Evaluating FM Transmission Audio Processors

Introduction

Broadcast audio processing is both an engineering and artistic endeavor. The engineering goal is to make most efficient use of the transmission channel while preventing its overmodulation. The station's management sets the artistic goal. It may be to avoid audibly modifying the original program material at all. Or it may be to create a distinct sonic signature for the broadcast by radically changing the sound of the original. Most broadcasters operate somewhere in between these two extremes, with the main goal of audio processing to increase the perceived loudness within the peak modulation constraints of a transmission channel.

If the transmitted signal meets regulatory requirements for modulation control and RF bandwidth, there is no well-defined right or wrong way to process audio. Like most areas requiring subjective artistic judgment, processing is highly controversial and likely to provoke thoroughly opinionated arguments amongst its practitioners. Ultimately, the success of a station's audio processing must be judged by its results. If the broadcast gets the desired audience, then the processing must be deemed satisfactory, regardless of the opinions of audiophiles, purists, or others that consider processing an unnecessary evil.

Any audio processor that performs weakly in either the engineering or artistic categories will put a radio station at a disadvantage by comparison to its competition.

Subjective Evaluation

We believe that professionally produced processing for popular music formats should be clean on voice and music, consistent (the hard one), polished, "produced," and punchy. The processing should sound good to the desired demographic on a vast majority of consumer radios and with all program material (except for the very occasional pathological case). It should be tuned to the station's target audience, which may require dayparting the processing.

In a matter of speaking, processing can be viewed as an art form, just like painting a picture. What one individual finds attractive another may find repulsive. However, the general trend towards commercially successful audio processing systems, aside from superb technical performance, is to achieve loudness and give the listener the illusion that the signal is not processed. The idea is to get the original artist's performance on the air, within the constraints and limitations of the medium, without gross modification of the original artist's and producer's intent.

Stations should consider how the original recording artists and producers intended a record to sound, and should not radically change this sound without careful consideration. All major-label CDs have been "mastered" in a mastering facility (like Bernie Grundman Mastering in Los Angeles, for example). The purpose of the mastering step is to make final tweaks to the master tape (such as cut-to-cut level matching, fine-tuning equalization and adding compression and peak limiting) to give it an "international, major-label" sound in cooperation with the original artist and producer. We believe that it is wise for the individual evaluating and adjusting an audio processor to be very aware of this contemporary, major-label sound texture. The evaluator should then try to reflect it in the station's sound, because this is what the radio audience associates with top-grade commercial CDs from the big international labels.

It is a waste of time to argue subjective preference beyond the issues above. You can go round and round forever and never do more than "agree to disagree." What should count to any professional broadcast engineer is not personal preference, but whether the processing appeals, in some statistical sense, to the target audience of a given radio station.

With the above in mind, we can give some specific advice about subjective evaluation of processing.

Listen on a high-quality home receiver, a typical auto radio, and a portable or tabletop radio. In particular, the lack of bass response in the portable may reveal problems like midrange pumping that were masked by the radios with good bass response.

Evaluating *consistency* requires you to devote considerable time to an evaluation. Listen to all of the different types of program material presented within your format. Listen to a wide variety of music and voice, both previously produced and from your own live microphone channel. Make sure that all of this material emerges from the processing with a consistent texture. Transitions from one item to the next should be smooth. The spectral balance should not shift unnaturally, particularly within one piece of program material. No material (particularly live voice) should exhibit raspiness or other overt distortion.

Evaluating *texture* is a multifaceted procedure. Most modern processors come with factory presets designated for different programming formats. Choose the preset closest to your format.

There are several subjective aspects that should concern you. The first is *density.* Dense audio has a small short-term dynamic range. It is a sound that you rarely, if ever, find in an unprocessed CD. Operating multiband compression with fast release times usually creates it. Very dense audio is generally appropriate for stations looking for maximum loudness and dial impact. However, it tends to be fatiguing in the long-term, so it is usually most appropriate for formats that emphasize cume over time-spent-listening.

The second is *non-linear distortion.* In a correctly designed processor, audible distortion is entirely determined by the amount of clipping or other very fast limiting process that is used for final peak limiting. The harder the final limiting process is driven the louder the sound but the higher the distortion. In general, it is unwise to have processing-induced distortion levels that are high enough to be audible on portable and auto radios. It is not unusual to have slight distortion audible on very high-quality receivers, although you should avoid this in any format designed for long-term listening.

A processor can sound fine on some program material, yet produce disturbing distortion on other material. Listen carefully for distortion on spectrally sparse material like piano, nylon-string acoustic guitar, and voice.

A poorly designed processor can introduce unnecessary distortion elsewhere in the chain. Listen carefully to pure low bass (typical of urban and rap recordings) for a low-level buzzy or clicky distortion. Improper smoothing of the gain-control signal in a compressor can cause this. Also, clipping distortion that seems highly correlated to the input signal level can be caused by headroom problems in the processing chain. In a DSP-based processor, make sure that the input A/D converter is never overdriven.

The third is *spectral balance.* Compare the on-air sound to the sound of recently produced major-label CDs. The spectral balance on-air should be similar from the upper midrange through the midbass. At very high frequencies, the limitations of the FM pre-emphasis curve (which is up 17dB at 15kHz in North America) may force the processor to reduce the high-frequency energy by comparison to the CD, which ordinarily is produced without pre-emphasis. This may be particularly true of contemporary CDs, because many current pop CDs are mastered very with bright balances.

The higher the on-air loudness, the more brightness you will have to sacrifice because high frequencies take up a great deal of modulation without delivering very much of that modulation at the receiver's loudspeaker due to receiver de-emphasis. Also, a very loud station will have to sacrifice low bass. This is because the ear is relatively insensitive to low bass by comparison to midrange energy, so there is little "room" for heavy bass in audio processed for maximum loudness.

The fourth is *dynamic distortion.* This includes classic *compressor pumping* caused by attack and release time constants that are not well matched to the program material. The resulting sound has a strained, unnatural quality. Also in this category is *spectral gain intermodulation*, caused by a dominant sound in one frequency range causing gain reduction that reduces the loudness of a second sound unnaturally. A typical example is heavy bass that modulates the loudness of midrange material in a wideband compressor. A further problem can be caused by "*clipper pumping*," where bass transients smash against the processor's final clipper, momentarily "shutting it down" and blocking other program material. This can sound like severe compressor pumping.

Any competent processor will have some sort of gating that freezes or slows the compressor release process to prevent the processor from pumping up low-level material or noise. The gate should operate unobtrusively, preventing these effects without introducing problems of its own. For example, in a multiband processor be sure that the gating does not cause the various bands to get "stuck" with widely varying gains so that the resulting frequency response unnaturally colors the low-level audio passing through the processor.

Technical Evaluation

This is easier than subjective evaluation because it is based on objective, repeatable measurements. The purpose is to be sure that the processor controls modulation correctly, protects subcarriers and the stereo pilot tone, and is compatible with the rest of the transmission plant. It requires a good modulation monitor (or digital storage scope) and a 0-100kHz spectrum analyzer, ideally with a "maximum peak hold" function.

Modulation control determines the uncertainty of the peak level at the processor's output. It should be measured with the processor configured for pre-emphasized output. Connect the input of the monitor or storage scope to the output of the processor that you will use in your installation. (If you are using a digital output, you may have to use your digital exciter's modulation meter.) Observe the modulation meter or scope over several minutes with a wide variety of program material and note how consistently peaks are constrained to 100% modulation. Different processors can show very different results on this test. Inconsistency of 5% or less is equivalent to approximately 0.4dB loudness loss, which is relatively insignificant. Inconsistency of 10% or greater will force you to reduce average modulation to the point of causing audible loudness loss.

Be sure that the monitoring device you use indicates dynamic modulation correctly and has no built-in overshoot. We use the Belar Wizard; there are other competent monitors available as well.

If you monitor off-air, you will be seeing the overall overshoot in your entire signal path and not just the overshoot at the audio processor's output. While minimizing transmission path overshoot is very important in achieving maximum loudness, we do not advise off-air monitoring for testing processors because you don't know the source of the overshoot. Further, if your FM exciter has a built-in overshoot limiter and this is active, the overall sound you hear off-air will include the coloration of the overshoot limiter and the results of the evaluation will then apply *only* to chains containing that exciter. Further, the exciter's overshoot limiter will mask the very real differences in peak modulation consistency among processors.

Spectral contamination is measured with a spectrum analyzer. If your processor has a composite baseband output, measure this so you can directly see any interference with subcarriers or the stereo pilot tone. If your processor has only left/right outputs, measure these. Some low-cost processors may not have built-in 15kHz lowpass filters. If this is so the spectrum measurements will clearly show it. Such processors must be used with stereo encoders with built-in lowpass filters, and special care must be taken to overshoot-compensate these filters.

Using program material, observe the output spectrum of the processor for at least 10 minutes, using the spectrum analyzer's "maximum peak hold" function. (This procedure is based on the recommended practice for verifying that NRSC-1 standards are met in AM.). If you are looking at the composite output, observe the area around the pilot tone for spectral contamination. Car radios with pilot-based variable blend detectors will read excessive contamination here as noise or multipath, and will cause them to prematurely blend towards mono. It will make the effects of multipath worse.

If you are using (or are planning to use) a RDS/RBDS subcarrier, look at 57kHz +/-2kHz to make sure that this part of the spectrum is protected. If you are using (or are planning to use) subcarriers in the 62kHz-100kHz region of the baseband, make sure that this area is protected as well.

If you are observing the left/right output, observe how quickly the spectrum falls off beyond 15kHz. Adequate RDS protection requires negligible spectrum above 17kHz. Pilot tone protection requires negligible spectrum above 18kHz.

If you are planning on using an uncompressed digital STL with a 32kHz sample rate, this requires negligible spectrum above 16kHz and a fast descent after 15kHz. Otherwise, the anti-aliasing filter in the STL will remove spectral energy and will cause overshoot. This can be a particular problem with older analog processors, although some digital designs may show it as well.

Interface

Analog interfacing is straightforward, although you must make sure that the signal path after the final peak control element in the processor is down 3dB at 0.15Hz or below to prevent tilt and bounce, which cause overshoot. Some processors have inadequate line amplifiers and will introduce tilt.

To preserve an all-digital path, the processor should have AES/EBU inputs and outputs. The sample rates accepted at the input and emitted at the output must be compatible with the rest of your equipment. If downstream equipment accepts AES status bits, to prevent surprises make sure that the processor implements these bits correctly.

32kHz is the ideal sample rate for a digital output because it is compatible with almost all digital exciters and STLs. A 32kHz output will pass transparently through a 32kHz digital link without overshoot. While a 38kHz rate can slightly increase frequency response above 16kHz compared to 32kHz, it has no benefit in a system that is bandlimited by choice to 15kHz and can cause overshoot in a 32kHz STL after sample rate conversion to the lower frequency. Any higher sample rate than 38kHz is of no benefit at all because the basic "sample rate" of the FM stereo system is 38kHz. The extra bandwidth that can be accommodated in a 44.1kHz or 48kHz data stream must therefore be filtered out anyway to avoid aliasing in the FM stereo transmission.

Summary

Proper evaluation of an audio processing system is hard work. It requires careful, long-term listening and rigorous measurement to ensure meaningful results. It requires a clear concept of the station's processing goals and the mental discipline to ignore marketing hype and spin. But it is crucial. Processing is one of the most important factors determining the overall impression that your station makes on its target audience. Make the wrong subjective choice and you can damage you chances to get that audience. Make the wrong *objective* choice and you can end up with a processor that is a nightmare to install in your plant that throws data errors into your RDS, that interferes with your subcarriers, and that reduces your coverage area because it interacts with the variable-blend circuitry in consumer receivers.

Good luck!