Using Multiple Audio Processors

Date: November 2015

Background

Questions often arise about what is the best way to process audio in a television facility. Answering these question is difficult because there is no single correct answer. The right way depends on management goals, technical requirements and the facility and transmission technologies. For instance, analog studio to transmitter links require processing to prevent over-deviation at the send side and to eliminate overshoots on the receive side. Analog transmission requires careful pre-emphasis and limiting to prevent over-deviation and, in some regions, strict MPX power limits must be controlled. Audio processing in end to end digital systems is somewhat easier to talk about. Overshoot and RF deviation are not controlled by the audio but are a property of the link itself and the bittream encoder. This Application Note will specifically address the use of multiple processors in a digital television broadcast facility. We will look at the audio consequences of using multiple processors in the audio air-chain.

Creative Processing vs. Air-Chain Processing

Audio processing is very much a part of the creative process. We consider it is an essential part of recording and presenting audio content. In a simple newscast each anchor microphone has gating to control noise on the set, EQ to help the microphones to sound natural, and perhaps, a touch of compression to even out levels. A compressor on the master bus to help maintain consistent levels and a limiter to protect against a mic being dropped or struck, are both common and should be considered essential. All of the above should be considered source processing and is a necessary and essential part of creating the sound of a program. Any audio presented, or recorded, live will benefit from the judicious use of gating, EQ, compression and limiting.

Prerecorded and post produced content may have layers of audio processing to achieve the desired sound of the finished product. Music, dialog, effects, and natural sound, are all mixed, and processed, to achieve the desired result. The use of audio processing in production is found in every kind of content. Examples include; simple public affairs shows, Hollywood movies, documentaries, TV dramas, promos, advertisements, and music videos.

Processing for delivery is the art and science of preparing content for delivery through the broadcast chain to a viewer. We should assume that the audio content delivered <u>to</u> the broadcast chain is already processed as the creator intended. The purpose of the final audio chain processing is to help ensure that all content will sound good to the viewer.

This brings us directly to the question what does "sound good to the viewer" mean? For digital television the simplest answer to this question is that audio levels should be controlled.

From EBU Recommendation R-128: *… peak normalisation of audio signals has led to considerable loudness differences between programmes and between broadcast channels;*
And *…the resulting loudness inconsistencies between programmes and between channels are the cause of the most viewer/listener complaints;*

The ATSC A/85 Recommended Practice for audio states: *Consumers do not expect large changes in audio loudness from program to interstitials and from channel to channel.*

So, the most basic requirement for delivery of television audio to the home can be expressed like this: Program to program, program to commercial advertisement, program to promotional item and channel to channel level changes should not be annoying to viewers. This is a very big and meaningful change from the requirements of audio processing for analog broadcasts.

Analog delivery requires audio processing to control audio bandwidth, carrier deviation, noise, and distortion. In a digital system the audio processing in the air chain can be all about providing great audio to the home viewer. The Linear Acoustic view of audio processing for digital delivery is that audio that sounds good in the home is naturally compliant. Our experience with broadcasters, all over the world, is that compliance with loudness regulations is (and should be) a consequence of making audio sound good for home viewers.

There are some countries with loudness regulations so restrictive that broadcasters are forced to make good sounding audio secondary to compliance. This is, fortunately, the exception and not the rule.

Digital delivery means that audio can be processed purely to meet viewer needs and expectations without concern for how processing affects the delivery channel. Linear Acoustic AEROMAX® processing was designed specifically to meet this challenge. Designed and built specifically to provide high quality audio to the viewer, Linear Acoustic® audio processors were used for 9 years before any loudness regulations were discussed or in place. An accurate representation of the original audio is presented in the viewer's home on the viewer's equipment with audio loudness properly controlled and in compliance with local requirements. Today, country after country is putting loudness regulations in place following the shift to digital broadcasting. Linear Acoustic 4th generation of AEROMAX processors continue to fill the need for very high quality audio delivery that also meets worldwide loudness regulations.

Having established the difference between production processing and processing in the air-chain we can look at using multiple processors in the delivery chain or air chain.

Air Chain Processing

It is very difficult to build an AGC amplifier that can operate over a very large range of input levels and also operate without any (or at least with few) audible artifacts. This task is even more difficult if the circuit must be entirely analog. As a result it became normal practice for engineers to place a dedicated AGC amplifier in front of a TV processor that already contained an AGC on the front end. The idea behind this practice is to use two AGC's with each one providing part of the required gain reduction. Each AGC was configured to do less gain reduction than a single AGC would be required to do by itself. This has long been done because the AGC, built into many processors, did not sound very good on their own.

Using two AGCs in series sometimes sounded better than a TV processor on its own. Sometimes. Often this arrangement did not sound very good at all. The biggest problem is that two independent AGCs could not be properly configured to operate together. If two AGC's are going to be used in series then the attack, release, ratios, thresholds and level detection methods must all be set to, and able to, operate in a complementary fashion. If the two AGCs are not chosen and configured properly then one AGC fights the other. One is increasing its output while the other is reducing its output. Likewise one AGC can be holding the levels low while the other AGC has returned to full gain. The results were often inconsistent. A loud sound in the program that cannot be adequately, or smoothly, controlled. Or, quiet dialogue that was reduced in level instead of being increased in level. One of the worst case situations is having a medium speed AGC changing in level with the input and then the change is amplified by the next AGC. The result is audio level that bounces up and down.

Time, great patience, and some luck, are all required to get this configuration to work with the widely varying content that most TV broadcasters handle day in and day out. Using two, or more processors in the air chain was, and remains, a difficult to manage solution to correct for inadequate processor design.

A legacy of success with the two processor approach may make some engineers continue to try this approach with new processors. However, Linear Acoustic digital audio processors are both intended to, and capable of, operating alone. Look ahead capabilities, coupled with clever analysis of the input signal, and real time adjustment of multiple processing parameters, allows AEROMAX processors to operate over an extremely wide input range and still provide a controlled and artifact free output.

The use of audio processors before or after a Linear Acoustic AEROMAX loudness controller is not recommended. Proper use of the unrivaled power and capability in any AEROMAX processor can handle the processing needs of any broadcast audio.

Exceptions to the Rule

Flexibility is the key when it comes to proper audio processing.

Multiple AGC's Inside AEROMAX

There are situations where the use of multiple AGC's can provide a benefit. One example is the presentation of live music. Sudden, fast rise time sounds, sudden level drops and audio peaks that are much higher than average levels all present unique problems for a single AGC. Another example is an attempt to create a wideband processor. In this case loud sounds, anywhere in the audio spectrum can modulate the loudness of dialogue and reduce the overall audio levels for extended periods of time. All of the AEROMAX processors in the current line-up, AERO.10/100/1000/2000, contain the processing capability to handle these difficult to control situations. Multiple AGCs within the AEROMAX processors are designed to work in complementary fashion. Using control side chains and other level and frequency controls, multiple AGCs can be properly configured and controlled to work together. This is a built-in feature of AEROMAX processing and is included in every program processor block.

Please note that there is no advantage, and all of the disadvantages, to installing an independent processor upstream or downstream of an AEROMAX processor.

Slow AGC and ITU Limiting

If a user wants to use a particular processor in front of their AERO™ processor the AERO can be configured with a slow AGC and ITU limiter. This configuration will allow an upstream air-chain processor to provide AGC level control, using wideband or multi-band processing, with the AERO maintaining very tight LKFS output level control. A very slow AGC setting in the AERO input AGC will not adversely affect the audio control of the previous processor.

Independent Processing for Live News

It is becoming increasingly common for television stations to give audio control of news programs to automation systems. Microphone processing and levels are preset and the automation system simply turns microphone channels on and off based on the camera shot. Whatever else may be said for this technique the result is audio that is not as consistent or dense as the mix done by live operators. Many stations that frequently switch between a network newscast, mixed by live operators, and their local automated mix find the contrast in the audio programs is obvious and not pleasing.

The AERO.lite®, or its replacement the AERO.10®, can be used on the studio output. Properly adjusted the processor will improve the consistency, increase the density and correct the level of the local news studio audio. The result is audio that is a much better match to network audio mixed by live operators. The success of this approach comes from the AEROMAX processor. This is not a simple gain reduction based compressor limiter. AEROMAX increases low levels and reduces high levels in a very natural manner. It sounds like a live operator is in control. It can, in fact, compensate for an inexperienced audio mixer. In this use the processor should be considered as a program processor and not an air chain processor. Only the news broadcasts pass through it.

The Future

We can no longer assume that broadcast delivery means terrestrial broadcast to the home viewer. Terrestrial, satellite, Internet and cell phone/wi-fi are, now, all delivery methods. Thin panel displays, handhelds, laptops, home theater, ear buds and headphones are all consumer playback platforms. Immersive audio, multiple languages, regional commentary, audio for the sight impaired, and emergency messages are all examples of the growing number of audio content types. Audio processing in the air chain will have to become very flexible in order to deliver audio that sounds great, and meets consumer expectation, on the ever growing range of consumer devices and viewer environments.

Linear Acoustic Intelligent Dynamics® is metadata for the future of broadcast audio control. Audio processing devices capable of reading and writing this metadata will be able to apply a specified amount of dynamic range control or simply pass audio that is already in compliance with local regulations. Maintaining quality audio and compliance, throughout the broadcast chain, will be less intrusive and more automated by the use of metadata. Linear Acoustic Intelligent Dynamics will be critical to the delivery of audio to the wide range of delivery channels and consumer receiving devices that are coming. Comprehensive, sophisticated and flexible audio processing is the key to the future.