Mixed Signals

Barry Cleveland
Nov 1, 2000

Mixers are about signal flow. A good recording mixer lets you route a variety of input signals wherever you'd like and combine them into a stereo, mono, or even surround mix.

Audio travels through a mixer on a signal path. However, signal paths is more accurate because a signal is often sent to multiple destinations simultaneously, and sometimes a signal leaves the mixer and comes back in again. In analog mixers, audio signals travel through wires and circuits and are routed to various physical modules along the way. In digital mixers, the audio signals are converted to digital information at the inputs, and software does all of the routing and processing. (Of course, you can also route already-digitized audio in and out of most digital mixers.) Audio signals never make it past the converters, much less down any wires (see the sidebar "Digital Differences").

Signals can flow in and out of physical components via actual wires (analog mixing), or a CPU can process a stream of zeroes and ones (digital mixing). Regardless, analog and digital mixers perform many of the same functions, and their user interfaces share many characteristics. For clarity I'll speak mostly of analog signals moving from one point to another along physical paths. Just bear in mind that the basic concepts also apply to digital mixers.

Although all recording mixers handle more or less the same tasks, none of them do it in exactly the same way. Manufacturers offer a variety of routing options and special features, and they even name controls differently, so generalizing is sometimes difficult. There will always be more than one way to accomplish the simplest task - a prescription for confusion. I'll look at the most common paths that signals take through mixers and briefly mention appropriate alternatives.

To keep things relatively simple, I'll avoid issues pertaining only to large production consoles costing tens of thousands of dollars. Instead, I'll focus on the mixers commonly found in personal studios. Though I won't talk about any mixer in detail, I'll use the Mackie 24/8 8-bus analog mixer and the Yamaha 02R digital mixer as examples. As two of the most widely used mixers on the market, they arguably represent the upper end of personal-studio mixers. That said, most of the principles I'll discuss also apply to larger and smaller mixers.

I'll take you along the signal path from one end of a hypothetical mixer to the other, stopping at major destinations along the way. But first, let's talk channels and buses.

GOING THROUGH CHANNELS

The number and type of input channels govern the number and type of devices that can connect to a mixer. For example, if you want to record a large ensemble using a dozen microphones simultaneously, you'll need a mixer with at least 12 mic inputs. Similarly, if you have a 16-track recorder, and you want to hear all the tracks played back at once, you need 16 inputs capable of handling the type of signals output by your recorder.

If you are intimidated by the sheer number of knobs, buttons, switches, and faders on larger mixers, take comfort in the fact that most of them are channel controls. If you know how one channel works, you understand them all. Typically arranged in vertical "strips," mixer channels have controls for modifying and routing input signals. The controls are organized in a manner that more or less follows the signal's path through the channel. This path varies from mixer to mixer, but there are more similarities than differences, so we'll refer to a generic "channel" (see Figs. 1 and 2).

THE IN CROWD

All sorts of things can be connected to a mixer's inputs, from cheesy high-impedance microphones to state-of-the-art samplers. But if a device has analog outputs, it usually falls into one of two groups: line-level sources and microphone-level sources. Channel strips typically have inputs for both types, usually accompanied by a mic/line selector switch. In many mixers that lack a mic/line switch, plugging a device into the line input automatically disables the mic input. However, some mixers allow you to use both types of inputs at once.

Line-level inputs are high impedance (see the sidebar "Impeding Progress") and typically employ 1/4-inch phone jacks or RCA phono connectors. Keyboards, samplers, drum machines, the outputs from most recording decks, effects processors, and instrument preamps can all connect to line-level inputs. For the most part, you use mic inputs only for microphones and direct boxes.

Because many line-level sources have stereo outputs, it is increasingly common for mixers to have one or more stereo input channels - that is, a single channel strip that handles two input signals. The two inputs in a stereo channel are most often treated as a single source - with one fader, one EQ section, and so on - making it unnecessary to dial in duplicate settings for the left and right signals. Stereo input channels sometimes have fewer features than regular channels because they are most often used for modern synths and samplers that have onboard effects, EQ, and level controls. The idea is to get the instrument's sound into the mixer and route it with little or no processing.

An additional type of line-level input is the tape return. This term originated when all recordings were made to tape, but it now designates inputs that route playback signals from any type of multitrack recorder back into the mixer. I'll deal with how signals get routed to and from multitrack recorders shortly.

Microphone-level inputs are low impedance and typically employ female XLR connectors. Because mixers operate at line level, low-impedance microphone signals must be boosted to line level using a microphone preamp. Channel-strip mic preamps generally include a trim (or gain) pot that adjusts the input sensitivity, a pad (or attenuation) switch for reducing very hot signals (typically by 10, 20, or 30 dB) to the range covered by the trim pot, and a clip (or overload) LED that lights as the signal approaches the upper limit of the input circuit's capacity, usually around -3 dB.

While these characteristics are the same for the majority of mixers used in professional applications, some confusing exceptions exist. First, microphones can be high impedance, and these must be patched to high-impedance inputs, usually on 1/4-inch phone jacks. Second, some older mixers and compact newer models use 1/4-inch jacks for low-impedance mic connections because 1/4-inch connectors are smaller and less expensive than XLRs. Some mixers use 1/4-inch, switchable mic/line inputs on certain channels; these mic/line channels include mic preamps that are in the circuit only when the input is set for mic-level operation.

If your mixer offers only 1/4-inch mic inputs, check the manual to determine their impedance and mic/line status. This is important, as low-impedance microphones function improperly when connected to high-impedance inputs, and vice versa. You can circumvent this problem by using "step-up" and "step-down" transformers. Besides altering the level, they also function as adapters, converting one type of connector into another. However, they can also affect the sound.

HIDDEN POWER

Many mixers deliver phantom power to condenser microphones. Condenser mics require a small amount of DC voltage to operate, and although many of them can be powered using batteries, powering them from the mixer is more economical and consistent. The voltage (usually 48 VDC, but it can be as high as 52 and as low as 12V) travels along one of the audio conductors in the microphone cable. Active direct boxes can also be phantom powered.

Mixers may offer individual phantom-power switches for each input, switches for groups of inputs, or simply a global switch that routes phantom power to all the inputs, regardless of whether it is required by more than one microphone. A global switch might complicate things if you plan to use condenser mics at the same time as other types of microphones; however, inadvertently routing phantom power to noncondenser mics usually produces no noticeable effects.

BUS TOUR
Before I discuss the rest of the channel strip, let's look ahead to routing the signals to external devices.

Signals enter a mixer through inputs and leave on buses. A bus is simply an independent route out of a mixer. The most common example is the master stereo bus, corresponding to the main left and right stereo outputs. Like a tour bus that picks up band members from various locations and drops them off at a gig, a mixer bus accepts signals from all channels assigned to it and takes them to a specific place, usually an output.

A very simple mixer might have just eight input channels, mixing down to a single stereo bus and its stereo outputs. This is fine if you just want to mix eight signals directly to a stereo recorder, but what if you have an 8-track recorder, and you want to simultaneously record each input signal to a different track?

You can accomplish this in a few ways, depending on your mixer. Some mixers have dedicated direct outputs (sometimes called tape outputs) on each channel. Simply patch each channel's direct output to the appropriate recorder input, and you're wired.

In addition to the main stereo bus, most mixers designed to work with multitrack recorders have four or more recording, or "group," buses. You can assign signals from one or more channel inputs to a single group bus and route them through that bus's output to a track input on the recorder (or anywhere else for that matter). The more buses you have, the more tracks you can record at the same time.

One common method of classifying mixers is by the number of channel inputs, group buses, and outputs they offer. If your mixer has 24 channel inputs, 8 recording buses, and a pair of stereo outputs, it is classified as a 24 by 8 by 2 mixer. The Mackie 24/8 (see Fig. 3) and the Yamaha 02R (see Fig. 4) are examples of this configuration, but you can give the 02R up to 16 additional analog or digital inputs using optional I/O cards.

The number of buses that you'll need depends on how many tracks you wish to record at the same time. For example, if you have a 24-track recorder and you want to record different signals on all the tracks at the same time, you'll need a 24-bus mixer. In practice, though, you probably wouldn't need to record 24 tracks at once, and most 24-track studios get by with an 8- or 12-bus mixer. You can also take some signals from the channel direct outputs and others from the subgroups, if your mixer offers both. Alternatively, you can work around the limitation by using DI boxes, stand-alone mic preamps, or other devices to route signals directly to the recorder inputs, bypassing the mixer altogether.

On the other hand, some mixers can send the same subgrouped signal to several different parallel outputs, letting you record the group to several decks at once. On the Mackie 24/8 mixer, for instance, you get 24 triple-bused submaster/tape outputs, which are actually three sets of outputs for the eight subgroups (aka "submasters"). This means the signal from subgroup 8 is routed to outputs 8, 16, and 24, allowing you to record to three 8-track decks simultaneously.

INSERT HERE
After the mic preamp, the signal path's next stop is the channel insert point. An insert point comprises an output (send) and an input (return). It commonly routes a signal from a channel to an external device such as an EQ or compressor and then returns the processed signal back into the channel's signal path. This lets you process just the signal on that channel. For instance, you could patch in a compressor to even out a lead vocal's level during tracking to make up for the singer's uneven mic technique. On some mixers, the insert point comes after the fader. However, digital mixers' insert points - when provided at all - come between the mic preamp and the A/D converter.

Insert points are implemented in one of three ways, but in almost all cases, when you plug into the insert's send, the signal is removed from the channel path. To send the signal on its merry way through the channel, you have to patch it back in via the insert's return. Sometimes there are separate jacks (usually unbalanced 1/4-inch) for the send and return signals. As with a patch bay, the send and return jacks are connected together, or normaled, unless you insert a plug. A variation on this arrangement uses individual jacks (usually RCA). These aren't connected internally; instead a jumper or dummy plug connects the send jack to the receive jack when no external devices are connected.

The third and most common method employs a single 1/4-inch TRS jack and requires the use of a special insert, or Y-cable. Insert cables have a 1/4-inch TRS stereo plug on one end and two mono 1/4-inch plugs on the other. Instead of carrying a stereo signal, the cable carries two mono signals, one from the mixer's insert send to the external device's input, and the other from the external device's output back to the mixer's insert return. Note that some mixer inserts are wired with the send signal going to the tip and the return going to the ring, while others are wired in the opposite way. In both cases, the ground signal is wired to the sleeve. Check the user manual to see your mixer's wiring scheme.

Another use for the TRS-type insert point - assuming it's postfader - is as a direct out. There are two common approaches to this use. The easiest is to insert a mono 1/4-inch plug halfway into the insert point, thus engaging the send part of the jack but not the return. This allows the channel signal to be sent to a recorder without returning to complete the circuit. However, it is not a great solution because a half-inserted plug often makes a relatively poor connection.

Instead, use a Y-cable to connect both the channel insert send and the return to a normaled patch bay. The trick here is that you patch the send to the top rear and the return to the bottom rear of the same patch-bay channel. When nothing is plugged into the front of the patch bay, the signals flow through it and back to the channel, as if the insert points were not in use. But when you patch the insert send from the top front of the patch bay to your recorder, the normal is broken, and the signal flows to the recorder. As a bonus, if you want to temporarily patch something new into the mixer's channel input, you can patch it to the insert return on the patch bay; that interrupts the channel and inserts the new signal in place of the signal at the channel input. Unlike plugging partway into the insert point, there's nothing half-patched about this approach.

In addition to individual channel inserts, insert points are commonly provided for certain buses, particularly the subgroups and the main and monitor output buses. This allows you to process an entire submix, monitor mix, or main stereo mix.

EQUAL OPPORTUNITIES

When people refer to the sound of a mixer, they often mean the sound of its EQ section. Equalization serves two purposes: to cut undesirable frequencies and to boost ones that need enhancement. Like any gain-affecting circuit in the signal path, EQ can disturb the overall balance of sounds and add distortion, so use great care when using it.

Channel EQ designs vary a lot with regard to number of bands, whether they are fully parametric, and so on. (For more on the types and uses of EQ, see "Equal Time" in the October 1999 issue of EM.) But in terms of signal flow, there are only a few considerations.

The EQ section often has a switch to insert it into or remove it from the channel's signal path, and sometimes (as with the Mackie 24/8) you'll find a switch for splitting the section and assigning bands to various functions. The 24/8 lets you assign the high and low shelving bands to the Mix B channel, leaving the sweepable mid bands assigned to the main channel signal.

YOU SEND ME

Next come the auxiliary-send buses. Although the basic function of aux sends is fairly straightforward - they route varying amounts of individual channel signals to a single bus - implementation and occasional labeling inconsistencies can lead to confusion.

Three types of aux sends exist: prefader, postfader, and those with a pre/post switch or internal jumper for configuring them either way. Prefader aux sends tap the signal before it reaches the channel fader (and usually before the EQ, though some aux sends also have pre/post EQ switches). Putting the aux-send control before the fader lets you set the send levels independently of the overall channel level. Postfader aux sends tap the signal after the channel fader and are therefore affected by the fader position. If an aux send is not labeled pre- or postfader and doesn't have a switch, it's probably postfader.

Two of the most common uses for aux sends are creating separate monitor mixes and routing signals to external effects processors. Here's where the situation can get confusing. Some mixers label sends "monitor" or "effect" rather than aux, meaning that they are configured for a particular application. Let's take a closer look.

Monitor sends route varying amounts of signal from individual channels to a monitor bus, where the signals combine to create a monitor mix. Because the monitor mix must be independent of the main mix, you must tap the individual channel signals before they reach the main channel faders (prefader). Assume that a send labeled "monitor" with no pre/post switch is prefader.

Monitor mixes usually provide a custom balance of sounds for a particular musician. For example, a drummer will likely want to hear less drum signal in the headphone mix, as the drum set's acoustic sound will already be quite loud. Therefore, some mixers have several monitor sends, each of which can be used to create a separate monitor mix. The Mix B bus on the Mackie 24/8 can create a stereo monitor mix, because each channel has a pan pot in addition to a level control.

Effects sends route signals from mixer channels to the inputs of outboard processors. For example, if you have a reverb processor connected to an aux send, increasing the aux send level for a particular channel sends more of that channel's sound to the reverb's input. (A master aux-send control determines the overall send level to the processor's input.) Some mixers, including the Mackie 24/8, offer stereo effects sends for feeding processors with stereo inputs.

Normally an effects send is postfader, which means changing the overall channel level also changes the amount sent to the effects bus. But if you have the option of using a prefader send to feed an outboard signal processor, you can create some interesting special effects. For example, if you are sending to a reverb processor, you can pull the main fader down and hear just the reverb returns, creating an eerie effect. If a send is labeled "effects" and has no pre/post switch, you can assume that it is postfader.

Most mixers have dedicated effects-return jacks in their main mix section (not in the channel strip). From there, signals from the effects unit's outputs return to the mixer and blend with the main stereo output mix, forming an effects loop. Some mixers let you assign effects returns to buses other than the main stereo output bus, such as recording buses. If your mixer does not have dedicated effects returns - and many digital mixers, including the Yamaha 02R, do not - you'll need channel inputs.

Conventional wisdom dictates that compressors, EQs, and other processors inserted into a channel or master bus should not connect to an effects send and return because 100 percent of the signal is supposed to pass through them. However, it is sometimes desirable to have some of the original signal combined with the processed signal to achieve particular effects. Famed producer Eddie Kramer routinely uses this technique.

An aux send with a pre/post switch can be used for any purpose that requires summing output signals from individual channels and routing them to a single bus. For example, it can pinch hit as an additional recording bus. It is also worth noting that auxiliary sends and monitor sends are usually balanced, while effects sends may be either balanced or unbalanced.

MUTED COMMENTS

Channel mutes remove individual channels from the main signal chain, allowing you to turn off a channel without pulling its fader down. Some manufacturers call the channel mute an "on" button, reversing the concept, but the same function is performed. (Note: you can also mute a channel by disengaging the L/R bus assignment switch.)

On the more sophisticated analog mixers and most digital mixers, channel mutes can be organized into groups. Pushing the mute button for one channel in the group engages all mutes in the group. Mute groups are useful whenever you need to turn several channels on or off simultaneously, such as when bringing a group of instruments into a mix. Say you need to record a horn section with six mics, with each mic on its own track and mixer channel. The horns play only on the bridge, and during the rest of the song there's nothing but extraneous noise on the tracks. Instead of pressing six mute buttons before the horns enter and again after they finish, you simply press one button.

FLYING SOLO

The solo button is in some ways the opposite of the mute button; it turns off everything except the desired channel or channels. Solo comes in three styles: PFL (Pre-Fader Listen, sometimes called cue), Solo-In-Place, and mono. PFL allows you to monitor the sound before the signal reaches the channel fader - and thus before fader level, panning, EQ, effects, and other settings affect it. Of necessity, PFL comes after the mic preamp, because an unboosted mic-level signal would be too weak to monitor. The PFL signal is usually routed to the headphone bus and is useful for setting the preamp input level or for any other task for which it is advantageous to hear the signal before it's altered.

Solo-In-Place also allows you to hear an individual channel, but it taps the signal chain after the fader, panner, and EQ, and it retains the stereo imaging. Most useful in mixing, this type of solo tells you exactly what the processed signal sounds like and where it will appear in the stereo field. Mono is identical to Solo-In-Place, but without the stereo placement.

PANNING YOUR MUSIC
The panoramic control, or pan pot, serves two functions. ("Pot" is short for "potentiometer," the type of variable resistor used for panning in analog mixers. Digital mixers use rotary digital encoders, but these are still conventionally referred to as "pan pots.")

You commonly use the pan control to position a signal on an individual channel within the 180-degree stereo spectrum. You accomplish this by adjusting the relative volumes of the signals being routed to the stereo bus's right and left sides. For example, when you increase the right side's volume and decrease the left side's volume, the sound "moves" to the right. Sounds placed entirely on one side are said to be panned "hard" right or left. When the pan pot is in the 12 o'clock position, the channel sends equal amounts of the signal to both sides of the stereo bus, which means you are essentially listening to a monaural signal.

In some mixers, pan pots also assign a signal to a particular group bus. Group buses are organized in pairs (for example, 1 and 2, 3 and 4, 5 and 6, and 7 and 8), with a single switch assigning a channel to a pair. The pan pot sends more or less signal to either bus in the pair, or sends it to only one of the buses (when the pot is turned to one side).

NOT FADE AWAY
Channel faders serve different purposes when you're tracking and when you're mixing. During tracking, the fader interacts with all the buses to which it is routed, particularly the group bus that feeds the record channel. If the recorder is fed by a direct out or tape out, the channel fader determines the record level. During mixing, the faders determine the level of each channel in a mix. In either case, the faders' relative positions supply a visual reference.

Also, in some mixers, faders can be grouped to operate as a single control. Once they're locked together as a fader group, moving one up or down moves all of them, and they maintain their level relationships. That is important because fader values taper logarithmically, not linearly. That means a greater volume increase occurs when you move a fader at the lower end of its range than when you move it at the higher end. To get a uniform group fade, the actual level relationships between channels - not the faders' visible relationships - must be maintained. If you group the first two channels, and channel 1's level goes up by 2 dB, channel 2's level will also go up by 2 dB, even though the physical positions of the faders may be quite different.

In some boards, especially digital consoles, you get a choice of level relationships in a fader group. You can maintain either the relative levels or the linear relationships, even though choosing the latter means that the levels will not increase identically in all the grouped channels. You also might be able to move one fader in the group without affecting the others until the first fader exceeds a certain threshold level, at which point the other faders start moving with the first.

ON DECK

If you record and mix using a multitrack recorder, you need to get signals from your mixer to the individual recorder tracks. You then must send them back to your mixer for stereo mixdown. When recording overdubs, you need to hear the previously recorded tracks while adding new ones. Mixers handle these tasks in different ways, and in almost all cases there are work-arounds for any problem or limitation.

As you have seen, channel signals can be sent to recorder tracks in two primary ways: through recording buses and via individual channels' direct outputs. Each approach has its limitations. Recording buses allow you to sum signals from several channels to the same bus and then route the summed signal to a single recorder track. Dedicated direct outputs route signals from only a single channel.

Say you have an 8-bus mixer, and you want to use four mics to record four-part backing vocals to two tracks. At the same time, you will record individual instruments on six tracks and a stereo lead vocal on two tracks. You could assign the four backing-vocal mics to group buses 1 and 2 and the stereo lead vocal to group buses 3 and 4, using group buses 5 through 8 for four of the instruments and two channel direct outs for the other two instruments. That way you utilize the mixer's ability to combine channels using the group buses while circumventing the eight-group limitation.

How you get signals from the multitrack back into the mixer depends on your mixer. If it has dedicated tape returns, and they connect to a separate mixing section with its own inputs, faders, and other controls - usually located to the right of the master faders - you have a split mixer. If your mixer has tape returns on the regular input channels, you have an in-line mixer.

With an in-line mixer, each channel does double duty by accepting signals from both the channel input and the tape input. The tape input generally has its own path (called the monitor path), complete with separate volume and pan controls. In many cases, you can split the channel EQ section, with the high and low bands assigned to the monitor path and the two center bands assigned to the channel path. Considerably more limited than the channel path, the monitor path cannot be assigned to any buses other than the main stereo L/R bus. That means signals arriving at the tape input can't be fed to external effects processors, included in discrete monitor mixes, routed to group buses, or processed with the entire channel equalizer.

However, many in-line mixers have a Flip button, which swaps the two signal paths, reversing the situation. This lets you use all of the board's channel-routing features with your outboard processors while mixing recorded tracks. The more limited signal path can then be used for virtual tracks from synths or samplers, which usually have onboard effects and EQ anyway. (The Mix B bus on the Mackie 24/8 can be configured for this purpose.)

WE HEAR YOU, MASTER

All paths lead to the master section. On the hypothetical mixer in Fig. 5, this section contains the master faders and meters for the group buses, along with their mute and solo switches; the master aux-send and aux-return level controls; the master stereo output fader; the control-room volume pot; and the headphone jack and level control. It also has buttons for collapsing the stereo mix to mono and for switching the monitor section between the main stereo output bus and the 2-track input so you can hear your recorded mixes. Some mixers also allow you to assign the aux returns to group buses.

Sounds travel from the mixer to your ears via the headphones and the studio monitors. The mixer's control-room monitor outputs (which generally use either XLR or balanced 1/4-inch TRS jacks) feed powered monitors or a power amplifier and passive monitors.

The 2-track mixdown machine (say, a DAT deck) connects to either the main (XLR) outputs or the dedicated recording outputs (XLR, 1/4-inch TRS, or RCA). Its output is patched to the 2-track recorder inputs (usually RCA, but sometimes XLR or 1/4-inch TRS).

HAPPY TRAILS

I've traced the basic signal paths commonly found in personal-studio mixers, but space does not permit me to address the host of routing options, esoteric functions, and applications.

The best way to really understand your mixer's capabilities - and learn what does and doesn't work - is to experiment. Happy mixing!

Although digital mixers perform nearly all of the same functions as analog units, they do so differently. They also offer many features not found on analog mixers. It's far beyond this article's scope to give a complete account of signal routing in specific digital mixers, but I'll point out a few essentials using the Yamaha 02R as an example.

Analog audio signals don't flow through digital mixers. Analog signals entering a digital mixer are converted to digital information in a data stream, and they stay that way until converted back to analog signals. If you track to a digital multitrack machine, using digital buses and mixing to a digital recorder, the only reason to convert the digital signals back to analog is for monitoring. (Monitor speakers must be analog.)

Digital consoles have no analog circuits between the A/D converters and the DACs. When you push up a digital fader, you're simply moving a vertical data encoder assigned to adjust amplitude. You can reassign this fader to adjust some other value - such as monitor-send level - and most digital mixers exploit this capability with several different fader assignments levels. For example, level 1 might assign faders 1 through 16 to channels 1 through 16, while at level 2 the same faders control channels 17 through 32, and at level 3 they act as master sends for the aux and recording buses.

Many digital mixers feature software-based effects. For example, the 02R has two complete onboard multi-effects processors and enough 4-band parametric EQs, dynamics processors, and delays to patch one of each across nearly every input and output on the mixer - all without plugging in a single cable. Yet most digital mixers have few, if any, analog insert points. Adding inserts (and direct outs) means more D/A converters. If you normally patch in your favorite vintage compressors on mixdown, you may not be a happy camper. This point is a deal-breaker for many people.

Because digital-mixer settings are information stored in memory, they are perfect for "snapshot" and dynamic automation. Once you set the mix parameters, you can take a snapshot of them and name and store them as a file. In most cases, you can back up these settings to a computer for safekeeping. You could return to the mix years later, press a few buttons, and shazam! Your settings load, and the mixer is ready. The same goes for dynamic automation, in which you usually store the fader, EQ, pan, and mute moves in memory (or in a MIDI sequencer, if the mixer has MIDI automation).

Like analog mixers, digital mixers have analog mic and line inputs, but they usually offer additional channels accessible only through various I/O cards. If you want to use a digital mixer with an analog multitrack deck, you need analog I/O cards. You'll probably connect the mixer to digital recorders, so you'll need the appropriate interface: ADAT, TDIF, AES/EBU, or S/PDIF. A few digital mixers boast lots of digital inputs, but most need optional I/O cards.

Impedance is a measure of a circuit's opposition (the sum of a DC resistance and the circuit's reactance) to a signal or current attempting to pass through. The practical difference between impedance and resistance is that impedance is the opposition to alternating current (AC), whereas resistance is the opposition to direct current (DC), and impedance changes as a function of frequency. Impedance is symbolized in equations by the letter Z, and its values are measured in ohms, represented by the Greek letter omega (Ω).

High-impedance signals are generally held to be in the 10 kΩ to 20 kΩ range, while signals in the 150Ω to 1 kΩ range are considered low impedance. If you plug a high-impedance source into a low-impedance input, you're likely to overdrive it into distortion; do the opposite, and not enough signal gets through for a satisfactory level. A line-matching transformer, usually in the form of a direct-injection box (more commonly called a "direct box" or "DI"), is often used to match signal levels and impedances. Direct boxes can employ either active or passive circuitry.