

Back to Basics: Gain Structure

We cover a wide-ranging list of topics in this column. But sometimes it's appropriate to return to look some of the industry's "basic" principles. This month we'll dust off the volume on gain structure. Once a frequent topic of conversation for the industry, gain structure has become less important as electronics have improved — although it should certainly not be ignored.

In addition to improved electronics, the explosion of digital mixing technology has changed much of the gain structure conversation. Issues like the gain behavior of inserts or summing buses are largely relegated to the dustbin. Additionally, more and more of the signal chain, before and after the mixing desk, is transitioning toward purely digital distribution. On the whole, this is a good thing. The resulting pro audio systems are quieter, the signal distribution easier and more portable, the connections less prone to ground loops, and the gear footprint smaller. Gigs are easier to set up, and the reduced infrastructure translates into business benefits for pro sound providers.

With better electronics and digital technology having removed many of the technological barriers that originally drove gain structure considerations, let's turn the conversation around and focus instead on how gain structure influences the creative processes involved in mixing. Through this, we will show that gain structure remains an important topic for those who want to mix quickly and effectively. But first, a review of the basics of gain and gain structure.

The Basics

Gain, at its simplest, is a measure of the relative amplitude of a signal after it goes through a process or piece of equipment; gain quantifies how much the volume changed. Because we have many different input sources of different volumes, and we would like to rebalance the volumes of those sources before returning them back into the wild through the P.A., we have many different gain levels throughout the mixing process.

We cannot, however choose gain values with total impunity. Circuits are only

capable of producing so much gain, and inputs can only accept signals less than a given amplitude. Further, all circuits have some inherent self noise. The interested reader can research Johnson-Nyquist noise for more details. Designers have reduced this noise over time, but it is always present. If we mishandle where and how we apply gain, it can accentuate the noise inherent in our electronics.

Thankfully, the process to keep electronic noise at bay is straightforward. The noise level, or noise "floor," of electronics is fixed by their design, but the relative gain of the incoming signal is not. Thus, the principle is to *maximize* the level of the incoming signal at every point in the electronic signal chain. This ensures the signal is as far above the noise as possible. In a completely analog signal chain, this principle is critically important. There could be many input and output stages in series, each contributing some electronic self-noise to the equation.

Digital processes, by contrast, do not contribute self-noise at each point of the signal chain, eliminating much of the noise buildup that has occurred historically. As digital processes will continue to spread throughout the industry, the challenges of maximizing the aptly named "signal to noise ratio" will continue to get easier and easier. Eventually, everything between the preamp and the speaker driver will communicate via digital interlinks. Because much ink has been spilled elsewhere on the nuances of maximizing signal-to-noise ratio, we will instead turn to the art aspects of gain structure in the service of mixing.

Gain Structure — An Artistic Approach

We have acknowledged the decreasing importance of traditional gain structure setup for modern, largely digital sound reinforcement systems. One place, however, where considering gain structure remains important is within the confines of the mixing desk, and, specifically, within the confines of the fader moves we use for mixing.

Historically analog mixing desks have had "log taper" faders, which is to say sliding the fader has a logarithmic influence on the signal level as one slides it. From a hearing standpoint, the use of log taper faders makes sense, as humans

perceive loudness in a manner similar to the logarithmic curve. But logarithmic faders impose a specific limitation on the mixing process. There is a “sweet spot” where fader movements result in precise control of the channel volumes. Far away from the sweet spot, a small fader movement results in a large change of volume. Look at the faders on a mixing desk, notice the signal level labels, and you will quickly see how this comes into play.

For the most part, digital mixing desks choose to emulate their analog brethren and implement volume encoders that behave logarithmically. Thus, most digital mixing desks exhibit the same sweet spot as their analog cousins, not out of design limitations, but out of interface considerations for the board user. The precise control of the fader, typically in the range from -5 dB to +5 dB, may drive the mix engineer to re-investigate whether to follow what theory says is optimal for signal-to-noise ratio. Specifically, theory indicates that the preamp gains be set to maximize the levels coming in from each source, running the preamp right below overload (i.e., clipping). Then, after this procedure is complete for each input, the fader levels end up wherever they may. Thus some faders may be high in volume and some very low.

There are a couple of limitations of the theoretical procedure in practice. First, the fader location may end up well away from the sweet spot, and thus, small movements of the fader will result in a large change in volume. Secondly, it can be very difficult to set the preamp gain precisely, as sources are often widely variable in output level from moment to moment, depending on the vibe of the performance. Set the gains too high, and in the middle of your set, suddenly multiple inputs start clipping as the performers dig in.

A Quick Approach

A quicker, more practical approach to gain structure in mixing is as follows:

- Set the master fader to the bottom of the sweet spot region (e.g. -5 dB).
- Set the channel faders for each input in the sweet spot, usually at the 0 dB point.
- Unmute the inputs and dial up a rough mix using the preamp input gains as if they were faders. Let the preamp gains fall where they may.

- Now, forget about the preamp gain knobs and focus on mixing the show using the faders. All the faders are now near zero dB for maximum mixing finesse.

Now, the procedure above is clearly not optimal for maximizing signal-to-noise ratio, as the preamp gains will be low compared to the theoretical approach. However, this approach is optimal for quickly setting up a mix, and for having maximum control over the mix during your event.

Does this alternative gain setting approach really hurt us in terms of signal to noise ratio, especially in the digital console age? As modern preamps designs are rather quiet, the noise penalty incurred by not optimizing the preamp gain is comparatively minimal. Also, the digital mixing process is not adding any more electronic self-noise to the mix. Digital fader encoders do not pass any signal, and do not contribute any self-noise or other contamination. Further, the ambient noise picked up by microphones before the preamp remains in the same relative proportion to the source signal *regardless* of preamp gain.

While our perception of the microphone self'-noise, plus ambient room noise, changes as the gain level increases, we should not let that fool us into thinking the signal to noise ratio of the microphone has somehow changed. Rather, "noise masking" — due to the inherent background noise of the venue — helps hide the microphone's inherent noise (self-noise plus ambient) at low preamp gains. Overall, our theoretically incorrect procedure yields practical benefits with little downside.

Mind Your dB's

In the above section, the mix gain setting procedure is not following the textbook. Does this mean we can ignore setting gain structure for maximum signal to noise entirely? Not at all! Do you have a guitar player with a noisy pedal board, or a hissy direct box? These are perfect applications for working through the gain structure to insure that each input stage is getting the maximum signal possible from the previous output stage. The trick is applying gain structure theory smartly at the appropriate times.

As a practical example, let's consider configuring a new amplifier rack. Amplifiers apply a lot of gain, usually more than 25 dB, to their input signals. Further, amplifiers have a lot of stuff going on inside the chassis, and their self-noise levels are typically higher than a mic preamp. These details are motivators for maximizing the signal to noise ratio of the signal that enters an amplifier. A prudent approach to configuring the amplifier gain must consider the input attenuators and the internal amplifier gain.

Generally, it's good practice to leave the amplifier's input attenuators wide open. This ensures that the amplifier can only be turned down, not up, during a gig. This is a protective measure to guarantee that fans, techs, or other sets of fingers cannot add gain at the amplifiers. If the attenuators are left wide open, then the overall input gain of the amplifier must be managed by the internal gain setting.

Most modern amplifiers are configured for constant voltage gain, and the most common values are 26 dB, 32 dB, and 38 dB. We assume that the next device upstream of the amplifiers is either a modern equalizer or digital signal processor and that either device is capable of applying a strong signal. This means that the input gain for the amplifier should be set to +26 dB, or the *smallest* amount of amp gain. By setting the amplifier for the lowest gain, we insure that the upstream processor must output a robust signal to drive the amplifier to full output, and that pushes the signal of the processor above the noise floor as far as possible.

To make the above amplifier gain determination, we did not perform any math. There was no calculation of dBu, dBV or dBm — and no use of logarithms. There are some circumstances, especially when matching amplifiers from different manufacturers or when using older amplifiers that do not have constant voltage gain, where the math is necessary. But smart application of the principle behind good gain structure can be done by the most math-phobic pro audio technician. One doesn't have to calculate every last dB value to provide a 90 percent solution for improving the signal to noise ratio.

Conclusion

The ever-improving equipment shouldn't obviate us from learning the principles of gain structure, but we also should not be placed in a mixing straightjacket, with faders all over the map, from trying to get every preamp input signal gain set just below clip. The trick lies in not confusing technique with results. If an input is obviously noisy, then physics dictates that we should reduce the input gain from that stage, and increase the strength of the output signal from the previous stage. If we persistently keep this general principle in mind, we can keep the signal noise ratio of all parts of the chain high without regularly resorting to the calculator to determine input attenuation. The big idea is simple and quick enough to teach to musicians, so they can be thinking about reducing the noise of their various inputs in the same fashion. In pro audio, we are only as good as our sources (and their noise), and having everyone on the same team is a great step towards a pristine show.