Line Up And Be Enlightened Part 1

Myths De-Confirmed and Gain Structure De-Mystified

The scene is far too familiar. The sound person, we'll call him Tom, has just f inished connecting up the sound system and is now starting to line up the board. Beginning with a normalized board (we wish!), Tom loads his favorite CD and presses play. Using the meter associated with the PFL selection, he adjusts the CD player L&R gain.

He turns it so that the majority of the time the needle is hovering around the -3dB VU mark with only the peaks slamming it all the way through the red zone, past the +3dB VU mark, and all the way to the end stop. Tom then brings the channel faders up to the 0dB mark and routes them to the main L&R outputs (f irst checking that the mains are “ out” of course!).

Tom's Tales Continue

Having already set the amps at full when he was loading them onto the stage, he now brings the main L&R faders up and plays sound through the system. By the time the main output faders reach the -30dB mark the level is already deafening.

But hey, it's his favorite CD! For a moment he allows himself to wallow in the sheer power and glory. Giddy with all the technology at hand Tom is deeply satisfied with the two-channel peak LED's flashing to the beat of the music.

As usual, the moment is short lived, due in part to the sudden realization that there is still much to do before they are ready for the show, and partly because of the angry stares from the rest of the crew.

Tom now turns his attention to the stage mics. He starts out by setting all the mic gains to the 12 o' clock position and PFL's them to make sure they are clean. After getting his stage tech to scratch each mic in turn so that he can identify them on his board, he next proceeds to set the gain for each of them.

He does this by getting his stage tech to either count into each of the vocal mics or play whatever instrument that particular mic is supposed to be picking up.

Starting with the channel fader down, he f irst adjusts the gain until the peak LED flashes, then backs it off slightly. Then he brings up the channel fader having previously routed them to the main outputs (still set at their -30dB mark).

Before Tom ever reaches the 0 dB mark, he is almost immediately confronted by ‘ The Voice of God' followed by screaming feedback. He quickly reaches over and turns down the mic channel gain while simultaneously pushing up the channel fader until he reaches the point where the fader is at the 0dB mark and the house sound is at a comfortable level.

Tom Chases His Tail

Suff ice it to say that the rest of the sound check goes on in a similar fashion. Tom gets through the f irst couple of numbers (or cues, or whatever) and is starting to succumb to the boredom that sometimes follows unexpected achievement. As usual, all the faders have ended up in very different positions than they were at the sound check. He thinks this has made the board look a bit messy so he goes about the subtle process of adjusting each gain so that the associated fader can return to the 0dB mark.

Aside from the distraction of actually mixing the sound, this pursuit occupies him for the rest of the show. If the mix requires the fader to be away from the 0 dB mark for more than one number (cue, whatever) the gain is tweaked to bring the fader back to the 0dB mark.

Not content with the input faders, the fact that the show is running with the output faders at the -25dB mark is equally as dissatisfying. This results in an attempt to creep them up while simultaneousl y turning down all the input gains throughout the course of several numbers!

OK. Let's end that story there. We all get the picture, because, to some extent or another, we have all been there. What should be apparent from the story is that none of the adjustment decisions were as a result of knowing what level the signal was in comparison to the limitations within each element of the sound system.

To help us understand the correct way of lining up a mixing console, or an entire system, we must examine than the individual sound system components.

So, you might ask: “ He got through the show and it sounded halfway decent, didn't it?” Well, the answer to that may be ‘ yes it did,' but is ignorance a good enough basis for a career in sound? And what is he going to do the next time when it doesn't sound as good?

To help us understand the correct way of lining up a mixing console, or an entire system, we must examine than the individual sound system components. We need to go all the way back to basic electronics because it is here where the challenges actually originate.

Our First Enlightenment

Everyone knows that's the last sound system component before the loudspeaker is the amplifier. But many of us forget that there are many other amplifiers in each piece of equipment in that system.

Unlike the power amplifier, which can generate lots of current so as to drive a low-impedance load, the loudspeaker, these other amplifiers only pass-on the signal as small voltages, with very little current. Some of these amplifiers, such as the mic pre-amps, are familiar. Yet conceptuall y we often ignore the other amplifiers, such as EQ cut/boost controls.

Another example of something we don't think of as an amplifier comes in the case of a group of input channels on a

mixing board, all routed to the same output fader. If the routing selection switch simply connected together all those faders then they would all interact with each other.
In some basic circuit designs, each time you move one of these faders you are effectively altering the resistance that all the other faders " see. " This would either increase or decrease the total resistance of all the other faders. The effect would be as if you had actually moved each of those other faders.
So, in practice, many of the amplifiers that contribute to our gain structure remain unaccounted for in our gain structure staging.
This is not good!

Think Unity (Our Second Enlightenment)

To stop this happening all quality consoles use input channel faders followed by a little amplifier with it's gain set at unity, which means it neither increases or decreases the signal. The signal voltage coming out of it is exactly the same as the signal voltage going into it. This has the effect of ' buffering' the signal that has been altered by the fader so that it is not affected by (and in its turn, doesn't affect) anything ' downstream' of it.
Which brings us to our second enlightenment, and a major cause of misunderstanding. All these little amplifiers have their gain set by components on the circuit board. The associated knob or fader does not alter the gain of that amplifier, it only alters the amount of signal being presented to that amplifier.
Even our big power amplifier driving our loudspeakers works this way. It has a f ixed amount of amplification (gain). The knob on the front panel just adjusts the amount of signal being sent to that low-level amplifier section.
In fact all amplifiers, whether big or small, work this way.
Let's say that our amplifier has been internally set to have a gain of 10. This means that it will increase whatever is being fed to it by 10 times. If I feed it a signal of 1 Volt, we will get 10 Volts out. If we use a potentiometer (variable resistor) to reduce that 1 Volt signal to a half Volt (0.5V) then we only get 5 Volts out of the amplifier (0.5V x 10 = 5V). If we adjust the fader or the knob so that we are reducing that original 1 Volt signal to a tenth (0.1V) before passing it to the amplifier. Then we are only getting 1 Volt out of the amplifier.
This actually looks like we are not amplifying it at all, but we are. We are amplifying a reduced signal. If we use the fader to reduce the signal to a twentieth, then we will only get half a volt out of the amplifier.
This appears as if we haven't amplified the signal at all since we started out with 1Volt. It actually looks like we reduced it. However, as with all the other examples, the amplifier is still amplifying it ten times.
We labor this point, because it is very important! It's not the faders or the knobs (potentiometers/v ariable resistors) that distort
in a sound system, it is the little (and big) amplifiers.

Now Comes our Third Enlightenment

No matter what the internal gain is set to, and no matter what the signal level being sent to it is, you cannot get out of an amplifier more than the voltage rails that provide it with its electrical power.
It's sort of like your garden hose. You can turn it on and get any amount of flow through the hose, but you can't get more flow than would come out of the faucet if the garden hose wasn't attached.
In practice you end up getting slightly less due to the hose constricting the water flow. In the same way, amplifiers can't quite give you an output signal voltage that equals their power rails. There is always a cost - sort of like life, really!
When an amplifier is operating within the limits of what it's capable of, any change in the input is mimicked by a corresponding
change in the output. If the level of an input signal goes down to half, the output signal (although a different level) will go down to half its level. Similarly if the input signal increases by a factor, the output signal (although different) will increase by that same factor. The amplifier is said to be behaving in a linear fashion.
However, when the amplifier receives an input signal that attempts to drive it's output above its upper limit, it can't do it. It cannot mimic the input, so it is said to be distorting. The output remains locked at the upper limit for the duration of the time that the signal is trying to be greater than the amplifier's limits until it once again returns to within it's limits. This has the effect of clipping the top off the output signal wave form, and for that reason is commonly called ' clipping' .

Up Against It

As with most things in life, our output signal is caught between ' a rock and a hard place,' or as they say in England ' the devil and the deep blue sea, ' So if the upper limit describes the ' rock, ' what is the ' hard place?' Well, the ' hard place' is noise, and boy is it hard!
Everything, and we mean everything, including us, our bicycle, our Grandmother , and oh yes, even electronic components, are made up of atoms with electrons moving around them. Electrons on the move also happen to be known by another term: Electricity.
This means that the mere fact of being made out of atoms creates low level random electrical signals. Now this doesn't really matter if you are a bicycle, or my Grandmother for that matter, but it does matter if you are an electronic circuit.
Even a potentiometer (what our fader knob is connected to), being an individual component made up of atoms, generates it's own random electrical noise. But being only one passi ve component,
the noise is so low in level that it can virtually be discounted.
However our amplifier, even if it just consists of a single integrated circuit or ' chip, ' is lik ely to be made up of dozens or hundreds or transistors, resistors and capacitors, etc.
The cumulative effect of the low-level random electrical noise in each of these many components suddenl y becomes something significant. Not only that, but remember I told you that the gain of these amplifiers was most lik ely f ixed and not dependent on the fader that precedes it? Well, you guessed it!
That not so insignificant random electrical noise is now being amplified, so now even internal or self-noise becomes significant.

Noise Floor, Our Fourth Enlightenment

Every circuit that our audio signal passes through inside a sound system has a low level of random electrical signal we call a noise floor. In some technologies, such as long distance spacesignal telemetry, this noise floor is so high in comparison to the signal being sent through it that engineers have to employ extraordinary means to keep it at bay.

My father spent his whole working life working for the same British Telecommunications Company. When I was 11 or 12 years old I remember him showing me around one of the ' Ear th Stations' (Satellite tracking stations) that he had built in Hong Kong. When we got to the amplifier room, I remember him telling me that because the satellite transmissions collected by the big dish were so small in level they had to cool the amplifier to close to 0 degrees Kelvin (-273deg C) just to keep its internal noise as low as possible.

Like most things, electrons slow down the colder they get, and therefore create less of noise, resulting in the fact that they had a better chance of amplifying the minute signal and not the noise floor. This story amply demonstrates how diff icult it is to reduce a noise floor. It's actually easier to increase the level before the onset of clipping by increasing the power rails, although this does suffer from it's own set of problems.

Luckily for us in the sound industry, we don't have to go to these extraordinary lengths, as there is still quite a usable range between the noise floor and the level of onset of distortion in most pieces of sound equipment. Which segues nicely to our f ifth enlightenment.

The Fifth Enlightenment, Desperately Seeking Gain Structure

For the best results we must maximize our audio signal within that range and keep it as far above the noise floor as is possible without clipping. Actually, isn't this what lining up a mixing board, or sound system, is supposed to achieve? And you thought it was just lining up all the faders at 0dB mark!

So how should we use our newly acquired enlightenmento line up our sound system? The f irst thing to do is forget that entire 0 dB VU thing. That worked when sound engineers and manufacturers of sound equipment were gentlemen, stopped for tea at 4 o' clock every day, and played cricket on the weekends. Since those days of 0 dB VU equaling +4dBm (+4dBu), we have had hi-f i and semi-professional manufacturers deciding that their VU meters would read 0dB when the signal voltage was at -10dBV. This is a drop of nearly a volt when we are only talking about 1.2 Volts in the f irst place. We have had CD player manufacturers who originally shipped their players with RCA (phono) outputs at 2Volts (whatever that meant).

I have a stack of Denon CD players whose outputs are too hot for line input of my Soundcraft even with cranked all the way down. We have computer sound card manufacturers who wouldn't know a linear IC if it hit them on the head. And what is with that 1/8in (3.5mm) connector? Is that a weak link in a chain or what?

OK, OK. I' m taking the medication! But this really is a gross case of faulty thinking. And so to our enlightenment #6.

Lining Structure

We do not line up signals, we line up systems! Yes, we line up s-y-s-t-e-m-s! We line up all those little amplifiers so that their usable range between the onset of distortion and their noise floor coincide with each other.

It's sort of like lining up windows. That way there is a clear passage for the audio to get through. Now some of those windows are taller than others, and some are shorter. So we have to decide if we are going to line them up with their middles, or bottoms are tops in a row, ie: with the noise floor all at the same level or with their onset of clipping all at the same level.

Recording may optimize noise floor. Sound reinforcement may optimize headroom (clipping). My vote is for onset of clipping, as it is easier to hear and measure, than noise floor, especiall y during a noisy loadin.

Now, for sanity's sake, don't go cranking up your sound system and driving it into clipping as you will destroy all the HF drivers. Clipping produces harmonics across the sound spectrum at levels your HF drivers wouldn't normally have to handle.

So just hang in there. In the second part of this article, (LIVE SOUND! International Oct/Nov 1999), we will reveal how you can build and use a snappy hand held device called a ClipCop. For around $50 U.S. in components and a rainy weekend armed with a soldering iron and a drill, you can have a hand held " line up" device that you cannot buy anywhere else.

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Armed with this and the ten enlightenments (four more to follow), the world of sound is your Toaster! (Oops sorry. Oyster). I had my second year undergraduate students each build one earlier this year, and if they can do it, so can you. So remember , order your next copy of LSI now - don't miss out on the DIY gizmo to end all DIY gizmos, the ClipCop!

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CORRECT THOSE INCORRECTIONS

One of the major reasons for incorrectly lined up mixing boards is inadvertently using the wrong control to do the wrong thing. This becomes even more confused because a lot of the times the same controls are used to not only accomplish engineering requirements, but are also used for artistic reasons.
You may hate me for this, but here is a selected list of some of them.

• Incorr ect use #1: The channel input gain (trim or preamp) control, whether mic or line input, is used to bring the signal into the mixing board at an optimum voltage level so as to be as far above the noise floor as possible without distorting.
There is no other reason for this control, and there is no appropriate artistic use for this control. Adjusting it at the expense of your signal to noise ratio so that your fader can look pretty at the 0 dB mark is not the mark of a good engineer .
If your channel fader is way off the 0dB mark, look for the problem elsewhere. And using it to get a desirable level in the house, as with our story, is way off the mark. A good indication of your input gains being incorrectly set is if you are not getting enough signal out of you Aux channels.

• Incorr ect use #2:You have no idea at what level the manufacturer has set for the input channel peak LED to come on. There is a modern trend for manufacturers to set it low so that sound engineers can get a kick out of being bad boys and girls and making it light up without their equipment sounding distorted. Grow up! Use the LED bargraph or meter associated with PFL. It will hopefully give you a better idea, and if you know the specs of your board, then you can do the calculation in your head.

• Incorr ect use #3: Channel faders are used for artistically mixing the program material from each input channel. Like cooking recipes, not all ingredients in a dish are used in the same quantities. Get used to it. This is life! If a couple of your faders are so low that they are becoming hard to subtly control without wild shifts in level, then assign them to a sub group and use it's increased range of control or vice versa.

• Incorr ect use #4: (Nothing to do with line-up but a pet peeve none-the-less) Faders are designed so that they can be more continuously and subtly controlled than knobs. If your f ingers aren't on them then you are not a mix engineer , you are just a level setter, sort of like people who set the levels for PA systems in train stations and airports.

• Incorr ect use #5: Main outputs (and matrix outputs if more than plain vanilla L&R) are for fading up, fading down, and fading out the level in the house. They should be used in a region that allows them a lot of travel so as to be able to accomplish these fades smoothly. They can be used to push the overall level for a musical button etc. but they should not be used for setting the overall level in a house (see #6, below)

• Incorr ect use #6: The knob on the front of the amplifier, crossover, compressor or EQ (read: snake return-line driver for the last two), is used for setting how much the signal from the mixing console (via the outboard equipment) is reduced so that the maximum signal from the board does not push the amp beyond what it is capable of. If this makes it too loud in the house then you have made the promoter pay for too big a sound system.
If it is not loud enough then you allowed him to pay for too small a system. As a professional sound engineer you should be able to gauge this, or calculate it from specs. If you are not in control of the size of your amp then you use this knob to set the overall level of sound in the house.

• Incorr ect use #7: Incorrect overzealous application of any of these rules without proper consideration of all the pros and cons is the sign of an unthinking engineer , and certainly not the sign of a true professional, let alone " artist!"