Sound System Gain Structure

Considerations and a Suggested Methodology for Setting Sound Reinforcement System Gain Structure

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TABLE OF CONTENTS

I. INTRODUCTION

Good gain structure is a necessity, especially if the system employs today's DSP devices in a live sound reinforcement signal path. These devices don't clip nicely, and avoiding that clipping while maintaining a low noise floor requires an accurately balanced gain structure.

The term Signal-to-Noise Ratio (S/N) is the difference in dB from a nominal signal level, usually +4 dBu or 0 dB VU, to the measured thermal noise floor of the device or system. Headroom is the difference in dB from the nominal signal level to the onset of clipping, which is the maximum signal amplitude limited by the output voltage of the device. The Signal-to-Noise Ratio plus headroom equals the dynamic range of the device or system. Proper gain structure is the process of maximizing a system's dynamic range.

Figure 1. illustrates a device with a dynamic range "window" of 105 dB. This dynamic range is arbitrarily calibrated to an electrical signal range of hearing and an acoustical signal range of hearing.

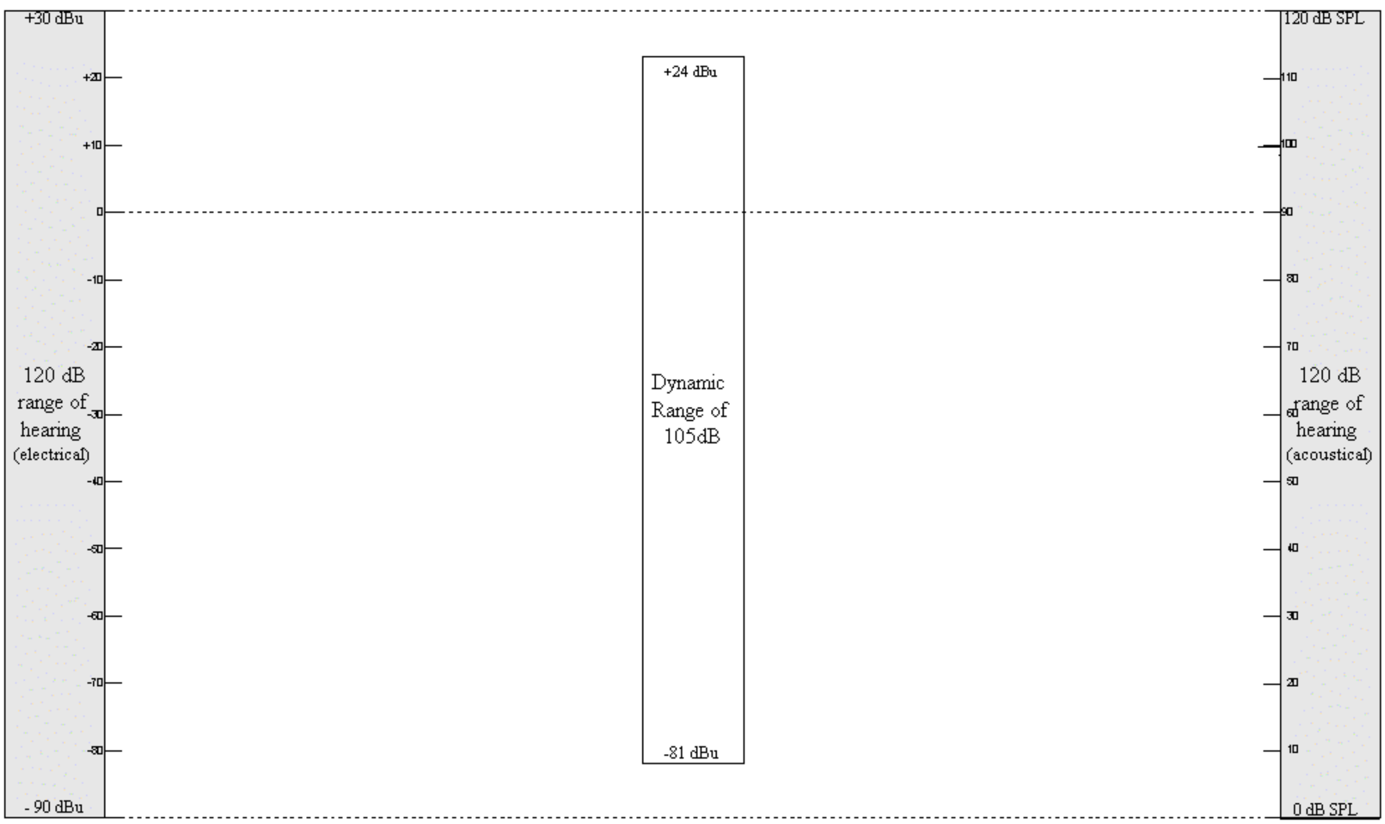

Figure 1.

Figure 2. is a sound reinforcement signal chain of devices' dynamic range windows. Note the electrical reference level of 0 dBu and that the input and output of both the mixing console and the DSP signal processor are set for "unity gain," meaning the input level and the output level are equal. The acoustical signal range is calibrated so that the amplifier, with its input attenuator set full up, clips at a speaker output level of 120 dB SPL at some reference listening position. This particular gain structure is a very popular one. The electronics are set to "unity" and the amplifier is set full up.

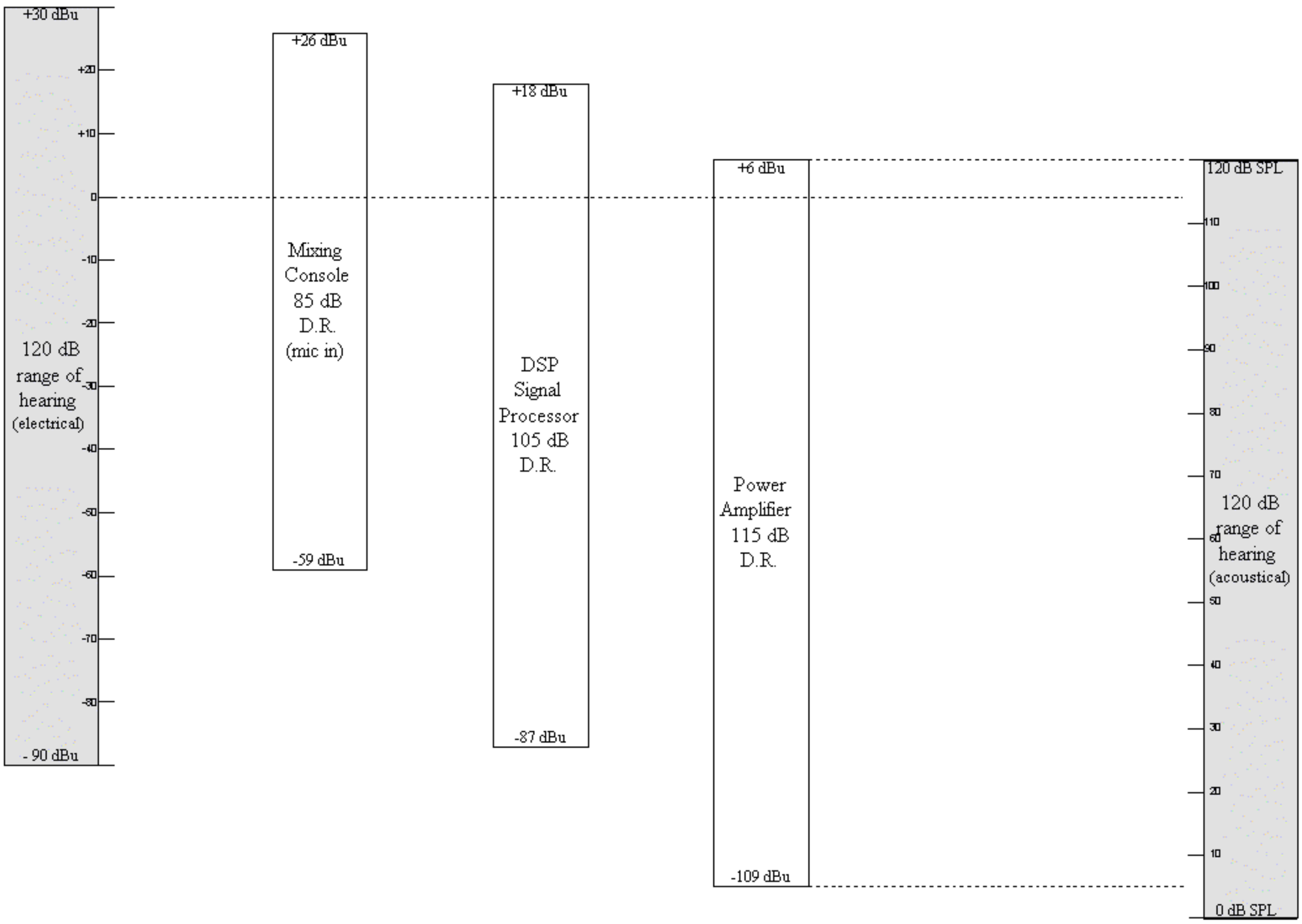

Figure 2.

Figure 3. illustrates that the resulting dynamic range for this popular gain structure is a mediocre-at-best 65 dB.

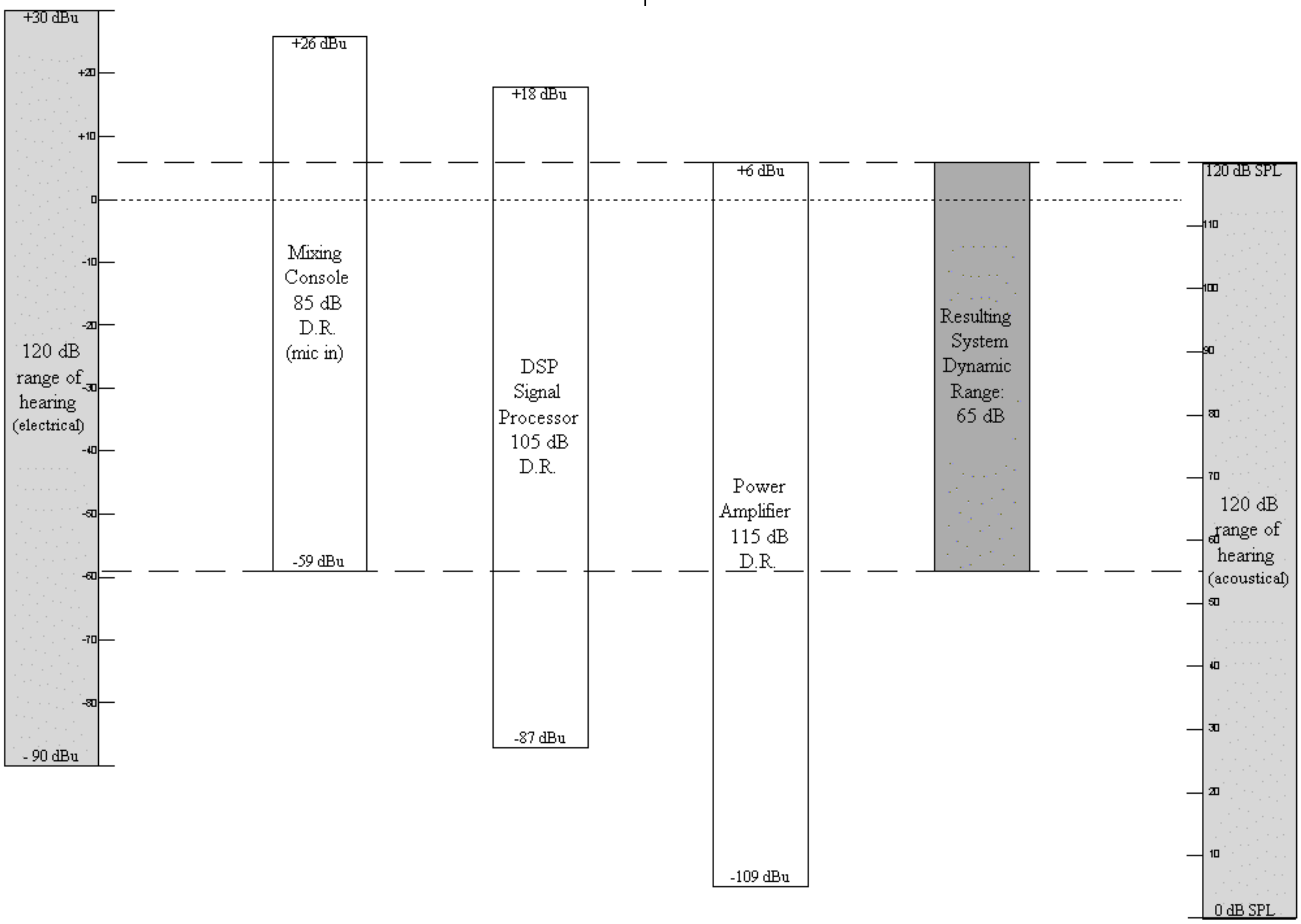

Figure 3.

It is the intent of this paper to explain how to properly set a sound system's gain structure so that it's dynamic range is no worse than that of the noisiest device in the signal chain. It will also be explained that, with some compromise to the noise floor, a system can sound as loud as a system many times its size, if that is the intent of the system technician. However, to cover this topic properly, we must start with the basics.

II. REFERENCE LEVELS

a. The Decibel

As confusing as the array of reference levels may seem in audio, one must have a basic knowledge of some of their relationships to properly understand gain structure. The decibel or dB is derived from the "Bel" to honor Alexander Graham Bell. The Bel was the amount a signal dropped in level over a one-mile distance of telephone wire. This level-loss was too large an amount to be useful as a single unit, so it was divided into tenths for practical application. The smaller unit was called the deci-Bel (tenth-Bel) or dB and has many values today, depending on the reference used. <u>The dB without a suffix only</u>

represents a ratio and must have a reference level, as indicated by the suffix to establish what value 0 dB is. For the purposes of this paper, we will only concern ourselves with those references that are involved in electrical audio signals, not acoustic ones.

b. Electrical dB References

The **dBm** is the original electrical dB power reference and 0 dBm is a power level of 1 milliwatt (mW). It is usually described as being dissipated across a 600-ohm load, however the load is of no real importance. 600 ohms simply happened to be the standard circuit resistance of a telephone line when the dBm was first used.

Most contemporary signal processors and amplifiers used in professional audio today have bridging inputs which are high impedance circuits that allow several units to be driven in parallel by one signal source. They also have op-amp outputs having very low output impedances that are designed to drive any load from 600 ohms to an open circuit (infinite impedance) without affecting the output voltage level. Since they are no longer based on load-sensitive power circuits like earlier audio electronics, the dBm is not as useful in current system calibration as a voltage-referenced dB. The dB has also been referenced to voltages, and the first used was 0 **dBV** = 1.0 volt (V). You may see this reference used from time to time, but the author prefers not to use it in his own calculations. It is enough just to know what it means if encountered in a specification.

The amount of voltage dropped across a 600-ohm load while dissipating 1 mW is approximately 0.775 V. This reference originally was the **dBv** and was often confused with the dBV, so now it is called the **dBu** (for unterminated), and 0 dBu = 0.775 V. To convert from dBV to dBu, just add 2.2 dB to the dBV value and that is the equivalent dBu value.

The last reference we must familiarize ourselves with is not an actual standard reference, but we spend more time looking at this level indication than any of the others. It is the VU meter "reference" **dB VU** (Volume Units) and basically 0 dB VU = +4 dBm if the circuit is loaded with 600 ohms. If the circuit is not loaded with 600 ohms, as is often the case in today's audio systems, then 0 dB VU = +4 dBu.

Chart 1. lists the dB references, their characteristics, and relationships.

Reference symbol:	reference type:	reference level:	comments:
dBm	power[1]	0 dBm = 1.0 mW	original electrical dB reference
dBV	pressure[2]	0 dBV = 1.0 V RMS = +2.2 dBm/dBu	currently not used often in professional audio
dBv	pressure[2]	0 dBv = 0.775 V RMS	older version of dBu, not currently used
dBu	pressure[2]	0 dBu = 0.775 V RMS	currently most useful in professional audio
dB VU	pressure[3]	0 dB VU = +4 dBu	pseudo-reference for VU meters & LED bar graphs

1. a 10-LOG power level ratio, twice the power = +3 dB increase
2. a 20-LOG pressure level ratio, twice the voltage = +6 dB increase
1. not a real reference, but a 20-LOG pressure level ratio, twice the voltage = +6 dB increase

Chart 1.

Since modern sound system signal chains are voltage sources, the author will use the dBu as the standard reference of measured voltages to set system gain structure and will relate it to the dB VU as viewed on the typical VU meter.

III. PEAK, RMS, and VU METER LEVELS

a. Peak Level

In order to set system gain structure, one must also understand the differences between peak and RMS levels. Peak level is very simple, it is the maximum signal voltage a circuit can produce. This is the level seen on an oscilloscope when a sine wave begins to clip. Because today's electronics have op-amp outputs, they are essentially voltage sources and peak level is reached when the power supply can no longer increase the voltage to the audio circuit. Capacitors in the circuit, usually in power amps, can sometimes provide instantaneous levels higher than the power supply voltage rails, but only for a short time and only a few dB above the continuous output level.

b. RMS Level

RMS (Root-Means-Square) level is a mathematical method for determining how much power is being dissipated by a complex waveform in an AC circuit. RMS gives the closest approximation of that power if a DC circuit was dissipating it. For this paper, it means 0.707 times the peak value of the sine wave as seen on an oscilloscope.

It is important to keep in mind that if, for example, a mixer's output voltage is rated in RMS value, the actual peak output where it begins to clip is that number divided by 0.707. A useful relationship that enables one to convert RMS voltages, as observed on a volt-ohm meter (VOM), to dBu, as given in specifications, is displayed in **Chart 2.**

In section IV, it will not be necessary to keep all these dB references or peak vs. RMS in one's head. The above information will enable one to properly understand and confirm product specifications involving levels. The methods employed here will enables one to visually or aurally set proper gain structure without complex calculations or conversions.

dBu	-20	-10	0	+4	+10	+15	+18	+20	+24	+26
V RMS	0.0775	0.2449	0.7746	1.2275	2.4495	4.3557	6.1531	7.7460	12.275	15.454

Chart 2.

c. VU Meter Level

The VU meter was originally designed to indicate a level that related very closely to the perceived loudness of the signal source. Today, many VU meters and LED bar graphs no longer have the proper meter ballistics due to cost limitations. However, the VU meter or LED bar graph on the output of the mixing console is what the system operator must depend on. Let's relate this to the peak output levels we will use to set gain structure.

Since a VU meter's movement is damped to reflect a signal's perceived loudness, it does a very poor job of indicating transient peaks. This is why many have employed separate peak indicators. Peak levels rarely exceed a level 15 to 20 dB above the level indicated by the VU meter. Mixing console designers are well aware of this fact and it is reflected in the consoles we have today. Consider that 0 VU = +4 dBu and most consoles have maximum output levels of +18 to +24 dBu. Their respective differences between 0 VU and maximum are 14 and 20 dB. This is the 14 to 20 dB of needed headroom for a safe average-to-peak ratio to avoid clipping. If one keeps the VU indication for program material around the 0 VU point, then the unobservable peaks will not exceed the maximum output, thus avoiding console clipping.

One word of caution: Do not trust a peak indicator to be the true clipping point of a mixing console. Manufacturers often set these indicators as much as 10 dB below actual clipping to "idiot proof" the mixer's signal quality. Don't get caught being the idiot!

IV. SETTING SIGNAL CHAIN GAIN STRUCTURE

The method in section IV requires an oscilloscope, or inexpensive piezo-electric transducer being fed a sine-wave signal below its passband, to determine the clipping point.

a. Preparation

Start with the input and output gains on all electronics in the signal path all the way down. Be sure that the signal path's equalizer, if it has one, has all its faders set to 0 dB. Equalization should be done after gain structure is optimized. If the system under test has a dedicated speaker "processor/controller," care needs to be taken to ensure that any internal equalization is accounted for by using the frequencies of maximum boost for crossover section test signals. Additionally, if the processor/controller has compression/limiting, it must not be operating during this procedure.

b. Adjustment Via Oscillator and Oscilloscope or Piezo-electric Transducer

Using an oscillator, send a 1 kHz sine wave signal out of the mixing console at a level just below clipping. The most foolproof way to confirm this condition is to look at its output with an oscilloscope. Clipping indicators and maximum level specifications cannot be trusted! As you turn up the level, the sine wave will "flat-top" once the maximum signal voltage of the circuit has been exceeded. This is the onset of clipping since the top and bottom of the waveform has been "clipped" off. Increase the input and then the output gain in turn on each device downstream in the signal path until the oscilloscope indicates the level in and out of each device is just below clipping. Be sure that the clipping threshold of the input attenuator, if the device under test has one, is found before adjusting its output attenuator.

If an oscilloscope is not in one's current budget, an inexpensive piezo-electric transducer, as manufactured by Motorola and used in low-cost speakers will do. In this case, one must use a 300 to 400 Hz tone, which is below the transducer's passband, for the sine-wave signal. Before clipping, the tone will not be heard through the transducer. Once a circuit has clipped, harmonics of the tone will be easily discernible, revealing that clipping has occurred.

This method has proven to be quite accurate, however, there are two limitations. First, it cannot be used on power amplifiers, as the signal is far too strong. Fortunately, the clipping indicators on most all power amplifiers can be trusted to indicate, within a dB or two, the true onset of clipping. Second, it cannot be used on signal chains after the high-pass portion of active crossovers where 300 to 400 Hz is not passed. Provided the crossover device has no built-in equalization, the other bandpass sections can usually be set the same as the 300 to 400 Hz bandpass section for gain structure purposes.

c. Active Crossover and Compressor/Limiter Adjustment

When adjusting the gain of a crossover network using an oscilloscope, a frequency in the center of each bandpass section should be used. Once the input gain is set for the first bandpass section, it should not need adjustment for the rest of the outputs (note the above concerns when processor/controllers are used). If compressor/limiters are in the signal path between the crossover outputs and the amplifiers' inputs, set them for no compression/limiting during this test. Concerning compressor/limiters, after setting gain structure, they may be appropriately adjusted according to the system technician's intent (see section **IV. f.**).

d. Power Amplifier Adjustment

After each output on the crossover has been set, start turning up the input on the lowest frequency amplifier with its output connected <u>only</u> to the oscilloscope until it is just below clipping. The output level of an amplifier is too strong for the piezo method. In this case, use the clipping indicators of each amplifier channel instead. Unlike mixers, amplifier-clipping indicators are usually accurate to within a few dB. If the amplifiers have input sensitivity switches (attenuator pads), set them for the highest voltage rating/least amount of input gain/sensitivity. Since today's amplifiers are constant-voltage outputs (if the load is = or > the minimum load of 2 - 8 ohms), the high impedance load of the oscilloscope will produce an amplifier output voltage sufficiently close (within a few dB) to the voltage when the speakers are connected. The remainder of the amplifier channels should be set the same way.

Once the speakers are connected, these amplifier channels will most likely be attenuated further to match the level of the lowest frequency section on the RTA's display during equalization via a test microphone. If a particular bandpass section cannot be appropriately matched to the other sections without exceeding the clipping threshold just set, the other sections must be further attenuated to match it. This will be the weak link in the system and every system has one. If overall maximum level is insufficient under these circumstances, see section **IV. f.,** or consider adding transducers and/or increasing power amplification to this bandpass section.

Please keep in mind that most all amplifiers need only a 0.775 to 1.545 V RMS signal (0 to +6 dBu) with the input attenuator wide open to produce full output power. <u>Understand that turning down an amplifier's attenuators does not reduce its potential power output. It simply decreases the input sensitivity, requiring more input voltage to get full output power.</u>

Since most mixing consoles produce from 6.153 to 15.454 V RMS (+18 to +26 dBu) before clipping, a sound system with its amps full up can send the amplifiers into 24 dB of clipping. Poof go the drivers and up go the sales of compressor/limiters!

Many sound companies tolerate clipping in their sound systems for two reasons: 1) the human ear does not readily detect clipping distortion of high frequency peaks due to their short time duration, and 2) we become accustomed to drivers sounding more harsh at high levels, attributing that harshness to driver distortion rather than amplifier clipping. The amplifiers' input attenuators must be turned down to just below clipping to avoid this condition. This way, when the console's output meters read an average level of 0 dB VU (+4 dBu), there is at least 20 dB of headroom for peaks in the console and all other devices downstream, including the amplifiers. This is the most economical driver protection (free), and it provides the greatest dynamic range for the system.

Remember **Figure 3.** and its marginal 65 dB of dynamic range for the sound system? Following the instructions of this section, the next figures step through proper gain structure setting for the example sound system
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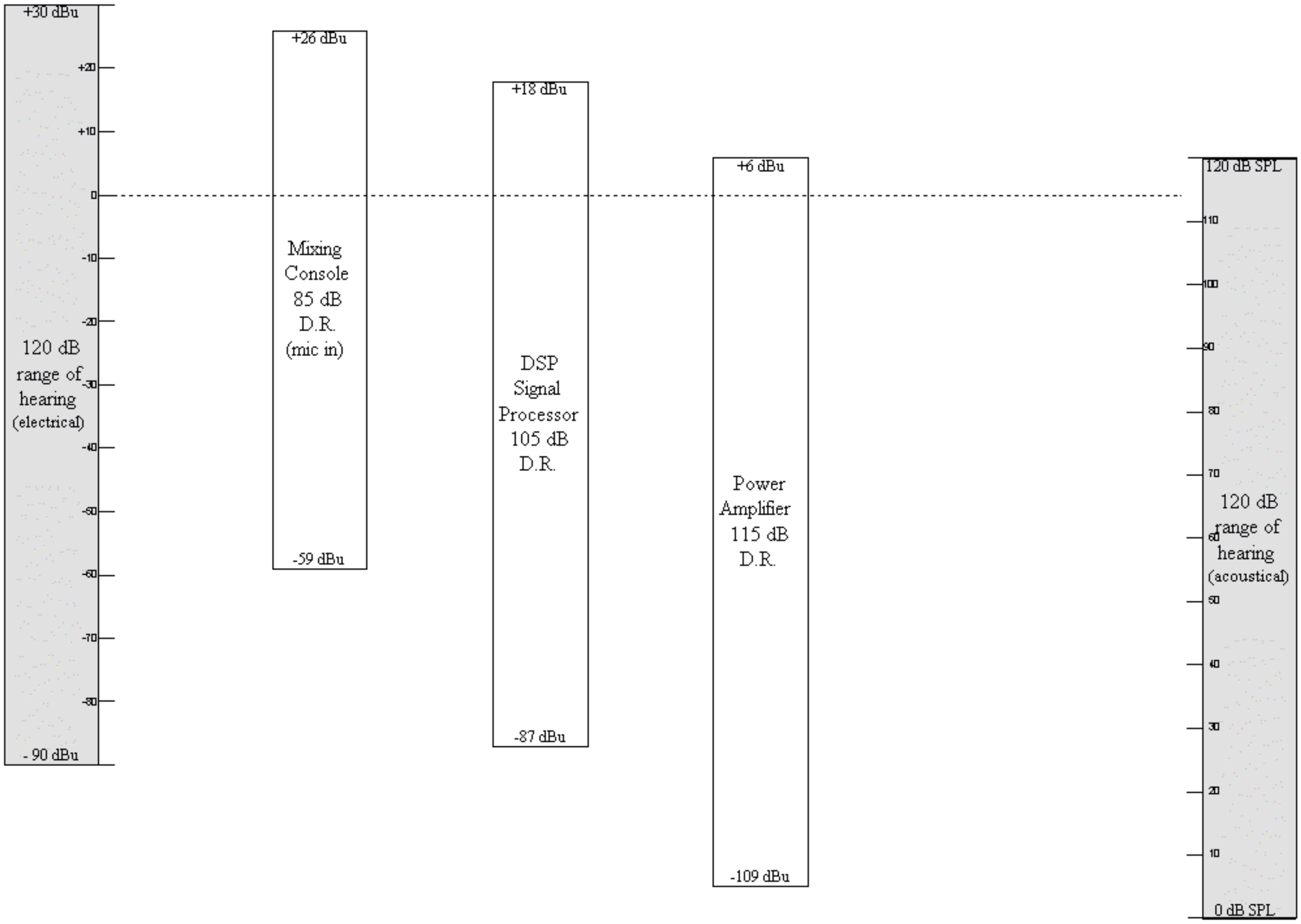

Figure 4.

Figure 4. starts with the gain structure of **Figure 3.** having unity gain for the line-level electronics and the amplifier's input attenuator full up.

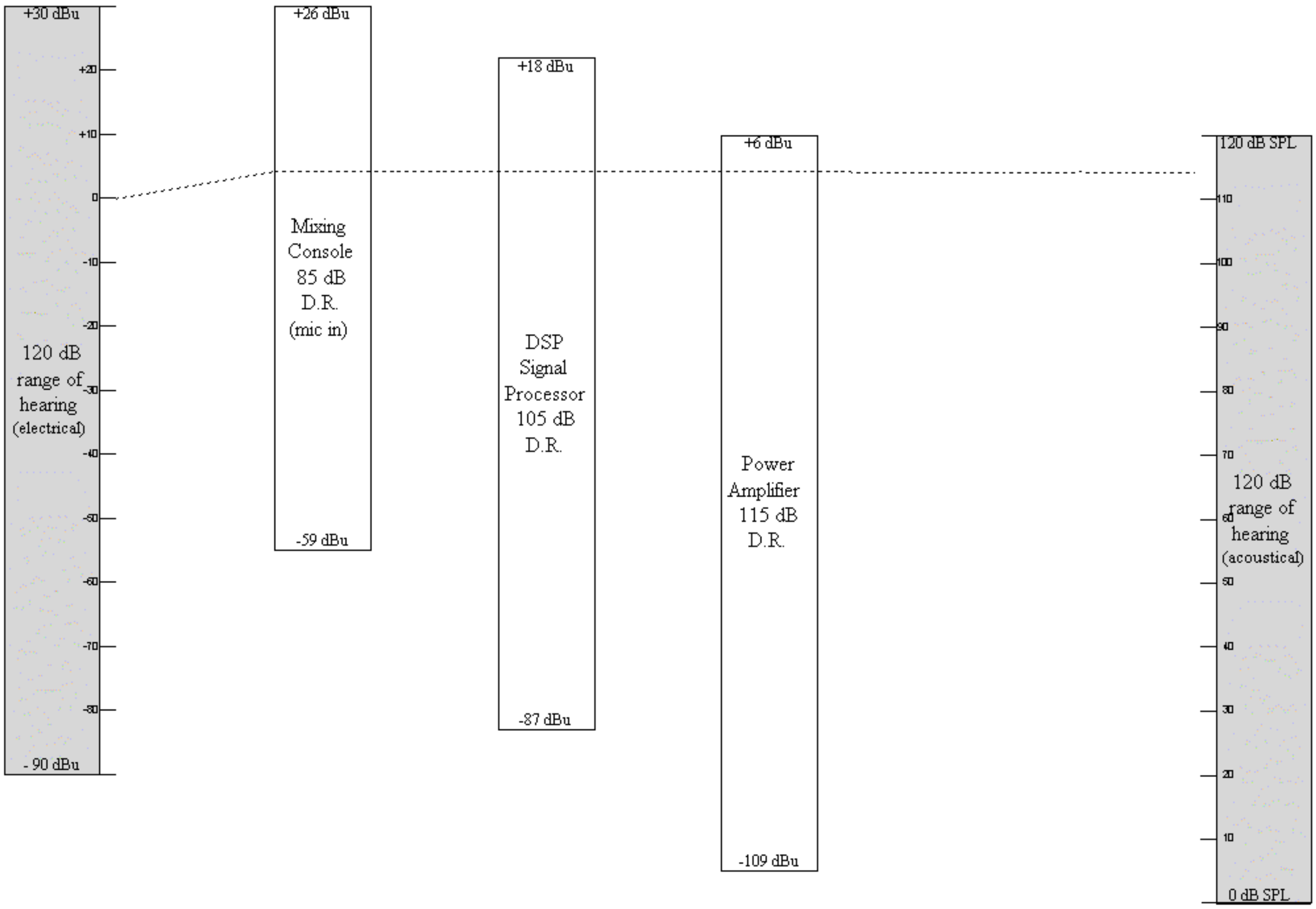

Figure 5.

Figure 5. adjusts the mixer's input for some arbitrary signal source so that it's input level is just below clipping.

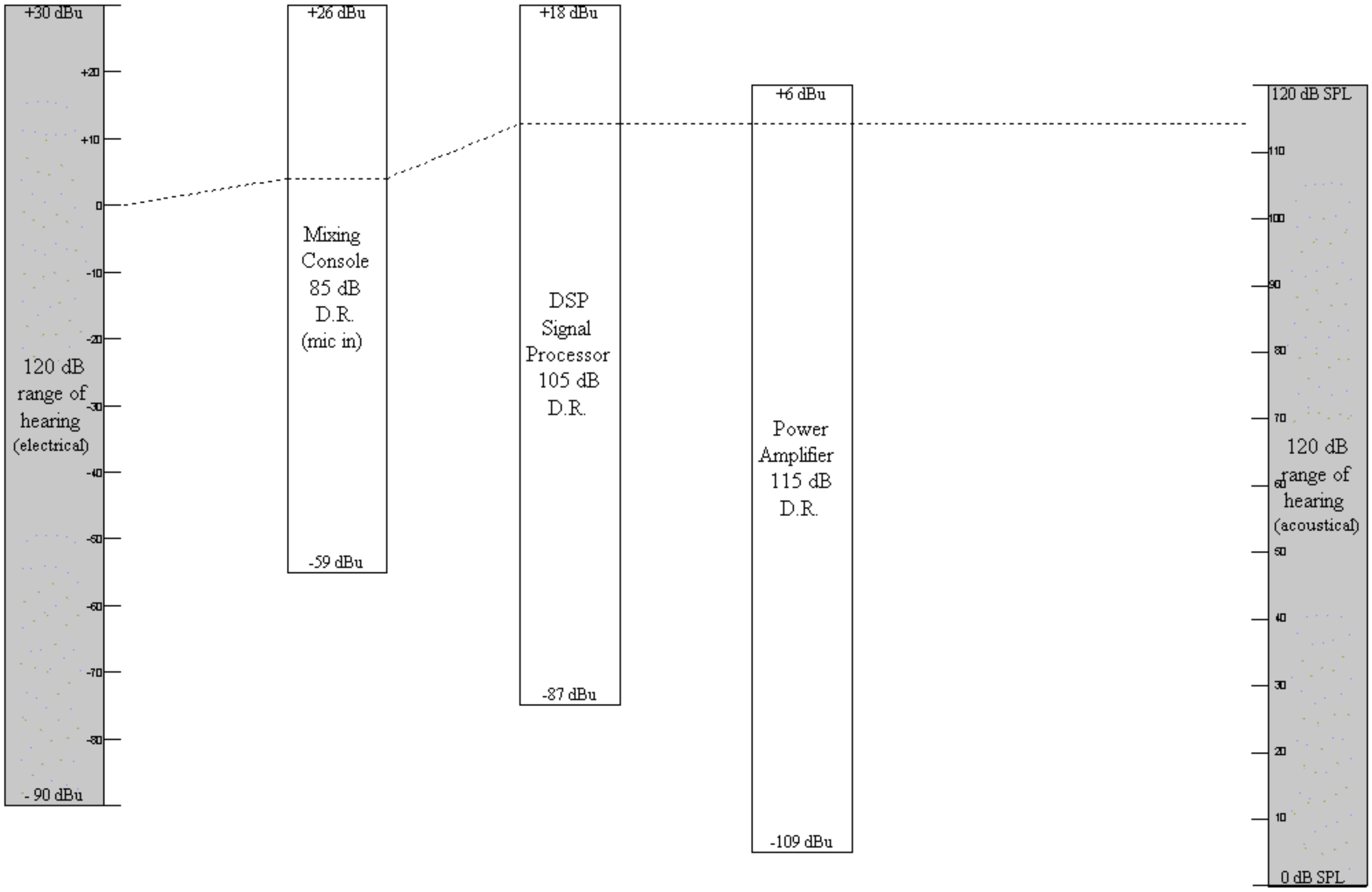

Figure 6.

Figure 6. adjusts the input of the DSP down 8dB so the +24 dBu output of the mixing console doesn't clip its +18 dBu input capacity. A word of caution here: Most active balanced inputs have the input attenuator located *after* the active-balancing circuit. If the active-balancing circuit chip cannot handle the input signal without clipping, and many cannot handle a +26 dBu signal, then the input can clip regardless of the attenuator setting. In this case, there are only two choices: 1) run the mixer a little lower in level, or 2) build an input pad for the DSP.

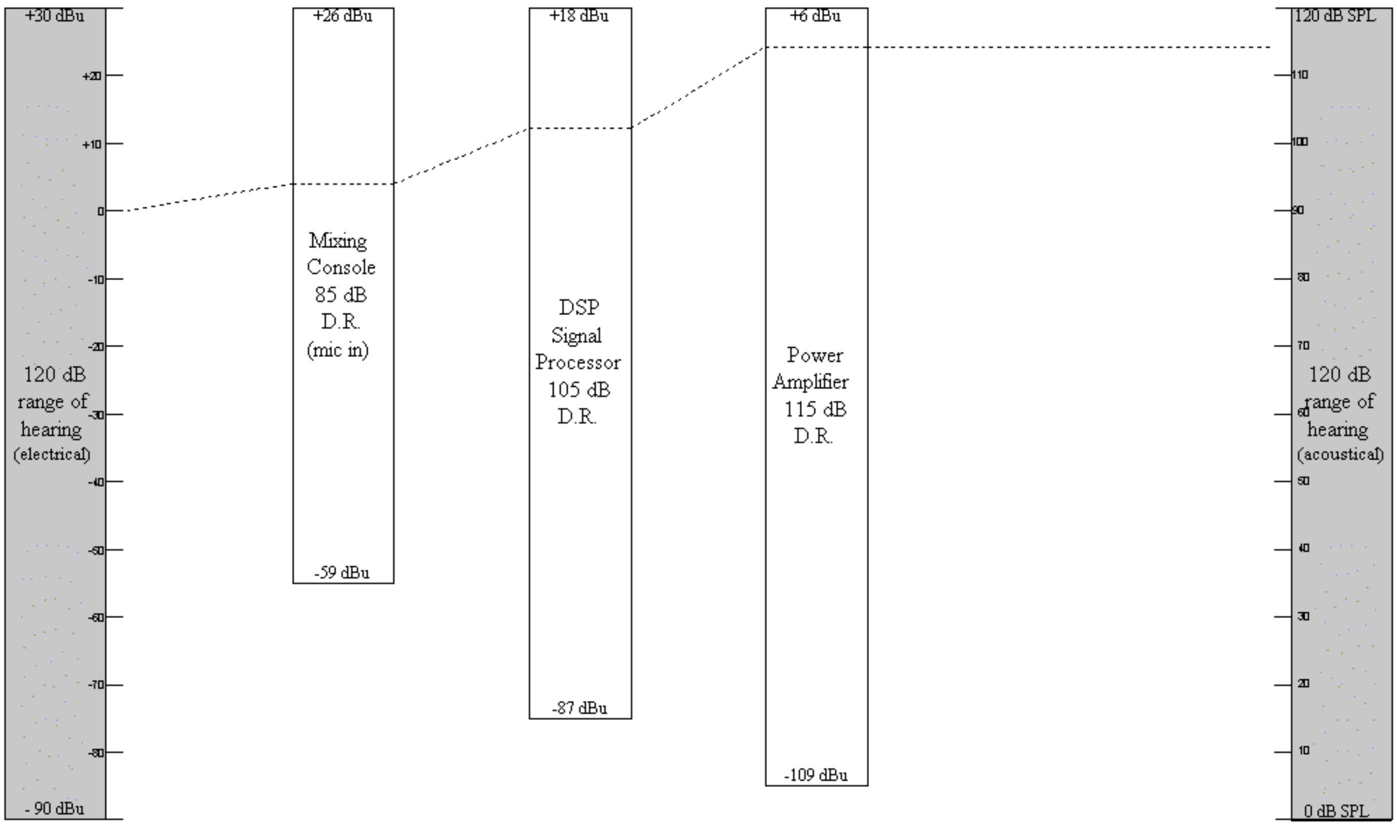

Figure 7.

Figure 7. adjusts the amplifier's input attenuator down by 12 dBu to reduce its sensitivity to match the +18 dBu output level of the DSP. It will now take a +18 dBu signal to drive the amplifier to full output instead of the +6 dBu signal that would do so when the input attenuator was full up. Notice that the arbitrary speaker level of 120 dB SPL can still be attained, but now the system cannot clip with a full level signal into the mixing console.

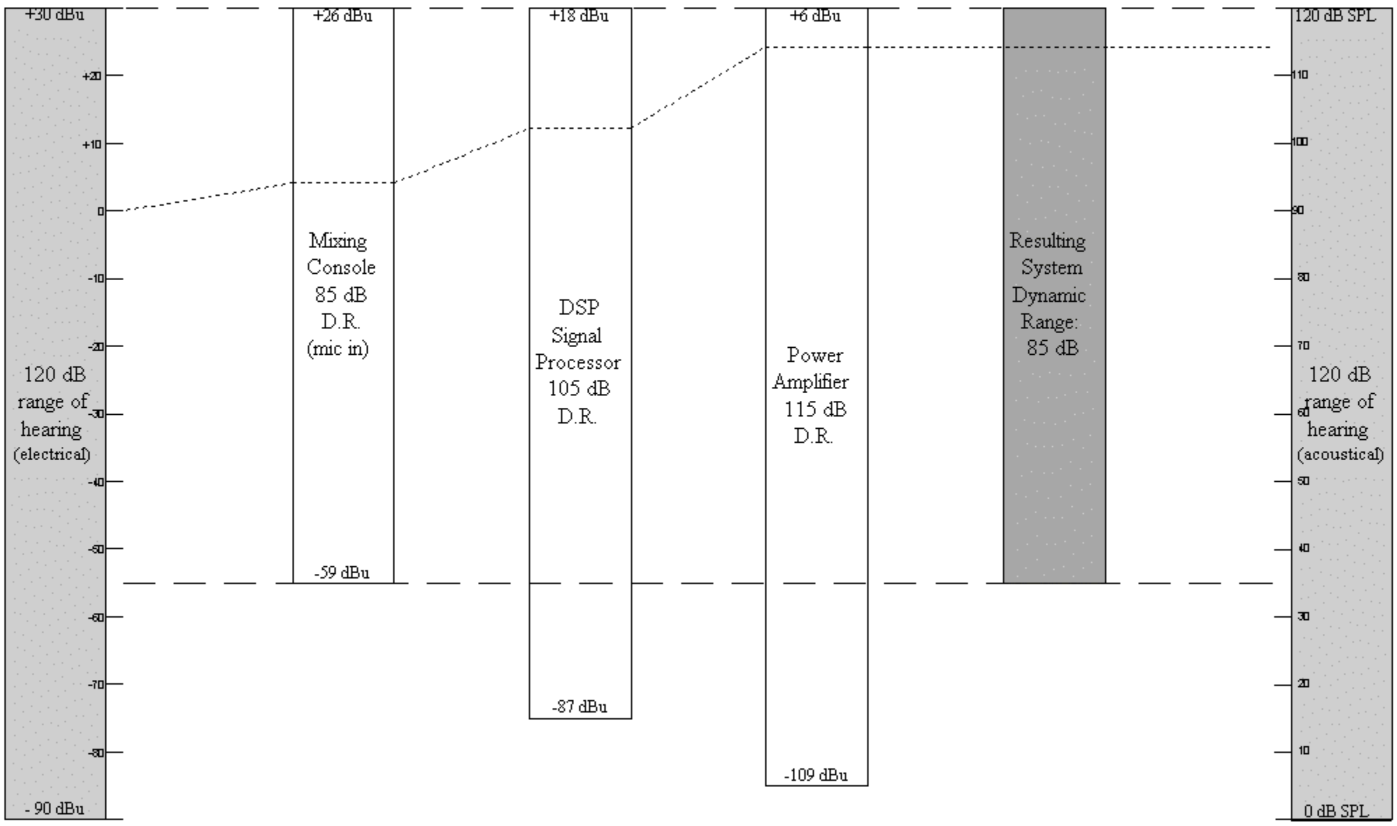

Figure 8.

Figure 8. shows the improved dynamic range to be had by properly adjusting the gain structure of the system. The marginal 65 dB of dynamic range for **Figure 3.** has now been improved by 20 dB, to 85 dB. Note that it is the same as the dynamic range of the weakest link in the signal chain. In this case, it's the mixer. Be aware that some noise floor addition due to the series circuit nature of the signal chain will make the system dynamic range *slightly* less than that of the weakest link, but it will be a minimal difference due to the way random noise levels add.

The system still produces the 120 dB SPL maximum peak level, but now is very difficult to clip and has a noise floor 20 dB lower than before with the gain structure properly adjusted.

e. Low Crossover Drive Level Option

One compromise to adjusting each and every amplifier's input attenuator individually is to turn down the output of the devices driving the amplifiers instead. This enables touring sound companies to have all amplifiers set the same (full up) without the worry of incorrect adjustment to any one amplifier. The trade-off to this compromise is that the potential interference from hum, noise, and RFI in the lines feeding the power amplifiers has been increased by the number of dB's the devices' outputs have been attenuated. Whatever the CMRR (Common Mode Rejection Ratio) of the amplifier's input specification is, it has just been compromised by that many dB's. The system technician in charge must decide if this compromise is acceptable. Often, it is the practical choice.

f. Higher Average-to-Peak Ratio/Lower Dynamic Range Option

For some applications, like live-band sound reinforcement, when one adjusts the gain structure as instructed above, the maximum level possible before the system clips is inadequate. It can be very expensive to have a system that provides a true 20 dB of peak-to-average headroom. A comprising method that is used often by large touring sound companies answers this dilemma. It is to turn the system amplifiers input attenuators up for greater sensitivity and use limiters just before the amplifiers' inputs to prevent amplifier clipping.

For example, if the amplifiers' inputs are turned up an additional 10 dB, when the console meters average 0 dB, limiters now must be set to reduce the peaks by 10 dB to avoid amplifier clipping and subsequent driver damage. This produces a higher average-to-peak output level ratio thus resulting in a 10-dB higher perceived loudness without clipping. This is the equivalent of having a speaker system 10 times as large!

Though the noise floor is now 10 dB higher than optimum, and the dynamic range has also been reduced by 10 dB, the economic gains of having a system 1/10th the size on tour is well worth the sonic sacrifice. This is especially true if the dynamic range of the source music by the touring heavy metal group is only 5 or 10 dB to start with. Just be sure to reset this system's gain structure to optimum before doing the local philharmonic!

V. SETTING MIXER GAIN STRUCTURE

a. General Considerations

The system's mixer, or mixing console in larger systems, also needs gain structure consideration, and this should be done <u>after</u> setting the downstream electronics' gain structure. The settings for this will change from day to day, depending on the signal sources, types of music, or the moods of the musicians that day for a live music system. As a result, in all but the simplest venues, one must adjust the mixing console's gain structure for each use.

The general rule of thumb for mixer gain structure is to present as much level at the beginning of the signal chain as possible without clipping and keep it that way to the outputs. This maintains the highest signal-to-noise ratio and greatest dynamic range for the mixed signal at the end of the signal path.

Another general rule to keep in mind is that when large numbers of sources are mixed together, the overall level increases. On some mixing consoles, one can mix several signals with their input gains properly set and their individual levels at zero and still clip the mix bus. Get to know the characteristics of the make and model being used.

b. Initial Adjustments

Be sure all controls are "zeroed out" with all levels at -∞ (off), all equalization controls at 0 dB (no boost or cut, unity gain) and all high-pass and low-pass filters off.

The input "trim" or "gain" control is the first level to set on the mixer, and it may be a fixed amount without a knob to control on less expensive rack-mount type mixers. For mixing consoles, it should be set just below clipping so the "clip" or "overload" LED does not illuminate on the loudest passage that source will be producing. For example, if the source is a mic'ed electric guitar, have the musician play their loudest passage and adjust the trim/gain down until it's associated LED stays off. Often an attenuator pad, usually -20 dB, must be used to prevent input overload. Do this for each source in the system. For mic- level inputs, this stage of amplification, the mic pre-amp stage, has more gain than any other in a sound system, including the power amplifier. Proper adjustment of this gain stage is the most important in the entire signal chain. If the mic pre-amp is clipping, nothing done downstream can remove the distortion induced here. Likewise, if the signal level is too low here, the result is the greatest amount of noise-floor hiss that can be produced in the system.

Next, if the mixer has sub-grouping capability that one can use to separate vocals, instruments, drum kits, choirs, etc., set the levels for these groups initially at 0 dB. As a mix is created, these levels will change to get a better balance between source sections. Always try to bring levels down to get the best balance rather than up so that mix-bus clipping can be avoided. Once again, be familiar with the board being used to know its characteristics.

Last, with a test source like a CD, put its input level
control fader, not its gain trim control, to zero and run

the applicable output levels up to zero. If this is too loud, this means two things: 1) Congratulations! The system has adequate headroom and output level, and 2) There is enough level to turn the system up later, if needed. If the level is not enough, consider the gain structure technique in section **IV. f.**

Following in **Sections c.** and **d.** are the two ways most professionals in the industry choose to set levels for mixing consoles. Each has an advantage and a disadvantage.

c. Technically Optimum Method

Once the input gains have been trimmed and the appropriate sub-group levels set to zero, set the master output levels to a comfortable listening level at or below zero. Now individual input channel faders are run up with the source intended to be the loudest, like a lead vocal mic, at zero and the others brought up somewhat lower for a balanced mix.

This provides the best signal-to-noise ratio and dynamic range for the system and is, in the author's terms "technically optimum." The disadvantage to this method is that the individual channel fader settings for a balanced mix can be difficult to recall for a repeating performance, or if one loses control of the mix and wants to return to a starting point.

d. Easily Recalled Method

For this method, all the individual input faders are set at 0 dB and the input gain trims of all but the loudest sources are reduced to a level well below the clipping point to provide a balanced mix. Once these initial levels have been set, DO NOT continue adjusting the input gain trims to rebalance the mix. Only adjust them if an input is clipping.

This method compromises the best noise floor for most of the sources and is problematic for levels of pre-fader monitor and effects-sends for each input. However it provides a balanced mix with all the channel faders at 0 dB, making it an easy benchmark to return to.

Choose the method that suits the situation. If mixing in a relatively quite environment, the technically optimum method may be preferred. If dramatic level changes occur throughout the event, or if it is repeated regularly, the easily recalled method may be the best for the situation.

VI. REFERENCE PUBLICATIONS

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3. New Ways to Set Up Gain Structure, a white paper by Chuck McGregor, TOA Electronics, Inc.