A systematic approach to audio processing

By Martin Acuff

Getting the desired sound for your station can be difficult and frustrating. To complicate the matter, there are many types of processors in use, both old and new. Moreover, there is not one audio aesthetic; several factors determine a station's ideal sound. Fortunately, you can follow a systematic approach to tailor your station's sound and achieve the desired results.

Tuning a station's audio processing requires artistic judgment and technical experience. The basic approach is finding your preferred sound incrementally, usually by adjusting past the optimum tuning point, then backing it off. It is unlikely that the first pass of adjustments will meet your objective, so you should employ a progressive approach to the task. The following sections detail this approach to achieving the on-air sound you are looking for.

Subjectivity and definition

If the general manager, program director and engineer have diverse preferences, define who will be the ultimate judge of your station's sound. This is an opportunity for interdepartmental teamwork. Secure the best outcome by uniting all players toward a common objective.

For a broader perspective on your station's sound, consider using audience research. Such research is unbiased and will prove more reliable than depending solely upon intuition or the preferences of one (or a few) individuals at the station.

There is no right or wrong audio quality, only a scale of preferences. The best sound for your station is processing that maximizes your target audience. Align your processing objectives to complement the overall audience, format and business plan. For example, if increased TSL (time spent listening) is a priority for your station's ratings, aggressive processing for dial dominance is not appropriate.

Formats such as CHR and urban often benefit from a highly processed sound that increases the energy level of the presentation. Compression and equalization can do this without excessive clipping.

Good judgment must be the rule. In comparison with audio processing, it's much easier to tune the RF section of a transmitter because there is a meter indicating when tuning is ideal. There is no meter to indicate that processing is in tune; you must rely on your artistic judgment. Defining the ideal point on the scale is perhaps the most difficult tasks in meeting your objective.

Legal limitations restrict peak modulation. Peak modulation and perceived loudness are not the same. A minimally processed classical format can have the same modulation peaks as an aggressively processed hot urban format, although the latter will sound much louder on the dial. Increasing loudness by simply cranking the modulation invites legal penalties; more effective (and less costly) alternatives exist. Loudness is perceived approximately as the average power level of audio. However, many short-duration peaks in unprocessed audio go much higher than the perceived loudness level. This difference is commonly referred to as the peak-to-average ratio. These short-duration peaks contribute nothing to perceived loudness but do determine the limits of modulation. The purpose of all broadcast audio processors is to reduce the peak-to-average ratio to elevate perceived loudness without exceeding the peak limitations.

System considerations

No audio processor can repair source audio that is noisy, distorted, excessively bit-rate reduced or suffering from other maladies. In fact, aggressive audio processing exaggerates defects in the source material. The best-sounding stations start with the best-sounding source material.

Installing more audio processing components in your system will not guarantee better-sounding audio. It does, however, guarantee greater difficulty in managing the audio processing to achieve the desired result. Additional components increase risk of failure and often degrade the audio. Changes made to the first

component cause downstream components to react in sometimes-unpredictable ways. In general, it's wise to minimize components in the audio chain.

Peak overshoot is the enemy of perceived loudness. Every percent of overshoot broadcast is about 0.1dB of loudness sacrificed. Therefore, every component in the system following and including the broadcast audio processor must have the lowest possible overshoot.

Not all audio processors are created equal. Probably the best measure of an audio processor's performance is how effectively it can reduce the peak-to-average ratio without introducing unpleasant artifacts.

Components in the system must be arranged to deliver the best peak control possible (i.e., minimal overshoot) to the transmitter. A common problem is locating the main audio processor prior to an overshooting studio to transmitter link (STL); if the audio with overshoots is applied directly to the transmitter, you must reduce modulation (and loudness) to remain legal. Two common solutions are to relocate the main audio processor after the STL or to fix the overshoot problem in the STL itself. While some newer exciters have built-in overshoot limiters, they should be used only as a last resort if you can't eliminate overshoot by other means. Moreover, composite clipping is not appropriate to compensate for overshooting STLs because it adds audible distortion and interferes with SCAs and the stereo pilot tone, reducing your stereo coverage area.

Receivers differ. What sounds great on a dollars 3,000 receiver will not sound the same on a dollars 100 boom box. Act upon the broadcasting system as a whole: from the studios to the audience. Appraise through the lens of the entire system: processing components, STL, transmitter and receiver.

Listen in varied environments, especially those typical for your audience. In addition, if your target audience typically listens at low volume, it's wise to do the same. It's prudent to have a high-quality monitoring system as a reference, but always keep things in context of the target audience.

Where and when to tweak

Pick a starting point. It doesn't matter whether you're walking in cold or correcting an ongoing problem. If you have an analog processor, start with one of the suggested settings listed in the operating manual. If you have a digital processor, try one or more of the manufacturer's recommended presets as a starting point. Some digital processors have the added advantage of a single control that scales many parameters to your preference, making the task of adjustment much easier than tweaking individual parameters, and less time-consuming.

Decide what to tweak. Make an effort to understand the operation of your audio processor(s). Study the operating manual(s). Understand the effect of each adjustment - what it will do and won't do for your sound. Adjusting the Automatic Gain Controls probably shouldn't be your first choice to correct a clipping or distortion problem. Furthermore, manufacturers implement traditional controls in different ways: AGC drive may not perform the same on different products. If you are uncertain about the description of the Throbulator control, you may wish to carefully sweep that control through its range while monitoring its effect. Be careful if you do so, as some controls can yield significant changes, and make sure to return it to its original position. If you're still uncertain, call the manufacturer's technical support.

Document everything you do and document each change along the way. At some point, you'll need to back up to a previous setting. For analog processors, create a chart tabulating all relevant parameters, and the date of each change (see Figure 1). If you have a digital audio processor, save each change to a preset. That way you can revert to any previous settings whenever needed.

Proceed one step at a time. Use an incremental approach to adjusting the audio processing. As mentioned earlier, taking a single huge leap is seldom (if ever) effective. Make small changes to perhaps only one or two parameters per adjustment session. Then listen. Making large changes to many parameters in one

session is difficult to manage. The downside to making small adjustments is that differences can be too subtle to notice.

Allow plenty of time to judge the effects of your changes. Sometimes the consequences will not be immediately evident. What sounds great on your dollars 10,000 monitoring system may sound wimpy on a car radio. So, listen to many selections, from many sources and program types. Listening over time will also give your ears an opportunity to rest and time for you to formulate an opinion.

Determine a stopping point. If the present on-air sound is far from your goal, it's relatively easy to judge changes to the audio processing. However, as the processed sound approaches the goal, subjective differences from the ideal become smaller and increasingly subtle. Take your time.

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