AUDIO PROCESSING FOR HD RADIO: ISSUES AND CONSIDERATIONS FOR A SEAMLESS TRANSITION

Frank Foti
Omnia Audio
Cleveland, Ohio

ABSTRACT

With Digital Audio Broadcasting (HD Radio) utilizing In-Band-On-Channel (IBOC) technology in the USA, there is the need to consider the process in how this new transmission service will be implemented. One aspect that demands consideration is audio processing. With HD Radio, processing will be employed more so as a tool instead of a competitive weapon. Due to the employment of a codec as part of the system, a dynamics processor can be used to actually improve the efficiency of an encoder as well as improve upon the intelligibility of the source signal. This presentation will provide insightful information relating to these aspects.

Will the days of the loudness wars be over? Maybe or maybe not, but the areas that need to be addressed for HD Radio are much different as compared to the competitive nature of analog broadcasting for FM and AM. This is the dawning of a new era, and we have the technology in place today, to insure a seamless transition.

A CHANGE UPON THE HORIZON

Recently, over the past few years, as development and testing has progressed regarding the proposed IBOC transmission methods, audio processing has been one of the key components being considered. It became readily apparent that dynamics processing would figure in both the aural and technical performance aspects of the transmission system, which has also been the case regarding signal processing for other bitrate reduced audio services like the internet. The interest here is in the implementation and usage for HD Radio. Opening up a dialog about audio processing for HD Radio requires more thought and insight than one might imagine. If you think it would be easy to just *pencil in* your favorite flavor of processor and be done with it, then I ask you to think again! There are some definite issues that must be considered, or the benefits will not be realized.

The item that must be understood is that *all transmission mediums are not the same!* This author is mildly amused at the number of individuals who think that using an old FM-Stereo Limiter will suffice to process a narrow band internet audio service. Then they do not understand why the coding artifacts seem to be enhanced rather than suppressed. This analogy holds water here for HD Radio. This is why we have dedicated processors for FM-Analog, AM, Television, and the Internet. For FM-Digital, or HD Radio, there is no exception either. As the change upon the horizon becomes reality, we as an industry must give thought to the complete transmission system, which includes the audio processor.

AUDIO DIFFERENCES REGARDING HD RADIO (IBOC) AND FM-ANALOG

Before considering processing for HD Radio, the technical differences between the two mediums needs to be understood. The most obvious difference between the two is that HD Radio (FM-Digital) has a wider audio bandwidth of 20kHz, as compared to FM-Analog, which offers only 15kHz. Later we will see why this is very important when considering the sampling rate of an audio processor.

Another difference, and this is significant, is that HD Radio does not employ any form of *emphasis* in the audio path, whereas FM-Analog does. The audio path for HD Radio employs a *flat* spectrum, which negates the hurdle of having to manage the high-frequency boost generated by 75µs preemphasis, as used in FM-Analog. There are two interesting issues relating to this: The first is that the high frequency content of the HD Radio system will sound cleaner. This occurs because there is not a 17dB boost at 15kHz that further drives the HF spectrum into the final limiter. This definitely changes the *effected* perceived sound of the transmitted audio, which leads to the second item. Due to the nature of the preemphasis boost, it creates what some industry insiders refer to as the *sounds like radio* effect. This author can already hear the desires of program directors who claim that HD Radio doesn't sound the same as FM. No matter how much processing they will employ, the aesthetic sound will differ and this is one major reason why. What transpires in the FM-Analog system is that the preemphasis boost is usually coupled into a final limiter design that employs distortion control. The effected sonic result of this method yields the *radio-like* sound that many of us are used to. Processing for HD Radio will change all of this, as the HF content will appear dramatically much more *open* and clear sounding. This is in addition to the added 5kHz of audio bandwidth.

PROCESSING FOR HD RADIO: THE LANDSCAPE HAS CHANGED!

The important consideration regarding processing for HD Radio is understanding what is needed in a processing system. The challenges for a processor are much different for HD Radio as compared to FM-Analog. Consider it a quick review, but in the FM-Analog system, the processor must provide precision peak control to guard against over-modulation, management of the preemphasis curve to avoid audible distortion generated by the processing, and offer a brick-wall filter to protect the 19kHz pilot signal of the multiplex stereo system.

A processor for HD Radio has a completely different set of requirements. The most important issue is in dealing with data reduced audio. This poses many different conditions than what is needed when processing for the FM-Analog carrier. A HD Radio processor must be able to manage the audio spectrum in an efficient manner. Just as the processor for FM-Analog must manage the preemphasis curve. The processor needs to be thought of as a *partner* with the audio encoder. This is analogous to how a processor for FM-Analog works as a partner with the stereo multiplex encoder. In this case, the processor has the ability to understand, in advance, what needs to be done in managing the audio spectrum so that the least amount of coding artifacts are created. It is possible to predict what spectral conditions will exist that can generate audible artifacts due to coding. Dynamic algorithms in the processor can offset these conditions and in many cases remove unwanted artifacts, especially at higher bitrates, such as 96kbps. In essence, the audio processor can improve the efficiency of the encoder.

While peak control is required in order to keep the modulation input from exceeding the full-scale headroom limit of the system, the aggressive function like clipping is not required, and actually becomes a deterrent to the codec process. There is the need for peak control, but there possibility for over-modulation is removed as the audio level can not exceed digital full-scale of the encoder.

The End Of Clipping? Precision peak control can be achieved using numerous methods. Probably the most common is the hard limiter, or peak clipper, as it is known. By truncating the peak segment of the audio signal, precision limiting is achieved, and over-modulation is avoided. Most audio processors designed over the last twenty years also employ some form of distortion masking means as a tool to suppress the Total Harmonic Distortion (T.H.D.) that is created by the clipper. This makes it possible to utilize more of the clipping function, which directly transposes into perceived over-the-air loudness.

Employing a clipper as a peak limiter in a HD Radio system will work with regards to precision limiting, but there are sonic penalties to be paid when considering the clipping by-products and the encoder. Any clipping process will yield harmonics of the fundamental source signal. Even with distortion masking employed, there will, at least, still be some second order harmonic content remaining. It is this added content that upon entering the encoder process, adds to the audio spectrum and aggravates the encoder, which in turn yields additional sonic artifacts. This is especially noticeable in the high-frequency range, which is also where most codec artifacts seem to exist. This problem is extremely noticeable when trying to process for low bitrate audio and trying to use a clipper as the peak controller. Therefore another form of peak limiter is needed.

Hello Mr. Look-Ahead Limiter! There is another form of peak limiter that is the perfect companion for the HD Radio application: The Look-Ahead Limiter. The reason that is suits this application so well is that while it provides excellent peak control, it does so with very little, if any, harmonic content that can adversely affect the encoder.

Following is a quick look at how a basic look-ahead processor operates. In essence, the processor has the ability to calculate the peak level of a signal over a specified period of time. While that is occurring, the audio is delayed by a like amount. Then as the control signal is applied to the audio gain function, the audio peak is reduced at the precise time that the control signal reaches the maximum control level and the crest of the peak is reduced without truncation. This is how clipping is avoided. The following diagrams show a simple view of how a look-ahead processor operates:

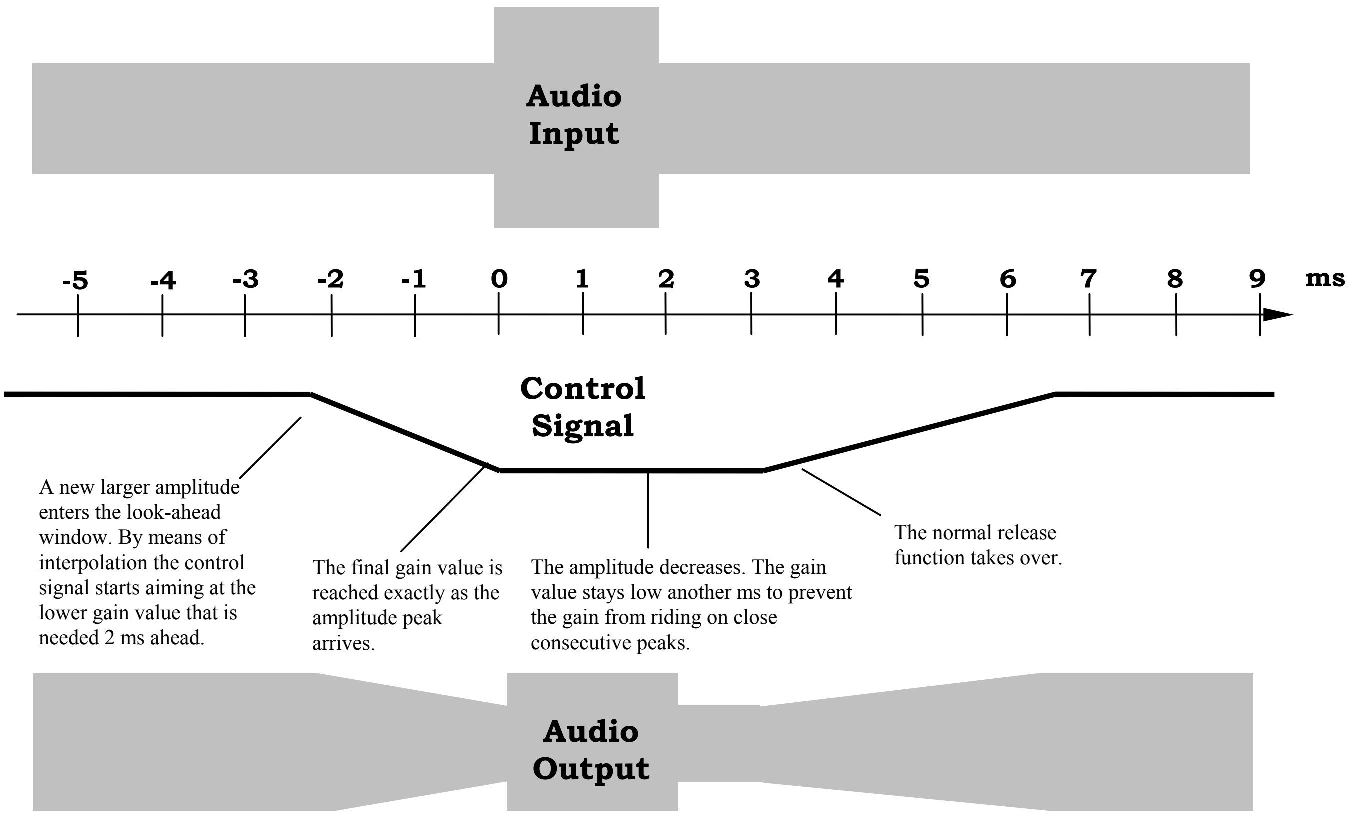


Figure-1

As the examples show, peak control is achieved without creating any harmonic distortion. If the diagrams were expanded to show detailed sinewaves, there would not be any peak truncation during the period of processing. Unfortunately there is no free lunch! A basic wide-band simple look-ahead processor will not create harmonic distortion, but will generate Intermodulation Distortion (IMD). This type of distortion has a different type of sound to it, sort of a *busy* quality and can be as annoying as harmonic distortion (THD) especially with music.

Innovative Algorithm Design! All is not lost! The difference between these two forms of distortion is that THD is hard to eliminate and will contain out of spectrum components. (Although some of those can be removed through precise filtering.) In the case of IMD, we know what will cause those products and how often. Therefore a processor can be designed to take advantage of look-ahead calculation time and add ancillary control signals that will monitor and remove IMD. Now music can be controlled in a precise manner, and the audio quality will not suffer from the *busy-ness* of IMD as it is removed.

Additionally, a processor designed where the time constants, *attack and release,* are optimized in a manner that provides a transient *feel* to its operation will sound very natural. In order to achieve this, the *attack and release* must be set faster as frequency increases. In the case of a look-ahead processing system, it will require different processing delays for each audio band. The following Block Diagram will provide a functional overview of a single band of processing:

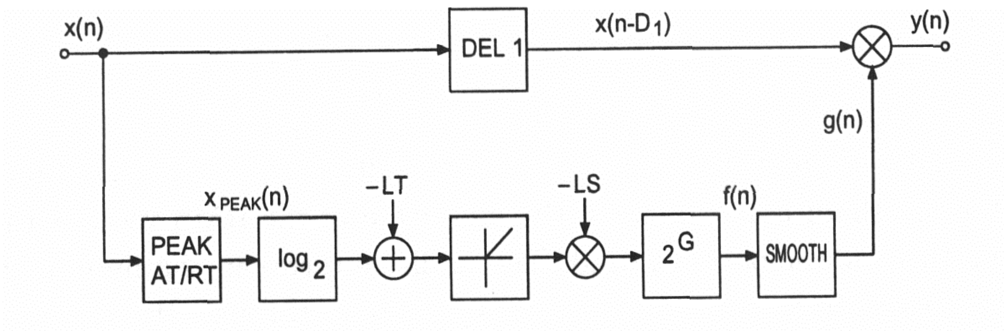


Figure-2

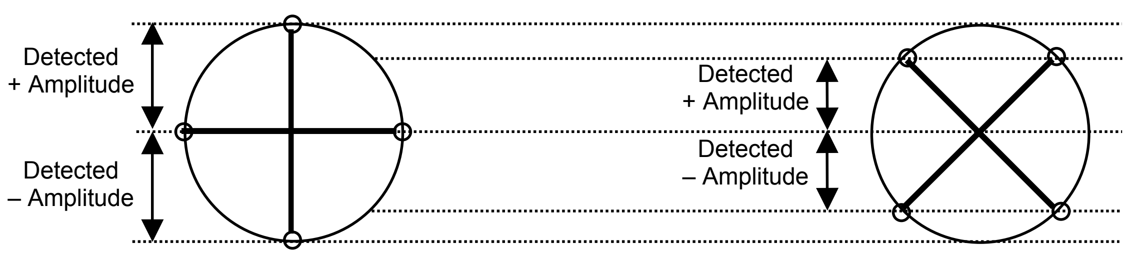
SAMPLE-SLIPPING: A MUST TO AVOID!

Not so very long ago in this same technical forum, discussion raged about peak control in digital processors for the FM-Analog system. "Concerns" were raised about sample-slipping with relation to how peak clippers were implemented in DSP. Well the exact same issue must be considered regarding the Look-Ahead limiter, or as Paul Simon once sang, those peaks will be "*Slip Slidin' Away!"*

This issue here is in reference to signals that fall near even multiples of the sampling frequency. What may occur is a rolling of the phase relationship of the sampling frequency multiple, and the specific frequency fundamental. It is possible that the peak level of signals near the even multiples of the sampling rate will not be fully detected. The illustration below provides more detail on this topic:

O = Sample Period Is Even Multiple Of Fundamental Frequency.

O = Sample Period Is Near Even Multiple Of Fundamental Frequency.



Control Signal
Figure-3

When the above scenario presents itself, overshoots will result on account that the correct peak level is not detected in the control signal of fundamental audio signals that approach even multiples of the sampling rate. By example, if the sampling rate is 48kHz, then the potential for this problem would occur at 16kHz, 12kHz, 8kHz, 6kHz, 4kHz, and possibly 3kHz. As the fundamental frequency becomes smaller in size, the degree of slippage is reduced. This overshoot characteristic can be observed in the following scope bitmap. The *ripple* is the difference frequency between the even multiple of the sampling rate, and the fundamental frequency where the sample slippage is occurring.

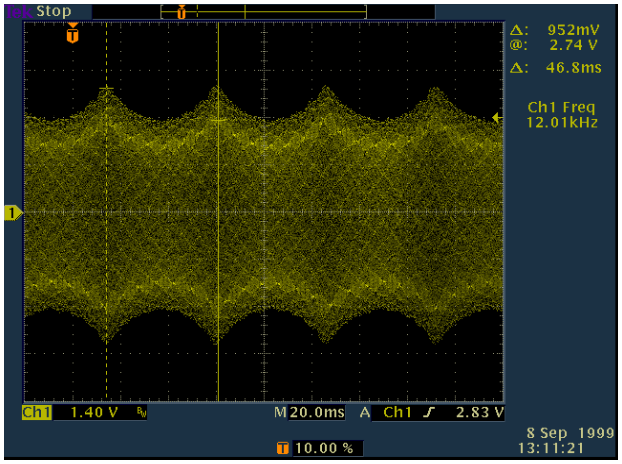


Figure-4

The above graphic is of a commercially available audio processor designed for HD Radio and Internet Audio applications. It is quite evident that overshoots exist due to sample slipping. The test for this was done using a tone set to 12,010Hz, which is an even multiple of 48kHz sampling. As the audio signal approaches 12kHz, one-quarter of the sampling rate, the *ripple* effect occurs. The next two captured scope illustrations are of the same processor's performance using program material. These displays were recorded using the storage persistence mode.

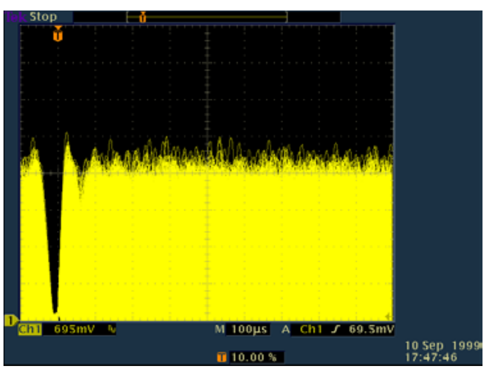


Figure-5, 15kHz Bandwidth

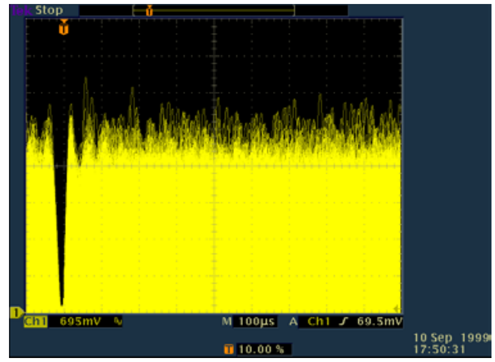


Figure-6, 20kHz Bandwidth

As a matter of reference, the peak level output was set up using tone that activated the look-ahead limiter. This is verified in the following scope capture:

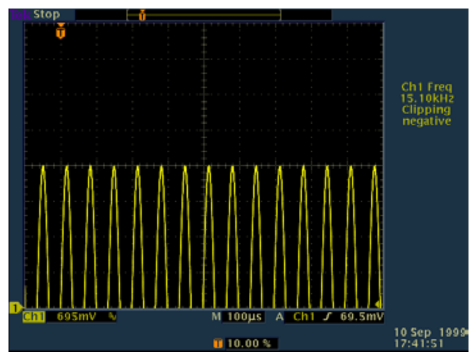


Figure-7, Reference Level

When compared to the two previous screen captures of program material, the overshoot content is quite severe! This is related to the above noted *ripple* issue in this limiter. This is not the fault of the look-ahead limiter concept, but of the algorithm design!

By contrast, the following screen captures will reveal a well-designed look-ahead limiter system. The first scope capture is of the audio signal that is near the one-quarter corner frequency of the 48kHz sampling rate. Note that the previously observed *ripple* is non-existent.

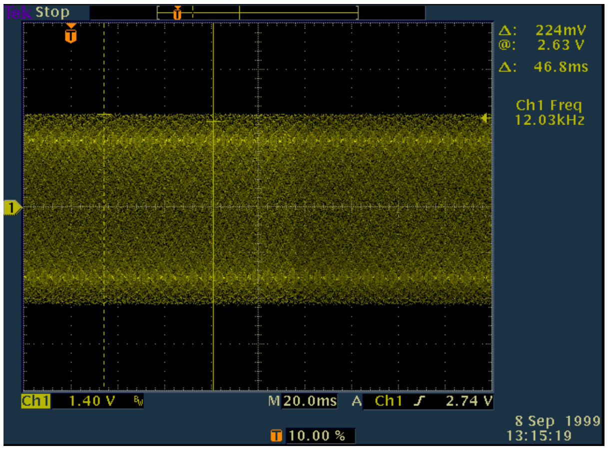


Figure-8, No Ripple

Now the same program is applied as before, and the scope captures are of the persistence mode. Notice the absence of overshoots in both the 15kHz and 20kHz spectrum captures.

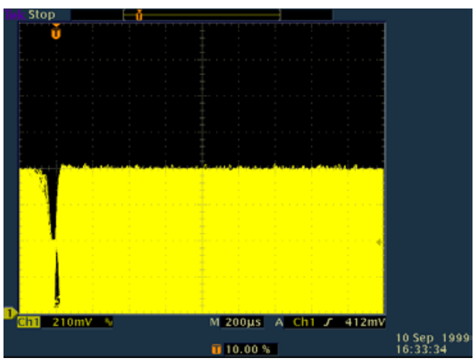


Figure-9, 15kHz Bandwidth

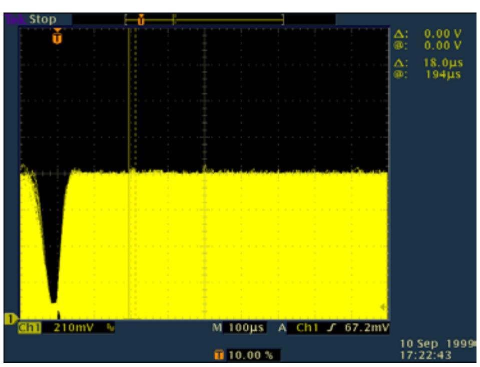


Figure-10, 20kHz Bandwidth

It stands to reason that a well-designed look-ahead limiter requires as much forethought as the conventional audio clipper. Both limiting means are required to provide precise peak control, yet accomplish this feat will minimal audio degradation. Notice that in the previous scope captures of the processed audio that contained overshoot, this system would exceed the HD Radio headroom of the encoder, as most systems are designed to operate within a few dB of digital full-scale. The overshoots that are displayed here are in excess of 2dB.

TRANSMISSION SYSTEM CONSIDERATIONS

Blend-To-Analog Integrating an audio processor into the HD Radio transmission path would appear to be straightforward, and in concept it is. But there are a few items that need consideration. Of importance is the issue regarding the *blend-to-analog* mode in the receiver. This function is designed to switch the receiver to the analog audio path whenever the digital signal is lost for an extended period of time. Even though the IBOC system has design time diversity into the transmit/receive path, so that the audio from both the digital and analog signals arrive at the same time, there is an issue of audio spectrum and phase relations that must be considered between the digital and analog transmission paths. Should there exist significant phase relationship differences across the audio spectrum of the HD Radio and FM-Analog signals, then the *blend-to-analog* mode will not appear as a smooth transition. If this occurs, it will be audibly perceived.

Therefore it is imperative that the two audio processors employed for the HD Radio and FM-Analog paths have the same, or very close to the same phase relationships across the audio spectrum. It is understood that the FM-Analog signal will possess less spectrum, as the FM-Stereo system will only allow a 15kHz audio bandwidth. So it is essential that phase linearity exist over that range of spectrum between the two transmission paths.

STL System As mentioned previously, HD Radio offers a broader audio bandwidth of 20kHz. This will require an STL Link that is capable of supporting the wider bandwidth. It stands to reason that the system should also employ a sampling rate of at least 44.1kHz or even 48kHz. (The iBiquity IBOC system operates at 44.1kfs.) The issue of sampling rate should also be considered for the processing systems too. Since it has already been mentioned the importance of linear phase relationships between the HD Radio and FM-Analog systems, this can be easily accomplished by using processors that employ a common design *and* sampling rate! Time alignment between two different sampled systems is possible, but it adds needless effort to the overall system.

PROCESSING AS A TOOL, NOT A WEAPON!

All too often the application of processing in the broadcast chain is employed to a level where it's thought more so as a weapon. The new landscape that HD Radio offers, requires it to be considered more to be a *tool* rather than the arsenal that normally transpires. Due to the ability of the processor to enhance or improve the efficiency of the audio encoder, it will act as more of a partner or tool to the transmission system. At low bitrates, processing will actually improve the intelligibility of the perceived audio. Listening to any low bitrate Internet stream that has a well-tuned audio processor applied to it will verify this belief.

Processing for *effect* is still possible; make no mistake! Creating the appearance of that larger than life big *phat* sound is quite possible. But, that synthetic smash-mouth sound that is characterized of many FM-Analog stations is far less possible in HD Radio. This is due to the removal of the clipping method and no need for preemphasis.

THE END OF LOUDNESS WARS?

Here's where most program directors that might be reading this are ready to have the author shot!! But, this statement might be true. In order to comprehend this, we need to consider how the HD Radio receiver operates. When selecting a HD Radio station, there is an internal buffer inside the receiver that must fill itself up before HD Radio audio is routed to the speakers. Depending upon the receiver design, this buffer may take a few seconds or longer. So, when switching between stations, the analog signal will be the first to be heard. HD Radio audio will follow, once the buffer has been restored in the receiver. When this happens, it becomes quite difficult for the ear to *remember* the loudness level of one HD Radio station as compared to the next. The ear normally has a retention memory for only a few milliseconds.

Also of note, the ability to gain added loudness via over-modulation is not possible as the encoders are designed to operate near digital full-scale. So the days of over-modulation will cease! Trying to exceed full-scale only results in drastic and annoying distortion.

FURTHER ON THE HORIZON…

As HD Radio deploys, the author is optimistic that the transmission system configuration for HD Radio and FM-Analog will evolve as both our industry and marketplace determine it's applicable needs. At present, we're enamored with the prospects of transmitting digital high-fidelity audio. But, we're all well aware of the additional possibilities of the available spectrum for data transmission too. There is even the eventual plan of eliminating the FM-Analog channel altogether at some point.

Thus, we may not know what the eventual packaging of the needed transmission equipment might be. At present, there are designs under development for combo transmission devices for the Internet. Where the dynamics processor is packaged along with the audio encoder. This type of device is possible for HD Radio too. Then, we need to consider what can be done in the processor so that the audio for both the HD Radio and FM-Analog channels are kept in proper alignment with regards to phase and latency. All of these issues are still under consideration by the developers of the transmission system, as well as those whom will provide towards this technology.

As an industry, all of us need to consider the best possible ways and means to deploy this system. Whether or not you are a proponent of HD Radio, it's still worth noting that we have the exciting opportunity of seeing a new technology come into being. Think back on how exciting it must have been to be involved with the launch of FM Stereo, or Color Television.

That same excitement now encompasses our industry once again. We have the rare opportunity to further the performance capabilities of radio broadcasting — this time for both FM *and* AM. I firmly believe that, with some thought and planning, the launch of HD Radio can be not only seamless, but a shining technological triumph that will benefit broadcasters and listeners alike in the years to come.