RADIOWORLD

TRENDS IN AUDIO PROCESSING FOR RADIO

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FUTURE

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Trends in Audio Processing for Radio

Paul McLane
Editor in Chief

Radio processing is like a passionate lover. It can be sexy and mysterious and tempt you to put your hands on it more often than might be good for you. But if not handled "just so," it can become really difficult. Also, while everyone loves to talk about it, few want to surrender the details of their own special dalliances with it.

And sometimes, again like a lover, processing just wants to be left alone.

Forgive my stream of simile, but processing does generate a unique kind of passion among radio people.

Broadcasters now prepare and deliver content over numerous platforms, to listeners in numerous environments including far beyond their local markets.

We wanted to learn what users, leaders and manufacturers consider to be the most important recent or pending developments in design of processors for radio's needs; how processing differs for those various platforms; and how the cloud, virtualization and software as a service affect the processing marketplace.

Tom Lawler, Matt Levin and Mike Cooney provide engineering user perspectives, with Cooney adding insight from his work with the NAB Radio Technology Committee. David Bialik and John Kean offer opinions about issues around streaming and loudness, including work being done by the Audio Engineering Society on a document for online audio parameters.

Then manufacturers including our ebook sponsors weigh in; we hear from processing gurus at Wheatstone, The Telos Alliance, Orban, Inovonics, Circuit Research Labs and WorldCast Systems.

And consultant Gary Kline wraps up our discussion with his trademark list of "things to think about."

I welcome your input on this or any ebook. Email me at radioworld@futurenet.com.

Cover image: Getty Images/Filo

Listening Has Come Almost Full Circle

Like small 1950s mono speakers, today's smart devices are fueling an audio revolution

Tom Lawler is a contract studio/RF engineer who builds and maintains analog and AoIP radio and home studios for broadcast; his full-time job is in field technical services for RCS.

Tom Lawler

Radio World: *Tom, what do you see as the most important trend in the design and use of processors?*

Tom Lawler: With the development by leaps and bounds in flexibility — between insert patch points for ratings watermarking, multi-mode boxes, as well as being able to do MPX over AES or AoIP — modern processors have become virtual Swiss Army knives. Not to mention that devices like StreamBlade from Wheatstone or software processors like StereoTool let your online presence have just as much punch as the OTA signal.

It wasn't that many years ago where the only option was to try and adapt an FM box or use a PCI card that couldn't be easily updated.

> *"If this is how your audience consumes the station/stream/podcast, make sure to give them a download or on-demand stream that is easy to listen to no matter the environment. Make the most of the 3-inch speaker without sounding smashed."*

RW: *What should readers know about the differences in processing needs for analog over the air, digital OTA, podcasts and streaming?*

Lawler: Every medium requires a different approach, but they all require you to have as clean a source material as possible.

With analog OTA you can get away with clipping/limiting to achieve loudness without introducing fatiguing artifacts — but that approach won't work with digital OTA as artifacts will cause issues with the bit-reduced codec. For podcasting, use a gentle multiband to smooth over transitions between segments/presenters — resist the urge to treat it like FM!

Streaming can be treated like FM, but make sure to use lookahead limiting instead of clipping — also, make sure everything is in-phase for when it's folded down to mono on a smart speaker.

RW: *How will the concepts of the cloud, virtualization and software as a service affect the processing marketplace?*

Lawler: I hope that it will lead to greater flexibility, redundancy and better quality. Imagine no STL issues to contend with (as long as your ISPs are up), and you now have the ability to make upgrades with the click of a mouse rather than having to physically rack up gear. This is a great opportunity for users as well as vendors alike — users gain as much flexibility as they are willing to pay for, and vendors can have a reliable subscription income stream. Plus, there is no single point of failure like in a traditional plant.

RW: *The pandemic is changing thinking about the need for big buildings and studios to make good radio. What does this mean for processing?*

Lawler: I think COVID-19 will accelerate moving to flexible software/cloud solutions for processing — and it will be more important than ever with the myriad of sources and level differences.

Given that more and more talent is working remotely from home rather than at the studio it will be a challenge to keep everything sounding consistent from source to source. Not every home studio has mic processing, and many automation systems do not handle ducking gracefully.

Adding processing in the cloud will be necessary to keep the audio consistent — more so now than before. This also means less in the racks to power and cool if done right.

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Audio Matters and Methods

George Lucas once said that sound is 50 percent of the movie-going experience. That certainly leaves little doubt to the importance of audio for those of you in broadcasting who don't have CGI and visual theatrics at your beck and call.

Wheatstone's processing guru Jeff Keith at work in his lab at Club Wheat.

It's why Wheatstone has an entire product line dedicated to creating bottomless lows, crisper highs, and warm, engaging vocals, both on-air and online. We have processors uniquely designed for miking, streaming, and for AM and FM. Processing is built into our AoIP I/O units, and we have add-on products for getting that perfect sound across links, around multipath, and through ratings watermarks.

Our processors share key building blocks of technology, algorithms and sound engineering that we've developed, and continue to develop, in the pursuit of good broadcast.

LimitLess clipper: High frequency pre-emphasis has always been the bane of the FM processor's existence; one of the main reasons why processors can't push the clipper harder is because of the pre-emphasized higher end frequencies. So we re-thought the relationship between peak control and HF pre-emphasis. Our LimitLess clipper for the X5 FM/HD audio processor uses proprietary high-frequency distortion cancelling technology to pass the highs, without the associated "spittiness," pops or other IM distortion from clipping.

Adaptive multiband AGC: You'll find iAGC in most of our processors as it's an effective first line of defense for the wide level variations found in today's source music. We call this our intelligent AGC because to be truly effective today, an AGC needs to make decisions about both the amplitude and dynamics of source material. If a particular cut is too dense, the iAGC knows to relax the processing so that dense material doesn't sound "double processed." If a song needs more punch, the iAGC knows to make real time adjustments to "program match" it to yield a consistent audio signature. Furthermore, by coupling the iAGC with compression or limiting, we are able to produce a much more consistent, spectrally-balanced sound regardless of density variations. Read on.

Multiband spectral compression: Since the dawn of the multiband AGC/Compressor/Limiter, the goal has been to improve tonal consistency and increase loudness by making algorithms smarter and, in turn, making the effects of multiband control less audible. Our iAGC is the first stage in this process. The iAGC uses a very smart algorithm and we collect a lot of data from it, as we mentioned earlier. With this, we can modify the processing behavior based on the dynamics and amplitude characteristics of the incoming program. But what if we could also use that data to affect compression and limiting down the chain? In fact, we do. We use the data from the iAGC to dynamically adjust multiband compression to yield a much better tonal balance, cut to cut, without it sounding over-equalized, artificial or "boxed in."

Unified processing: For many of our later model processors, functions in the processing chain interact closely with each other so they can be informed by and react according to what each is doing. Sharing information between the various stages of audio processing reduces 'over-processing.'

Specific tools for streaming: Streaming lives by a different set of processing rules. Codecs used for streaming content can multiply the byproducts of aggressive clipping and limiting, which is why we use a special AGC and limiter in our StreamBlade to get a fuller sound through encoders.

WHEATSTONE PROCESSORS

StreamBlade. Combining AoIP, audio processing and codecs for streaming.

This WheatNet-IP audio appliance is designed specifically to play to the psychoacoustical characteristics of lossy codecs to get a fuller sound, crisper highs and deeper bass out of streamed content. StreamBlade accepts eight input sources of native WheatNet-IP audio directly from a soundcard or AoIP driver, each capable of four outputs for a total of 32 output streams.

Meet X5, Wheatstone's flagship FM and HD audio processor

This is one serious FM/HD audio processor! X5 has all the latest in limiting, clipping, stereo enhance, and mitigation tools. If you're up against serious on-air competition, this is the processor. It includes every advantage we can think of and has advanced but intuitive onscreen tools that make it easy to program.

Four-Channel Mic Processor on the AoIP network

The M4IP-USB mic processor combines four high-quality microphone preamps, four channels of mic processing, four independent USB ports, and a WheatNet-IP BLADE interface for mic processing anywhere in the WheatNet-IP network.

WHEATSTONE PROCESSING OPTIONS

MPX SyncLink for extending HD/FM signal alignment

You can keep the processor at the studio and still keep HD and FM audio in perfect alignment at the transmitter with our optional MPX SyncLink receiver. MPX SyncLink is compatible with every existing FM transmitter in the field, unlike other HD/FM time alignment options.

SG-192 FM Stereo Generator for repeaters and SFNs

The SG-192 is the first standalone FM stereo generator capable of passing full AES MPX composite baseband to the exciter, including RDS and SCA, and is equipped with an intelligent stereo multipath controller that helps mitigate the effects of multipath-induced receiver blending.

Bass you can feel: We love deep, bottomless bass. But we also love crisp, clear highs, and we have found a way to get both. The result is increased depth, feel, and clarity of bass, without affecting mid and high frequencies.

Watermark tip-in: This is for inserting the ratings encoder after the processing chain instead of before it. This increases the likelihood of ratings meters picking up the signal without audibly interfering with the listener experience.

Total Linear Phase Chain: Our X5 FM/HD audio processor uses linear phase filtering exclusively throughout the chain to increase energy and dynamics without adding the ugly coloration of phase distortion that fatigues listeners. With coloration no longer a byproduct of filtering, dynamics and depth are increased for a more natural sounding program.

Super Quiet preamps for mic processing: For our mic processing line we use Super Quiet preamplifiers that have an extremely low noise floor, very wide dynamic range, faithfully accurate transient response, and ruler flat frequency response.

Baseband 192: Our FM audio processors include baseband192 technology for direct AES/EBU output into any FM transmitter equipped with a digital baseband input. Baseband192 digitizes the entire multiplex spectrum, including RDS and SCAs, clearing the last obstacle to a 100% digital air chain.

HD/FM alignment: Integrated HD and FM analog signal alignment keeps listeners and people meters tuned in to your station, even during extreme HD/FM blending conditions.

Multipath mitigation: We use adaptive stereo width management to reduce the multipath blending in car stereos and to give the listener a predictable soundstage.

AoIP connectivity/routing: Many of our audio processing products have a WheatNet-IP interface. In addition, our WheatNet-IP audio network I/O BLADEs have AGC, limiting and EQ dynamics built in, making these processing tools available at every I/O connection point in the AoIP network.

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Processors Tasked With Repairing Damage

A view from the field with veteran engineer Matt Levin

Matt Levin is chief engineer for River Radio in Columbus, Ohio, and does contract engineering for several stations.

Radio World: *We're asking users and manufacturers for their take on key trends.*

Matt Levin: I think the biggest development in processing is the shift from conventional dedicated hardware boxes to software that can run on a server with an alternative method for the MPX audio to get to the transmitter.

By shifting to software, it allows you to do your processing on your own server hardware, either on a physical box or in a virtual machine, or in the cloud via hosted services. Virtualization is the direction pure IT infrastructure went years ago and now the radio industry is finally embracing this concept from automation vendors to now processing vendors.

One of the keys to allowing this to work fully was the invention of the MicroMPX codec by Hans van Zutphen and his employee Mathijs Vos, and now through their collaboration with the Telos Alliance, we're seeing products employing this technology. We are seeing further innovation by Telos and Nautel to synchronize the HD Radio and FM audio across the internet, which was the last major problem to solve before this becomes the norm for processing moving forward.

The other major benefit to this model is that it brings the cost of good processing capability down, as there is no expensive hardware box to design, build, maintain and support by the manufacturers. It's just a server that most IT savvy engineers can maintain on their own, so really it's a win-win for everyone!

RW: *What should we know about differences in processing for various types of platform?*

Levin: The needs are very different.

The worst thing an engineer could do would be to take the OTA FM signal and feed it into a web encoder. Low-bitrate webstream encoders do not deal well with a lot of density, or clipping, both of which are employed for FM OTA.

For FM OTA processing we are trying to overcome both the inherent noise in the FM analog broadcast system, and the road noise in automobiles, as studies have proven that most FM OTA listening is done while driving. Even with FM HD OTA we want some density there to overcome the road noise I spoke of, although you obviously don't want all the clipping designed for the FM analog system.

Streaming in my opinion always needs its own separate processing which uses gentle, low-ratio compression, mainly for consistency between each piece of audio, and with some light lookahead limiting for peak protection on the encoder.

The other thing I've discovered through my own experience with low-bitrate webstream encoders, both MP3 and HE-AAC, is that they don't deal well with excessive stereo enhancing or excessive warm bass/low mid-range material. This seems to muddy everything in the codec, and too much stereo energy also causes havoc in the encoder, so careful shaping of the audio to pull some of the muddy area out, and use of very light spatial enhancing should be employed here. Since podcasts deal primarily with speech, but are still typically low-bitrate-encoded audio files, the same rules apply from my previous streaming comments with the added aspect to keep the

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Cooney: PPM Integration Is Coming Soon

NAB Radio Technology Committee has been working with Nielsen and processing makers

Michael Cooney is CTO and EVP engineering for Beasley Media Group. He's been chairman of the NAB Radio Technology Committee for the past five years.

Mike Cooney

Radio World: *What would you say is the most important development in the design or use of processing?*

Mike Cooney: I personally believe we have reached a plateau where large processing improvements to the sound can no longer be accomplished. The three largest processor manufacturers all make a great product and each has its own distinct sound.

I believe the most important development is the integration of PPM encoding in the processor. The NAB Radio Technology Committee has been working with Nielsen, Wheatstone, Orban and Omnia for the past couple years to develop this very important enhancement, and I believe we are very close to having a product in the field.

This future integration should also allow us put the PPM signal in the cloud, which will help some broadcasters with their long-term goals to have a full cloud-based solution.

"I think our listeners are less and less concerned about the audio quality and more concerned about the content."

RW: *What about the cloud, virtualization and software as a service?*

Cooney: I know those three processor manufacturers are developing a cloud or virtual solution and think it will become more important in the future. I don't see Beasley rushing out to replace our processors with a cloud-based solution but there will be needs in the future where it makes great sense.

I also believe the PPM encoding portion of this development will also be important, primarily the larger broadcasters.

RW: *With so many broadcast teams working remotely, what are the implications for processing?*

Cooney: I don't personally see an impact from someone working remotely, because most of us have been doing our processing adjustments remotely for years.

I think our listeners are less and less concerned about the audio quality and more concerned about the content. During COVID we were forced to send many of our on-air talent home and in some cases our audio content was not up to our normal standards. While I know it bothered the engineer and PD, it did not seem to negatively impact the listener.

RW: *COVID-19 has shown that expensive office buildings and myriad studios may not be necessary for good radio. What do you think the future holds for processing's place in air chains?*

Cooney: For those broadcasters who plan to put everything on the cloud, it will be important to have a virtual processing solution. Beasley does not plan to move towards the cloud solution or centralize our programming, and we prefer to differentiate ourselves by staying live and local. While processing hardware costs will probably go down, I believe the processing vendors will be forced to implement a subscription-based service for support.

Your Guide on the Road to Virtualization

Broadcast audio is shifting toward virtualization, accelerated by circumstance and the demand for more flexible broadcast-from-home workflows. From virtual mixing, processing, routing, control, telephony, or comms, Telos Alliance gives you broadcasting options that are familiar, create new ways of working, and deliver on virtualization's promise of added scalability, adaptability, cost efficiency, simple deployment, and reliability.

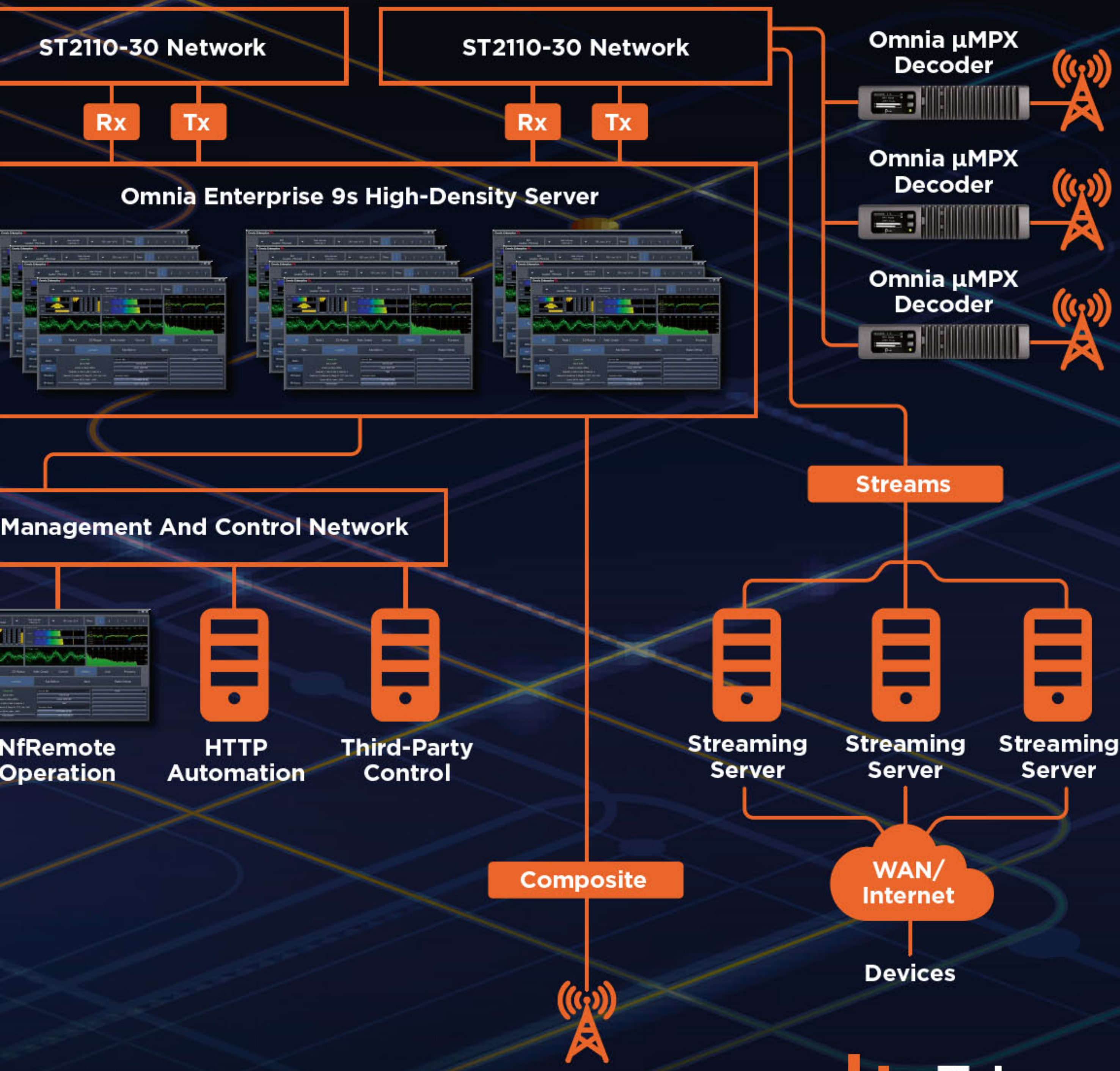
Broadcasters are tapping Telos Alliance for the virtual solutions they need today. Omnia Enterprise 9s software delivers audio processing for high-density applications for customers with a large volume of signals that feed a wide range of broadcast formats including FM, AM, HD, DAB, DAB+ and Streaming.

Telos Alliance virtual solutions equal your real-world success.

Contact us today for a consultation.
inquiry@telosalliance.com

Omnia Enterprise™ 9s

Virtualized High-Density Radio Processor with Redundant ST2110 Networks



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Streaming Has Its Own Processing Needs

Bialik says stations and engineers are learning that one size doesn't fit all

David Bialik consults to stations on their streaming and audio processing. He is an AES Fellow and award-winning engineering leader who has held technical positions with Entercom, CBS, Bloomberg, United Broadcasting, Bonneville International and the National Association of Broadcasters.

David Bialik

Radio World: *What would you say is the most notable trend in processing?*

David Bialik: An important development in the use of processors is the awareness that streaming requires different processing than "over-the-air." While broadcasters want to be the "legally" loudest, streaming does not have to be the loudest, but they can be the clearest; and with commercials originating from various locations, matching loudness levels is extremely important.

The current recommendation from the AES' recommendation for Streaming Loudness (currently being revised) is –19 LKFS.

Stations (and engineers) are now understanding that one size of processing does not fit all. You should not use the same processing for over-the-air that you do for streams. Use a loudness meter. Orban released a free one that is quite good!

As far as features: Many processors are good and have good features. Bob Orban and Frank Foti have often joked that they make the gun but you do not have to shoot it. Do not process so aggressively that you cannot identify the instruments. You should always be able to hear the cymbals! Ask yourself if the artist wants to hear their audio clipped or not.

RW: *How will the cloud and virtualization affect the processing sector?*

Bialik: Stations are hoping this will cost less and take up less real estate. Hopefully the benefit of a cloud architecture will create redundancy and eliminate a point of failure.

It also makes your internet connection more important and the need for backup more critical.

RW: *With so many people working remotely, what are the implications for managing processing today?*

Bialik: Security will be important, of course. Routing will be incredibly important since a station will have to set a "Quality of Service" to guarantee that the audio always has the bandwidth needed.

Stations will want remote facilities to sound the same as studios. Remote users will need good acoustics, and be able to produce high-quality audio — we do not want 1K telephone sound.

RW: *Content comes at us from so many locations. What role do loudness and LRA (loudness range) play?*

Bialik: This will be more important, especially for streaming where Direct Ad Insertion is being used. You do not want to be listening to content (at –19 LKFS) and then have commercials and interstitials played much louder. I have heard this happen at 6 dB louder at times. You will be knocked off your chair.

"You do not want to be listening to content (at –19 LKFS) and then have commercials and interstitials played much louder. ... You will be knocked off your chair."

RW: *Are listeners, especially younger ones, moving toward greater fidelity because of their use of on-demand services and personal downloads?*

Bialik: Stations with a good dynamic range will always sound more appealing to the listener.

RW: *There are committees at the Audio Engineering Society working on standards for audio metrics for all online audio content. What would you like to see from this work?*

Bialik: I am chairing much of this. Loudness issues invite the

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Target Loudness Is Changing Online

Will FM stations take note of the interest in greater dynamic range?

by **John Kean**

The author is senior engineer with Cavell, Mertz & Associates and head of Kean Consultants. He has held technology positions with NPR Labs, XO Communications, Moffet Larson & Johnson, and Jules Cohen & Associates.


John Kean

Audio processing has reached a level of performance where audio content can have high loudness without the traditional artifacts of audible clipping, pumping, intermodulation distortion, etc.

Of course, audio processing in a broadcast medium is justifiable for over-modulation protection and combatting noisy listening environments. Due to freedom from distortion in processors and loudness wars, however, much of radio has reached a state of hyper-compression where already-compressed popular music is fed to multiband compressors and limiters that aggressively reprocess the audio.

This situation is hard to reverse in broadcast, where competitive loudness remains a concern, but I believe minimal processing may be the right direction for online radio media.

LOUDER BECAME "BETTER"

I hate to be nostalgic, but FM was once considered a "high-fidelity" medium (I'm old enough to remember!). Consumers used to buy exquisite, expensive tuners to get the best FM sound for their living room systems.

Today a number of my non-technical friends don't even hook up the antenna on their multimedia receivers. What happened to that reputation, and is it connected to FM's gradual loss of listeners to online media?

A look at a General Electric transmitter two-page ad in a 1945 issue of Broadcasting magazine says a lot about FM's change.

The signal-to-noise ratio of the new FM system promised to deliver "double the Dynamic Range" of AM and remove "the unreality of artificially controlled sound levels that compress a fortissimo."

Using an ingenious size comparison between AM and FM (via a photo of an all-woman orchestra during World War II), G.E. touted the "contrasts of sound intensities … in all its glorious realism."

Along the way, years ago, FM radio got the idea that dynamic range had no value, and louder was better. The development of stronger and stronger FM audio processors began.

That seemed to work for FM for many years; after all, it was a portable and in-car medium with lower noise and wider frequency response than AM, as well as stereo.

> *"Younger listeners play music and shows online and from digital personal collections. My research finds that this music is distributed almost entirely in its original, unprocessed form."*

However, the 2000s brought a newer medium: online digital audio that could be delivered to smartphones as well as home computers.

While FM's decline of listeners may be due to a combination of causes, online audio (streams, podcasts and on-demand playout) have flourished. Online audio is a 16-bit digital system having a dynamic range greater than 90 decibels, regardless of the bit rate, and lossy compression codecs have continued to improve in sound quality.

AUDIO EVOLUTION

Younger listeners play music and shows online and from digital personal collections. My research finds that

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MediaWorks align with Orban

The Author is Blake Beale, Radio Engineering Manager for MediaWorks New Zealand

AUCKLAND, New Zealand - MediaWorks Radio is a New Zealand broadcaster with over 180 unique content streams broadcast over a network of just under 300 transmission points. MediaWorks broadcast 9 national brands: MoreFM, The Breeze, The Edge, The Rock, The Sound, MagicTalk, MaiFM, George FM and Magic Music.

Due to the unique way MediaWorks radio came to be through many years of acquisitions and mergers, there have been inconsistent technologies used from broadcast consoles, automation and transmitters, resulting in a difficult support model to maintain and stations that were once local, but now network brands sounding different in each market, eg, a MoreFM could have an Orban 8500 in one market and a 25 year old Unity2000 in another, creating a network which sounded different in each of the 28 markets the network broadcasts to.

A nationwide audit was undertaken with the primary brands and models that made up the network all being worthy options for comparisons. Our requirements were one processor supplier with the flexibility to sound great on an Oldies format to a CHR, Rock as well as a Dance format to mention a few. These comparisons resulted in our choice to partner with Orban, who were more than happy to assist with a long-term procurement schedule. Having the reputation of rock-solid reliability that we required for remote sites with difficult to access due to New Zealand's challenging and volatile geography also played a part in our decision.

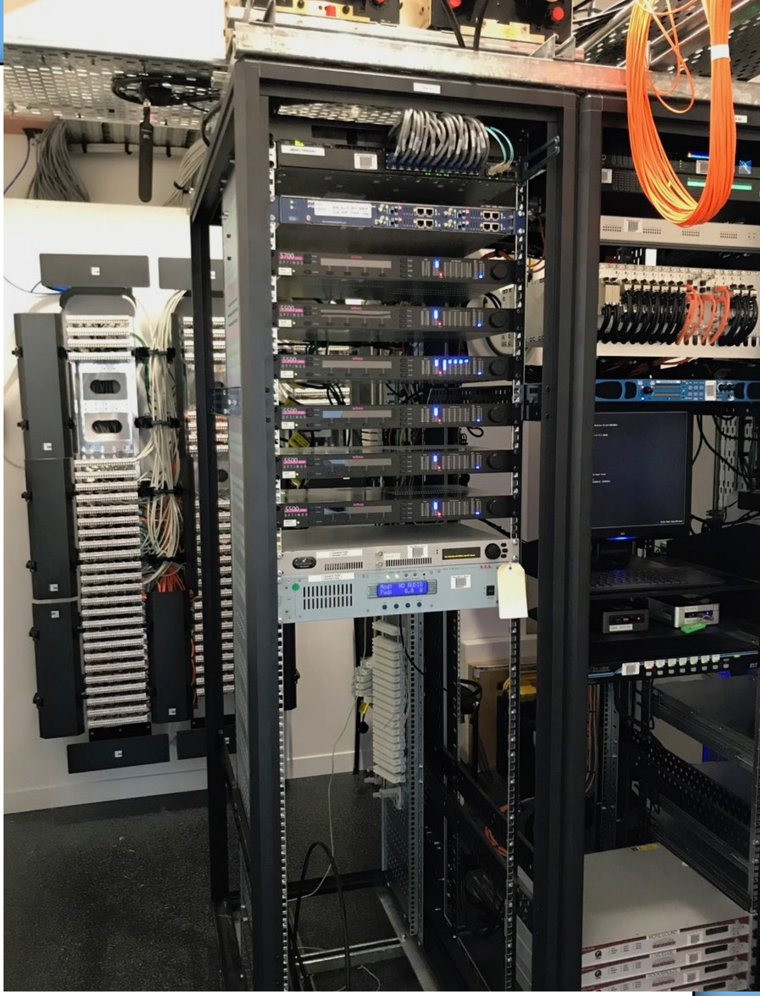
Three models were chosen to rollout, depending on the size of the market. Metro cities like Auckland, Wellington and Christchurch would receive the 8600 (now 8700i), Regional cities the 5700i and small provincial markets the 5500i. Having the added extra of integrated RDS generators has enabled MediaWorks Radio to not only up the game sound wise, but also introduce RDS with an AF table that hands over seemlessly when transitioning frequencies from the next broadcast point.

A major milestone was reached just before Christmas 2018, when major stations in the MediaWorks Radio group were aligned with Orban technology, delivering a consistent sound and RDS frequency handover along 95% of New Zealand's

MediaWorks "The Breeze" studio in Christchurch primary highway, from the top of the North Island to the bottom of the South Island, that is covered by MediaWorks Radio brands. This means along the primary route down New Zealand, All major brands mentioned at the top of this article sound exactly the same at each broadcast point, resulting in the processed sound now being a part of the brand itself. More FM, a Hot AC, has some compression, whereas The Sound, a Classic Rock format is very open with wide stereo separation, making the most of the production techniques used at the time of recording.

With studio automation raw audio being linear and satellite delivery with limited compression, we have eliminated the listener fatigue that had plagued our brands for many years, Listeners can hear the difference, clients love the wide friendly quality of our broadcasts and listener TSL has risen for each brand in the markets where we have deployed the Orban processors, resulting in higher ratings.

The partnership MediaWorks NZ has with Orban for this project has been second to none. The professionalism, the speed of support and the willingness to introduce enhancements based on our requirements has resulted in the decision to align with Orban becoming one of the most successful vendor relationships I have had the pleasure of working in and continue to work with as we continue the plan and work towards aligning the remaining markets in NZ by Q2, 2020 resulting in a 100% Orban processing solution across our growing FM Network.



Listeners Deserve a Smooth, Comfortable Ride

Jeff Keith says processing will continue to get smarter and more powerful

Jeffrey Keith is senior audio processing product development engineer for Wheatstone. He has been working in processing development since 1999 and joined Wheatstone in 2007.

Jeff Keith

Radio World: *Jeff, what would you say is the most important development in processors?*

Jeff Keith: The radio broadcast medium is in the process of reinventing itself. While over-the-air radio is still important, especially with the ability of HD to simultaneously carry multiple program types, technology now makes the delivery of other information not just a fad but the soon-to-be norm.

I can see a time where those huge broadcast towers we've seen for many decades are all but gone, and replaced by high-speed internet or cellular technology — technology that will allow listeners to carry their favorite programs and stations not just out of the local market, but to anywhere in the world.

RW: *What should readers know about the differences in processing needs for various platforms?*

Keith: Each transmission medium requires different audio processing treatment in order to deliver the best quality audio to the listener.

I've seen many stations that are still using retired on-air processing for their internet stream, or worse yet, feeding the internet stream encoder from the output of a radio or modulation monitor. I can't think of a better way to make a nasty sounding internet stream!

Purpose-built streaming processing will always sonically outperform any other form of processing not specifically designed for streaming codecs.

RW: *How will the concepts of the cloud, virtualization and software as a service affect the processing marketplace?*

Keith: There is no question that it is possible to run anything software-based, including audio processing, on cloud servers. It'll be awhile before we see the end of this movie, though — how broadcasters will handle redundancy, encryption and failover to an alternate when the main goes offline.

Software as a service will be the norm, and I can see a time when radio stations will no longer "own" their audio processing, at least in the form of today's hardware box. Audio processing will be a chunk of software running on a cloud server somewhere and licensed by instance, probably annually, on a recurring schedule.

The whole game will be different.

RW: *With audio originating from so many locations, what role do loudness and loudness range play?*

Keith: It is my personal wish that the United States would adopt some form of over-the-air loudness regulation.

Listen to stations in countries where they need to adhere to ITU BS.412, for instance. Those stations are much more pleasant to listen to because the processing hasn't been tuned to the singular goal of "louder than everyone else on the planet."

I think many stations have forgotten that it isn't loudness, it's program content. Every radio made in the last 100 years has had a volume control …

> *"I can see a time where those huge broadcast towers we've seen for many decades are all but gone, and replaced by high-speed internet or cellular technology."*

RW: *What recently introduced new features or capabilities in processors are most notable?*

Keith: Wheatstone has made the job of time alignment easier for FM/HD broadcasters by including fully automatic time alignment in the X5 FM/HD on-air processor.

In fact Wheatstone's X5 has two methods by which perfect long-term time alignment can be had: the built-in FM/

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The Recipe for Processing Is Never Finished

Frank Foti: It's how one achieves a loud signature that determines listenability

Frank Foti is executive chairman of The Telos Alliance and founder of Omnia Audio.

Frank Foti

Radio World: *Frank, what would you say is the most important recent or pending development in the design or use of processors?*
Frank Foti: The recipe for audio processing is never finished.

Aside from ongoing development to subjectively improve sonic performance, the function of processing has crossed over into the virtual realm. This concept was first fostered by Steve Church, and myself back about 1994, as our early efforts began on Livewire, our audio over IP platform then under development.

Today, we have the tools to provide processing in the software-as-a-service (SaaS) format, as well as a container. Yet we also know that there are those in the marketplace whose comfort level remains having their processing running in a dedicated appliance. Our work will always support that platform as well.

RW: *What should we know about differences in processing needs for analog over the air, digital OTA, podcasts and streaming?*
Foti: Telos Systems was first to introduce data-reduced audio more than 25 years ago. Steve Church and I were also the first to recognize the need of dedicated processing for conventional broadcasting, and audio streams.

In reality, digital OTA, podcasts and streaming are all basically one form or another of the data-reduced technology. Thus, all conventional analog OTA transmissions for FM or AM need to employ a processor for that function, and digital OTA, podcasts, streaming, need to use processing designed for data reduced audio.

The main difference between conventional and data-reduced audio transmissions is the final limiter function. Suffice it to say, a processor designed for one system will not "play" well with the other type of system.

RW: *How will cloud, virtualization and SaaS affect our processing marketplace?*
Foti: It already has! The pandemic of 2020 escalated efforts that were already in place regarding this topic. If anything, now we'll observe refinements to what's already in place.

The concepts of the cloud and virtualization present flexibility to the broadcaster that was never possible before. Processing can be installed, adjusted, modified as a system, moved, updated and a host of other utilities from basically anywhere in the world. We even have the ability to transport monitor audio back to remote locations that might be outside of the listening coverage.

RW: *Six years ago we had an ebook where we wondered how processors could advance much more, given how powerful their hardware and algorithms were. What about today?*
Foti: This question gets asked fairly often. The Achilles Heel of broadcast audio processing has always been the final limiting system. As much as we'd all love a free lunch, it does not apply here, and there is a breaking point.

I'm constantly evaluating our own efforts, as well as those from others. Using choice content, which is challenging for any algorithm, it is easy to discern a good limiter design from another. Sadly, there are some current designs that leave a lot to be desired in this area.

Recent ongoing development from my own workstation has produced a new final limiting system that further reduces and in some cases eliminates sonic annoyances. Those being harmonic and intermodulation distortion components that are audible.

RW: *Has radio reached a point of "hypercompression," with little further change in how loud we can make over-the-air audio? How do we break out of that plateau?*

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Moving Audio in the Cloud Brings Challenges

Bob Orban: Processing options help stations that seek to navigate these new scenarios

Robert Orban is a consultant to Orban Labs Inc. He has been developing audio processing algorithms and hardware for broadcast and studio use for more than 50 years.

Bob Orban

Radio World: *What's the most important new development in design and use of processors for radio broadcasting?*
Bob Orban: There are several possible answers. For some operations, virtualization of processing software has become significant, although putting processing software in the cloud is constrained by the need for reliable, high-quality audio connections with 100% availability. For other operations, compatibility with audio over IP connections and digital composite connections to the transmitter are more important. Others may value the ongoing refinement of processing algorithms that improve stations' sound.

RW: *How different are processing needs of analog broadcast, digital OTA, podcasts and streaming?*
Orban: The processing for these transmission channels can be very similar except for the peak limiting technology.

For analog AM and FM, peak limiters must not pump or compromise loudness when faced with preemphasized signals, which implies clipping-like limiting with sophisticated distortion control.

For the other transmission channels, all of which include lossy codecs with no preemphasis, it is more important not to waste bits by encoding limiter-induced distortion spectrum, so limiters for these services should be very clean spectrally.

Additionally, some streamers may wish to use static file normalization to a target loudness instead of radio-style processing, although static normalization does not handle transitions and voiceovers nearly as well.

RW: *What is the impact of the cloud, virtualization and SaaS on the processing marketplace?*
Orban: There is considerable interest in these concepts. However, moving the audio in and out of the cloud without dropouts, glitches and/or unacceptable latency is challenging.

Broadcasters must make a choice between the reliability and low latency of the current hardware processor infrastructure and the potential convenience of not having to own and maintain processing hardware. Orban offers products for both scenarios.

I find it interesting that there seems to be a backlash developing regarding putting everything in the cloud, with some players moving infrastructure requiring high performance back from the "cloud" to the "edge."

RW: *With audio coming from so many locations, what role do loudness and loudness range (LRA) play? Will future audio processors have monitoring capability for both on-air and streams?*
Orban: As a member of the AES committee working on revising the AES TD1004.1.15-10 "Recommendation for Loudness of Audio Streaming and Network File Playback," I am familiar with how industry experts in this area are thinking. We all agree that it is important to have consistent loudness between streams so that consumer can switch between streams without uncomfortable loudness jumps, and the ITU-R BS.1770 loudness measurement algorithm has been standardized for that purpose despite some limitations.

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Solid, Innovative Processing for "Regular Guys"

Ben Barber uses the analogy of driving a car when comparing audio processors

Ben Barber is president/CEO of Inovonics. He says that when his parents bought him a "Radio Shack 65-In-One Kit" when he was 12, there was no looking back.

Ben Barber

RW: *Your take on the most important development in processors?*

Ben Barber: Everything needs to be remote controllable and "monitorable."

With fewer and fewer people actually being onsite, if there is an issue, broadcasters want to know about it right away. All of our newer audio processors are web-enabled, which means you can log into them and control them via their web page and not a proprietary app or program that may run on your PC but not on your smartphone.

With web-enabled products, everything can be controlled from any device with a web browser. You can also get emails, text messages and SNMP alerts as well as stream the audio back over the web.

"Sure, programs can be done remotely; but in my opinion, if we think this is the new normal and we continue doing everything from scattered offices with little human interaction, then we are not giving our best."

RW: *What should we know about the differences in processing for various platforms being used by today's radio media companies?*

Barber: Today's processors are all DSP-controlled and most can sound very good while at the same time controlling peak modulation as well as density. All that is great; but if you start with an MP3, especially at a lower bitrate, there is little you can do to make that source material sound great.

Always start with great audio, which will in the long run save you so many headaches down the road.

RW: *What are the implications for managing processing, now that so many people have been working remotely?*

Barber: I think what COVID has shown us is the resilience of both broadcast personalities as well as engineering to be able to do "everything" remotely. But just because it can be done begs the question, "Is it best for radio?"

Our medium is a very personal one, where the synergy between hosts is evident on nearly every show. Sure, programs can be done remotely; but in my opinion, if we think this is the new normal and we continue doing everything from scattered offices with little human interaction, then we are not giving our best.

As for processing, its primary purpose is to control peaks in order to protect your transmitter's modulation, and also to keep you from splattering on your "neighbor." Our industry should strive to give that processing air chain the best possible content that we can produce; to do that, I think in-person energy is what stimulates the mind, and the product shows it.

RW: *What tools are available to mitigate issues involving synchronization of HD Radio and analog signals?*

Barber: Back in the day when HD Radio was introduced, the system could be stable if everything was collocated and set up properly.

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It's All About Good Audio Source and Level Control

Brentlinger: Understand the interplay of codecs and hard limiting or clipping

Jay Brentlinger is president, CEO and chairman of Circuit Research Labs Inc. He owned Orban for 16 years and now does service, repairs and trade-ins on older Orban legacy products. Robert William Leembruggen is an Optimod technician and former broadcast engineer.


Jay Brentlinger and Robert Leembruggen

Radio World: *What is your view on the role of the cloud?*
Jay Brentlinger: Personally, as a station owner, I would not want to depend on the cloud for my audio. I feel compromised even depending on the internet to deliver audio to the transmitter and feel there must be a backup.

My feeling might change with time, but for now a full local backup audio playout system is required at the transmitter site when using the cloud or audio over IP from the internet. I have proved this to myself many times with my stations.

> *"Everyone must remember that using low-bitrate MP3 audio as a source and then to use a codec of any kind makes the sound more distorted and it gets worse every time it happens."*

RW: *What is the most notable development in processing?*
Brentlinger: Using HD audio processing for streaming and audio over IP.

RW: *What about differences in processing for various analog and digital platforms?*
Brentlinger: Digital processing for analog audio transmission is totally different from digital audio processing for any digital transmission. This is entirely due to all-digital transmissions using some form of perceptual codec to reduce the size of the data being transmitted. If any hard limiting, or even worse, clipping is used, the codec simply cannot handle this and results in high distortion and wasted bits of information.

RW: *Many broadcasters are working remotely, what are the implications for processing management?*
Brentlinger: A station using remotely generated audio files or live source must have proper automatic gain control and good final audio processing before the transmitter. Levels can vary from one DJ, announcer or source and this can cause ducking as a result to affect the next audio cut. This happens all the time when the producer of the remote audio does not use some sort of AGC or limiting.

RW: *The health crisis has shown that costly buildings and studios may not be necessary; how does this relate to processing?*
Brentlinger: All major groups are looking for ways to save money. Brick-and-mortar buildings are a huge cost and allowing DJs and announcers to work from home is a real cost-saving move.

But once again good audio source and level control is critical. Everyone must remember that using low-bitrate MP3 audio as a source and then to use a codec of any kind makes the sound more distorted, and it gets worse every time it happens.

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Product Lines Reflect New Service Models

Gregory Mercier says WorldCast has seen a shift in customer mindset

Gregory Mercier is director of product marketing and pre/post-sales support for WorldCast Systems and co-designer of its new sound processor line.

Gregory Mercier

Radio World: *What's notable in processing from your viewpoint right now?*

Gregory Mercier: WorldCast Systems' new five-band sound processor (Version 2) for FM broadcasting, with an integration into our Ecreso FM transmitter lines. It provides powerful processing algorithms, presets, adjustment capabilities, high loudness for those looking for it, and an unrivalled signal clarity.

"Reducing op-ex is not a new topic at all for broadcasters; however, the market was traditionally conservative and tended to refrain from software innovations."

RW: *How do you view differences in processing needs for various OTA and non-broadcast platforms?*

Mercier: The audio needs to be adapted to each broadcasting format and to the reception conditions. Each format has its own specificities.

Here are a few examples. Digital broadcasting usually implies lossy audio compression, which will unlikely sound good with heavy clipping. In FM however, there is a 15 kHz filtering and pre-emphasis and the loudness may change the reception quality.

Only with these basic examples can we clearly understand the need for specific final processing to ensure the station's sonic signature through all the formats.

RW: *What is the impact of the cloud, virtualization and SaaS?*

Mercier: In the context of the ongoing crisis, we are clearly observing with our customers the growing importance of reducing their operating costs, or more precisely, it has now become a priority.

Reducing op-ex is not a new topic at all for broadcasters; however, the market was traditionally conservative and tended to refrain from software innovations. With the crisis, we are now seeing a shift in customer mindset with, for example, an increasing demand for solutions based on software licenses.

Based on this new service model, our five-band sound processor is being met with a lot of success. Other WorldCast examples I could mention: SmartFM is a software license for FM transmitters to reduce energy consumption by up to 40%. APTmpX is a software license for APT codecs enabling high-quality MPX/composite transport over IP while saving bandwidth (under 900 kbps) and removing the need for on-site processors. KYBIO Media, offered as an SaaS license, is for centralized and scalable system remote monitoring and control.

RW: *What recently introduced features or capabilities in processors are notable?*

Mercier: A major innovation is the way we integrated the processing in the broadcasting chain. With the five-band sound processor, the algorithms run inside the transmitter's FM direct-to-channel digital modulator. There is no additional board inside the device and no cabling, and the result is a huge simplification of the traditional chain. From audio input to RF output, our robust DSP/FPGA platform provides unprecedented control of the signal and its purity while reducing hardware, consumption and maintenance costs, which is more than ever the challenge for radios.

Recently, we also launched SmartFM, our "green" innovation capable of predicting the listeners' perceived quality in the field and reducing transmitter energy consumption by up to 40%. Program content characteristics, including its processing, obviously plays a role in SmartFM. Our customers' feedback is that they are improving their audio while considerably reducing operating costs.

11 Processing Things to Think About

Here are some best practices as well as some questions you should consider

As we have in several ebooks, we conclude by asking veteran engineer and consultant Gary Kline to create a list of key topics to consider.

Gary Kline

I think the processor may be one of the most discussed pieces of equipment that a station owns. Everyone you meet in any country will be glad to give you their opinion on which is the best for a particular market, format or budget. If you put 10 PDs or engineers in a room, it would be rare that they agree on the "best processor out there."

That said, there are points most processing gurus will agree on. Here are some based on my travels and experience.

1 KNOW THE MARKET, KNOW YOUR COMPETITORS.

Get to know everything you can about your competitors and their technical setup.

This goes beyond listening to every station on the dial carefully (you should), but also objectively. Don't be reluctant to admit that another station sounds — in your opinion — better than yours. Do your research, which may require intelligence gathering. Get to know everyone's transmission path including console, STL, transmitter, age of equipment, and, of course, the processing they use. Don't forget to listen to HD or DAB channels too.

You should know *your* equipment; once you know what the competition is using, you can balance your objectiveness.

For example, say you think the CHR competitor sounds better than you. Is that because they have a cleaner transmission path? Stronger/newer processing? Better source material? Greater RF over the coverage area? If any of those is true, your processing concerns may expand to fixing other things too.

Whether you have a direct competitor in the market or not, still get to know each station's particular sound. This will help you rate the market overall and help you in designing your custom audio signature sound. Some markets are softer. Some are loud and very competitive with high MPX density levels. Some just sound poor across the dial.

2 KNOW YOUR GOALS.

Too often there is a desire to purchase a new processor without a clear reason. Understanding your reasons and budget constraints will go a long way in making an informed choice.

Is your processor older and not as competitive or clean-sounding? Do you need to feed a new DAB or HD channel, and your processor does not support that? Did your current unit die of old age or a lightning strike? Is it time to standardize processing or stereo generators across the network? Are ratings slipping That's a common reason given, but a processor is not always a ratings cure.

What's the budget? How much processing can you afford? Or better yet, do you need to buy the top-level box when something less costly might do?

I frequently get into a discussion about goals and budget with an operator only to find out that what they already have meets their goals; in other cases, I may determine that while an operator thought they could make do with what they already own, it becomes clear they cannot. Each situation is unique.

3 KNOW THE LANDSCAPE OF CURRENT PRODUCTS.

If you are going to make a purchasing decision you should know what your choices are.

Sure, most of us in the radio business know the top brands and may even know the current model(s). But do you know about processors designed, manufactured and sold around the world? Processing philosophies and design varies around the world; perhaps there is a "sound" you can import that your listeners will gravitate to.

Do you know how each brand sounds or the benefit of one versus another? Do you know "street price" for every model? Do you know which features require an additional fee for extra outputs like one for HD or an internet stream? Do you know if there are forthcoming firmware updates which may add improvements which could influence your decision? Do you have contacts at the manufacturer or their reps who can explain these things or set up a demo?

"Do you process your stream with as much thought and attention to detail as your terrestrial signal? About half of the stations I listen to online are not paying attention to their digital asset audio processing."

4 CONSIDER HIRING A SEASONED AUDIO EXPERT IF YOU ARE NOT COMFORTABLE DESIGNING YOUR AUDIO SIGNATURE.

I visit broadcast facilities that have PDs or engineers who are adept at processing and know how to install and tune a box. I also run into places where outside expertise can add considerable value.

There are many important and critical steps in setting up a new processor. There are the technical transmission settings such as input, output, pilot injection, sample rates, input switching, network IP parameters, and other interfaces. Then there are the hundreds (yes, hundreds) of individual processing settings to tailor the audio to your preferences.

Even in situations where a station has in-house processing expertise, it can help to get an objective opinion from individual who has a toolkit of presets and starting points to speed the adjustment process. It is also good insurance to have a consultant to ensure the transmission parameters are set correctly and legally.

Many newer processors have non-expert modes that make the tailoring of the sound easier with fewer settings. However, in some instances, such as very competitive situations, "expert" mode may be the best way to achieve that perfect signature sound.

A consultant can help with the selection process as well as performing a full technical review of the plant.

5 UNDERSTAND THE FEATURE SETS OF MODERN PROCESSING.

Local stations may need one set of features while network, enterprise or state-owned broadcasters may require a different set. Here's a series of questions you might consider, and topics to research.

Do you know what MPX over IP is? (Hint, it is one the latest techniques for sending your composite MPX over IP to your transmitter.) Do you know who offers that and in what configuration(s)?

Do you know what composite EQ is? What is pilot protection? What is SSB and DSB and why might that matter to you?

How many digital and analog inputs and outputs do you need? Which boxes offer how many of each?

Do you want a box that can generate dynamic RDS? How many bands of AGC and limiting would work best for your format and desired sound?

Do you want dual power supplies or some form of additional redundancy? Are you interested in processors that can run in a virtual environment and is that something you should be interested in?

Do you need GPS sync for your stereo generator, say for an SFN? Did you know that many processors sold today have hard-drive storage to hold music and imaging so that if your studio playout system (or studio altogether) goes offline, you'll still be on the air?

Do you need SNMP monitoring? Do you know what de-clipping is? Phase correction? Do you want to feed your analog transmitter, digital transmitter and internet stream simultaneously? Do you need ratings encoding or a ratings encoder patch-point?

I could fill pages with features you might consider. Do your research and get to know what features matter and why they matter, then overlay that with your market research.

6 KNOW YOUR AIR CHAIN.

Understand you air chain from microphone to speaker. Literally.

I visit many stations whose managers complain about their sound and ask for processing adjustments or a processor to "fix" it. I almost always find weak links in their audio path that contribute to the quality issue.

Sure, they may need a new processor. Sure, they may need careful adjusting and tweaking. However, other things need to be addressed too. At the top of the list and most often is source material.

I still find plenty of MP3s on the playout system hard drive. I've been to stations with hundreds of MP3s (with bit rates between 96 and 192) and they wonder why their sound is not as clean or lush as the other stations in the market. I very rarely find a hard drive that doesn't have at least a few MP3s.

Beware — several playout systems rename MP3 to WAV and increase the file size; that will fake you out. You need special tools to scan the library and find these fake files. More on source material below.

I also see STL paths that have issues. Does your feed to the transmitter use an uncompressed audio path or is it something lossy? Is your sample rate 44.1 kHz or 32 kHz? How many A/D and D/A conversions are in the path?

Also I still find digital consoles that use their analog output to feed a digital STL. I see playout systems using their analog outputs to feed a digital console. Even with AoIP systems — which you'd think by definition would be all-digital — it is possible to find analog ins/outs used for playout systems, emergency alerting interrupt boxes and feeds to the transmitter.

Each analog to digital conversion (or the other way around) is another point of degradation. These weak points between console and transmitter add up; while one thing alone might not be noticeable, several together can be.

The road to excellent sound is not just about the box, it's about the entire system.

7 BE A PERFECTIONIST ON SOURCE MATERIAL EVERY STEP OF THE WAY.

You know this saying but it certainly applies to processing: Garbage In = Garbage Out.

In over 90% of stations I visit, I find at least several source material violations: MP3s, low sample rates, recordings from imperfect masters, etc. This is what I tell every PD, MD, and APD I meet.

We all know MP3s are a no-no. Resist the urge to download material from YouTube or iTunes or some other source.

I often hear that a particular MP3 files is the result of not knowing where to find the older material. There are sources for CD quality (or better) versions of almost anything — many are online for download. There are companies that can provide a fully loaded hard drive with your specific music and in true PCM WAV uncompressed format. Do your research, put the effort in, and ensure you have the best material.

Sometimes the issue isn't the file format but the actual source. There are plenty of forums online that discuss the best masters, greatest hit collections and top picks by audiophiles for various artists. Google is your friend.

Did you know that among several discs by ABBA, some are considered far better quality than others? Read this interesting discussion.

Use your ears. If you hear a song on your station that doesn't sound quite right, go back and research the cut. If you can't determine where it came from, get a copy of known quality.

If you run HD or DAB, you already know those digital signals use a codec. If you play an MP3 file — which already is a lossy format — over an HD/DAB channel, you end up with cascading codecs. In other words, the sound quality may even be worse when listened on the digital carrier.

"Know your competitors. Get to know their transmission paths including console, STL, transmitter, age of equipment, and, of course, the processing they use."

8 DON'T RUSH THINGS.

It takes time to perfect an audio signature. It can take days or weeks to get that perfect audio signature. Take your time.

Some of the best sounding stations around the world have taken their time to "dial in" their settings. It is rare to design a sophisticated, nuanced and consistent sound in one day.

Yes, processors come with presets that get you in the ballpark. And, yes, as a result you can have a decent sounding station quickly, assuming you don't have other severe audio chain problems, very poor source material, etc. But, for that perfect market-leading sound, it takes time to "process beyond the preset."

For example after a processor is adjusted, all parties should take a break, sleep and then listen again. Your ears get fatigued after hours of listening and adjusting. They can confuse you. Sleep on it and see how things sound when your ears are fresh. In some cases, as you get closer, it's helpful to wait a week or two and listen afresh. The longer period also allows you to listen to various content samples to ensure that the sound is consistent across sources.

If you are an oldies or 80s station or play music across several decades, finding a balanced sound that works for every cut can be challenging. The way music was mastered and produced in the 1970s is different from how it was done in the 80s and way different then say, Dua Lipa in 2020.

And, yes, there are stations that play Madonna, Van Halen and Dua Lipa in the same hour. I worked on one recently and it required careful attention to detail to sound consistent throughout the day.

Fortunately, many of the modern processors have great toolsets to help with this issue.

9 UNDERSTAND THE POLITICS.

The process of processing can be complicated by the need for more than one person to agree on the results.

You may be working alongside a PD, OM, GM, programming consultant or owner who "thinks" they know audio. And perhaps they do— but will they all agree on what good sound is? Chances are, no.

Audio processing is very subjective. What one person thinks is the perfect low-end or vocal mix might sound horrible to another.

If you are the person with fingers on the knobs, your bedside manner and maturity will become crucial in these situations.

Don't be offended if someone says they don't like the sound. Don't be frustrated if it takes several iterations to get consensus. And you may never get 100% agreement.

To avoid problems I'll ask, at the beginning, to know who makes the final call. It may be the PD of the brand or the PD plus the general manager. Sometimes, it's the owner too. Try to limit the decision-making team to very few people.

I've been asked as a processing consultant to be the one to make the final call. I inquire about the goals; for example, maybe everyone thinks the high-end needs to be cranked yet they've also said that TSL matters. In a situation like that, I may advise that too much high-end could risk tuneout and lower TSL.

10 LISTEN TO YOUR PRODUCT ON SEVERAL DEVICES AND IN VARIOUS TYPES OF VEHICLES.

Listen in your car, your GM's car, your PD's car and your best friends' cars. Listen on a clock radio (especially in mono), on different smart speakers, and through the type of headphones/earbuds typically worn by your listeners.

Your signature *will* sound different depending on where and what you are listening to. Ensure that the sound is acceptable across most devices and speakers. It's OK to tweak based on what you hear. The goal is a nice balance so that the station sounds great on small speakers and big ones alike.

Be honest with yourself. You may have achieved what you think is the best low-end you've ever heard … in your car. Then, you listen in another car and wonder why it's overwhelming. Don't ignore it, go back and carefully find the right balance.

Remember the politics. The PD may be listening in their car and will legitimately hear things differently than you do. The audience will too. This is another reason finding that perfect sound takes time.

That being said, you can chase your tail forever if you make an adjustment every time you receive a comment or listen to a new device; know when to stop. Keeping the decision team small will help with this.

11 DON'T FORGET YOUR DIGITAL ASSETS.

These include your streams, Alexa, YouTube, FB, IG, etc. There are smart speakers, mobile apps and other ways in which your product is distributed.

Do you process your stream with as much thought and attention to detail as your terrestrial signal? About half of the stations I listen to online are not paying attention to their digital asset audio processing.

Consider using your main processor if that's technically feasible. If not, use a separately purchased processor designed for digital streaming, an older model laying around the station left over from a previous upgrade (something is better than nothing), or one of the many great software-based processors you can find online.

Some manufacturers do offer software that can be used for streaming; many will run on the same PC your streaming encoder resides on. There are also hardware-based streaming appliances with built-in processing.

But don't forget, processing for streaming requires a sound that's great across various devices.

Digital processing design does not have all the same considerations as AM or FM transmission. For instance, analog FM is limited to 15 kHz frequency response while your stream might go out to 20 kHz. There is no 50us or 75us equalization curve.

Pay attention to your bit rates — don't dip too low. There are several very good white papers on streaming across the web and located on the sites of audio processing manufacturers.

Streaming audio, done properly, will sound amazing — better than the analog terrestrial signal.

The author is owner of Kline Consulting Group LLC. He has held technical positions with several major broadcast organizations, most notably as senior VP of engineering at Cumulus Media. He has provided engineering support and consulting in the United States, Canada, China and several South American countries. He is a past recipient of the Radio World Excellence in Engineering Award.

LAWLER

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RW: *How is consumer behavior changing; for instance are younger consumers moving toward greater fidelity?*
Lawler: Things have come full circle almost. In the 1950s and '60s you had a 3-inch mono speaker that went hand-in-hand with the explosion of top 40 radio. And now, we have smart speakers that are about the same size fueling another revolution in how audio entertainment is consumed. Apple and their just released new iPod touting greater fidelity, and the ability to pair them and create stereo, similar to other smart speakers.

If this is how your audience consumes the station/stream/podcast, make sure to give them a download or on-demand stream that is easy to listen to no matter the environment. Make the most of the 3-inch speaker without sounding smashed. Do your content creators have access to the tools to process voice without making it sound unnatural? That is the trick with modern listening — making it pleasing while taking into account less than perfect conditions.

RW: *In 2014 we did a story asking if processors had become as powerful as they could get. In 2020, where might further dramatic improvements come from?*
Lawler: Never underestimate the ability to go further. My grandfather once believed that Cadillac would go no further than a dual points ignition setup — now look at what can be done with engine management!

Tools like the limitless clipper in Wheatstone's X5 or being able to generate a perfect composite FM signal from a 192 kHz PC soundcard with StereoTool. Six years ago everyone was asking what was next after the big three (Orban, Omnia, Wheatstone) took FM to as loud as could be asked — and all went in the direction of how to put the quality back in with such hyperprocessed source material from record producers. I can't wait to see what the next six years bring!

RW: *What else should we know about processing for radio?*
Lawler: Look at your entire audio path — from the quality of the files you are playing (WAV vs MP2/MP3), the STL, the exciter/transmitter, and even the antenna. Any one of those could be the reason you cannot achieve the sound you are looking for. And as the old saying goes "Garbage in, garbage out."

RETURN TO STORY

LEVIN

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voice region clean, intelligible, and consistent.

RW: *With "hybrid" platforms, a listener might tune to an FM but then drive out of market and the receiver switches to the online stream. What "matching" challenges does this present?*
Levin: As this technology becomes more prevalent, paying attention to your web stream processing becomes more and more important, as it won't just be in homes and offices anymore, but now in cars as well and for the masses.

This is where creating your "sonic signature" on both your OTA and your stream is so important. While the needs of processing for streaming differ greatly, you can still create a certain "sound" for your station that stays consistent on all platforms.

Take the time to listen to your FM, HD and web stream and come up with something that sounds comparable on all platforms.

"I'm finding as I travel that most modern DSP-based HD capable receivers start to induce distortion on anything over 110%."

RW: *Where might further dramatic improvements in processing power come from?*
Levin: Unfortunately, I think the needs today are more about trying to repair the damage done to the music by poor mastering techniques used by the record labels, and/or the damage done by using lossy codecs in the distribution process. Processing has become more than just compression, limiting and clipping.

Modern processors of today also have to repair the audio before it ever hits the compression stages. Different manufacturers are finding different ways to do this; these tools aim either to declip and add dynamics to audio that the mastering process has over-processed and over-clipped, or restore missing spectrum and remove artifacts from lossy compression.

Those that implement these repair tools in their processors have a cleaner product going into the compression stages, and will end up with a much-better-sounding product on the output, and I think we will continue to see more of these kinds of tools. Additionally, there has been effort put into preparing the output audio or processors feeding low-bitrate codecs (i.e. streaming or HD) to prevent artifacts from being generated in the codec itself; all in an effort to get the best sounding audio to the user.

RW: *We've also been asking folks if radio processing has attained such a condition of "hypercompression" that there has been little further change in how loud one can make over-the-air audio.*
Levin: I have actually seen a significant amount of development from several of the leading processor manufacturers to create cleaner and cleaner clipping structures. Each employs different techniques to do this, so each has different side effects, but as a whole, the loudness levels we are able to achieve today while still keeping the audio clean and free of clipping grunge, distortion, and artifacts out of the top boxes on the market is actually a huge improvement over the boxes of 10+ years ago.

Now, how the engineers are turning the knobs on these boxes at their individual stations is another story. I think in some cases engineers are still abusing even these modern clippers and driving them past the point of sounding good, and further damaging the end user experience by over modulating significantly, causing massive amounts of distortion in modern DSP receivers.

I'm finding as I travel that most modern DSP-based HD capable receivers start to induce distortion on anything over 110%, and while many markets and engineers stay below this and can maintain clean audio, there are others that choose to carelessly overmodulate by as much as 140%, and you can imagine how bad that can sound on a modern receivers.

As much effort as the manufacturers have put into cleaning up the audio and providing a better product for the end user, it's still up to the engineer installing and setting up their air chain and processing to make sure that they are using the tools at their disposal to provide the best possible product to their listeners.

I remember a day when radio sounded better than the music you would buy and listen to on your own, when processing actually improved the sound. With the power of modern processors, this is still possible today, but so many markets I've driven through recently this is sadly not the case. I long for the day when we as an industry strive for that goal once again, to sound better than the other streaming services and listening options out there.

RW: *Could radio see loss of potential audience due to listening fatigue?*
Levin: We as an industry are driving listeners away by bad practices, not only by overcompression, overclipping and overmodulating, but let's add overusing Voltair to that list as well. I've traveled to some markets where all I hear is PPM tones adding flange effect and reverb effect to everything going over the air.

We have to do a better job of caring what our product sounds like if we hope to stay relevant in the future. Now sure, there may be some listeners out there who don't care; but there are a lot that do.

While they may not be able to tell you why they can't stand to listen to a particular radio station for more than a few songs or a few minutes before it drives them crazy or makes them want to turn the volume down, I wager that if you had the same content on a much cleaner-sounding delivery system, they would suddenly find it much less annoying and actually find themselves turning the volume up, instead of down or off.

Give listeners a reason to turn the volume up, make your station sound good!

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BIALIK

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listener to constantly adjust the level. If they are adjusting one control it is as easy to turn the content off. How will that help the TLH?

RW: *As AES loudness metrics are moving to a lower target level for content, streams, podcasts and on-demand file transfer, could radio lose audience due to listener fatigue?*
Bialik: I believe the lack of dynamic range will cause listener fatigue. Hopefully the content will have good dynamic range and good loudness levels. The level of audio-only streams is being targeted at –19 LKFS while video is at –24 LKFS. Within the short term future, loudness could be controlled by metadata. Yes we are talking a 5 dB difference. The recording industry is also pushing for –24 LKFS. This allows for more headroom as well.

RW: *With new "hybrid" radio platforms coming out, a listener might tune to an FM signal in a market but then drive out of it, with the receiver switching to the station's online stream. What matching challenges does this present?*
Bialik: Stations that have to cover ads and sports blackouts will sound worse.

RW: *What else should we know?*
Bialik: If everyone say streaming is the future, why not invest in the future now and do the best audio you can?

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This ad for General Electric transmitters appeared in Broadcasting magazine in 1945. Courtesy www.worldradiohistory.com.

KEAN

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this music is distributed almost entirely in its original, unprocessed form.

This is true of major on-demand music services, and some are now are offering high-fidelity channels with higher bit rates and even "lossless" coding. The tracks are simply normalized (gain offset) to a common loudness target, without touching the dynamic range of the content.

In a recent project for a major radio group, I found that some online distributors of live station audio are using substantially less processing than their on-air broadcasts. Perhaps some are learning that "artificially controlled sound levels" are not preferred by listeners.

Similarly, podcasts — the fastest growing segment of online audio — are produced and delivered with little or no audio processing.

The target loudness of the online industry is changing to a lower value, to permit greater dynamic range. I have the privilege of chairing a drafting committee at the Audio Engineering Society, which is writing a new technical document for online audio parameters. These interim specifications will evolve to a profile with even wider dynamic range to match audio-for-video standards — and we know how much dynamic range video services deliver!

TIME TO RECONSIDER?

Broadcasters are now faced with another choice if they adopt "hybrid radio," which provides a streaming alternative to radio reception as listeners drive outside the broadcast coverage.

FM stations could choose to match the audio processing of their stream to the (hyper-compressed) broadcast audio, to avoid changes as the dashboard receiver switches between off-air and stream.

Or should they? Perhaps radio should reconsider what they broadcast and move with the audio industry and away from heavy compression.

When hyper-compressed audio is normalized to the same Integrated Loudness as lightly processed audio, a heavily-compressed stream sounds weak and flat by comparison. Compressing a stream to sound like air can't compete with natural, dynamic sound.

Considering this, wouldn't it be wonderful if the FM stations, too, returned their own air audio to a high-fidelity condition, as FM promised 75 years ago?

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KEITH

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HD automatic time alignment, and our SyncLink product, which takes the X5's FM and HD audio signals and packetizes them into one data stream for delivery to the transmitter. FM and HD *cannot* get out of sync, ever. Further, SyncLink's various signal outputs are compatible with every FM transmitter and exciter ever made.

RW: *In 2014 we wrote that processors were so powerful that it was hard to imagine further dramatic improvements. How do you answer today?*

Keith: We've made tremendous strides since 2014 (and in the past 20 or so years), and I think algorithms will continue to improve. Over time developers have learned more about what people prefer to hear and how subtle differences can make or break the perception of what is "good" processing.

We've also learned more about masking distortion from the ear and what we can get away with as far as different forms of distortion. Evolution will continue, processing will continue to get smarter, and the availability of wickedly powerful hardware will enable us to do things that were only imagined five years ago.

Oops, did I say hardware? Remember … what you have "running in the cloud" is actually running on somebody's hardware.

RW: *One expert says, "My perspective is that radio processing already attained a condition of 'hypercompression' years ago and there has been little further change in how loud one can make over-the-air audio." Do you accept that, and how do we break out of that plateau in the loudness wars?*

Keith: My goal, and I suspect that of most audio processor designers, has been to deliver to broadcasters a new processor that can be as loud on the air as their previous processor was, but be much cleaner while generating that same loudness.

Unfortunately, what most stations do is crank the new processor up until the distortion is back to about where it was before … and now they're 2 dB louder than before.

Don't be a wimpy station on the air but there's no need to blast listeners out of their car, either.

RW: *We understand AES loudness metrics are moving to a lower target level for content, streams, podcasts and on-demand file transfer, like metrics already established for online and over-the-top video. If radio stays with the current environment of modulation limiting, reception noise and lingering loudness wars, could radio see loss of audience due to listening fatigue?*

Keith: Loudness wars only seem to serve the egos of the individual stations, and I'm not aware of any research showing that louder wins even when the program content is poor.

I do agree, however, that a loud signal helps overcome noise. And I'll also agree that we should carefully manage the audio so that listeners aren't lunging for the volume control every time a new song comes along.

Listeners should get a smooth and comfortable ride with our station's audio; and the better and more pleasant that ride is — accompanied by something worth listening to, of course — the longer they are going to listen.

As professional people who have dedicated ourselves to this industry to perform our art, we intuitively know what can turn listeners off; and yet sometimes we still do it. Puzzling.

RW: *We read about how processing can mitigate FM stereo multipath distortion and reduce clipping distortion in source content. How can equipment buyers evaluate such claims, and could there be some kind of third-party scientific testing?*

Keith: The problem with evaluating anything that's not actually running in the field is that it's not actually running in the field, i.e., lab tests can only show what things do under lab conditions.

Stereo multipath mitigation is a good example, and one must understand that it is receiver behavior that needs to be modified.

The technique that Wheatstone uses is something that I designed back in the '90s for solving a different problem; mono loudness when airing ping-pong stereo recordings (oldies). It cured that problem very nicely but it also had a greater-than-expected effect on multipath on most stereo radios. Customers have reported similar findings in the field and while it doesn't help everyone, it appears to help most.

RW: *What's your take on the demo from Nautel and Telos to eliminate alignment issues by locking the FM and HD1 outputs from the processor through the HD air chain to the transmitter?*

Keith: Great idea, and extremely similar in function to the SyncLink product Wheatstone demonstrated at NAB 2017. A guaranteed way to preserve FM/HD synchronization over an IP STL is to ensure that the two audio signals always look like one signal to the link. That way, even if packets are dropped the two signals can never get out of sync.

We also recognized that not every station can afford shiny new state-of-the-art transmitters so we designed SyncLink to be compatible with every single FM transmitter and exciter ever made.

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FOTI

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Foti: Loudness is really only a problem if it's accomplished in an annoying fashion. That's not being said to promote loudness. It is possible to create a "standout" loud on-air signal that is not annoying.

It comes down to the processor involved, as well as who sets it up. The term "hypercompression" can be defined differently based on interpretation.

I know there are some who absolutely love the sound of "deep compression" and the effect the added intermod it creates, whereas there are others who use less dynamic compression and rely on the final limiting system for their end result. Both are capable of generating large levels of RMS modulation, yet result in dramatically different effected signatures.

Is one better than the other? It's all very subjective, as well as what is truly to be defined as hypercompression.

"Processing can be installed, adjusted, modified as a system, moved, updated and a host of other utilities from basically anywhere in the world."

RW: *As John Kean writes elsewhere in this ebook, AES loudness metrics are moving to a lower target level for content, streams, podcasts and on-demand file transfer, like existing metrics for online and over-the-top video. If radio stays with its current environment — modulation limiting, reception noise, loudness wars — could radio see loss of audience due to listener fatigue?*

Foti: Any broadcast facility that has lost audience due to listener fatigue needs to realize this occurred due to their approach to audio processing.

Loudness is not the issue. It's how one achieves a loud signature that determines the listenability of a signal. There is a difference between the perception of a good clean loud signal, and another which sounds like your head is squashed within the jaws of a vice. Both are loud, but both are not bad.

It really comes down to choices made by the broadcaster. Analogy: A car that goes fast is not necessarily a reckless auto. It comes down on the driver of the car. Same applies here.

RW: *We read that processing can mitigate multipath distortion and reduce clipping distortion in content. How can users evaluate such claims?*

Foti: Great question! I've done significant work in this area, and have recently created a method to test, and observe the effects of induced multipath, based on audio processing. Surely, it could be further developed, as a tool for broadcasters.

As of this writing, there is nothing on the market, but there are technical papers that address it. Suffice it to say, I'd be very weary of those who make *ad hoc* statements about multipath, exaggerated by processing, that were done without any technical evidence or test criteria or employed good engineering practice.

RW: *Nautel and Telos recently did a joint demo aimed at eliminating alignment issues by locking the FM and HD1 outputs from the processor through the HD air chain to the transmitter. What's your take?*

Foti: Having been in some of the discussions about this method, this is a solid design that negates outside/ancillary devices to monitor and adjust the time alignment. This is the first systemic approach, which further solidifies the digital transmission infrastructure. It's very straightforward in design, and reduces the level of complexity within the digital transmission system.

We need to remember that as HD Radio evolved and refined itself, the overall system and infrastructure has had to change. Now that the tech has become mature, it's possible to create a method that efficiently and reliably creates the broadcast signals for conventional and digital transmission.

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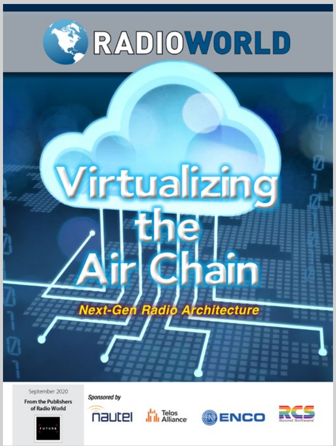

The concepts of virtualization and putting parts of the air chain in the cloud were already on the minds of forward-looking managers; the 2020 health crisis has accelerated the trend.

We sat down with leading industry technologists for a roundtable discussion. Find out what Roz Clark, Alan Jurison, Shane Toven, Philipp Schmid and Greg Shay had to say about this pressing topic. Read it at [radioworld.com/ebooks.](radioworld.com/ebooks)

ORBAN

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For example, its relative simplicity causes it to handle speech and music such that speech needs to be normalized about 3 LU below music for an esthetically pleasing balance between speech segments and music segments in a program.

As for LRA, its main values in the context of processing are, first, to help users assess if a single BS.1770 integrated loudness measurement corresponds well to perceived content loudness (high-LRA content will have parts whose short-term loudness is very different from its integrated loudness value), and second, to help users decide if dynamic range reduction for high-LRA content will provide a better listening experience to listeners in typical environments.

As for monitoring capability, most of Orban's Optimod-FM processors and all of its streaming processors — Optimod 6200, 1101e, and 1600PCn — have had built-in BS.1770 loudness metering for several years, and some also include the CBS loudness measuring algorithm, which uses a more sophisticated psychoacoustic model than BS.1770. Additionally, Optimod-TV 8685 provides loudness measurement and automatic logging.

No Orban processor displays LRA, but our free loudness meter software for Windows and MacOS (http://orban.com/meter) does this and more, and also allows logging and file analysis.

"I find it interesting that there seems to be a backlash developing regarding putting everything in the cloud, with some players moving infrastructure requiring high performance back from the 'cloud' to the 'edge.'"

RW: *Has processing attained a state of "hypercompression" from which there has been little change in how loud one can make over-the-air audio?*

Orban: I agree that this is true for FM processing, and most improvements in FM processing are refinements. However, our new XPN-AM incorporates our MX limiter technology for the first time in an Orban AM processor, and this has enabled as much as 2 dB of increased modulation density for a given perceived distortion level compared to previous Orban AM processors. This provides meaningfully improved ability to increase coverage, to reduce power bills when using AM transmitters with dynamic carrier control technology, or to split the difference.

Given the ever-increasing amount of noise in the AM band and the financial challenges of maintaining an AM operation, we feel that XPN-AM processing helps support the economic viability of the AM service.

For both AM or FM, more sophisticated processing algorithms enable higher levels of perceived quality for a given loudness level, and these advantages remain if broadcasters choose to back off average modulation levels to improve quality.

RW: *We read the processing can mitigate FM stereo multipath distortion and reduce clipping distortion in source content. How can buyers evaluate these claims, and could the industry develop third-party psychoacoustic testing to learn how listeners rate these features?*

Orban: Orban backs up its claims in this area with white papers and conference presentations that show objective measurements supporting our claims. Several of our product manuals include the white paper "Measuring the Improvements in Optimod-FM xxxx's FM Peak Limiting Technology," and I have been doing presentations at local SBE chapter meetings that include measurements showing how our "Multipath Mitigator" phase corrector reduces the peak and average L–R stereo subchannel modulation. This reduces multipath distortion because it is well-established that the stereo subchannel is much more vulnerable to multipath distortion than the stereo main channel.

While it is of course possible to do third-party scientific testing that further backs up these claims, we believe that each station's situation is unique, particularly regarding its multipath environment, and that the most significant testing is on-air testing at a given station's own facility. Our processors offer user the ability to turn the improved algorithms on and off, so it is easy to do comparison testing.

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TELL US WHAT YOU THINK

Radio World welcomes your opinions about the comments expressed throughout these articles. Send email to radioworld@futurenet.com with "Letter to the Editor" in the subject field.

BARBER

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Unfortunately, the problem was exacerbated by splitting up the system and not keeping the importer and exporter at the same location, nor keeping them time-locked together via GPS. In addition you had latency and packet issues that would wreak havoc on the FM and HD1 alignment.

Though there are new processors and equipment on the market that should keep things in alignment, the majority or equipment still in service still has huge drift issues.

Shown here is a picture of FM/HD1 drift over a 20-day period on a local station. This is not a small market off in the corner of some small city or county. They either need to replace *all* their HD Radio equipment, or get a JUSTIN 808 Time Alignment Processor from Inovonics. Our box goes in-line with the HD1 audio and continuously monitors the alignment of the two audio signals. When the alignment drifts, samples are slowly added or subtracted from the air chain until the FM and HD1 audio is aligned. It's really that simple to fix.

That picture shows a drift of 20,000 samples which is nearly 0.5 second!

RW: *In 2014 when we visited processing in an ebook, we thought radio processors were so powerful and had such incredible algorithms, that it was hard to imagine where further dramatic improvements would come from. How do you answer that today?*

Barber: I more or less agree. Today's DSPs are so powerful that the issue no longer becomes processing power, but the intellectual property of making algorithms function in a way that makes things sound exceptional.

Inovonics' goal in designing and manufacturing audio processing has been to design a quality product that is innovative and gives exceptional results at an affordable price.

I like to use the analogy of driving a car when comparing audio processors. It would be hard to argue that a McLaren 720S, Lamborghini Aventador or Ferrari 488 are not incredibly magnificent automobiles and take driving to a whole new level; but, for most of us, a solid Mercedes, BMW, Chevy, Ford or Toyota are probably quite sufficient to get the job done of a "daily driver."

Again, taking nothing away from the supercars of today; but you will see a lot more "regular" cars on the road as we go about our daily tasks. The honest truth? That's where I see Inovonics fitting into the processor market: a solid, dependable, reliable, innovative audio processor for the "regular" guy.

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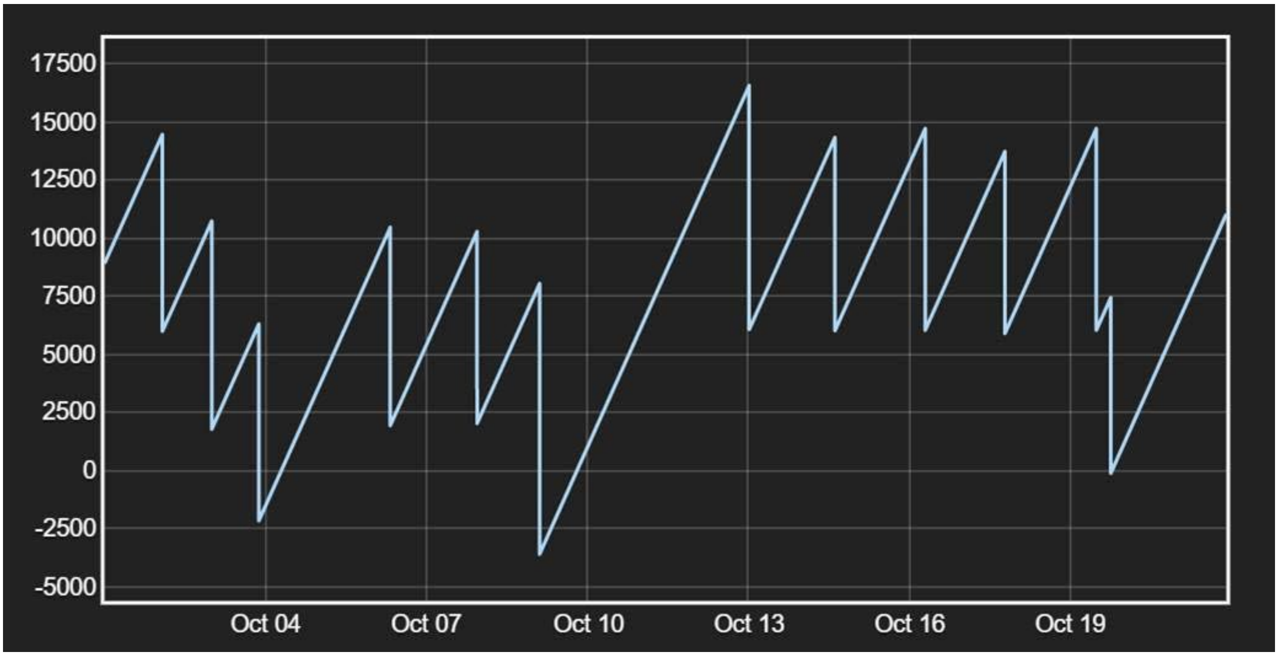

FM/HD1 drift over a 20-day period on a California station.

BRENTLINGER

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RW: *With audio coming from just about everywhere, what role do loudness and loudness range play?*

Brentlinger: Most new modern digital audio processors for digital transmission are able to monitor loudness and to comply fully with BS.1770 safety limiter for the CALM Act for TV transmission. However, I've compared both with Bob Orban and I feel the CBS Loudness controller is much more effective and outperforms the BS.1770, and this has been confirmed by double blind tests.

RW: *Are younger listeners moving toward greater fidelity because of their use of on-demand services and personal downloads?*

Brentlinger: I don't think that many young people really understand how to achieve quality audio, unless they are exposed to a studio or home environment with professional-quality speakers and amplifiers. From my experience with my own children and their friends, most young people are only concerned with the bass and the distortion caused by the rattling of their trunk lids.

RW: *Has radio processing attained a condition of "hypercompression," and how do we break out of that plateau in the loudness wars?*

Robert Leembruggen: Look-ahead limiting and clipping catches everything. Minimizing distortion is the job now.

RW: *Is it possible to make further dramatic improvements in processing, given the work that has come before?*

Leembruggen: We can always improve on distortion, number of bands, attack and ratio adjustment capabilities. Choosing 2:1 AGC with 50 mS limiter attack is a great place to start on an Orban 8600 MX preset.

RW: *What's your view on the demo by Nautel and Telos that they say eliminates alignment issues by locking the FM and HD1 outputs from the processor through the HD air chain to the transmitter?*

Brentlinger: I think it's a great idea.

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