

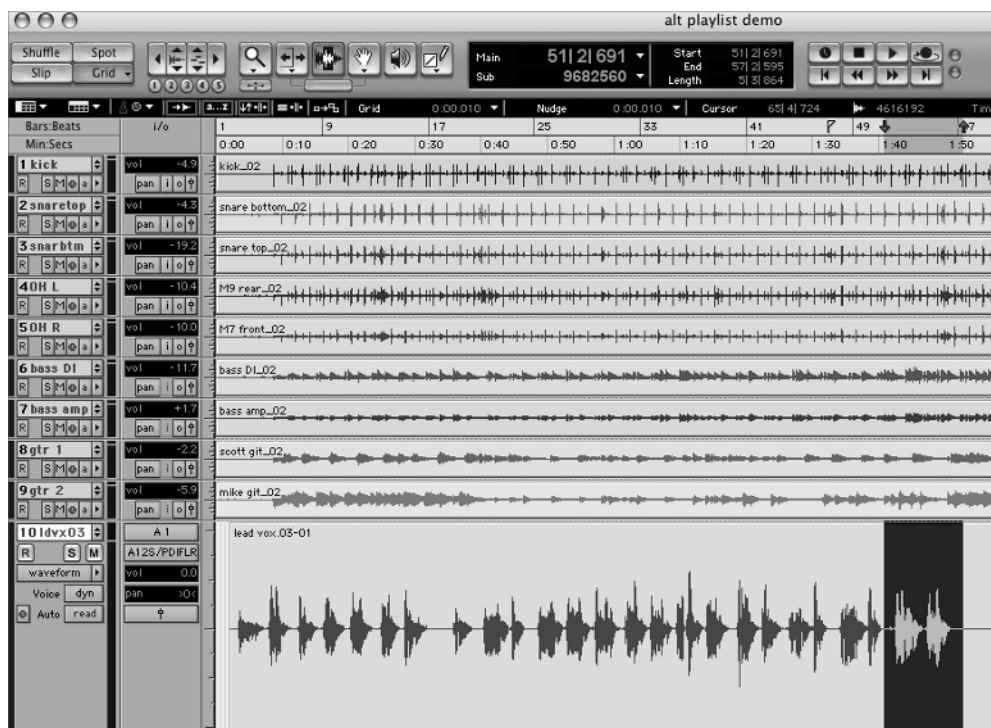


Figure 1.29 A selection from an alternate playlist

5. Copy to the Clipboard.
6. Move to your comp or master playlist track. Notice how the selection stays in exactly the same spot between playlists.
7. Paste your selection (see Figure 1.30).
8. Repeat as many times as necessary to make a solid track.
9. When you're finished, it is a good idea to check all of the transitions, cross-fade them together, and consolidate the selection. You'll learn more about this in Chapter 3.

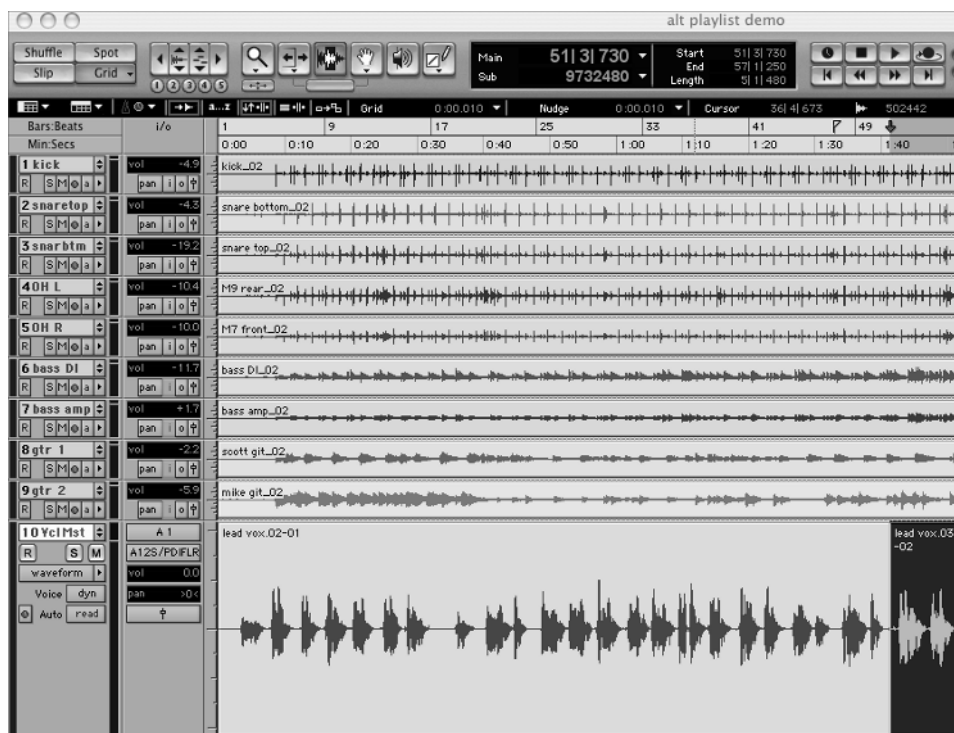


Figure 1.30 Paste from one playlist into another

Comping

Comp is short for *composite*. The term is used often in recording to refer to tracks that are assembled from a slew of performances of an instrument or vocalist recorded in different takes and then put together to form a final track. This technique has been around since the beginning of multi-track recording. In the tape-based analog world, comping required the use of many tracks, which then had to be bounced and mixed through the mixer onto a final comped track. This was a tedious, time-consuming process that ultimately resulted in a final comped track that was always one generation away from the original recording, thus increasing noise floor and tape hiss. In the digital world, you can comp tracks together quickly and easily without any generation loss.

Wait for Note and the Lonely Guitarist

Just about every piece of music software since the mid '80s has a function called Wait for Note. It's a MIDI function, but we are liberating this technique to use in an audio application. You need something that will trigger MIDI to do this. A keyboard is the obvious choice, but there are many other options today. A MIDI controller is incredibly

valuable in a modern studio because of its ability to work as a control surface as well as a keyboard. So you need to track a guitar part but everyone else has gone home. The guitar rig and the computer live in different parts of the room. Hmmm... This is a fairly typical situation in a home studio as well, where people often work alone (and late at night).

Set up a MIDI keyboard as close as you can to the spot where you will track the guitar. If you have a sustain pedal, this works best, but anything that will trigger MIDI will work. Now, back in Pro Tools, record-enable your audio track, no MIDI track required, and then if it's not visible, show the transport bar (⌘+ Keypad1 / Ctrl+Keypad1), enable the wait for note button (the pause symbol over a MIDI cable) and start recording (see Figure 1.31).



Figure 1.31 Wait for Note enabled

At this point the computer is ready to record as soon as you trigger the keyboard. It will wait for as long as you need to get comfortable and ready to play. Step on your sustain pedal or just press any key on your MIDI controller, and red lights for everybody.

Dealing with Latency while Recording

Latency is the dirty word of digital audio. Note that the first four letters of the word spell *late*. This is what latency is all about. When you experience latency, you hear things later in time than you are supposed to. Although there are many advantages of digital audio over analog, latency is an example of a new set of problems and challenges that didn't exist in the analog world. Everything essentially moves at the speed of electricity in the analog world, which is fast enough not to cause any problems. In digital audio, especially the host-based systems like Pro Tools LE, in which all of the audio processing is handled by your internal CPU, the number-crunching required to run plug-ins, combine streams of audio in the mixer, and record in real time actually takes a few milliseconds to process. For LE systems, the length of the time lag will differ depending on the speed of your processor and the amount of RAM you have. Ever wondered why the higher-end Pro Tools TDM systems cost so much more than LE systems? This is one of the reasons. *TDM* stands for *Time Division Multiplexing*, a technology used by Digidesign that solves many of the problems of latency and lands audio perfectly on time, even with a lot of plug-in processing and many streams of audio happening at once. But do not fret; there are a few ways to deal with latency to make it a non-issue. As long as you identify and acknowledge the problem of latency, you can get around it to make seamless and professional-sounding recordings even in the LE world.

Listen to Your Host: Adjusting Host-Based Settings in Pro Tools LE

Pro Tools LE is a host-based system, which means it is the only system that has the internal CPU handling all plug-in processing. The first thing to look at when dealing with latency on an LE system is the playback engine preferences in the Setup menu (see Figure 1.32).

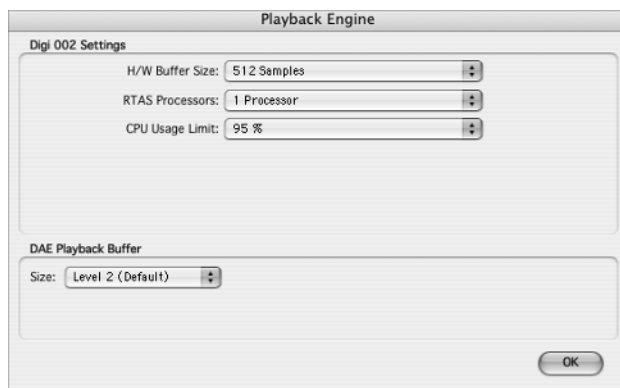


Figure 1.32 Host-based performance settings

In here you will find H/W Buffer Size settings, which are the keys you will need to unlock the latency issues. The lower you can get these numbers, the lower the latency will be. In other words, you'll be playing that killer software synth plug-in via your MIDI controller and you won't experience that crappy time lag that can occur with a high latency. The trade-off is performance. The lower the H/W Buffer Size setting is, the harder it is for your computer to keep up with all the processes it is running. The key here is that you can change these settings anytime, even with a session open. So when you are recording and having no latency is really crucial, you can lower the numbers. Then, when it is time to mix and you need all the plug-ins you can shake a stick at, and latency is not an issue, you will want to set those numbers to the maximum amount. If your computer buckles and you get an error message like the one in Figure 1.33 while playing back audio, then you have hit rock bottom and won't be able to continue at this buffer setting.

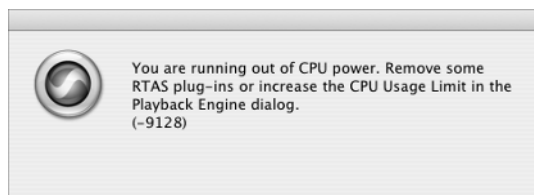


Figure 1.33 You get this dialog box when CPU usage is too high for the playback engine settings.