TECHTOPIC : Gain Staging

Cleaning up your sound. ***by Bruce Bartlett***

ARE YOU GETTING the best possible sound from your mixing console or sound system? By setting up your mixer and system for proper gain staging (gain structuring), you might be surprised how clean they can sound.

Gain staging is the process of setting the gain of each stage of a mixer, or of a sound system, to achieve the best compromise between noise and distortion. The goal is to have every stage in the mixer (or the sound system) operating at its optimum signal level.

What's optimum? The loudest peaks of the signal should be a few dB below clipping in each stage, even if you apply a boost in an equalizer, or makeup gain in a compressor. That way, the peaks are not audibly distorted, and the signal is well above the noise floor of the mixer.

In every stage, you want the signal level high enough to cover up the noise, but low enough to avoid distortion. Every audio component works best at a certain optimum signal level, and this level is usually indicated by a meter or LED built into the device. When gain staging is correct, all the components in the system clip at about the same time.

In this article I'll suggest some effective ways to set the gain structure in a mixing console and in an entire sound system.

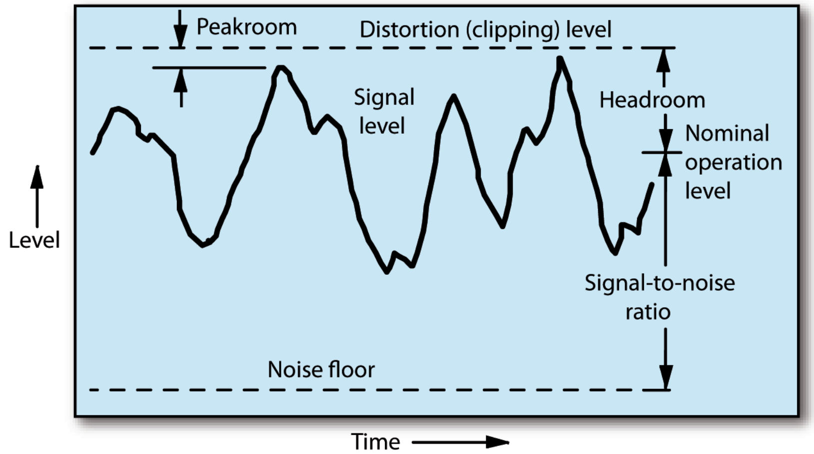


Figure 1: The range of signal levels in an audio device.

BACKGROUND

First, let's review the concept of signal levels in an audio device (**Figure 1**). At the bottom of the graph is the noise floor of the device – the level of noise it produces with no signal. At the top is the clipping level – the point at which the signal peaks flatten and distortion become easy to hear. In between is a range in which the signal sounds more or less clean.

As shown in **Figure 2**, a musical or speech signal changes in level (voltage) continuously as it plays. Imagine a musical passage with a low-level synth pad, but with high-level drum hits. The average level or volume of the passage is low, but the transient peak levels are high.

Peak levels may be up to 24 dB above average levels depending on the type of signal. Percussive sounds have much higher peaks than continuous sounds do (synth pads, organ, flute) – even if the two signals have similar average levels.

The meters in your console might show signal levels in two modes: RMS and peak. RMS (or root-mean-square) readings correspond to the average levels. A VU meter approximates average levels. Peak readings, such as displayed on LED bargraph meters set to peak mode, show the level of peaks or short transients. The average or RMS level indicates approximately how loud the sound is, while the peak level shows how close the signal is to clipping.

Generally you want the signal level to be as high as possible without the peaks clipping or distorting, but there are exceptions, which we'll cover in this article.

In **Figure 1**, signal-to-noise ratio (S/N) is the level difference in decibels between the average signal level and the noise floor. The higher the S/N, of course, the less hiss you hear.

The level difference in dB between the nominal (average) signal level and the clipping level is called "headroom" (**Figure 1**). The greater the headroom, the greater the signal level the device can pass without running into distortion. If an audio device has a lot of headroom, it can pass high-level peaks without clipping them.

Syn-Aud-Con instructor Pat Brown uses the term "peakroom" to mean the dB difference between signal peaks and the clipping level (**Figure 1**). *Headroom* is the space between the average level and clipping; *peakroom* is the space between signal peaks and clipping.

You want to set your mixer controls so that, in every stage, the signal has some peakroom so that peaks don't distort, and the signal is well above the noise floor.

SETTING GAIN STRUCTURE IN A MIXER

Let's look at each mixer stage from input to output, and tweak the settings for best gain staging.

Mic Preamp

Put the most gain in the console up front. Try to get the maximum gain in the mic preamp, not farther along in the chain. Here's why.

Suppose you turn down the preamp's gain trim more than it needs to be. If you try to make up that gain by turning up the submaster and master faders near the top, the noise from the previous stages will be amplified along with the signal. But if you turn up the preamp's gain trim as high as possible without clipping, you can turn down the submaster and master, reducing console noise in the process.

Imagine that the noise floor in a mic preamp is -86.5 dBu. If you adjust the gain-trim pot so that the loudest signal peak is just below clipping, you'll get the maximum S/N that the preamp can provide. Let's say that figure is 90 dB S/N referred to +4 dBu at the console output.

Now turn down the gain trim 20 dB for the same input signal. (That's 20 dB lower than necessary). You've just reduced the S/N by 20 dB, so now the S/N is 70 dB at the console output. If you reduce the gain trim by 40 dB, the S/N becomes 50 dB, and so on.

In most mixers, an LED labeled "clip", "peak" or "OL" flashes just before signal peaks reach the clipping level – typically when the signal is 0 to 3 dB below clipping. So turn up the gain trim until the clip LED flashes on the loudest notes, then back off 10 to 15 dB to create some peakroom. That way, when the band plays louder than expected, or when you apply some EQ boost, you won't run into clipping.

Note: Some engineers recommend that you let the clip LED flash occasionally. When that happens there is little to no peakroom between the peak signal level and the clip level, but there is maybe 15 dB of headroom between the average signal level and the clip level (depending on the signal's crest factor). The peaks are almost clipping when the LED flashes, but the average level is 15 dB below clipping. This procedure results in maximum signal-to-noise ratio at the expense of peakroom. It might be the best method to use with noisy mixers.

Mackie recommends another way to set the gain trim for their mixers:

1. Set EQ flat.
2. Solo the mic channel and press the MODE switch to the Level Set position.
3. Adjust the trim to get 0 dB on the meters.
4. If you apply EQ, re-do step 4.

According to my measurements, that trim setting makes signal peaks fall about 15 dB below the clip light's flash point. That's equivalent to turning up the trim until the OL light comes on, then backing off 15 dB. So the Mackie procedure creates about 15 dB of peakroom between signal peaks and the clip point.

What's the reason for all that peakroom? You can apply plenty of boost in the equalizer without clipping. And you are unlikely to clip the input signal if a musician plays louder than anticipated. Also, you can sum many channels at unity gain without overloading the stereo mix bus (more on this later).

Some mixer inputs have a switchable pad, which is a resistive attenuator before the mic preamp. A 20 dB pad drops the signal level by 20 dB (1/10 the voltage) before it reaches the mic preamp input. But it also reduces the S/N by 20 dB. If you turn down the gain trim all the way but the channel is still clipping, switch in the pad. Otherwise leave it out so you don't degrade the S/N.

This advice applies to standalone mic preamps as well as console mic preamps.

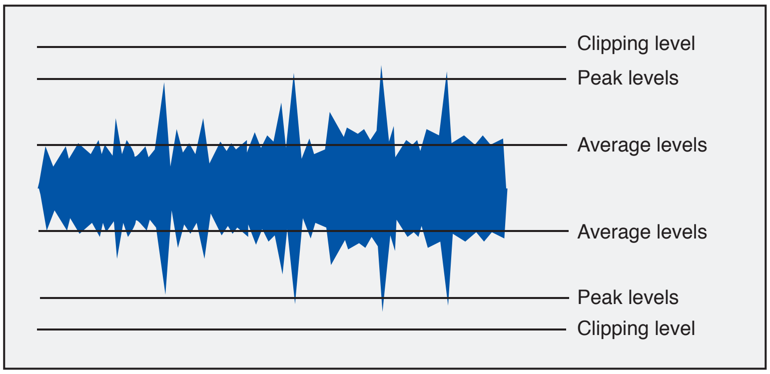


Figure 2: Average levels and peak levels in a musical signal.

Equalizer

In most consoles the signal enters a mic preamp, then goes to an equalizer, then a fader. If the signal is near clipping coming out of the preamp, and you boost that signal over some frequency band in the equalizer, the signal is likely to clip in the equalizer. So as we said, you need to create some peakroom – some level difference between the peak signal level in the preamp and the clipping level. To do that, set the gain trim about 15 dB below the clip point.

If an EQ boost results in channel clipping, turn down the trim until the clip LED goes out. That's assuming the clip LED is post EQ, which is not always the case.

Faders

The fader settings are important too. For starters, set them close to unity gain (design center). Do the same for the master faders.

The signals from all the channel faders sum into various sub-mixes that you set up. Every time you double the number of faders set to unity gain, feeding the same mix bus, their summed signal level increases by 3 dB. That's when each fader is receiving a different, non-correlated signal. In other words, summed level in dB = 10 log N, where N is the number of faders set to unity gain (design center).

Number of Faders	Summed level
1	0 dB
2	3 dB
4	6 dB
8	9 dB
16	12 dB
32	15 dB

Suppose you set 16 faders to unity gain and route them all to the stereo output bus. The summed signal will be 12 dB higher than the signal of one channel alone. So for example if you have 16 faders in use, it's a good idea to set them 12 dB below unity gain. That will prevent overload in the mix bus.

If you set the master very low, say 20 dB below design center, you will tend to push up the channel faders to get enough volume and a meter reading near 0. That will overload the mix bus, giving a harsh, fatiguing sound. If you can't set the master at design center because the house sound is too loud, simply turn down the power amplifiers. More about this later…

If the signal going into several faders is the same (for example, hum), the signal increases 6 dB for every doubling of faders. If you have equal hum pickup in every channel (not likely), a 16-channel mix will boost the hum by 24 dB! That's one reason to mute or turn down unused faders.

Mixing With the Gain Trims

Syn-Aud-Con member Rick Chinn suggested an interesting way of optimizing a mixer's gain staging:

1. Set all the channel faders and master fader to design center.
2. Set up a preliminary mix with the gain trims, making sure the output bus meters read near 0.
3. Fine-tune the mix with the faders.

The claimed advantages are:

This results in approximately correct gain staging in the mixer and maximum headroom in the input channels.

"Noise theory tells us that for best system noise figure, the first stage wants to get all the gain, and everything afterwards runs at/near unity."

You can move faders for solos, then line them all up to get back to the basic mix.

The faders are at their highest-resolution positions.

In other words, you set the input trims by ear to get a good mix, rather than setting them to get a certain signal level. I don't know whether this would result in optimum gain staging for each channel, but I'm offering it for your consideration.

A disadvantage of this method is that it changes the monitor mix as you tweak the trims.

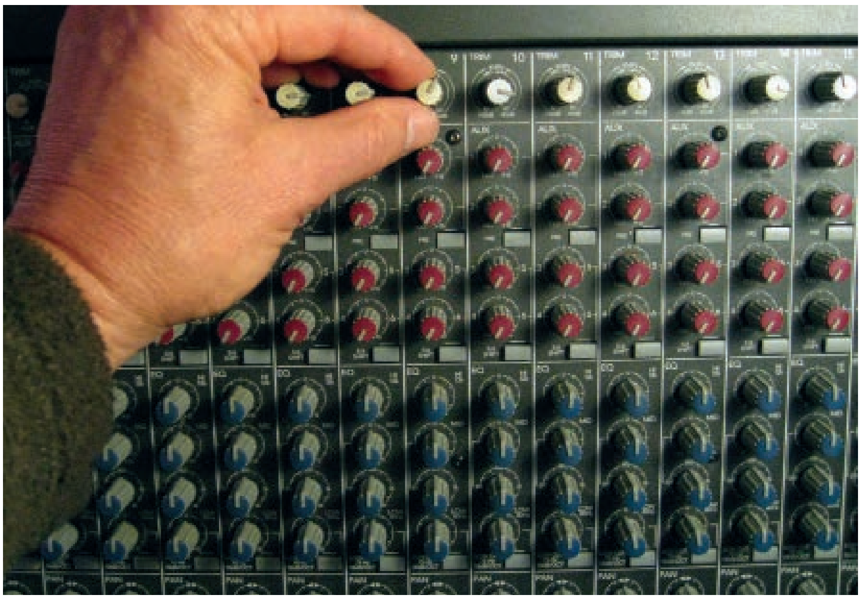
SETTING GAIN STRUCTURE IN A SOUND SYSTEM

Now let's examine an entire sound system of these components:

1. Musical instrument
2. Mic
3. Mixer (already covered)
4. Effects devices
5. Compressor/limiters
6. Crossovers
7. Outboard graphic EQs
8. Power amplifiers

I'll suggest how to adjust each component.

Musical instrument: Electric guitars and keyboards should put out a strong signal to override hum and RFI picked up by the instrument cable. So turn up their volume controls about 3/4 from the top (assuming that allows enough player expression). The more you can turn up the guitar, the more you can turn down its amp to achieve the same loudness. Turning down the amp reduces its hiss and any guitar-cable hum.



Take care that the level is nominal in each stomp box in a chain. For example, wah-wah pedals boost the signal several dB at certain frequencies, and this can result in distortion if the level feeding the pedal is too high. Before going on tour, listen to the output of each stomp box in normal use and make sure it's not more distorted or noisy than it should be. As always, a noisy output means the input signal is too low, and a distorted output means the input signal is too high.

In general, put signal-boosting devices near the beginning of the chain of boxes, and vice versa. A compressor should go up front because it keeps the signal level high throughout the chain. If you put a compressor at the end of the chain, it will bring up system noise during the release time. Similarly, a noise gate or volume pedal should be at the end of the chain to reduce noise.

The louder a vocalist sings into a mic, the better the signal-to-noise ratio of the mic signal. A loud singer overrides the self-noise of the microphone; a quiet singer or distant-miked singer exposes the mic's noise floor.

Microphone: Some condenser mics have a built-in pad. It attenuates the signal between the mic capsule and the mic's internal circuit. Because the pad degrades S/N, switch in the pad only if you hear distortion in the mic signal, and the mic's signal is not clipping in the console.

Effects: Most signal processors have an input level knob and output level knob. For starters, set the knobs to unity gain. Do the same for the effects-return control in your mixer. When your mix is nearly complete with effects sends in use, check the processor's input meter or LED. If it is showing an overload, turn down the processor's input level, and turn up the output level to compensate. Do the opposite if the input levels are low. The goal is to have nominal signal levels in the outboard effects devices, and unity gain through them.

Look for +4/-10 level switches on your console effects sends and in your effects units. Set them to match.

Compressors/limiters: Similarly, don't clip the input of a compressor, and don't feed it too low of a signal level. Generally you set the makeup gain to match the measured gain reduction. This achieves unity gain through the compressor on the loudest notes.

Electronic crossovers and graphic equalizers: Make sure their clip lights flash only occasionally. Generally they are set to flash a few dB below clip. Some manufacturers say to set the input and output levels to maximum; others say to set them 1/2 to 3/4 up. Check your manuals.

Power amplifier: Start with the amp's level controls halfway up. Set your mix to read around 0 on the mixer meters, then turn up the power-amp level until the sound is as loud as you want it.

It's a mistake to run a power amp wide open (full volume). If the amp's input level controls should always be set to maximum, amp manufacturers would not include them. They are there to set proper gain staging.

Turning down an amp's level controls does not reduce its power. Doing so varies the signal level needed to drive the amplifier to full power. You can still get full power from an amp even with the level controls partly turned down, if the input signal level is high enough. Those level knobs don't adjust the power level; they adjust input sensitivity.

If you turn an amplifier up to maximum, you amplify any noise coming out of the sound system. Also, you might run the console well below 0 on the console output meters so the house sound is not too loud – but this degrades signal-to-noise ratio. So, start with your console running at an optimum level near 0 on its meters, then adjust the power-amp input levels for the desired loudness.

If your system includes an electronic crossover and multiple power amps, set the level of each amp for the desired loudness from the lows, mids and highs.

SUMMARY

In the console, use a pad only if the gain trim can't prevent clipping.

Get the maximum gain up front, in the musical instrument and in the mic preamp.

If the mixer is not too noisy, turn down the input trim about 15 dB below clipping to create some peakroom. This prevents overload in the following stages.

In any mixer stage or audio component, avoid too-high levels (which sound distorted) or too-low levels (which sound noisy).

Try to keep channel faders, sub faders and master faders near design zero, the shaded portion of fader travel about 2/3 up.

The more channel faders in use, the lower they should be set to prevent overloading the mix bus.

With the mixer meters peaking around 0, set the power amp input levels to achieve the desired loudness. Don't set the power amp levels fully up unless the volume is too quiet otherwise.

By following these tips, you can make your system sound as clean as it can be. Goodbye noise and distortion. ■

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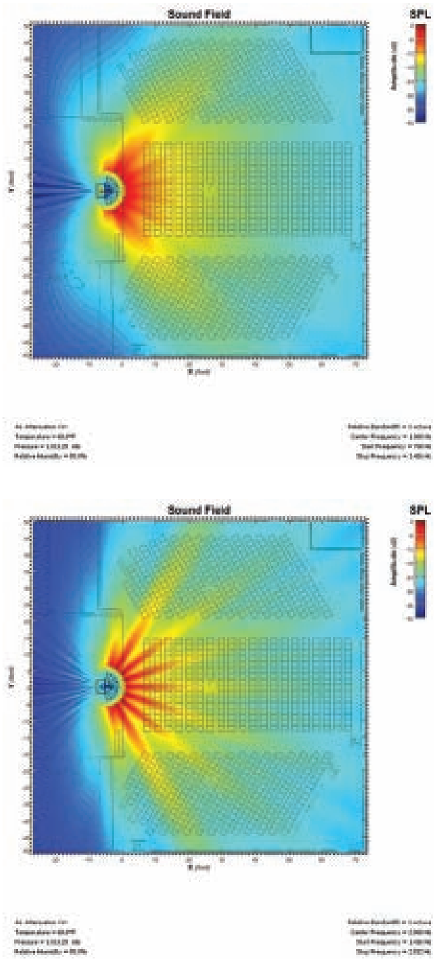
John Meyer with Meyer Sound's JM-1P point source loudspeaker.

Meyer Sound JM-1P

Reclaiming the horizontal through a point-source system

by John Meyer

This Factory Direct was submitted by Meyer Sound. Live Sound International makes every effort to eliminate any use of marketing-inspired hyperbole.



Two Soundfield diagrams, showing the JM-1P at 1 kHz, and then 2 kHz.

IN HIS SEMINAL 1940S WORK, Elements of Acoustical Engineering, Harry Olson clearly laid out the fundamental concepts behind loudspeaker arraying, specifically describing the behavior of a series of acoustical point sources. Yet, even with the rise of rock music in the 1970s, which brought about the birth of high-level sound reinforcement, these concepts were not fully deployed except, occasionally, in horns.

Meyer Sound has been working with point source arrays for some 30 years, earning patents for advances such as an array method relying on a common frequency-independent acoustic center, a horn design that brought a major improvement in distortion levels, and the trapezoidal cabinet shape, which is descended to the JM-1P, to name just a few.

The 2009 release of Meyer Sound's JM-1P arrayable loudspeaker marks a significant evolutionary step in the development of point source arrays.

Point source arraying is a systems-based approach that takes into account the interaction between loudspeakers. It offers the advantages of predictability and consistency, critical qualities in the field of live performance. Further, horizontal coverage can be shaped with equalization and level tapering to fine-tune coverage for a given application.

WHAT A DIFFERENCE 29 YEARS MAKES

Meyer Sound released the UPA-1 in 1980, ushering in the beginning of full-range point source arrays in sound reinforcement. The UPA-1 was revolutionary in at least three aspects. First, it was the first time a loudspeaker had been created and sold with a dedicated electronics processor that provided frequency and phase correction, as well as driver protection. Second, it featured a horn design patented by Meyer Sound that lowered distortion levels well below those of existing designs. Third, the loudspeaker was housed in a cabinet with a trapezoidal shape, also patented by Meyer Sound. The trapezoidal cabinet enabled multiple UPA-1 units to be arrayed in a configuration that implied a virtual acoustic center behind them, causing the array to radiate similarly to a true point source.

With the new JM-1P horizontal array loudspeaker, Meyer Sound has combined the latest technology advances in horn design, the ease and performance of self-powered systems,