

ADVANCED SYSTEM GAIN STRUCTURE

Note: This does not cover how to set the gain in a mixing console but is concerned with overall system gain structure.

INTRODUCTION

BACKGROUND YOU NEED TO KNOW

Realistically, audio signals at or near the noise floor of a system are not useful because the signal will not be significantly louder than the noise. Therefore, some minimum usable level must be assumed below which the electronic noise is considered objectionable. A signal to noise ratio of 20 dB is considered minimally acceptable for good intelligibility. For a high quality system 30 dB would be a better figure to use. Using this value, the range from this minimum signal level (30 dB above the noise floor) to the clipping level is the usable signal range window for the system (also called the dynamic range in my way of thinking). However, for purposes of this paper, the maximum output to noise floor is used as the dynamic range.

Every audio system with more than one electronic component has a "system gain structure". Gain structuring for a system occurs in the signal processing chain between the mixer or another signal source and the power amplifiers. One usual scenario is to set all the signal processors to unity gain and turn the amplifier inputs to maximum. Unfortunately as you will see, given the different maximum outputs and noise levels of typical signal processors, this method will may not come close to the best gain structure.

We will be dealing with the signal voltage levels on the interconnecting cables from the output of the mixer (or signal source if there is no mixer) up to the input of the amplifier. For the convenience of using simple numbers, this analysis uses relative dB, as a voltage ratio where $\text{dB} = 20 \times \log (V1/V2)$, and dBu , where $0 \text{ dBu} = 0.775\text{V}$. $V1$ and $V2$ are simply two voltages.

To set proper gain structure, the interconnections between devices must be constant voltage interfaces. This means an output device's voltage at any point in time is unaffected by whether or not it is connected to the device(s) it is driving. This type of interface is characterized by the output impedance of a device being 1/10 or less of its load. For example, if the output impedance is 100 Ohms, the total load it drives must be 1000 Ohms or greater. Virtually all professional audio equipment meets this criterion when a single device drives only one other device. However when one device drives multiple devices, such as a mixer feeding a number of power amplifiers, this may not be true. In this case a distribution amplifier may be needed to divide the load between its multiple outputs.

The last thing to consider is the power handling of the loudspeaker(s). As long as the amplifier does not exceed the loudspeaker's power handling capability and the system is operated without clipping, you should never blow a properly manufactured loudspeaker. The safest criteria to use

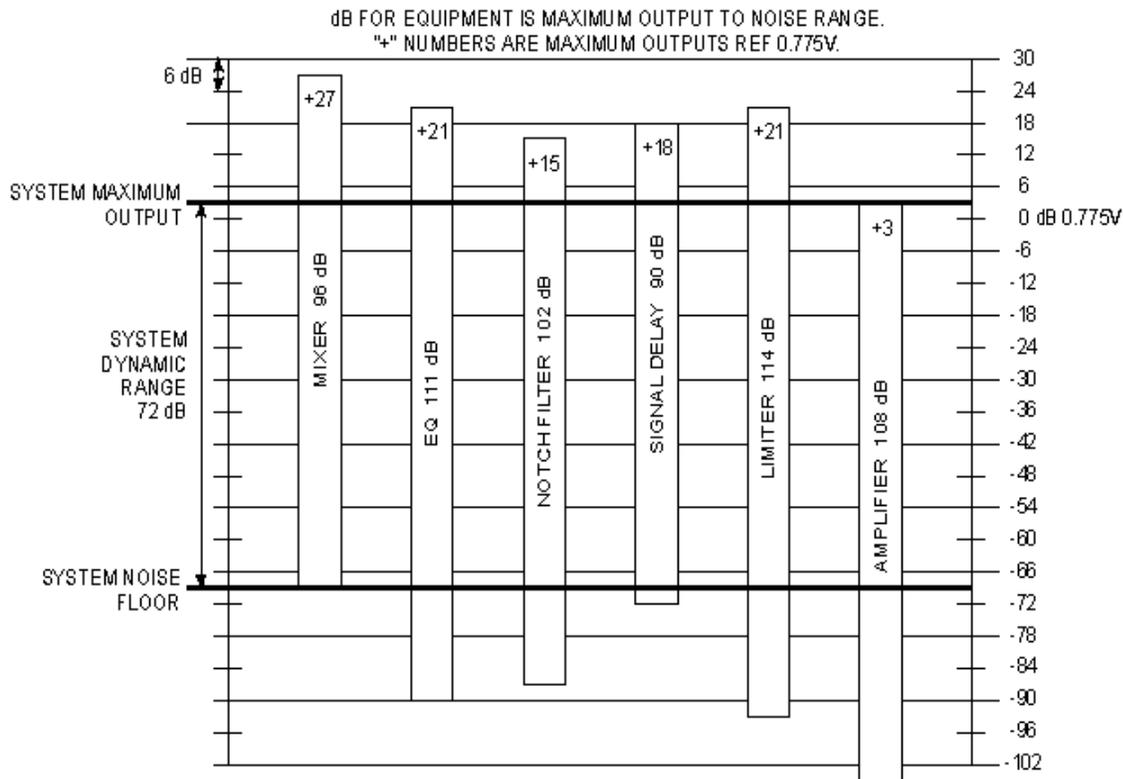
in selecting an amplifier is the RMS rating of the loudspeaker. In reality, most loudspeakers can handle peak signals in excess of this rating. A reasonable choice is an amplifier whose rating that is 2 times (+3 dB) the RMS rating of the loudspeaker. The RMS sine wave used to rate amplifiers has an inherent peak power component of 3 dB. So this all works out to a 6 dB allowance for power peaks over the loudspeaker's RMS rating. This is a pretty safe figure for the way most professional loudspeakers are rated (pink noise with a 6 dB peak factor) and given the peak to RMS content of most audio signals. However, sustained sine wave signals from the likes of a synthesizer could exceed the loudspeakers RMS capability by 3 dB without clipping the system. If you expect these kinds of signals and you expect to drive the system to maximum output levels with them, use the loudspeaker's RMS rating as the power rating for the amplifier.

With these basics in mind, we're ready to examine how to achieve proper gain structure in detail.

PART I - PICTURING GAIN STRUCTURE

Before you get out your equipment and start setting gain structure you have to learn just what it is you are trying to accomplish. Go through the following "on paper" analysis of a typical system. After you understand this can you appreciate where to actually set the controls on equipment to achieve optimum gain structure.

FIGURE 1 shows a simple system consisting of six pieces of equipment. The device clip level (maximum output) is listed for each device as published by the manufacturer. For this example, all devices between the mixer output and the amplifier input are set for unity gain and the amplifier input is set for maximum sensitivity.



Each device is represented by what looks like a bar. Rather than a bar, picture it as a tall, narrow window. The maximum output or clipping point from the specifications for each device defines the top of the window using the absolute dBu scale on the right. The published noise floor (or signal-to-noise ratio) specification below maximum output determines the height of the window. The relative dB scale on the left is used to determine this height. All usable signals must pass between the top and bottom of the window. However, remember that your low level signals won't be near the noise floor. Realistically the minimum usable signal is one that is at least 30 dB above the noise floor.

Next, a horizontal line is drawn across the top of the lowest window (in this case the amplifier). This is the system clip level, and for the rest of the analysis this line stays in the same place. Another line is drawn across the highest bottom window sill (in this case the mixer). The relative dB scale is used to measure the distance in dB between the 1st and 2nd lines. As you can see, it is only 72 dB for this set of devices and gain structure. That's equal to the performance of your average consumer cassette deck -- and you thought that professional equipment automatically guaranteed a professional grade audio system. Oh well, live and learn!

Now subtract 30 dB to find the "true" dynamic range (30 dB above the noise floor to the clipping level). The result is 42 dB. Measurements of the maximum dynamic ranges for acoustic instruments and voice yield maximum figures in excess of 40 dB. This means our system really doesn't have enough dynamic range to reproduce them.

PART 2 - THE MOST COMMON APPROACH TO GAIN STRUCTURE.

As seen in FIGURE 1 from the absolute scale on the right, the amplifier input sensitivity limits the maximum signal level in all the other devices to +3 dB. Above +3 dB the amplifier will clip - period. It doesn't matter how much "headroom" is in the mixer, you can't use it without distorting the amplifier. Well, you say, the obvious step is to put a pad (usually the amplifier input attenuator) so the amplifier will clip at about the same point as the next least capable device. In this case it is the notch filter. Using a -12 dB pad, the notch filter and amplifier will both clip at once and the signal level will be 12 dB higher through the other devices at the amplifier's maximum output.

The chart, as seen in FIGURE 2, was changed from FIGURE 1 by moving all the device windows (except the amplifier) down by 12 dB using the relative dB scale on the left. A +15 dB signal (the notch filter clip level) is now attenuated to +3 dB by the amplifier's input attenuator.

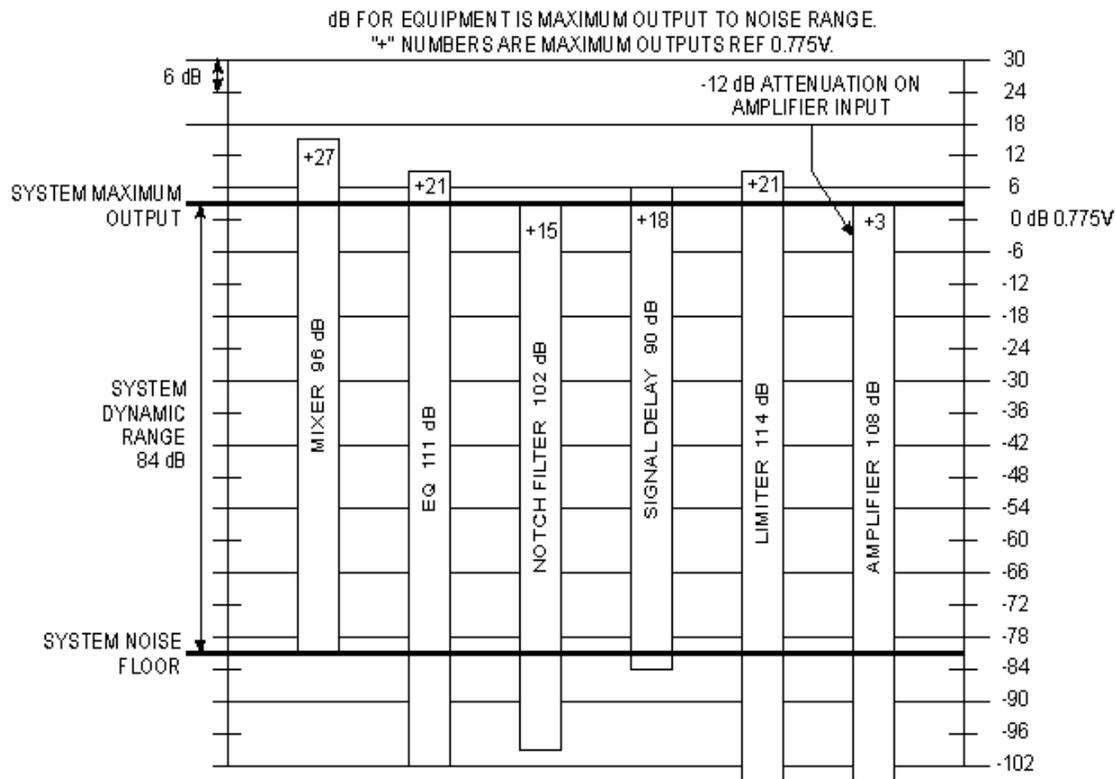


FIGURE 2

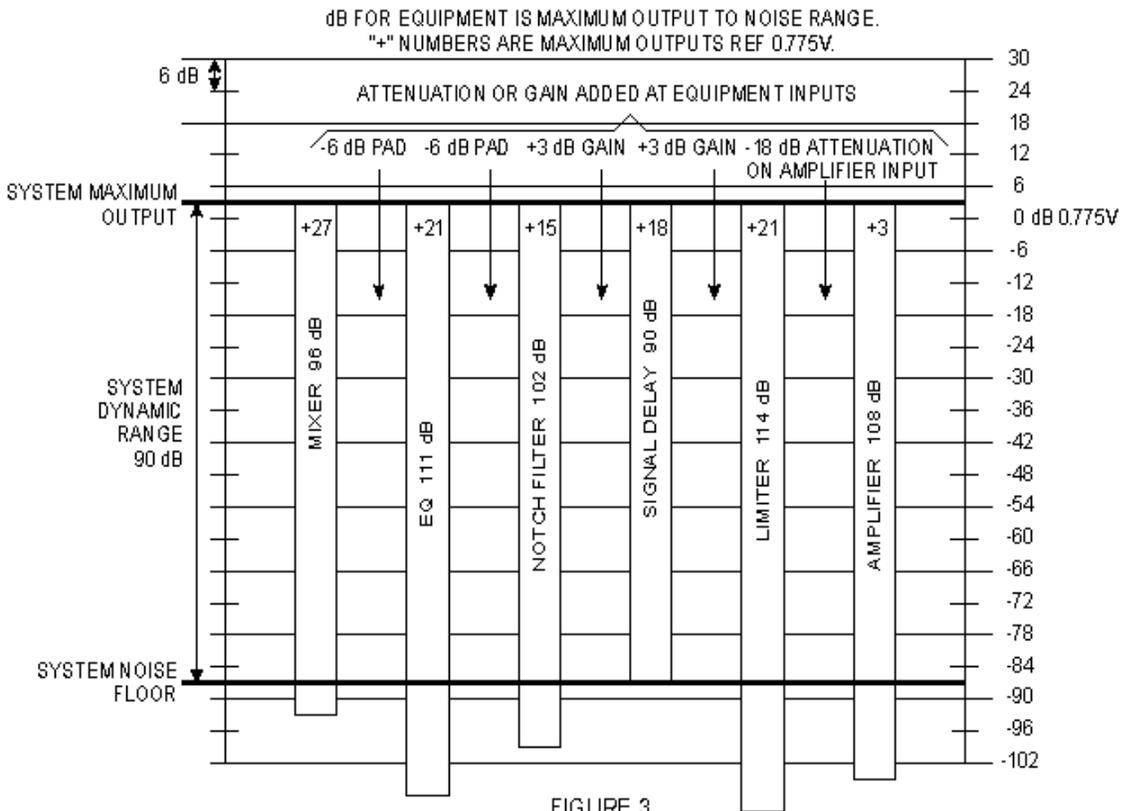
The noise floor line is redrawn through the highest window sill (in this case still the mixer). Because this ends up 12 dB lower than in FIGURE 2 relative to the system clip level, we see that the system's overall window height is now 84 dB. This is a 12 dB improvement - much better. Note that the absolute device clip levels no longer relate to the absolute dB scale except for the amplifier's input after its input attenuator. Our usable signal range (from 30 dB above the noise floor) is 54 dB. This means our system is now able to squeak out enough range to reproduce the dynamic range of instrumental and vocal sources.

Unfortunately, the mixer is still the primary noise source by 3 dB over the signal delay. However, according to their published specifications, the mixer should have some 6 dB better noise performance than the signal delay. It should also be obvious the signal delay is the weakest dynamic range link because it has the shortest window. Therefore, we must conclude that there is more that can be done to optimize the system's gain structure.

PART 3 - YOU CAN MAKE IT BETTER

To optimize the system, pads or gain must be added at the input of each device so that its clipping level and the clipping level of the preceding device occur at the same point. Think of the following procedure as a graphic picture of what would happen to the signal on a volt meter as you work your way through the system.

To create the chart shown in FIGURE 3, the windows are shifted up and down as needed so that all the tops are lined up on the system clip level line. To do this, start with FIGURE 1 and work from left to right in signal flow fashion. If you move a window up you need gain between it and the next device. If you move a window down you need a pad.



First, move the mixer window down so its top is even with the graphic EQ window. This movement is measured on the relative dB scale, which in this case is -6 dB. Therefore, you need a 6 dB pad at the input of the graphic EQ.

Next, move BOTH the mixer and graphic EQ windows down together so the graphic EQ window is even with the top of the notch filter window. This also turns out to be -6 dB. Therefore, a 6 dB pad is needed between the EQ and notch filter.

Now, move the mixer, graphic EQ and notch filter windows together so the top of the notch filter window is even with the top of the signal delay window. To do this, move ALL the previous devices up 3 dB. This means you need 3 dB of gain between the notch filter and signal delay.

Repeat the process again by moving the windows of the first four devices together so the top of the signal delay window is even with the limiter window. This distance equals 3 dB. This means 3 dB of gain is needed between the signal delay and limiter.

Lastly, you must lower ALL device windows to line up with the input to the amplifier. They are moved the distance between the top of the limiter and the top of the amplifier. In this example the distance is 18 dB. (In the actual system this would usually be done with the amplifier input attenuator.)

After completing all these steps, the tops of the windows end up on the system's clip level line as shown in FIGURE 3 (+3 dB on the absolute dB scale). Looking back from inside the amplifier after its input attenuator, all devices appear as though they are clipping at +3 dB. In reality, they are all clipping at their specified device clip levels. If one device is clipping -- everything is clipping.

The Results Are Worth It

Measure the distance between the system's clip level line and the bottom of the shortest window using the left-hand scale. This result is 90 dB which is 18 dB better than the raw system gain structure in FIGURE 1. The "true" dynamic range, considering a 30 dB above the noise floor signal as the minimum, is now 60 dB. Also, the primary noise source is now the signal delay. This is the weak link, which agrees exactly with the published specifications for all the devices. It also should be apparent that more of the usable signal window within each device is being used. What a concept.

In some cases, pads or gain will not have any effect on the overall usable signal range. For example, in FIGURE 3, the 3 dB of gain at the output of the signal delay could be omitted. The 18 dB pad for the amplifier input would become only 15 dB, and the top of the limiter window would end up 3 dB above the system clip level. The key here is that the bottom of its window is still well below the noisiest device (in this case the signal delay). You can use this reasoning to save yourself the hassle of making up small pads or small amounts of gain. If you omit one of these along the chain you MUST move all the devices preceding it up or down in your chart by the dB of gain or loss that you are omitting. Otherwise, you will not see the effects of the omission on the noise floor.

Background Noise

Now that you have set up proper system gain structure on paper, it is time to hook-up the system and do the same thing for real. Once completed, audibly evaluate the noise floor heard from the speakers. If all is quiet, pack up and go home. If the noise floor is too high, there are two possibilities:

A. The maximum sound level is higher than necessary, which means you over-designed the maximum capability for the system. If this is the case, turn down the amplifier input attenuator. You will lower the noise, and the maximum output level for the system will be reduced by the amount you decide is over-kill.

B. The maximum sound level you can get out of the system IS necessary, which means your system does not have enough usable signal range. You now have three choices; the first two are compromises.

1. Accept the noise and achieve the maximum sound level you need.

2. Turn the amplifier input down to make the noise acceptable. This will, of course, reduce the maximum output level capability for the system. (Sorry, you can't have it both ways unless you pick choice 3.)

3. Change the primary noise source in the system to something with lower noise performance.

Doing Your Own Analysis

A similar chart for setting up proper system gain can be created for any system. Using graph paper, make a vertical absolute dB scale from about +30 dB to -120 dB so you can plot increments for 3 dB or less. The relative dB scale simply uses the same graph increments for plotting and measuring distances in dB.

You could also follow this procedure by using some simple math. If you don't trust your addition and subtraction, or would rather work with pictures (they are more dramatic and will quickly show errors in your thinking) cut out rectangular paper bars (windows) like those shown in the figures. The length of each should equal the distance in dB between the device clip level and its noise floor. Be sure to convert noise figures to noise below maximum output.

Write in the clip level for each device on its window. Using these numbers and the absolute dB scale, position the top of each window on the graph paper in signal flow order from left to right. Move these "paper cut-outs" up and down on the chart as outlined above, by measuring the distances using the relative dB scale. You can very quickly determine the necessary pads and gains -- probably faster than with a calculator.

A way to check your work is take the maximum output for the first device and subtract the dB for the all the pads and the gain to that number, including the pad before the amplifier. The result should equal the maximum input sensitivity for the amplifier. This calculation should give math mavens an interesting insight into the gain structure process.

PART 4 - DOING IT FOR REAL

To actually adjust a system you need to do exactly what you did on paper except you are now doing it for real. You start from the console output and find out what you need (gain or loss) to adjust its maximum output signal so that it just drives the next device into clipping. And so on.

You don't need to know the specifications of the equipment. When you go through the system you'll find out what those specifications are in terms of maximum output levels. As you should

have understood by going through the exercise on paper, the noise floors of the equipment will take care of themselves. Some device (like the signal delay in the above example) will be the weak link. There is nothing you can do to make this better except to replace it with a device with a better maximum output to noise floor window (better signal to noise ratio specification).

Because of production variations and possibly conservative specifications, you may be able to pick up a few more dB of dynamic range by adjusting the pads or gain values you determined on paper. If things are not reasonably close to your on-paper calculations, you have a problem such as bad wiring or a mis-adjusted or defective device.

What To Adjust

When you set gain in the system, the attenuation or gain needed between devices can be added externally or by using a device's input level control, if it has one. **DO NOT ADJUST ANY DEVICE'S OUTPUT LEVEL CONTROL** - this should be left at maximum. This is because it is rarely the last thing in the internal circuitry before the output connector. Unlike some input level controls, it usually does NOT adjust actual gain. Therefore using it will squash the dynamic range in that device's output stage and you may end up making things worse, even though the signal level is matched up to the next device. Use an output control only if you **KNOW ABSOLUTELY** that it is a simple attenuator feeding its output connector. The reason it usually is not is that this topology would cause changes in the output impedance when the control is set for anything other than maximum. Among other things this could wreck havoc with - guess what - the gain structure. If the device has a noise floor below other devices when you have set the overall gain structure, you can use output gain. But reduce it only by 3 dB less than the amount between the device's noise floor and the device that determines the noise floor of the system. This is because if you bring its gain, and hence its noise floor, up to the worst case device its noise will add to the worst case device and give you 3 dB less dynamic range.

The Tools You Need

To find the clip points in a system, you need to use an oscilloscope and a pink noise test signal. There is really no good substitute for this equipment to set gain structure. Sine wave signals are not recommended as they only show one frequency at a time and you can easily miss something. The pink noise should be full-range (20 Hz - 20 kHz) and have at least a 6 dB peak to average ratio. If you can find one with a 10 dB peak to average ratio, you will more closely simulate real audio signals.

If you must use sine wave signals, you will have to check each and every EQ boost frequency or range of frequencies very carefully. When measuring electronic crossovers or other frequency response limiting devices, only a full-range pink noise signal will allow you to see full-range signal energy losses easily. (See sections on crossovers and band limited devices.) If using sine waves you must set the frequency to the center point of each frequency band of the crossover or the center of the band pass for a band limited device.

For simple systems (e.g. no electronic crossover), there is "poor man's" method where you use a Piezoelectric tweeter and a 400 Hz sine wave to find clip levels. Basically, you connect the tweeter directly to the output of each device. When the device hits clipping, the tweeter will emit a very noticeable buzzing sound due to the harmonics in the clipped signal. For high-powered

amplifiers, a resistive pad should be used to avoid burning out the tweeter. This method is detailed more rigorously by Pat Brown of Syn-Aud-Con. You can find this information as a PDF file on the internet at <http://www.synaudcon.com/downloads/piezo.pdf>.

Doing It

You start the whole procedure by inputting the pink noise test signal into to mixing console. Set it so that it's output just clips as seen on the oscilloscope. Make sure it is the output of the mixing console that is clipping. Determine this by reducing the master fader. The clipping should stop. If it doesn't, you are clipping something before the output fader. While you're at this point, note the reading on the output meter. This is a good indication of what the meter will read when you have reached the system's maximum output after you set its gain structure. If you are using sine waves this will NOT be a reliable indication.

Once completed, if the system noise levels are low enough, you may want to increase the setting of the amplifier(s) input level control. This will make the mixer more sensitive for operation. If you reduce the amplifier input level control, something in the front end of the system will clip first. This means the amplifier will not reach full output. But it WILL reproduce that clipped signal and possibly damage the loudspeakers. Either way -- if you choose to increase or reduce the amplifier's input sensitivity from the optimum gain structure setting -- you really don't gain (pun intended) anything.

There is possible exception to this: by reducing the amplifier's input level control, the output meters on the console will indicate you have reached the system's maximum output before the amplifier's clip. This is useful so that a less than capable mixing engineer will THINK he's pushing things to the limit but there will still be something left in the amplifiers. This may help protect the loudspeakers but, bear in mind, it will limit the maximum output of the system to something less than it could be.

Note that to reach a system's maximum output analog Vu meters on mixing consoles may "peg" before the system clips. If you can afford the reduction in dynamic range, operating the system so the meters don't peg means you'll never clip the system. Generally, this means you won't ever blow the loudspeakers assuming the amplifiers are chosen not to exceed the loudspeaker's maximum ratings.

PART 5 - MORE COMPLEX SITUATIONS

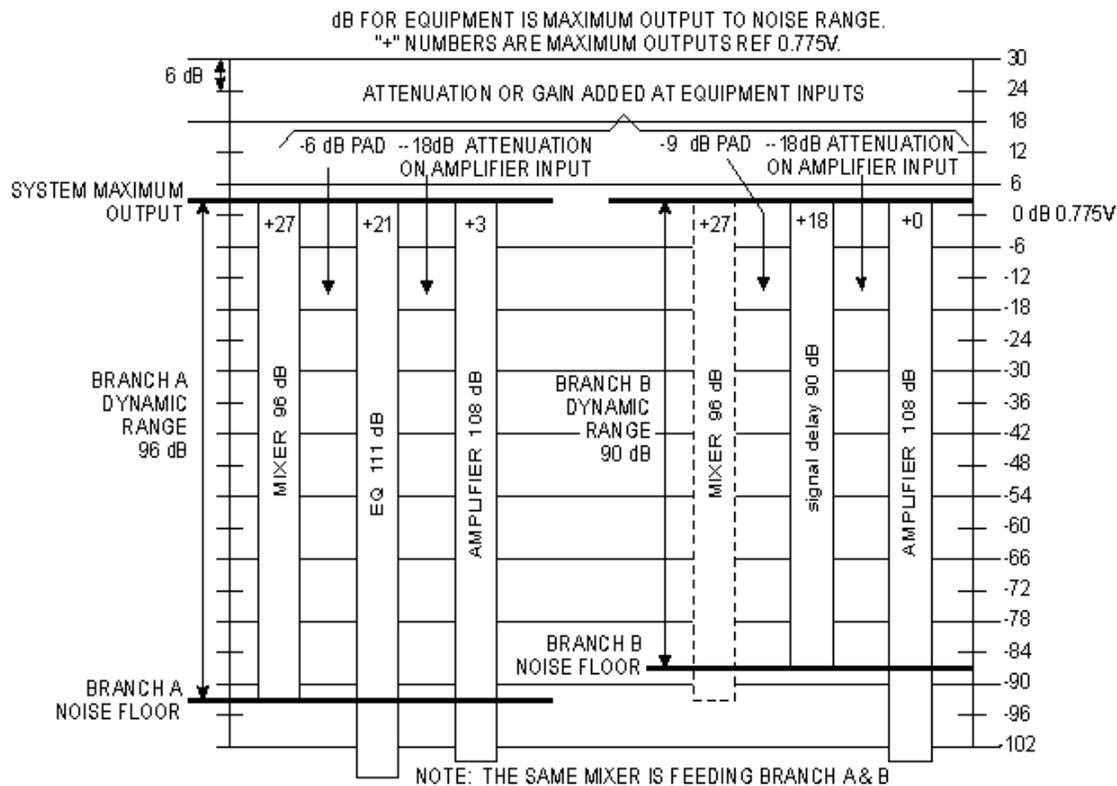
Up to now we've looked at a simple systems. Here is where gain structure gets more complicated. However the ideas are exactly the same. You just have to think about what specific pieces of equipment do and/or about more signal paths.

Devices with Gain/Loss and EQs:

Parts 1 - 3 assumed devices in the signal chain have no gain (unity gain devices). However, a device may have gain or loss, or you may want to allow for boosts in an EQ, which may be needed to tune the system. EQ boosts are like adding overall gain to the device. In such cases, as illustrated in FIGURE 4, input of the device's window is shifted down below the output of the

Multiple Signal Paths, Arrays and Delays:

Another variation in this procedure is when a system has several branches, such as a mixer feeding multiple sub-systems. You have to separately analyze each branch and include in each analysis the source common to all branches (the mixer in the example FIGURE 5). This will automatically optimize the system so that the common source and all the branches clip at once. To do this, the mixer in FIGURE 5 must feed each branch through a separate pad. Note that the dynamic range is different in each branch.



To balance the multiple branch systems acoustically in the actual system, you will probably need different operating levels in the branches than what the optimized electronic gain structure provides. An example would be a central cluster with delayed balcony speakers. To balance operating levels in these instances, use the branch that is lowest in acoustic level as your reference branch (i.e. the one you are itching to turn up because it isn't loud enough - but don't touch that dial). Use the input attenuation on the amplifiers for each of the OTHER branches. This will reduce their output levels and achieve proper acoustic balance with the reference branch. This will also have the effect of lowering the noise levels and reducing the maximum capability of the other branches. In this case, less capability is acceptable because you have determined that the maximum capability can't be used in these branches unless you drive the reference branch into clipping.

However, if you find, for example, that you have to significantly reduce the maximum output capability of the central cluster so you won't clip the balcony system, then your balcony system

is under-powered. Instead of attenuating the central cluster, you could add gain prior to the balcony system amplifiers (or "un-attenuate" the amplifier input). While this will balance the system, the balcony amplifiers will be driven into clipping before the central cluster amplifiers.

In this situation, the only way you can have your cake and eat it too, is to increase the size of the balcony amplifier, which translates to more voltage (power) capability for the balcony speakers. You will not spot this problem by analyzing the electronic gain structure. This could only have been spotted on paper with proper analysis of the acoustic output for each branch based on speaker sensitivities and listening distances.

Electronic Crossovers:

Electronic crossovers require special attention. Consider a full-range signal with equal energy per octave (e.g. pink noise). A crossover will divide the total energy of such a signal among two or more frequency bands. This causes an inherent signal loss at each band-limited output, compared to the full-range crossover input signal. In effect, crossovers are NOT unity gain devices when fed a full-range signal. You can approximate these losses by calculating how much of the total energy is in each frequency band by using the following procedure:

Example: A 3-way crossover with frequency bands of 50 Hz - 125 Hz, 125 Hz - 500 Hz, 500 Hz - 10 kHz.

1. Multiply the lowest frequency in each band by 2 until you get to the highest frequency for that band. The number of times you multiplied = the number of octaves. Round off the results for each band to the nearest whole octave [= 1, 2, 4].
2. Add up the total octaves from all bands [= 7].
3. Divide the octaves in each band by the total octaves [= 0.14, 0.29, 0.57]
4. Push the LOG key for each result [= -0.9, -0.6, -0.2].
5. Multiply each result by 10 to find the approximate losses [= -9 dB, -6 dB, -2 dB].

Note the low frequency output is down almost 10 dB. That is why many systems have problems achieving enough drive levels for the subwoofers.

Now you must draw horizontal lines on the output side of the crossover's window. Draw these lines at a distance below the top of the window equal to the loss in dB for each output as found above. This line for each crossover output is used to match up the crossover window to the top of the window of the device it feeds (usually an amplifier). In the example, a different pad would be needed for each output (assuming the amplifiers have equal input sensitivities). The top of the window of the device feeding the crossover is still matched to the top of the crossover's window.

In the actual system, the amplifier input levels are adjusted to acoustically balance the system similarly to a multiple branch system. Use the frequency band that you want to turn up the most - typically the subwoofer (but of course you won't turn it up - right?) as the reference output. Balance the other bands to it by turning DOWN their amplifier input level controls.

Once you have the system balanced to your acoustical liking, you may find that amplifier input level controls, in particular for horn amplifiers, may be set too low for them to reach full output - even with a single frequency sine wave in their pass band. You can increase all the amplifier input level controls by the same amount to get some or all of this unusable capability back for

limited frequency range signals. Keep in mind, however, that this will have two consequences: It will raise the acoustic noise floor of the system and the capability for full-range signals will remain the same. However, some amplifiers will clip before the signal processing in the system.

This is another situation where you must accept a compromise or change amplifier sizes to get a better match in gain and capability between the different frequency bands.

Other Band-Limited Devices:

There is a more general case, similar to the crossover scenario. If you have full-range signals at the input of a device that limits the frequency response -- such as with high or low pass filters -- there will be an energy loss from its input to output. Calculate this loss using the same procedure outlined in the previous section on electronic crossovers. The significant energy of full range music signals effectively spans about 9 octaves (approximately 30 Hz to 15 kHz).

Example: An under balcony system band limited from 150 Hz to 5 kHz.

1. Multiply the lowest frequency limit of the device by 2 until you get to the highest frequency limit for the device. The number of times you multiplied = the number of octaves. Round off the results to the nearest whole octave [= 5].
2. Divide the number of octaves by the 9 full-range octaves [= 0.56].
3. Push the LOG key for this result [= -0.3].
4. Multiply this result by 10 to find the approximate loss [= -3 dB].

Now you must draw a horizontal line on the output side of the device's window. The line is drawn at a distance below the top of the window equal to the loss in dB as found in #4 above. This line is used to match up the device's window to the top of the window of the following device.

PART 6 - SYSTEM LIMITING

The purpose of a system limiter in a properly gain structured system is to prevent any signals from exceeding the system's maximum level. As such, it is used as an "emergency" device meaning it is intended to provide a hard, never-to-exceed maximum output level.

Limiter/compressors with soft-knee thresholds are not as ideal for protection. You really want something that doesn't do anything up to a certain point then stops any further increase cold in its tracks. Because you need some margin between the device's maximum input and its limiting threshold they are a bit tricky to implement properly without compromising the system's dynamic range.

Just as with any other device you must introduce the limiter using its input, output, noise floor specifications, and gain setting just as with any other device in the system. Because of the way they work, the threshold setting is used as the maximum input. For proper functioning the threshold should be set to at least 3 dB lower than the maximum output signal from the device preceding it. The limiter's output gain should be used to adjust its maximum output at threshold to about 2 dB below the input level of the device it feeds. This allows a little "margin for error" in the protection.

If you think about it, the only point you can put a limiter in a properly gain structured system that will truly work perfectly is at the output of the signal source. It would be set so that the input level to system would never allow the first device it feeds to clip. This is technically practical if only one signal source is used at a time (i.e. not mixed with others). Therefore, if you are simply switching between multiple input sources, put your limiting device at the output of the switcher. The system sees one input source and really doesn't care which one it is and no input signal can drive the system into clipping.

With multiple mixed sources and a properly set system gain structure, the next best place to put the device is at the output of the mixer. This is because any mixer output voltage at any frequency exceeding the system's maximum output will clip the system somewhere. With multiple branch systems you might think to use a limiter on each branch. But with proper gain structure they would either all work at once or compress some part(s) of the system and not others, thus upsetting the acoustic balance. Thus a single limiter for each main output that is controlled by the operator makes the most sense. "Controlled by the operator" means outputs such as a separate sub-woofer output where the acoustic balance is actively "mixed" by the operator based on the input signal content. You should make sure the operator can see when a threshold is exceeded to avoid clipping the mixer.

If the noise floor of the mixer is low enough compared to the other devices in the system you can allow more than the 3 dB margin the mixer has above the limiter's threshold. You can do this by reducing a pad between the mixer and the limiter. You can also lower the limiter's threshold level (and increase the output of the limiter by the same amount) if the noise floor of the limiter allows this.

Because it is used as a "hard-line" device, you should set the compression ratio to maximum (10:1 or higher if available). As to any attack and release settings, they do not affect the gain structure. However, as the limiter is intended to function only as an emergency protection device, there is every reason to use the fastest attack and release times. You are not going for sound quality here; you are protecting the system from any overdrive. So get into and out of protection as fast as possible. If you think sound quality IS important, then you are not thinking correctly. What you should have thought about is a more powerful system that would rarely be pushed into limiting. In other words if the system is constantly pushed into limiting, it is under-designed.

SUMMARY

Gain structure is only a problem because invariably we use equipment with different input/output capabilities and noise floors. There is no easy way to properly set system gain except to analyze each device in relation to the device that feeds it and in relation to the entire signal path. Fortunately, the little information you need is readily available on equipment specification sheets. By working out proper gain structure on paper before you purchase and wire up the equipment, you can spot potential problems and make appropriate substitutions. In any case, with the gain properly structured, you can make significant, or in some cases, spectacular improvements in the system's dynamic range and its noise floor. In the example system, 18 dB is certainly a spectacular improvement.

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Sep 99
