Maximizing Your Console

Taking a logical, signal flow approach

by Curt Taipale

Let's take a poll: how many of you use a road map to find the most efficient route to get where you're going? Wow, a couple of folks actually raised a hand! It may seem odd, but one of the best ways to understand how to operate a mixing console is to learn its signal flow. Figure out a simple road map that illustrates how the signal gets from the input to various outputs.

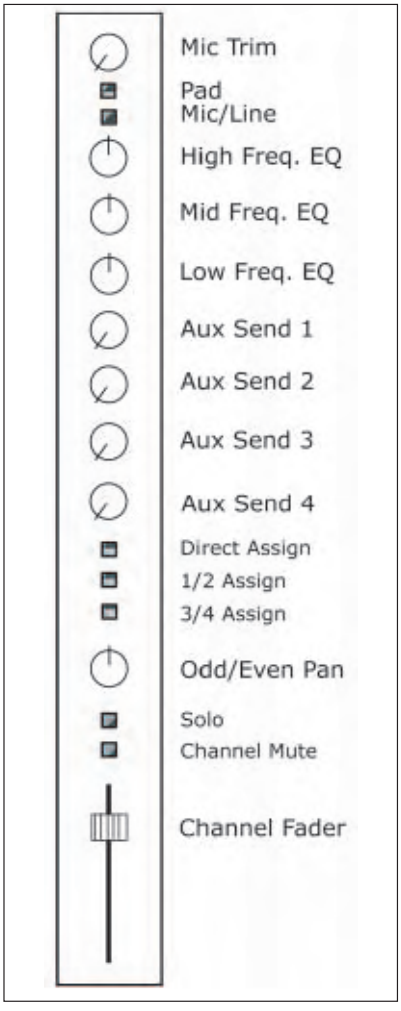

Figure 1: Input strips can vary, but most look roughly like this.

Without this understanding, then the likely approach is one of, "Well, if I twiddle these three knobs just so, push this fader up to here, and never ever let the master fader go past this indelibly engraved red mark that someone has carved into the face of the console, then maybe it'll work today!" There really is a better way, and it's the concept of signal flow logic. Let's get down to work.

Figure 1 shows the input strip from a hypothetical but typical console. We'll talk about each of those controls as we make our way through the signal flow. On the surface, one might think that the sound flows through each control in sequence from top to bottom. Actually, manufacturers have arranged the controls in a way that they think will be the most efficient way to operate the console.

For example, controls that you would for the most part "set and forget" are placed at the top of the input strip, well out of your reach, such as, say, the microphone trim. Controls for immediate and frequent access are placed right at the user's fingertips, like the channel faders. As we move on to the signal flow diagram later, it's a good idea refer back to **Figure 1** from time to time. It helps tie the signal flow into the related controls.

The signal flow diagram (**Figure 2**, next page) is again from a hypothetical console. This arrangement of the controls is fairly common, and once one understands this concept, it can be applied to all consoles.

WHAT THE SIGNAL SEES

The diagram reads from left to right, top to bottom. So we start at the upper left corner at the mic input - a very good thing to have on a console! The first component that the signal sees is the input transformer. Now, the reality is that on most consoles we work with today, especially on lower cost consoles, the input transformer is actually replaced by a less expensive electronic circuit that accomplishes much the same task. The purpose of this input stage on each channel is to receive the balanced signal from the mic and to cancel any extraneous noises.

The strength of that mic signal is quite low, and needs to be brought up significantly to operate with a quiet signal-to-noise ratio throughout the rest of the console and beyond. That gain increase is accomplished with the mic preamp. Note that there is a related control called the mic trim (or mic gain). This is a gain adjustment for the mic preamp that allows you to adjust for differing signal strengths coming into each channel.

If there isn't a control labeled mic gain, it simply means that the manufacturer has chosen to keep the price of the console down by replacing that variable resistor (called a potentiometer) with a fixed resistor. The manufacturer figures that you're going to plug in mic "A" and mic "B", and you'll place them on these various instruments or voices; then a ballpark gain setting is chosen that should serve the needs of all of the inputs. (That fixed resistor costs about one-tenth as much as the pot - multiply that times one for each channel.)

However, the tradeoff for lower cost is flexibility. For example, you might decide that you get the best sound if you mic a certain instrument in a certain way. If the incoming signal is too hot or too soft, you could use that sensitivity control to adjust for the difference. Without that control, creativity is required; i.e., what's

Just like any rugged terrain, this territory can be road-mapped.

another way to drop the strength of the signal coming into the console on a particular channel? One could move the mic farther from the instrument.

The main path of the signal goes next to the equalizer section of the channel. This is really just a glorified volume control which allows you to turn the sound up or down in various frequency bands. Your intent should be to improve the tonal quality of a particular voice or instrument.

Remember to be subtle with EQ adjustments. A small EQ change can make a huge improvement in the quality of a person's voice. But there are times when altering EQ is simply not needed. Part of a mixer's job is to know the difference, and then to not twiddle the EQ knobs just because they're there.

HIP TERMINOLOGY

The signal then flows on to the channel fader. Fader is the hip term - I want all of you to be hip, so stop using the term "slider". Here's where most of the action of the mix happens. The fader provides the means to make adjustments in the balance of each voice or instrument during the song or sketch.

Faders generally have a logarithmic taper; that is to say, the rate that the volume changes will vary along the throw of the fader. For example, a quarter of an inch move of the fader near the top of its travel may account for a 5 dB change in level, whereas the same quarter of an inch move of the fader near the bottom of its travel may account for a 20 dB change in level.

As a result, the mix will be smoother and quieter if the faders are operated in the upper one-third of their travel for the most part. This deals with gain structure; hang on, we'll get to that point soon.

Now our signal path is at the summing point. The Greek Sigma symbol with a circle around it is a common notation for a "summing" or "combining" point. The signal flow diagram here shows just one channel of the console, but let's say that we've really been talking about a 16-channel console. Each channel is identical up to this point, and then they all combine or sum.

Signal then goes to the master fader where the overall level of all the signals is controlled with one fader. Signal then goes through the output stage of the console circuitry, and then to a connector/cable on the back of the console where it's fed to the input of the power amplifier(s), for example.

That's the main path through any console. No matter how expensive, that's it. But we still need some more information on how the controls interrelate, so look again at the signal flow diagram (**Figure 2**) and note that right after the mic preamp, but before the EQ, there is a pickoff point. Think of it as a side road on your map.

The path drops down to a control I've labeled auxiliary 1. Follow the signal path down, and you'll see that it comes in contact with the aux 1 bus. This is basically a piece of wire that runs from one side of the console to the other. Signal then flows on to the right to the aux 1 master control, which controls the overall level of all the signals fed to the aux 1 bus.

THE FADE ROUTE

Let's imagine for a moment that we're using aux 1 to feed a set of stage monitors. We're looking at channel 5 that is the lead vocal mic, and aux 1 feeds the lead vocalist's stage monitors. If you were to make an adjustment to the EQ on channel 5, would that change be heard in the vocalist's monitor mix? No, because the feed for the monitor mix is picked off before the EQ.

If making adjustment to the lead vocalist's channel fader, would that change be heard in the vocal monitors? No. We're using what is known as a "pre-fade" auxiliary send, because the signal is picked off before the fader. It also happens that the signal is picked off before the EQ as well, but the proper term is "pre-fade".

But what if we make a change to the mic trim on channel 5? Yes, because it's upstream in the main path, changing the setting of the mic trim would affect the signal to both the main path and the auxiliary send.

This relationship is usually the preferred setup for mixing monitors. Most vocalists and musicians prefer that once their monitor mix has been established during soundcheck and

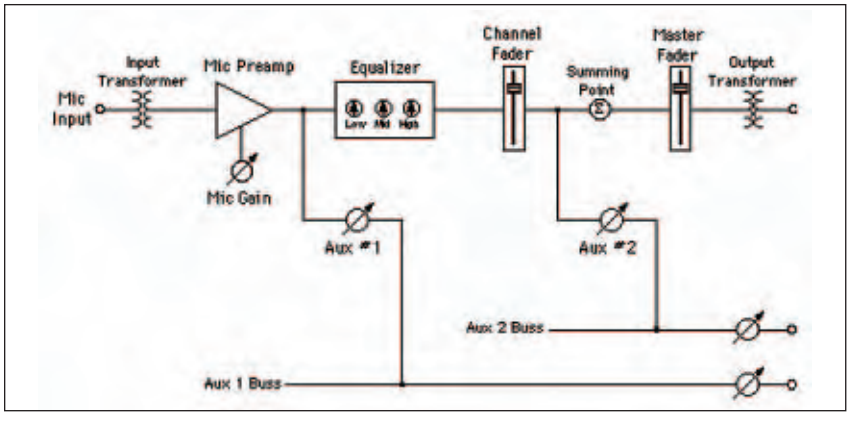

Figure 2: Apply knowledge of the signal flow to attain console zen.

rehearsal, that their mix not change during the course of the worship set (or show) unless they specifically request it. Unexpected changes in the mix can be very distracting.

Just past the fader is another pickoff point that goes to a control labeled auxiliary 2. As before, this also drops down to an aux 2 bus and its related master control. In this case, let's imagine that the aux 2 output on the back of the console is patched to the input of a reverb unit sitting in an equipment rack next to the console.

In order to hear the reverb, the output of the reverb device is connected to a line level input on the console called the auxiliary return. We establish our dry-to-wet ratio by adjusting the send on aux 2 of channel 5, and of course the return level of the reverb unit is adjusted with the aux return. The "dry" sound is the direct signal from, in this case, the vocal mic, and the "wet" sound is the reverb effect.

DRAW AND REDRAW

Now that the reverb sounds the way we like... If an adjustment is made to the lead vocal fader, does this affect the signal fed to the reverb device? Of course it does. What if you make a change to the EQ on channel 5? Yes, that change will also be reflected in the reverb sound.

Does adjusting the master fader affect the feed to the reverb device? No, because the feed to aux 2 is picked off before the master fader. What if the aux 1 send on channel 5 is changed - will that affect the signal fed to the reverb unit? No. Even though aux 1 is picked off upstream from the aux 2 feed, it's still just a pickoff point and will not affect the main signal path. Realize that the aux 2 send described here is commonly called a "post-fade" send.

It's typical to use a post-fade send to feed effects devices because doing so allows us to easily maintain that dry-to-wet ratio. It is generally a more musical sound for the loudness of the reverb to track with the direct sound. If you push the fader up, the direct sound increases, as does the feed to the reverb unit, so the dry-to-wet ratio is maintained. If you pull the fader down, the reverb sound diminishes as well.

The owner's manual for your console will include a block diagram. These drawings are often awkward to read, so redraw it! Not only will it be easier to read and simpler to understand, but the point of signal flow is reinforced understanding of its flow.

All you have to do is get past the manufacturer's silkscreen on the console. For example, what one manufacturer calls "solo" another labels the same control "cue", and still another labels it "PFL"; what one manufacturer calls "monitor send" another labels "foldback".

But it doesn't matter if the console cost $600 or $60,000. The reality is that they both possess the same basic signal flow.

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