History of broadcast audio processing

BY DAN ROACH

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In the beginning, there was audio… and transmitters had a lot of trouble with it! Audio levels varied all over the place, particularly with the large amounts of live broadcasting done "back in the day". And transmitters, especially AM transmitters, really don't like that.

Bell Labs responded by developing the ubiquitous VU meter, still with us after almost 90 years. Broadcasters strove to make various devices to control audio, with varying degrees of success.

Somebody noticed that some announcers' voices display a remarkable amount of asymmetry. In an age when broadcasting was inherently symmetrical, this could have been a job liability for announcers, but instead Leonard Kahn invented the Symmetra-Peak©, which smoothed out audio and made the positives and the negatives equal but opposite. Len also started the long tradition of dipping chunks of his invention in potting compound (to keep them from prying eyes, and maybe to add an impressive heft to his product) a practice that lives on to this day in audio processing.

CBS Labs finally solved the level control problem for all intents and purposes, with the two-box "Max" twins: the Audimax© gain-rider for the studio, and the Volumax© peak limiter at the transmitter site. The year was 1975, and our problems were all essentially solved… or so it seemed.

Robert Orban took a look at the FM program chain, and discovered there was a great deal to be gained by combining the low-pass filters, the audio limiters and the stereo generator into one box, which he called the Optimod©. He split the audio into two bands to better deal with pre-emphasis loudness issues. No longer would excessive high frequency content cause overall levels to drop.

AM broadcasting became asymmetrical, and it became legal to modulate 125% positive, but only 100% negative. Volumax© solved this by adding a peak detector and a relay to reverse polarity and make sure the big peak was always on top. It was time to torch the Symmetra-Peak©, and hire back all those out-of-work asymmetrical announcers, and maybe contemplate surgery for the now-unfortunate symmetrical ones to make them louder on the radio.

Next came Mike Dorrough. He had the brainwave of splitting the audio into frequency bands, processing each separately and then joining 'em together again. All of a sudden, everything got a lot louder, and brighter, and better—if we could just figure out what to do with all those extra controls on his Discriminate Audio Processor: the DAP©.

Mike started with three bands, but before you know it, others had as many as 10 or 12, and things got a little out of hand. But if the processing was adjusted properly, a bass drum couldn't "punch a hole" in the audio anymore.

Not content to take advantage of natural asymmetry in audio, Circuit Research Labs put phase scramblers back in the front end of their processors (the Symmetra-Peak© rides again!), and added adjustable asymmetrical clippers at the output. Again, this made everything a wee bit louder.

Texar introduced the Audio Prism©, which introduced a gated "dead band" into audio compression… instead of continuously raising and lowering gain around a threshold, the Texar had a *neutral* zone for each frequency band, allowing us to cling to existing levels until they were out of range.

On the FM side, Eric Small of Modulation Sciences started clipping the *output* of the stereo generator to create even more loudness. Others tried to copy his composite clipping approach, perhaps with a little less attention to what was happening to the stereo pilot, and all hell broke loose for a while. Eventually, it was learned that you had to do your clipping first and add the pilot later.

Just as we had reached what we thought was the pinnacle of audio processing, digital technology came along and set everything on its ear again. Now, we have latency, aliasing, sample rate and dithering to consider as well. And if there's any bit-reduction along the way, make room for psycho-acoustic masking, noise-shaping and of course, more latency.