RaneNote 135

Setting Sound System Level Controls

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IMPORTANCE

Correctly setting a sound system's gain structure is one of the most important contributors to creating an excellent *sounding* system. Conversely, an improperly set gain structure is one of the leading contributors to *bad* sounding systems. The cost of the system is secondary to proper setup. The most expensive system set wrong never performs up to the level of a correctly set inexpensive system. Setting all the various level controls is not difficult; however, it remains a very misunderstood topic.

The key to setting level controls lies in the simple understanding of *what* you are trying to do. A few minutes spent in mastering this concept makes most set-ups intuitive. A little common sense goes a long way in gain setting.

A dozen possible procedures exist for correctly setting the gain structure of any system. What follows is but one of these, and is meant to demonstrate the *principles* involved. Once you master the fundamental principles, you will know what to do when confronted with different system configurations.

DECIBELS, DYNAMIC RANGE & MAXIMIZING HEADROOM

Audio-speak is full of jargon, but none so pervasive as the *decibel.* Those unfamiliar or rusty with decibel notation, and its many reference levels, are directed to the sidebar offered as review or introduction. Mastering gain, or level control settings also requires an understanding of *dynamic range* and *headroom*.

Dynamic range is the ratio of the loudest (undistorted) signal to that of the quietest (discernible) signal in a piece of equipment or a complete system, expressed in decibels (dB). For signal processing equipment, the maximum output signal is ultimately restricted by the size of the power supplies, i.e., it cannot swing more voltage than is available. While the minimum output signal is determined by the noise floor of the unit, i.e., it cannot put out a discernible signal smaller than the noise (generally speaking). Professional-grade analog signal processing equipment can output maximum levels of +26 dBu, with the best noise floors being down around -94 dBu. This gives a maximum *unit dynamic range* of 120 dB—a pretty impressive number coinciding nicely with the 120 dB dynamic range of normal human hearing (from just audible to painfully loud).

For sound systems, the maximum loudness level is what is achievable before *acoustic feedback*, or system squeal begins. While the minimum level is determined by the overall background noise. It is significant that the audio equipment noise is usually swamped by the HVAC (heating, ventilating & air conditioning) plus audience noise. Typical minimum noise levels are 35-45 dB SPL (sound pressure level), with typical loudest sounds being in the 100-105 dB SPL area. (Sounds louder than this start being very uncomfortable, causing audience complaints.) This yields a typical useable *system dynamic range* on the order of only 55-70 dB — quite different than unit dynamic ranges.

Note that the dynamic range of the system is largely out of your hands. The lower limit is set by the HVAC and audience noise, while the upper end is determined by the comfort level of the audience. As seen above, this useable dynamic range only averages about 65 dB. Anything more doesn't hurt, but it doesn't help either.

Headroom is the ratio of the *largest* undistorted signal possible through a unit or system, to that of the *average* signal level. For example, if the average level is +4 dBu and the largest level is +26 dBu, then there is *22 dB of headroom*.

Since you cannot do anything about the system dynamic range, your job actually becomes easier. All you need worry about is *maximizing unit headroom*. Fine. But, how much is enough?

An examination of all audio signals reveals music as being the most dynamic (*big surprise*) with a *crest factor* of 4-10.

Crest factor is the term used to represent the ratio of the peak (crest) value to the *rms* (*root mean square* — think *average*) value of a waveform. For example, a sine wave has a crest factor of 1.4 (or 3 dB), since the peak value equals 1.414 times the rms value.

Music's wide crest factor of 4-10 translates into 12-20 dB. This means that musical peaks occur 12-20 dB higher than the "average" value. This is why headroom is so important. *You need 12-20 dB of headroom in each unit to avoid clipping.*

PRESET ALL LEVEL CONTROLS IN THE SYSTEM

After all equipment is hooked-up, verify system operation by sending an audio signal through it. *Do this first before trying to set any gain/level controls.* This is to make sure all wiring has been done correctly, that there are no bad cables, and that there is no audible hum or buzz being picked up by improperly grounded interconnections (See RaneNote 110).

Once you are sure the system is operating quietly and correctly, then you are ready to proceed.

Turn down all power amplifier level/sensitivity controls.

Turn *off* all power amplifiers. (This allows you to set the maximum signal level through the system without making yourself and others stark raving mad.)

Set all gain/level controls to their *off* or *minimum* settings.

Defeat all dynamic controllers such as compressors/limiters, gate/expanders, and enhancers by setting the Ratio controls to 1:1, and/or turning the Threshold controls way up (or down for gate/expanders).

Use no *equalization* until after correctly setting the gain.

CONSOLE/MIC PREAMP GAIN SETTINGS

A detailed discussion of how to run a mixing console lies outside the range of this Note, but a few observations are relevant. *Think about the typical mixer signal path*. At its most basic, each input channel consists of a mic stage, some EQ, routing assign switches and level controls, along with a channel master fader. All of these input channels are then mixed together to form various outputs, each with its own level control or fader. To set the proper mixer gain structure, you want to maximize the overall S/N (signal-to-noise) ratio. *Now think about that a little:* because of the physics behind analog electronics, each stage contributes noise as the signal travels through it. (Digital is a bit different and is left to another Note and another day.) Therefore each stage works to *degrade* the overall signal-to-noise ratio. Here's the important part: *The amount of noise contributed by each stage is* (relatively) independent of the signal level passing through it. So, the bigger the input signal, the better the output S/N ratio (in general).

The rule here is to take as much gain as necessary to bring the signal up to the desired average level, say, +4 dBu, *as soon as possible.* If you need 60 dB of gain to bring up a mic input, you don't want to do it with 20 dB here, and 20 dB there, and 20 dB some other place. You want to do it all at once at the input mic stage. For most applications, *the entire system S/N (more or less) gets fixed at the mic stage.* Therefore set it for as much gain as possible without excessive clipping. Note the wording *excessive* clipping. A little clipping is not audible in the overall scheme of things. Test the source for its expected *maximum input level*. This means, one at a time, having the singers sing, and the players play, as loud as they expect to sing/play during the performance. Or, if the source is recorded, or off-the-air, turn it up as loud as ever expected. Set the input mic gain trim so the mic OL (overload) light just occasionally flickers. *This is as much gain as can be taken with this stage.* Any more and it will clip all the time; any less and you are hurting your best possible S/N.

(Note that a simple single mic preamp is set up in the same manner as a whole mixing console.)

OUTBOARD GEAR I/O LEVEL CONTROLS

All outboard unit level controls (except active crossovers — see below) exist primarily for two reasons:

- They provide the flexibility to operate with all signal sizes. If the input signal is too small, a gain control brings it up to the desired average level, and if the signal is too large, an attenuator reduces it back to the desired average.
- Level controls for equalizers: the need to provide make-up gain in the case where significant cutting of the signal makes it too small, or the opposite case, where a lot of boosting makes the overall signal too large, requiring attenuation.

Many outboard units operate at "unity gain," and do not have *any* level controls — what comes in (magnitude-wise) is what comes out. For a perfect system, *all* outboard gear would operate in a unity gain fashion. *It is the main console's (or* preamp's) job to add whatever gain is required to all input signals. After that, all outboard compressors, limiters, equalizers, enhancers, effects, or what-have-you need not provide gain beyond that required to offset the amplification or attenuation the box provides.

With that said, you can now move ahead with setting whatever level controls *do* exist in the system.

Whether the system contains one piece of outboard gear, or a dozen, gains are all set the same way. Again, the rule is to maximize the S/N through each piece of equipment, thereby maximizing the S/N of the whole system. And that means setting things such that your *maximum system signal* goes straight through every box without clipping.

RaneGain Test Set

The RaneGain (RG) test set is a handy tool kit based on techniques first developed by Pat Brown of *Syn-Aud-Con* for use in quickly setting sound system gain controls. It consists of two pieces: a self-contained, phantom-powered 400 Hz generator and a separate audio Transducer housed in an XLR connector. The RG Generator plugs into any mic input on a mixing console (or separate mic preamp) having phantom power in the range of 12-48 VDC, providing a convenient sound source. The RG Transducer plugs into the output of each unit and sounds a warning whenever the output level is clipped. See the RaneGain data sheet (*www.rane.com/ranegain.html*) for additional details.

SETTING SIGNAL PROCESSING LEVEL CONTROLS

First, a sound source is connected to the mixing console (or separate mic preamp) to provide the *maximum system signal* output, then this signal is used to set the outboard units.

The most convenient sound source is one built into the mixer or preamp. If a built-in generator is available, use that; if not, use an external oscillator, such as the RaneGain generator or other test equipment. Connect the generator to an unused channel in the mixing console or to the input of the mic preamp. Carefully set the generator level and the channel input fader so the mic stage does not overload. Next, adjust the master output fader (or preamp output level control) for the largest level possible without clipping the output stage. Determine this maximum level using any of the four methods: *RaneGain Test Set*, *OL Light*, *Oscilloscope*, or *AC Voltmeter* described below.

- **RaneGain Test Set** Plug the RG Transducer into the console's (or preamp's) master balanced output XLR jack. Turn up the master output fader (or preamp output level control) until the Transducer first sounds; reduce the level until the Transducer stops. This is now the *maximum system signal* output.
- **OL Light** Adjust the sound source until the master output overload (OL) indicator just begins to light (or the output meter indicates an OL condition). This is now the *maximum system signal* output, although it is a conservative maximum since most OL indicators come on several dB before actual clipping.
- **Oscilloscope** Using the RG Transducer or OL light are fast and convenient ways to set levels. However, a better alternative is to use an oscilloscope and actually *measure* the output to see where excessive clipping really begins. This method gets around the many different ways that OL points are detected and displayed by manufacturers. *There is no standard for OL detection.* If you want the absolute largest signal possible before real clipping, you must use either the RG Transducer or an oscilloscope.
- **AC Voltmeter** If the RG Transducer or an oscilloscope is out of the question, another alternative is to use an AC voltmeter (preferably with a "dB" scale). Here, instead of relying on the OL indicator, you choose a very large output level, say, +20 dBu (7.75 Vrms) and *define that as your maximum level.* Now set everything to not clip at this level. This is a reasonable and accurate way to do it, but is it an appropriate maximum? Well, you already know (from the above discussion) that you need 12- 20 dB of headroom above your average signal. It is normal pro audio practice to set your average level at +4 dBu (which, incidentally, registers as "0 dB" on a true VU meter). And since all high quality pro audio equipment can handle +20 dBu in and out, then this value becomes a safe maximum level for setting gains, giving you 16 dB of headroom — plenty for most systems.

Outboard gear gain/level controls fall into three categories:

- No controls
- One control, either Input or Output
- Both Input & Output Controls

Obviously, the first category is not a problem!

If there is only one level control, regardless of its location, set it to give the maximum output level either by observing the OL light, or the RG Transducer, or the oscilloscope, or by setting an output level of +20 dBu on your AC voltmeter.

With two controls it is very important to set the *Input control* first. Do this by turning up the Output control just enough to observe the signal. Set the Input control to barely light the OL indicator, then back it down a hair, or set it just below clipping using your oscilloscope, or until the RG Transducer buzzes. Now set the Output control also to just light the OL indicator, or just at clipping using the scope, or just buzzing. (Note: there is no good way to optimally set an input control on a unit with two level controls, using only an AC voltmeter.)

For Rane digital audio products, like the **RPM 26v Multiprocessor** where input A/D (analog-to-digital) metering is provided with the RW 232 software, setting the input level gain is particularly easy and *extremely important*: Using the maximum system signal as the input, open up the Input Trim box and simply slide the control until the 0 dBFS indicator begins lighting. This indicates the onset of "digital clipping," and is definitely something you want to avoid, so this is the maximum gain point.

SETTING POWER AMPLIFIERS

If your system uses active crossovers, for the moment, set all the crossover output level controls to maximum.

Much confusion surrounds power amplifier controls.

First, let's establish that power amplifier "level/volume/gain" controls are *input sensitivity* controls. (no matter *how* they are calibrated.) They are not *power* controls. They have absolutely *nothing to do with output power*. They are sensitivity controls, i.e., these controls determine exactly what input level will cause the amplifier to produce full power. Or, if you prefer, they determine just how *sensitive* the amplifier is. For example, they might be set such that an input level of +4 dBu causes full power, or such that an input level of +20 dBu causes full power, or whatever-input-level-your-system-may-require, causes full power.

They do not *change* the available output power. They only change the required input level to produce full output power.

Clearly understanding the above, makes setting these controls elementary. You want the *maximum system signal* to cause full power; therefore set the amplifier controls to give full power with your maximum input signal using the following procedure:

1. Turn the sensitivity controls all the way down (least sensitive; fully CCW; off).
2. Make sure the device driving the amp is delivering max (unclipped) signal.
3. Warn everyone you are about to make a LOT of noise!
4. Cover your ears and turn on the first power amplifier.

5. Slowly rotate the control until clipping just begins. Stop! This is the maximum possible power output using the maximum system input signal. In general, if there is never a bigger input signal, this setting guarantees the amplifier cannot clip. (Note: if this much power causes the loudspeaker to “bottom out,” or distort in any manner, then you have a mismatch between your amplifier and your loudspeaker. *Matching loudspeakers and amplifiers is another subject beyond this note.*)
6. Repeat the above process for each power amplifier.
7. Turn the test signal off.

ACTIVE CROSSOVER OUTPUT LEVEL CONTROLS

Setting the output attenuators on active crossovers differs from other outboard gear in that they serve a different purpose. These attenuators allow setting different output levels to each driver to correct for efficiency differences. This means that the same voltage applied to different drivers results in different loudness levels. This is the loudspeaker *sensitivity* specification, usually stated as so many dB SPL at a distance of one meter, when driven with one watt. Ergo, you want to set these controls for equal maximum loudness in each driver section. Try this approach:

1. Turn down all the crossover outputs *except for the lowest frequency band*, typically labeled “Low-Out.” (Set one channel at a time for stereo systems.)
2. If available, use pink noise as a source for these settings; otherwise use a frequency tone that falls mid-band for each section. Turn up the source until you verify the console is putting out the maximum system signal level (somewhere around the console clipping point.) Using an SPL meter (*Important: turn off all weighting filters; the SPL meter must have a flat response mode*) turn down this one output level control until the maximum desired loudness level is reached, typically around 100-105 dB SPL. Very loud, but not harmful. (1-2 hours is the Permissible Noise Exposure allowed by the U.S. Dept. of Labor Noise Regulations for 100-105 dB SPL, A-weighted levels.)

Okay. You have established that with this maximum system signal this driver will not exceed your desired maximum loudness level (at the location picked for measurement). Now, do the same for the other output sections as follows:

1. *Mute* this output section — *do not turn down the level control; you just set it!* If a Mute button is not provided on the crossover, disconnect the cable to the power amp.
2. Turn up the next output section: either “High-Out” for 2-way systems, or “Mid-Out” for 3-way systems, until the *same maximum loudness level* is reached. Stop and mute this output.
3. Continue this procedure until all output level controls are set.
4. Un-mute all sections, and turn off the test source.

Congratulations! You have finished correctly setting the gain structure for your system. Now you are ready to adjust EQ and set all dynamic controllers. Remember, after EQ-ing to *always reset the EQ level controls for unity gain*. Use the Bypass (or Engage) pushbuttons to “A/B” between equalized and un-equalized sound, adjusting the overall level controls as required for equal loudness in both positions.

SUMMARY

Optimum performance requires correctly setting the gain structure of sound systems. It makes the difference between excellent sounding systems and mediocre ones. The proper method begins by taking all necessary gain in the console, or preamp. All outboard units operate with unity gain, and are set to pass the *maximum system signal* without clipping. The power amplifier sensitivity controls are set for a level appropriate to pass the maximum system signal without excessive clipping. Lastly, active crossover output controls are set to correct for loudspeaker efficiency differences.

REFERENCES

1.Murray, John & Pat Brown, “A Gain Structure Guide,” *LIVE SOUND! International*, pp. 18-24, Mar/Apr 1997. Thanks to John and Pat for inspiration and some content for this RaneNote.
2. “Rane Professional Audio Reference,” *http://www.rane.com.* Its our website; visit it.
3. *The Syn-Aud-Con Newsletter*. Various issues; you need them all — subscribe: 1-800-796-2831.

DECIBELS

decibel *Abbr.* **dB** Equal to one-tenth of a bel. (After **Alexander Graham Bell**.) The preferred method and term for representing the *ratio* of different audio levels. It is a mathematical shorthand that uses *logarithms* (a shortcut using the powers of 10 to represent the actual number) to reduce the size of the number. For example, instead of saying the dynamic range is 32,000 to 1, we say it is 90 dB [*the answer in dB equals 20 log x/y, where x and y are the different signal levels*].
Being a ratio, *decibels have no units*. Everything is relative. Since it is relative, then it must be relative to some *0 dB reference point.* To distinguish between reference points a suffix letter is added as follows:

0 dBu A voltage reference point equal to 0.775 Vrms (u = *unterminated*, i.e., the impedance is irrelevant).

+4 dBu Standard pro audio voltage reference level equal to 1.23 Vrms.

0 dBV A voltage reference point equal to 1.0 Vrms.

-10 dBV Standard voltage reference level for consumer and some pro audio use (e.g. TASCAM), equal to 0.316 Vrms. (Tip: RCA connectors are a good indicator of units operating at -10 dBV levels.)

0 dBm A *power* reference point equal to 1 milliwatt. To convert into an equivalent voltage level, *the impedance must be specified*. For example, 0 dBm into 600 ohms gives an equivalent voltage level of 0.775 V, or *0 dBu* (see above); however, 0 dBm into 50 ohms, for instance, yields an equivalent voltage of 0.224 V — something quite different. Since modern audio engineering is concerned with voltage levels, as opposed to power levels of yore, the convention of using a reference level of 0 dBm is obsolete. The reference levels of 0 dBu, or -10 dBV are the preferred units.

0 dBr An arbitrary reference level (r = *re*, or *reference*) that must be specified. For example, a signal-to-noise graph may be calibrated in dBr, where 0 dBr is specified to be equal to 1.23 Vrms (+4 dBu); commonly stated as “dB re +4,” that is, “0 dBr is defined to be equal to +4 dBu.”

0 dBFS A digital audio reference level equal to “Full Scale.” Used in specifying A/D and D/A audio data converters. Full scale refers to the maximum *peak* voltage level possible before “digital clipping,” or digital overload (“overs”) of the data converter. The Full Scale value is fixed by the internal data converter design and varies from model to model.

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