SOUND PROCESSING

A History of Audio Processing

by Jim Somich
with Barry Mishkind

[September 2009] Before his untimely death, Jim Somich and I had a number of conversations by phone and email as we discussed the history of broadcast audio processing and laid the basis for this article.

Both of us had been involved in the production of some of the processors that gave radio a loud, clean voice in the 70s and 80s, and we had watched the changes over the years, from simple levelers to the microprocessor-driven digital processors of today. While this article covers a lot of interesting history, it also takes a peek at the current state of the art – and where we may be headed. Jim Somich took the lead on this guided tour. I get to help finish it off as a tribute to a great radio engineer.

The point of audio processing is change the program line audio into something that will modulate the transmitter in the best possible way: loud and clean. Way back when, audio processing was a manual affair: someone sat at a console with the sole job of keeping the audio level – and preventing the program audio from constantly knocking the station off the air.

When audio processing gear first appeared in the 1950s, its purpose was to automate the transmitter protection. Over time, improvements raised the average levels transmitted dramatically – and the life-span of the modulation transformers in 1930s and 1940s era transmitters was often short. At the peak of the Modulation Wars in the 1980s, audio dynamic range was almost eliminated and TSLs started to drop dramatically.

My prediction is that the processing business will change radically over the next few years - and in a decade you will not even recognize it!

THE PURPOSE OF THIS HISTORY

The importance of the arcane field of broadcast audio processing is evident by the extensive work done in the field since the earliest days. From PROGARS through Uni-Levels and Sta-Levels into the Audimax years, processing was evolving into an artform to be mastered by a select adventuresome few.

But this is not merely an historical discussion. To be prepared for the acutely competitive future, we are better off when we know a little about the past and a lot about the present.

Our goal with this discussion is to remember and review the past, take a lingering look at the present, and provide a glimpse into the future of broadcast audio processing - written by someone who has "been there, done that."

Today, the audio processor is, in many ways, the heart of a modern broadcast station. It takes whatever audio input it gets and turns it into a consistent "sound signature." In many cases, it is possible to visit a market and immediately identify who owns or set up the processing at stations just by listening.

One of the reasons this is possible is that virtually every current audio processor is now controlled by one or more microprocessors. From wide-band leveling to multi-channel, stereo-generating, look-ahead limiting, and diversity delay capable, the user has a tremendous set of tools at hand to craft advances in sound processing.

THE GOLDEN EARS

One thing is for sure: we do not know everything as yet, regarding the there will be new players on the scene along with many of today's superstars. I, for one, cannot wait to hear what they accomplish. While some actively attempt to re-create the sound of the tube processors, the leading designers already are looking ahead to new horizons of audio control.

The names are legendary and relatively few: Emil Torick, Mike Dorrough, Bob Orban, Greg Ogonowski, Eric Small, Ron Jones, Steve Hnat, Donn Werbach, Glen Clark, Jim Wood, and Frank Foti. Each had a vision of the way radio should sound and made it happen. What a heritage: Audimaxes, DAPs, Optimods, Aphexes, and Omnias!

Through the year, the men pioneered more and more sophisticated and creative ways to process audio and snag listeners. Just like the record producers, each had their own ideas on how to build the best sounding radio stations. Some opted for a wide, open dynamic range, while others looked to enhance the "wall of sound" format popular among the record producers. Inevitably, a few simply wanted to be loud and turned their processsor up to "11." The controversial world of modern audio processing was born.

ADAPT, IMPROVE, OVERCOME

All through the years, many exceptional stations engineers continued to find ways to overcome the technological limitations of the time by modifying boxes to perform tricks never anticipated by their developers. Another group went further and designed their own processors from the ground up. Many of these custom boxes were literally built in garages and sold to the industry.

As you will see, successful audio processing has not always been the province of large companies or groups of engineers. In most cases, it was a lone engineer with a vision who struck out on his own to capture his aural imagination in his own "magic box."

It is not as easy to hotrod a DSP processor today the way one used to be able to change some component values in an Optimod or a DAP. But I do know one thing for sure: as I said before, there will be a new generation of processing gurus and they will have new ideas. They will do things with their boxes that we have not even dreamed about as yet.

Who will be these gurus of tomorrow? And what will they have to work with? They are already out there, working in the trenches. Guys like Scott Incz, Corny Gould, and John Burnill have dreams that just might come true.

ENTERING THE PROCESSING TIME MACHINE: YESTERDAY

As mentioned earlier, in the beginning there was no audio processing. AM radio stations used the technique of “manual gain riding” to avoid over-modulation - a *real live person* sat with his hand on the knob, trying to anticipate what was to come next. The result was low modulation levels– perhaps 30% on average.

In some ways, the practice was reasonably successful, but hardly efficient. Failure to properly anticipate just one spike in the audio level could result in the transmitter overloading and dropping off the air – or worse.

Even the huge 500 kilowatt transmitter at WLW operated in the mid-1930s with manual control to prevent over-modulation. If you are able to see some of the transmitter logs from that time, you will see they have many descriptions of outages caused by modulation peaks. Most were brief, but some notes indicated blown-up capacitors, tube failures, and other problems that took longer to repair.

FROM MANUAL TO AUTOMATIC

Gain riding really was an art, and it was practiced diligently by the studio engineers of the 1930s, 1940s, and 1950s. When I started at WGAR, in 1959, almost all gain control still was done manually. Fortunately, by that time, we had a GE BA-5 peak limiter at the transmitter for over-modulation protecttion. It was not quite a brick wall, but it really helped. But since one of the primary duties of a studio “engineer” was to ride the gain, there was not a compressor in the entire studio plant.

Talk about pressure: the chief engineer at WGAR had installed a chart recorder in Master Control to make a permanent record of the program level going to the transmitter every minute of every day. Each morning, one of his first stops was at the chart to check up on the gain riding of the engineering staff during the past 24 hours. Each engineer was held completely accountable for his shift.

However, with the advent of post-Television radio broadcasting, with its combo operation, fast-paced shows, short jingles, and other multiple elements, the need for an automatic form of gain riding became inescapable.

EARLY PEAK LIMITERS FOR AM

It was the mid-late 1930s when the first peak limiters came to market.

RCA introduced their model 96A in 1936, perhaps the very first commercial peak limiter to hit the market. Just three years later, in 1939, Western Electric introduced the 1126A.

But you could hardly call these boat anchors audio processors. They were basically mundane tools to eliminate over-modulation, pure and simple.

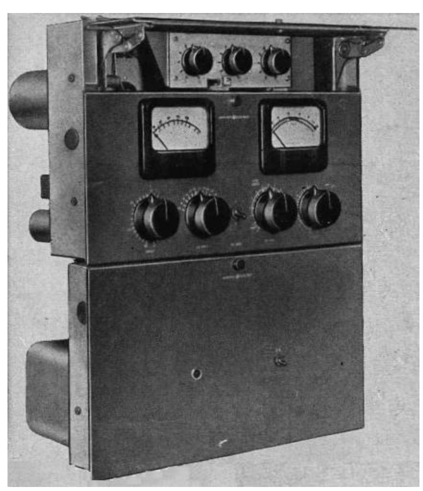
The PROGAR (PROgram GuARdian), developed by Al Towne at KSFO, San Francisco in 1935 really was the first known actual audio *processor:* a combined, intelligent compressor (automatic gain control plus AGC) and peak limiter. But it took more than ten years for Towne to patent it, sell it to Langevin and bring it to market.

The PROGAR brought automatic gain control and peaking limiting together. (We are still looking for a better picture of the PROGAR Can you help?)

Then something exciting happened. Peak limiting became much more sophisticated. Anyone with a processing background would have to agree that it was ahead of its time – to the extent that it would take decades before another product would match its key feature.

DELAYING THE PEAKS

When General Electric introduced the BA-5 delay-line peak limiter in 1947, it took the broadcast industry by storm. Its feed-forward limiting scheme used a delay line to "give the audio a change to catch up to the bias generator." The result was cleaner audio than anything that had come before it.



The GE **BA-5**

How clean was the BA-5? Back in those days, there was an iron-clad rule at NBC to use only equipment manufactured by parent company RCA. Yet, even NBC bought BA-5s - removing any evidence of it being a GE product by repainting them in the "standard RCA umber gray" – and adding the RCA meatball.

Magically, the new "RCA Peak Limiter" was born! Very rapidly, every NBC Owned and Operated radio station began sounding much better thanks to the "midnight engineering."

IMPROVING THE TECHNOLOGY

Meanwhile, GE was already working on the next step. Research and innovation helped them continue their dominance of the peak limiter market with the introduction of the BA-6 in the early 1950s and the BA-7 in 1957. These boxes were truly unique.

To summarize why these limiters were so good is easy: GE used the input audio to modulate an RF carrier - and then all peak limiting was applied to this carrier. After demodulation, the audio was fed to the transmitter.

Many processing artifacts were eliminated by this scheme, but as with many of the early processors, it was an absolute bear to keep in alignment – taking two engineers or one bodybuilder to wrestle one of the processors into a rack!

DIFFERING ATTITUDES

Ironically, throughout the 1950s, while the big AM stations were getting louder and louder, many FM stations eschewed audio processing entirely. I even remember one old-timer studio operator telling me that *"you really couldn't overmodulate the FM."*

As time went on, some FM stations began to install a Fairchild Conax pre-emphasized clipper to tame the pre-emphasis. But that was about it. In fact, it was not at all unusual in those days to watch the modulation monitor "pin" on muted trumpets even when using a (conventional) peak limiter. There was quite a way yet to go in developing effective FM processors.

This divergent attitude toward processors was to continue for a while. As important as a good limiter was in preventing overmodulation, the focus of the audio processing developers was to make the station louder.

BUILDING EFFECTIVE COMPRESSORS

For them, the most important part of a compressor or peak limiter was the gain-control element.

During the early years, this function was usually performed by a tube. The PROGAR used a 6L7 heptode tube. (The 6L7 was designed as a variable mu (amplification) tube – that was the purpose of the extra grid.)

The 6L7 came in a metallic style and the 6L7G glass envelope

All tube compressors and limiters functioned by mixing a DC control voltage with the audio at the grid of a variable mu tube. These amplifiers used push-pull operation so the control voltage could be effectively canceled at the output. This reduced the "thumps" that were common when these boxes got out of balance due to tube aging.

The Gates Level Devil (M5546A) added a level-dependent expansion gate that released about 10 dB of expansion when the input level was above "noise level." Unfortunately, this gate did not work very well and resulted in a lot of "sucking and wheezing." However, those were the humble roots of intelligent audio processing.

The Gates Level Devil

Jim Tonne recalls the Level Devil "… used a pair of 6BA6s. The expander sounded poor because it was reverse-acting. You had to reach an input signal level corresponding to 10 dB of compression before the expander would open up - at which time you suddenly had 10 dB of compression. If it would have been forward-acting it would have sounded much better."

THE 6386 RULES!

There were a few remote cutoff tubes designed before the GE-6386, but this tube became the rock star of the 50s in audio processing. It was the basis of the GE Uni-Level, Gates Sta-Level, and the CBS Audimax. The 6386 was a remote cutoff dual-triode, which made it ideally suited to push-pull gain-control operation.

A 6386 dual-triode tube

A remote-cutoff tube has a grid that is wound in a nonlinear fashion and this gives the tube the unique characteristic of reducing its mu with increased signal levels. This was a valuable characteristic in a compressor or limiter. Conventional sharp-cutoff tubes tended to operate with substantially more distortion and artifacts.

The Gates Sta-Level was a straightforward compressor, using the 6386 as a gain control element. A 6AL5 dual-diode was used to rectify a sample of the output from a pair of 6V6 tubes operating push-pull. This DC control voltage was fed, via an R/C time constant network to the grids of the 6386 tube. The circuitry was very similar to the GE Uni-Level, which preceded the Sta-Level by a year or two.

The Gates Sta-Level, a solid performer, was a common sight in stations well into the 1970s.

A CLEVER MARKETING SCHEME

CBS Laboratories introduced the Audimax I in 1959. Designed by Emil Torick – and marketed specifically as a "gain rider," or compressor – the Audimax I made no pretense to being an "audio processor." Yet, it was the unique design of the Audimax – and its marketing plan – that ushered in the era of audio processing.

The Audimax I
A straight-ahead audio compressor

The Audimax I was also the first broadcast audio processor to be sold on a 30-day trial basis. A broadcaster could submit a purchase order for a unit and put it on the air for a month. If they were not happy with the sound they could return it at the end of the trial period with no questions asked.

The original price was something around a kilobuck (in 1959 dollars). While I am sure CBS got a few of the Audimax's back, there was no doubt but that most of those who gave it a try became true believers in the Audimax concept – and the vast majority of users were quite satisfied with this box.

THE MAXX BROTHERS

As with the Uni-Level and Sta-Level, the Audimax used the 6386 dual-triode to control the audio gain, but some enhancements were made. For example, a "platform mode" kept the Audimax gain constant over a 6 dB gain platform. This resulted in a lot less "busy-ness" in the sound. When the input audio moved outside the current platform range, the system gain was quickly readjusted to define a new platform.

The Audimax II version quickly followed, adding an adjustable noise gate that froze the gain when input level fell below a user adjusted threshold. The next model in the line was the Audimax II-RZ, which featured a "return to zero" function that did exactly what you might think: when the input audio was below the threshold of the gate, it slowly returned the system gain to zero.

The Audimax really started the era of aggressive processing. Over the years, it transitioned into solid-state versions, and later added a biased-diode peak limiter called the Volumax. In its later years, Thompsen acquired the Audimax line and produced a thin, one rack-unit version of the Audimax and Volumax.

There were several modifications applied by engineers who just could not accept the parameters that were fixed in the units. Most of these were attempts to speed up the release action, but there were many others. It seemed like every creative engineer had his own set of Audimax and Volumax tweaks.

The Volumax provided a limiter to complement the Audimax

The development of the Audimax was indeed a mega-event in the history of audio processing for broadcast. The "Maxx Brothers," Audimax and Volumax, ruled for a decade – and continued to be in demand for another ten years after that! Truly a remarkable record.

A SYSTEMIC APPROACH

In the early 1960s, General Electric and CBS Laboratories ruled the roost when it came to state-of-the-art audio processing. But their success spurred the imaginations of many engineers who thought they could see a better way to create a distinctive sound on the radio.

The Audimax 4440/Volumax 4000 package

Among them was George Frese, a consulting engineer in Washington State when the FCC mandated that stations do yearly Equipment Performance Measurements. Frese began doing EPMs, eventually doing as many as 50 each year. This gave him an opportunity to analyze many different stations with their varied combinations of transmitters and audio processing. Additionally his work as a 1st Violin, 2nd Chair in a symphonic orchestra gave him a experience with clean audio and how things should sound.

From all this Frese began to see clearly what the limitations in the audio and transmission system were and how they could be overcome by properly designed processors. His work – giving attention to the entire audio and transmission chain - would soon change the whole concept of “loud” – and sell a lot of replacement transformers.

UNDERSTANDING ASYMMETRY

Frese was inspired by the request of a client – a 5 kW station – to explain why a 250 Watt station in the same small market sounded better and noticeably louder than they did.

Both stations were basically fully modulated. However, after looking through and analyzing the audio chain at the 250 Watt station, Frese realized how important it was to have both good frequency response and correct audio symmetry. Lack of either would rob a station of loudness.

Although a few engineers had already noticed the natural asymmetry of the human voice, most did not know how to take advantage of it. Often station audio could be seen as a mixture of waveforms with the highest peaks on the negative side, forcing the processors of the time to react to the wrong peak.

That, Frese decided, robbed stations of significant potential audio power.

SOLVING THE PUZZLE

One popular approach at the time was to use products like the Kahn SymmetraPeak to create an effect not unlike a long phone line, which tended to average out the positive and negative peaks and bring them close in level.

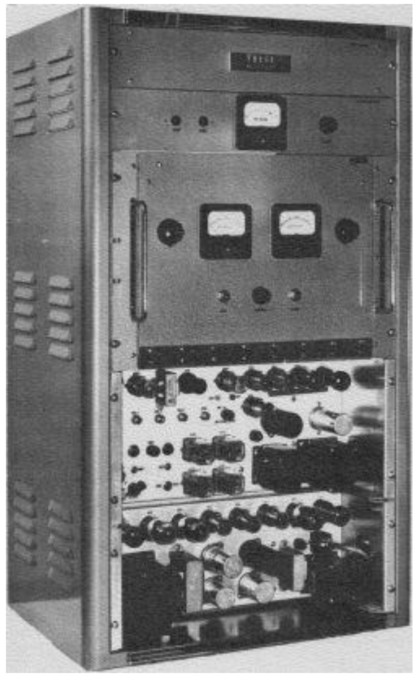
That solution worked to an extent, but in making most audio symmetrical, it lost the potential advantage of positive peak modulation. Adding devices that artificially increased positive modulation did increase peaks to a degree, but also usually introduced audible distortion. A different solution, automatic phase flippers to keep the highest peak going positive, usually caused audible problems, too.

Over time, Frese developed a list of nine things that would make a station sound better. Each of these aspects made only an incremental .5 to 1 dB difference, but all together they would make a station stand out on the dial. These nine improvements came to be the key features of the processor he would build: The Audio Pilot.

THE FRESE AUDIO PILOT

According to Frese, the Audio Pilot got its name from the original design. At first there were two signal channels, the audio channel and a “pilot” (or, control) channel. The plan was to use an audio delay line so the pilot channel could detect and control the audio peaks without excessive clipping.

However, Frese found that “the audio delay line wrecked the symmetry.” So his design was modified to use only one audio path, controlled by the processing system. The product name remained.



The Frese Audio Pilot

The result was an audio processor that was instantly recognizable, as soon as someone tuned past it. In fact, when operating on a transmitter with sufficient modulation capacity, a station running the Audio Pilot could hit 200% without sounding bad - and you can be certain *that* got the attention of engineers, program directors and station managers.

A DISTINCTIVE SELLING STRATEGY

Frese had a unique sales technique. He did not just send the processors out to stations. “My procedure was to go to the stations and make the installation myself at no charge to the customer,” Frese says.

By making the installs himself, he ensured each of his nine points were checked, so the audio would be as clean as possible before and after the Audio Pilot. Frese noted “… the audio always performed more than I expected by sounding better than what they were using and covered more additional area than even I had expected.”

Once the installation was complete: “Then I would stay in the city one day and let them listen to it. The following day The Question was: ‘do you want me to leave it in, or shall I take it out?’ If I took it out there was no further charge. If they kept it, I asked for $2,500 before I left. I lost just one sale …”

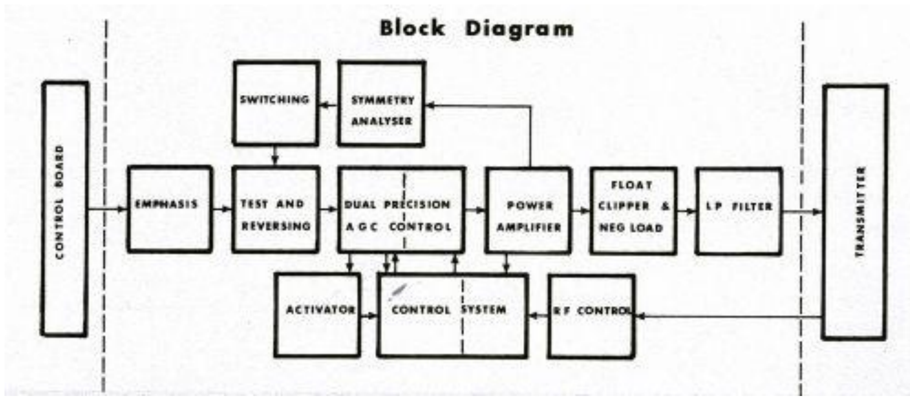
Eventually 41 units were sold. Frese recalls: “I installed Audio Pilots on just about every make and power of transmitters there were.” They were installed from coast to coast and even in Mexico at the famous XELO, a 150 kW flamethrower across the border from El Paso, TX. “Each Pilot went into a different town … but everywhere I went, I met with surprising success at the results.”

A DISTINCTIVE SOUND

As with most good products, Frese kept refining the operation. “With each one I installed I experienced something new that I hadn’t known before that they would do,” he said.

When set up, most of the control by the Audio Pilot was on the negative peaks. The processor adjusted modulation by reference to the RF carrier level, letting the positive peaks run through while it held the negatives to 99%. This permitted it to work correctly at any power level, even when the transmitter was cut back significantly for night time operation. Similarly, power line fluctuations had no effect on the modulation level.

The Audio Pilot was conceived to handle a wide range of input. Frese said: “you could leave the console alone and Audio Pilot would ride gain.” The specification in the brochure and spec sheet is impressive: “Output level is constant with input signal of -40 dBm to +5 dBm.” Such control, while maintaining an apparently wide dynamic range, was a hallmark of the way the Audio Pilot’s control circuits handled the audio.



The Audio Pilot did not have the fastest recovery time, but that was on purpose. Rather the custom rate of recovery, even when the dynamic range was virtually 0 dB, gave the listener then impression that there was a much greater dynamic range.

Looking the block diagram shows another key feature to the processor: the “Floating Clipper.”

Since the Audio Pilot was designed to enhance positive peaks, it had a clipper designed to automatically adjust to the program level. It would “float” on the most recent peak, slowly dropping its activation point to the next-highest peak. When a sudden, higher, sharp peak occurred, the clipper would grab it and clip, instantaneously moving the clip point upward so modulation was maximized, protecting the transmitter, yet reducing the need for heavy clipping. Distortion figures typically were under 1%

The History of Audio Processing Continues with PART 2: MULTIBAND AUDIO

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