Plays Well With Others: Staying out of the Red
By Myles Boisen
Sep 1, 2001

In a perfect world, every device in your studio racks would integrate seamlessly with its neighbors, your levels would always be hot without a hint of distortion, and routing signals would be like driving down a smooth road on a sunny day. If that ever happened to me, I’d know for sure that I was about to awaken and find myself facedown on the mixer, drooling into the faders.

At my studio, I have collected some of the least expensive and most incompatible audio gear ever made. Connecting that equipment is just the beginning of a never-ending set of challenges. Once the signals start flowing through those rackmounted mongrels, it can be a long, bumpy ride down the treacherous road of gain staging.

In any integrated recording and mixing studio, including uncomplicated cassette 4-tracks and self-contained digital-audio workstations (DAWs), maintaining optimal gain at each stage of the signal chain is crucial to the ultimate goal of clean, undistorted recordings. (Any point in the chain at which the signal level can be changed is called a gain stage.) If your signal dips too low at any stage, noise becomes part of the audio. In the analog domain, that noise — whether it's high-end hiss, radio-frequency interference (RFI), 60 Hz ground-loop buzz, or some combination of the three — is often amplified at one or more stages, especially when the audio is being compressed or equalized. (You should strive to minimize or eliminate those noise sources in any event.)

Digital quantization noise behaves in much the same way as analog noise, but it’s harder to eliminate or filter out. The loss of resolution that results from low digital-recording levels can never be recovered or corrected. At the other extreme, adding too much gain at any step of an analog or digital signal path introduces irreversible and harsh-sounding clipping distortion.

Because a variety of reference standards are used in vintage, semipro, professional, and software-based audio gear, keeping your signal levels consistent throughout a studio-processing chain is a little like driving on a rocky mountain road. When you route the audio from one device to the next, it can be a smooth turn, or you might feel a little bump in the level. If you adjust the gain too much (or if gear set at a different operating level adjusts it without asking you first), you risk a grisly crash.

ON THE LEVEL, NEVILLE
Fortunately, that motoring scenario includes a centerline designed to keep you from running into a mountain or careening off the side. Within the analog domain, where most audio signals originate, look for a zero mark —
either 0 VU on a VU meter or 0 dB on an LED meter, fader, or knob. If you steer the level toward zero (allowing for normal fluctuations in dynamic level), your signal will have a safe ride to its final destination.

The zero calibration of a device is referenced to its internal operating level. A sine-wave signal that produces a zero-level reading on a compressor's output might show up as a different level on the meter of a multitrack tape machine to which it is routed. Patch that signal from tape to a DAW, and you'll get yet another meter reading.

Theoretically, a zero reading on a device's meter indicates a specific AC voltage within a unit, calibrated to one of two prevailing audio standards. The professional audio standard is +4 dBu. An analog device calibrated to that level references its 0 dB meter reading to an input/output voltage of 1.23 VRMS, and the connection is generally balanced. Stereo equipment, and some home-recording gear, is referenced to the consumer-level standard known as -10 dBV, in which a 0 dB meter reading indicates an input/output voltage of 0.316 VRMS and the connection is typically unbalanced. (See “Square One: Decibels Demystified, Part 2” in the August 2001 issue for more about those standards.) When interfacing both kinds of equipment, it’s important to know there is an 11.8 dB voltage discrepancy between -10 dBV and +4 dBu reference levels.

Tape-machine inputs are referenced to -10 dBV or +4 dBu (some include inputs for both), but their meters are commonly adjusted so that 0 VU actually indicates a level that is 3, 6, or even 8 dB higher than the reference level. That convention lets engineers print maximum levels on tape without continually pegging the meter. Film, video, and broadcast equipment, especially older types, may also have unique reference level requirements and other quirks.

If the devices in your studio are the same type (-10 dBV or +4 dBu) and each one is properly adjusted and performing no gain changes, a 0 dB test tone applied to the first unit's input should appear at the final stage as a 0 dB level (see Fig. 3). But remember, I'm speaking in theory. In the real world, components age or aren't precisely calibrated to begin with, and somewhere in your chain is probably a box of the other type that you can't do without.

**GO WITH THE FLOW, JOE**

The initial source device, which is the first and most important gain stage, also has the greatest impact on the overall signal-to-noise ratio. A careful level adjustment there — well above the noise floor yet conservatively below clipping — will result in a clean, healthy signal throughout the rest of the chain as long as you intelligently use devices with different levels and observe the principle of unity gain.

*Unity gain* is a simple concept. When the input and output levels in a device are equal, resulting in no net gain or loss (technically, a gain ratio of one),
that unit is operating at unity gain. The zero mark on a mixer's fader indicates unity gain between the corresponding circuit's input and output. Running gear at or near unity gain preserves the optimal signal-to-noise ratio of the source and downstream devices.

If your studio is set up and calibrated for one operating level (-10 dBV or +4 dBu) and you follow the principle of unity gain (even with heavy-duty processing devices such as compressors and equalizers), you'll have few problems establishing proper gain staging. However, the reality is that most personal studios, and some world-class facilities as well, use a combination of -10 dBV and +4 dBu equipment. That mismatch is where most analog gain-staging woes arise.

ADAT, DTRS (the Tascam DA-88 and its siblings), and DAT machines usually offer -10 dBV and +4 dBu input options with dedicated connectors to eliminate confusion (see Fig. 4). Connector types aren't strictly standardized, but RCA jacks typically carry unbalanced -10 dBV analog signals, and 3-pin XLRs or proprietary multipin connectors commonly carry balanced +4 dBu signals. Quarter-inch connectors (TS or TRS) are often used in the shadowy nether region, where mixed levels with balanced or unbalanced lines coexist. Unlike their analog cousins, no level differences exist between the two common digital connections: S/PDIF (RCA jack) and AES/EBU (XLR connector).

In my experience as a mixed-level signal herder, it's easiest (though not essential) to keep signals consistent at -10 dBV or +4 dBu throughout a chain. My main multitrack recorder — a Tascam MS-16, 16-track analog tape deck — offers -10 dBV and +4 dBu ins and outs. My mixing board, a Soundcraft Spirit, is set up to run at -10 dBV, with mix bus outputs referenced to +4 dBu. (The deck that I use for mixdown or mastering has +4 dBu inputs.) Accordingly, my tape returns, board, and reverbs normally run at -10 dBV. The mixer's channel inserts are also -10 dBV, and consequently, a lot of the rack gear usually used for mixing operates at -10 dBV.

However, I also have many mic preamps and compressors that I like to run at +4 dBu directly into the tape machine, bypassing the board and any extraneous connections. In situations like that, gear with switchable -10 dBV or +4 dBu levels is obviously a godsend, and I will gladly take a trip to the back of my rack to change input or output level switches when necessary. Nonetheless, combining mixed-level gear in a chain is certainly not out of the question and can still result in pristine audio quality. One particularly handy device in that regard is the Ebtech Line Level Shifter (www.cymation.com), which converts -10 dBV to +4 dBu and vice versa (see Fig. 5).

**STEP UP THE GAIN, JANE**
I rarely worry about decreasing a signal from a +4 dBu output to a -10 dBV input. Remember that zero on a -10 dBV unit is about 12 dB less than zero on +4 dBu equipment. If that arrangement produces a distorted signal,
attenuate the output level of the +4 dBu device by 10 or 12 dB; that should solve the problem with minimal impact on the audio quality.

However, I'm less carefree about stepping up signals from -10 dBV devices to those operating at +4 dBu. When that gain change is unavoidable (for instance, from a -10 dBV keyboard to a +4 dBu compressor, or mastering from a consumer-level recorder to a pro deck), I step up the signal in one of two ways: boosting the internal amplifier of the source device or patching in another unit just for amplification.

Boosting the internal amp of a -10 dBV device by 12 dB is generally not the best way to go; it can substantially raise the noise level and the signal. In addition, the device may not have enough internal amplification to reach the required level. Finally, overdriving the -10 dBV device's output amplifier beyond its available headroom results in clipping. When that happens, a meter on the downstream +4 dBu unit's input stage will read at or below zero and still sound distorted because the -10 dBV unit doesn't have enough output voltage to drive it at an optimal level. The far better approach is to use an external amplifier to boost the level of the -10 dBV device by 12 dB.

Returning to my Soundcraft board for a moment, inquiring minds might ask, “If the board has -10 dBV inputs, inserts, and effects sends and +4 dBu mix outputs, isn't there a step up involved?” Yes, but in that case, the console's high-quality amplifiers provide a gain boost with little extra noise. Another factor is that the signals coming from my tape machine, referenced to -10 dBV, can easily have peak values from 10 to 20 dB above zero. That brings the general signal level at the faders within range of 0 VU at +4 dBu.

Once you become adept at analog gain setting, cautiously raise levels through your system to take advantage of the various devices' headroom. When you're ready to do that, though, remember that the weakest gain link in the chain (invariably a -10 dBV device) will limit the system's headroom. When you advance to that point, you will think of zero calibration as a guideline, but never an absolute or universal value.

**WATCH THE METER, PETER**
Setting the level controls of all the devices in a signal chain is a balancing act; for example, if you increase the level at one point in the chain, you must decrease the level somewhere downstream to maintain a constant average signal level throughout the chain. (The opposite is also true: if you decrease the level at one point, you must increase the level somewhere downstream to maintain a given signal level.)

In general, try to avoid setting any input- or output-level control to its maximum value, which causes the circuit to operate at the extreme of its range and invites distortion and increased noise levels. Keep most gain controls somewhere in the middle of their range, which lets the circuits operate in their most linear region, minimizing the potential for clipping
distortion and maximizing the signal-to-noise ratio. (That rule has some exceptions.)

Typically, the first gain stage in the chain is a mic preamp or an electronic source, such as a synth, sampler, or CD player. Most electronic musical instruments don't offer output metering, nor do some mic preamps (even fully professional ones), which makes it difficult to set their average output levels to 0 dB. In that case, follow the signal flow to the next downstream device — typically a compressor, recorder, or mixer — and set the initial gain according to what those meters tell you.

If you route the output of the source device directly to a tape deck that has been calibrated to accurately reflect signal levels on its meters, simply use those meters to set the output level of the source. It's important to match the reference levels of both devices (that is, -10 dBV to -10 dBV or +4 dBu to +4 dBu). If the reference level of the source does not match that of the deck, take steps to correct the mismatch.

On the other hand, if you send the output from the source into another device with the capability to change the level, such as a compressor or mixer, that device's meter might not give you a clear picture of what's going on. For example, if the input level of the downstream device is set very low, its meter might show a level well below zero, even if you crank the output level of the source to its maximum, which increases the potential for clipping and noise. The problem is compounded if you try to bring the meter reading up by boosting the output level of the downstream device to its maximum.

That is especially important in a mixer, where there are several gain stages before the signal reaches the output meters (input gain/trim, input fader, subgroup output, main output). If the mixer includes a meter bridge with input-level meters, you still have to think about the effect of the input gain/trim and fader on the meter reading.

Most +4 dBu devices have ample headroom to handle signal peaks above +20 dBu, but you can often hear clipping distortion at lower levels. The most common reason for audible clipping at levels that appear acceptable is inaccurate metering. VU meters are particularly slow to react to transients, which makes them unreliable for gauging fast-rising peak-voltage levels. Peak LED meters are more reliable in that regard, and they can be calibrated to indicate clipping, levels above clipping, or a “safety zone” from 3 to 6 dB below the onset of clipping. If a manufacturer's specifications don't address that issue (and even in the rare cases when they do), setting the levels according to peak-level meter readings involves trial and error.

Another factor that complicates level setting by the meters is asymmetrical waveforms. Speakers, drums, and other common acoustic sound sources tend to move equally far in both directions as they vibrate, generating electrical waveforms that have equivalent positive and negative voltage
values. Those waveforms look symmetrical when viewed on an oscilloscope or DAW screen.

However, some vocalists and most brass instruments and saxophones generate waveforms that are decidedly asymmetrical with far greater negative voltages when viewed onscreen (see Fig. 6). Older peak meters designed to read only positive voltage may not accurately reflect the level of those signals, letting the wave's negative voltage component (which may be double the indicated voltage) clip and produce audible distortion. Some signals, such as drum transients, mask some distortion, whereas the identical amount of distortion is easily audible in a piano or nylon-string guitar note.

**DOWN AND DIRTY, GERTIE**

When routing a microphone signal through a mixer, it is standard practice to set the channel fader at 0 dB and send the signal from the direct or subgroup outputs to the destination recorder. The direct signal path is simple, and the input gain/trim control is the only gain stage that needs to be adjusted.

In that scenario, gain through the mixer channel is commonly monitored by engaging the channel's prefader level (PFL) or solo button, which routes the selected channel to the monitor bus and displays the level on the main stereo meters. On most mixers, it is also possible to route an individual channel to a subgroup for level-setting purposes, even if that signal is sent out of the board through the direct output. Many modern studio mixers include signal-present or peak-overload indicator lights on each channel to aid the gain-setting process. In addition to using those indicators, check the level at the recorder for any track being recorded.

When you're mixing one or more channels to a subgroup output, gain-staging procedures become more complicated. First, it's important to establish the gain through each input channel by setting the channel fader at 0 dB and adjusting the input gain/trim control to maintain a usable signal level as indicated by the board's meters. Once that's done, assign the channels to the appropriate subgroups; their faders should be set to 0 dB.

To create a mix of two or more channels in a subgroup, simply adjust the channel faders and perhaps bring the overall subgroup level up or down. Use the meters on the subgroup and the destination recorder to determine the subgroup master level. Once the submix is set, all faders should ideally be within ±10 dB of zero. If not, adjust the input gain/trim to bring the channel fader into the recommended range and maintain an optimal signal-to-noise ratio.

Within a mixer are several other gain stages related to the auxiliary effects buses. Those controls are typically rotary pots that may have a central or unity-gain setting flanked by numerical plus and minus values, or they might use a simple one-to-ten numbering scheme. In addition to aux-send controls in each input channel, each aux bus should also have a master-send control
and an after-fader level (AFL) or solo button that lets you check the output signal to make sure it's within a usable range (that is, an average level near 0 dB).

Most engineers maximize the headroom within outboard reverb units by setting input and output controls to their full 100 percent levels. In most instances, they also set the mix controls (which adjust the ratio of dry to processed sound) to full or 100 percent “wet.” Most digital effects processors have their own metering with clear peak-level indication. For best audio quality, send the hottest possible signal to the units but avoid peaking and listen carefully for distortion.

The final gain stage in the aux-effects chain is the effects return at the board. That control can be a rotary pot or fader, and it adjusts how much effected signal is added to the stereo mix bus. In most cases, that effected signal is far less than a 0 dB level. Effects-return levels must be adjusted by ear rather than by metering. Solo or AFL switches at the effects return let you determine if the processed signal is undistorted and at a usable level. Comparing the effects-send signal to the return signal can also supply valuable insight into the character of an effects program. In addition, such a comparison can indicate overall gain changes through the unit, which may be due to reference-level mismatches or extreme regeneration in delay effects.

Compressors also present unique gain-setting challenges. As a general rule, you should compensate for the amount of gain reduction indicated on a compressor's meter by setting an equivalent boost at the makeup gain control. For example, if the compressor regularly cuts a signal by -4 dB, a makeup gain boost of +4 dB keeps the peaks at roughly the same level through the unit. The signal will sound louder because the dynamic range is reduced and low-level signals are boosted. But in terms of peak metering, the maximum signal levels should be the same from input to output and more consistent as well. At extreme compression settings, that method may not be so predictable, and you should watch the meters in a downstream device after the compressor.

**FIDGETY DIGITS, BRIDGET**

When it comes to setting digital levels, 0 dBFS (decibels full scale) is an absolute ceiling value that should never be reached until the final stage of mastering. At that level, the highest peaks of the waveform are represented by binary numbers consisting of all ones. Think of 0 dBFS as the guardrail on your digital gain-staging road; it's always there to guide you, but you sure don't want to risk running up against it!

There are various philosophies about what levels should be maintained when mixing from analog to digital or recording from an analog board into a modular digital multitrack (MDM) or DAW. Like the producers of analog gear, manufacturers of digital equipment use various digital-reference standards.
meant to keep engineers from maxing out their dBFS levels when making analog-to-digital transfers.

The conventional wisdom is that 12 dB of headroom is generally acceptable in a digital-audio recording system. That means an average meter reading of 0 dB on an analog mixer's outputs should equate to about -12 dBFS on a digital meter (assuming that the mixer's outputs and the MDM's analog inputs are at the same reference level). I say “about,” because no established reference-level standard exists within the industry for the crucial operation of converting analog audio voltage into digital ones and zeros.

Many equipment manufacturers reference 0 dB to -18 dBFS, whereas others set their standards from -12 to -24 dBFS. To keep overzealous recordists from pegging the meters (as they did in the old days of analog), some companies also include a safety margin — a few decibels of “hidden” headroom — between the 0 dBFS peak indication and the point of digital clipping. That lets you scoot along the mythical guardrail without scraping up your paint job.

Many MDMs offer -10 dBV and +4 dBu inputs, and the smoothest ride is always on the road where your reference levels match. In many studios, outboard mic preamps referenced to +4 dBu run straight into digital recorder inputs, following the philosophy of a minimal signal path. That routing presents few gain-staging problems as long as you remember that such a mic preamp needs to provide roughly 12 dB less output when driving a -10 dBV input.

On rare occasions, I’ve found that a +4 dBu preamp running at its lowest output still provided too much level for a -10 dBV ADAT input, and it was not feasible to use the multipin +4 dBu Elco snake. In those circumstances, it is easiest to engage the pad switch (if one is provided on the mic or preamp) or physically move the microphone farther from the source. If that doesn't do the trick, the remaining options are to choose another mic or preamp with lower output level or to insert a compressor or other gain-reducing device between the preamp and recorder.

There is also the possibility that a -10 dBV unit, such as a keyboard or CD player, may be called upon to drive a +4 dBu input on a digital device. An intermediate line amplifier, such as the Ebtech Line Level Shifter or the internal amp in a +4 dBu mixer, is the best way to avoid step-up problems. The increase in resolution attained by keeping the signal as hot as possible generally outweighs the potential increase in noise you may get by boosting levels in that manner, especially in a 16-bit digital-recording format.

**FIX IT IN THE MIX, TRIX**
Stereo mixdown to a digital medium such as DAT is another common procedure. When mixing from an analog console, remember that a signal reading 0 dB on the mixer's meter typically measures between -12 and -18
dBFS on the digital recorder's meter, depending on the analog-to-digital converter's calibration. If your program material has a wide dynamic range, set an average signal level of 0 dB at the mixer's master faders (once you've established proper gain staging throughout the rest of the board) and then play the selection from beginning to end, noting the highest peak level (not merely the average level) registered on the digital recorder. The peak-hold function on many digital meters is particularly helpful for logging maximum levels.

After the level-checking pass, you may want to adjust the DAT's input level, keeping in mind that it's always good practice to keep peak signal levels 2 to 3 dB below 0 dBFS to retain the best resolution. If you find that one transient peak in your piece hits -2 dBFS and the rest of the program sits around -8 dBFS, consider compressing or manually fading the track with the offending transient. That lets you boost the entire mix level by 6 dB and utilize more available bits.

On average, a highly compressed rock mix might have only 4 to 8 dB of dynamic range. Transferring such a mix at 0 dB equal to -12 dBFS is a waste of perfectly good bits. For example, if the dynamic range of the mix is 8 dB, set the DAT's input level so that 0 dB (the average level of the mix) corresponds to -6 dBFS, which lets the dynamic range vary by ±4 dB while leaving 2 dB of headroom below 0 dBFS.

Whatever kind of mix you have, do not be afraid to nudge the level so that you have a strong average digital level and no peak goes higher than -2 or -3 dBFS. Don't forget to use your ears on the final product. Some digital meters are a bit slow; they can let a distorted transient slip by undetected if every sample of the incoming audio is not measured.

In an all-digital mix — whether from a digital mixer or within a self-contained recording and mixing DAW such as Pro Tools — pay special attention to digital headroom. In a 16-bit system, it's advisable to keep peak levels between -2 and -3 dBFS. Higher-resolution systems can reproduce a greater dynamic range, and peaks near -6 dBFS are acceptable in that case. The internal processing of a digital mixer might allow for extra headroom and longer word lengths, but its signal must ultimately pass through a digital-to-analog converter to be heard. Also, the ultimate destination is usually a 16-bit CD. Those facts often mandate a gain reduction at the output faders to keep the final level below the 0 dBFS ceiling, especially with dense multitrack mixes.

**TIME TO REVIEW, SUE**

The best mindset for proper analog gain staging is to be aware of the potential consequences of stepping the gain up or down in professional and consumer-level gear. Even if the analog portion of your studio is set up and precisely calibrated for only one reference level, you should still take the time
to maintain and periodically check zero levels wherever possible in the processing chain, following the signal flow's direction.

Within the digital domain, it's equally important to maximize your levels, but keep the peaks at -2 to -3 dBFS at all stages before final mastering. It is also crucial to listen for distortion and noise at the end of an analog or digital signal chain, regardless of what the meters tell you.

To avoid nasty potholes and detours on the gain-staging road, steer your signals toward zero, keep your eyes on the gauges, and drive with your ears. Armed with the knowledge thus acquired, your signal-routing trip can indeed be like driving down a smooth, level road on a sunny day.

------------------------------------------------------------------------

Myles Boisen (mylesaudio@aol.com) is a guitarist, producer, composer, and head engineer and instructor at Guerrilla Recording and the Headless Buddha Mastering Lab in Oakland, California. Thanks to Karen Stackpole, Bob Smith of BS Studios, and Lawrence Fellows-Mannion of Rance Electronics.

INITIAL GAIN CALIBRATION

When setting up a studio system, start with the sources — in most cases, synths or samplers with -10 dBV outputs and one or more microphones connected to mic preamps with +4 dBu outputs. To ensure that the synths have adequate gain, connect their outputs to direct injection (DI) boxes and connect the DI boxes to mic preamps. (For mics and synths with DI boxes, it's best if the preamps have output-level meters.) If DI boxes are not available, connect the synths' outputs to the -10 dBV inputs on the mixer.

In that procedure, you will use one of the sources as a signal generator to set the levels throughout the rest of the signal chain. A test-tone generator with a known output level is more reliable for that purpose, but it's not pleasant to listen to for an extended period of time. A sustained keyboard chord is fine, as is a microphone in front of a radio or other compressed music source.

If you use a phantom-powered condenser mic, connect it to the preamp and apply phantom power first. Then, connect and power up all studio devices except for the final power amp or powered monitors. Set the gain on outboard devices to unity. Start with all mixer controls at minimum or detented settings; EQs should be flat or bypassed.

If you use an outboard preamp as the source, raise the gain so the preamp meter reads 0 dB. (If you connected a synth to the preamp, set the synth's output to the highest level that does not overload the preamp's input.) Connect the preamp directly to each outboard compressor, setting the controls for unity gain, no gain reduction, and 0 dB on the output meter. Any other outboard insert devices used in tracking (gates, equalizers, and so on) should also be calibrated in that manner.
Once it's been calibrated, the final device in the chain can be routed to one channel of the multitrack recorder to check its meter reading. The preamp's 0 dB level might not read zero on a digital or analog recorder's meter. For calibration or adjustment of the deck's input level, consult a qualified technician.

Next, route the preamp's output to an appropriate line input (not mic input) on the mixer. Set the mixer's channel, subgroup, and master faders to zero or unity; engage the channel's PFL or solo switch; and raise the channel's gain/trim control until the meter reads 0 dB. (Repeat for all other input channels.) At that point, the subgroups and master buses should read zero as well. If you're not using an outboard preamp, connect the mic or synth to an appropriate input on the mixer and follow the same procedure.

Connect the channel-direct outputs or subgroup outs to the multitrack recorder and check the levels on the multitrack meters. (As before, the mixer's 0 dB level might not read zero on the recorder's meters. For calibration or adjustment of the recorder's input level, consult a qualified technician.) Connect the recorder's outputs to the mixer's tape returns, engage the deck's monitor function, and adjust the mixer's tape-return levels so that they indicate zero on the mixer's meters after pressing the appropriate PFL or solo switch.

Connect the channel inserts to devices used for tracking or mixing. Generally, those devices are compressors, gates, equalizers, digital delays, and other single-channel effects not connected to an aux bus. Make sure the reference levels of those units match the mixer insert's operating level. Check for unity gain on the PFL or solo meters or by monitoring the channel level by ear while physically engaging and disengaging the insert connector from the mixer.

Set the inputs and outputs of all aux effects units at maximum and set the mix control at 0 percent (completely dry, noneffected signal). On the active mixer channel, set the individual aux-send levels at a repeatable midway point (for example, unity, 5 o'clock, or 12 o'clock). Raise the master aux-send control until the effects unit indicates a 0 dB level.

If a PFL or AFL meter is available to indicate the level at the mixer's master effects return, check it to make sure that a usable signal (that is, near 0 dB) is coming back into the board with the master effects-return control set between the midway and maximum points. Then, monitor those levels with the effects unit's mix control set to 100 percent. Various effects units will probably calibrate somewhat differently through those steps, but as long as levels are in the usable range (near 0 dB) without distortion, that should not be cause for alarm. Note the positions of all effects-send and -return masters so they can be preset before a mix.
Next, route the active channel to the 2-track mixdown recorder (DAT, CD, hard disk, or analog) and adjust its input levels so the deck's meters read zero when the mixer's stereo output bus reads zero. (An outboard sampler can be calibrated the same way.) To establish unity gain through the mixdown deck, connect the deck's outputs to the mixer's 2-track inputs (or other inputs if it has no dedicated inputs), record the calibration signal you're using, and then play it back and adjust the deck's output until the mixer's meters read 0. Make sure the reference level of the 2-track or sampler matches the mixer's output and 2-track input reference level (-10 dBV or +4 dBu).

During those procedures, it is possible (though not necessary) to monitor the audio signal with headphones, which can also be used to check the gain in any headphone mix buses following the procedures outlined for the aux buses. In addition, adjust and note comfortable levels through headphone distribution amps. Bear in mind that different headphones have widely varying efficiency ratings, and one brand or model may be much louder than another.

For the final stage, make sure your mixer's stereo master and control-room output faders are all the way down. Then, power up your main monitor amplifier(s). If the amp has gain controls, set them at 100 percent. Raise the mixer's main faders to zero and slowly increase the control-room level until a comfortable listening level is established. If a sound pressure level (SPL) meter is available, set the listening level between 80 and 90 dB. Clearly mark the position of the control-room master for future reference as a safe setting.

If the safe control-room setting seems low (say, below three on a pot marked from one to ten) or the monitors are noticeably noisy, you might want to attenuate the amp gain. If you do that, turn down both sides of a stereo amplifier by the same amount and test for audible stereo balance using a high-frequency mono source panned to center. In addition, be aware that trimming a power amp's gain lowers its headroom, which may interfere with transient response. Clipping distortion may also occur if the mixer has to overdrive the amp's input to achieve the desired listening level. For those reasons, most professionals run their amps at 100 percent and take care to keep the control-room pot at a conservative setting.

Another approach...

by Scott Wilkinson

Another approach to gain staging also works well in the studio. The goal is to minimize the audible noise level while maximizing the signal level at each gain stage. The procedure yields a wonderfully quiet sound system with plenty of signal level.
The idea is to work backward from the control-room amp to the mixer and aux effects and, finally, to the sources. EQ should be completely flat as you work through these steps.

1. Connect everything.
2. Set volume controls to minimum.
3. Power up all devices in order: instruments and preamps, effects, mixer, amplifier.
4. Raise mixer control-room volume to maximum; back off until noise disappears.
5. Raise amp level to maximum; back off until noise disappears.
6. Raise mixer master level to maximum; back off until noise disappears (hopefully, unity gain).
7. Raise each mixer subgroup fader to maximum; back off until noise disappears (hopefully, unity gain).
   For each aux effect bus:
8. Raise effects mix or wet/dry control to 100 percent or full wet.
9. Raise effects input and output levels to 100 percent.
10. Raise mixer effects-return level to maximum; back off until noise disappears.
11. Raise mixer master send level to maximum; back off until noise disappears.
12. Raise each channel aux-send level to maximum; back off until noise disappears.
   Adjust as needed for each channel during mix but try not to exceed that level.
   For each mixer input:
13. Raise the mixer input fader to maximum.
14. Raise level of instrument or mic preamp to maximum.
15. Raise mixer input gain/trim to maximum; back off until noise disappears.
16. Lower input fader to unity gain.
17. Lower instrument or preamp level to minimum before playing.
18. Slowly raise instrument or preamp level to desired level while playing, as indicated on mixer or tape-deck meters.