Enhancing the Digital Path: Digital Multiplex (D-MPX) Connectivity

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ABSTRACT

The all-digital transmission path is quite common in today's FM radio broadcast facility: from content-source on through to the modulation and RF generation stages. With the above system, stereo generation (MPX) must be done in the exciter, when it should be performed in the audio processor. Present systems, while they work electronically, pose problems with regards to modulation overshoots due to sample rate converters and the connection method of using AES/EBU. This has been researched and documented with numerous exciters and audio processors. This problem is not limited to just one type of configuration or specific product.

This paper offers an in-depth look at these problems, and states their cause. Discussion will reveal a new concept for high performance MPX generation and interconnectivity to an audio processor. In addition, with the advent of HD Radio a new concept for a high performance digital path is presented.

OVERVIEW

The rollout of HD Radio presents a whole new level of challenge for digital FM exciters and related audio technology. What is the point of a clear, digital transmission if the source is "dirty"? Modulation performance of digital exciters for the analog, or conventional channel, continues to under-whelm broadcasters. We have found that using AES/EBU between the output of an FM audio processor and input to a digital exciter can cause modulation overshoots. This is not a unique problem to one specific processor or exciter.

The fundamental problem with AES/EBU connectivity is that the audio is in separate left and right audio channels, forcing the exciter to perform the multiplex stereo generator function. When ancillary operations such as sample rate conversion, additional low pass filtering, and exciter based limiting, are added to the system, overshoots occur.

To avoid this, the audio processor and stereo generator need to operate together and the composite multiplex (MPX) signal connected directly to the modulator stage of the exciter. Because this has not been possible, the audio processor has had to generate the composite signal and then convert the signal to analog in order to be passed on to the analog MPX input on the exciter.

This paper describes, in detail, a proposed interconnection of an audio processor and digital FM exciter. The methodology and protocol is open source, and could easily be standardized for the broadcast industry.

The importance is precision modulation peak control and spectral management of the mpx signal. If precise peak control is not achieved, loudness will be lost. The present AES/EBU interconnection method can generate peak overshoots have been measured as significant as +120% modulation. This relates to a loudness loss of approximately 1.5dB!

STATE-OF-THE-ART

Before we talk about new concepts for enhancing the audio path, we should review what most would consider today's state-of-the-art in terms of transporting audio information from the studio to the transmitter. We should also discuss the problems that still exist in this path and justify the rational for further improvements.

Clearly, it is possible today to achieve a fully digital audio path from the studio to the transmitter. Our analysis of the audio path will make the assumptions as described below.

Studio-Transmitter Links (STL's): *There are a wide variety of STLs available today, some RF based and some not. To achieve best audio path performance it is essential that the STL meet two criteria. It must transport audio digitally and it must not compress the* *audio data as some do in order to operate within a narrow channel bandwidth.*

The reason for digital transport is primarily due to the desire to eliminate ADC and DAC transitions and the associated distortion. Compression of audio data, particularly if the compression ratio is high, will degrade the audio data because it cannot be perfectly recovered at the STL receiver.

Additionally, it has been proven, through experience, that passing processed audio through a codec, creates added audible distortion. This occurs due to the effects of added harmonic products, caused by processing/clipping, being passed through a codec. Even if the audio processor is placed after the coded STL, aural anomalies can occur when data reduced audio is passed through heavy gain control, as is performed within a processor for broadcast.

Component Interconnect: *It has already been implied that transporting audio digitally is state-of-the-art. The standard protocol is AES/EBU and is regarded as perfectly adequate provided that reasonable sample size and rates are used. While some may argue that 32kHz sampling rate is sufficient, there can be little debate that higher is better. There seems to be some unknown reason why 32kHz sampling rate is used in broadcast paths. It has been discussed and demonstrated that 48kHz sampling is far superior in performance. The importance is not so much the added spectrum that 48kHz sampling provides, but that 32kHz sampling makes it too easy to cause aliasing distortion in specific path functions. This is clearly demonstrated by any DSP based audio processor designed before 1997, and extensively detailed by one of the authors in a NAB presentation*[2].

With 32kHz sampling, there is also the problem of filter delay. Considering that conventional FM Stereo broadcasting requires 15 kHz of audio bandwidth, this leaves only 1 kHz of guard band spectrum before the Nyquist point. To facilitate this, a filter of very large magnitude must be employed in order to suppress all energy by at least 96 dB at the Nyquist, or aliasing occurs. This can be done digitally using a finite impulse response filter (FIR). The only drawback is that it will require many 'taps' within the filter to achieve this level of stopband rejection. The significance of the 'taps' is that for every two taps in the filter, it requires one sample to perform its duty. For a 15 kHz FIR filter of this magnitude, it will need 101 taps. This in turn results in 50 required samples which equates to 1.56 milliseconds of propagation delay through the filter.

Start adding up the number of 15kHz filters employed in a 32kHz sampled transmission path, and those alone will create enough time delay to disorient on-air announcers.

The use of at least 48kHz sampling as the AES/EBU input rate in the exciter insures the best sonic performance of the signal being applied. In addition, any input that needed to be converted up from 32kHz sampling would not create any overshoot component in the modulator. All of which are benefits to the broadcaster.

If we now apply these assumptions to the studio-transmitter audio path we end up with something like that shown in Figure 1.

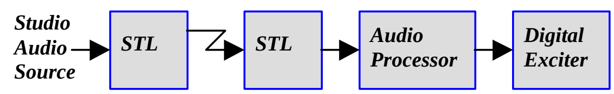


Figure-1

Whether the path includes HD-R is irrelevant for the purpose of this paper because it is a proprietary system with defined data transport. It is treated as a parallel system that does not affect the transport of non HD-R audio (other than the need to apply a delay).

The placement of the audio processor is shown to be at the transmitter site although it could just as easily be placed at the studio. While it may be easier to access the audio processor at the studio end, it will be shown that from a sonic performance point of view, the closer it is to the digital exciter the better the overall sonic performance. With a modern remote control system, there isn't much to be gained by having the audio processor at the studio.

The audio path shown in figure 1 should perform very well. Assuming that the components before the audio processor are good quality, there should only be minor degradation of the audio data as it passes through the sample rate converters in the STL's and audio processor input (an explanation of this will follow shortly). The audio processor will modify the audio data because that is what it is supposed to do. The audio processor 'processes' the audio so it will pass through the transmission path (modulation -> RF -> demodulation) and end up with a rich, pleasing sound to the listener.

Since the audio processor works hard to process the audio it would be beneficial if there were no additional sources of degradation after the audio leaves the audio processor. In fact, as we will see there are many opportunities for degradation, all inside the digital exciter. Let's take a closer look.

Digital Exciter

These are latest entry to the digital path. Capable of incredible modulation performance, the digital exciter offers two forms of signal input, analog composite (MPX) for non-digital transmission, and AES/EBU. The figure below shows both paths in a typical digital exciter.

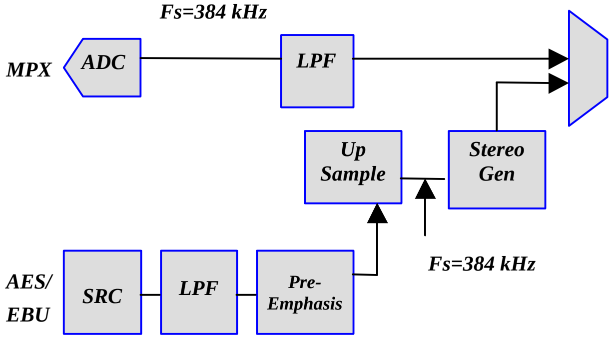


Figure-2

The analog MPX input must be passed through an analog-to-digital converter (ADC) before it can be digitally modulated. The sampling rate of the ADC must be relatively high due to the wide bandwidth of the MPX. Consider for a moment the audio spectrum of FM is 99kHz, any analog spectrum that is connected to a digital exciter must provide, at the least, a sampling rate of 200kHz or higher. Once digitized, no other processing is required before modulation.

The AES/EBU path is quite a bit more complicated than the MPX path because it must condition the audio signal in preparation for stereo generation and herein lies the potential for audio degradation. Some of the blocks, such as LPF and pre-emphasis, can be bypassed if the particular function is performed in an audio processor. Let's examine each in detail.

Sample Rate Converters (SRC): Consider for a moment the signal that is arriving at the AES/EBU input of the exciter. It might be a different sampling rate than the exciter is expecting. If so, a sample rate converter is employed to make the proper transition. This item can pose problems as the digital filter in the rate converter can generate overshoots to the already tight peak controlled audio data that is being adjusted. The following example illustrates how a synchronous SRC would work, however interfaces between equipment are not synchronous and therefore require asynchronous SRCs which are functionally similar but considerably more complex. In order to synchronously change 48kHz sampling to 32kHz sampling, the conversion is accomplished by scaling up, or interpolating the original sampling rate, usually by a factor of ten. Then, at the 10x rate of 480kHz, filtering the signal with a low pass filter that is set to the Nyquist of the new *desired* sampling rate. This filter is required to 'smooth' out the 10x rate. If it was not used, aliasing products would result. Finally, the signal is scaled down or decimated by the factor needed, in this case ÷15, to achieve the new rate of 32kHz. Figure-3 is a block diagram of a SRC. While this sounds quite simple, and basically it is, there are a few issues to consider. Of main interest is the interpolation filter.

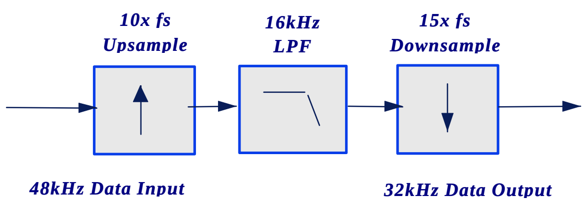


Figure-3

All audio processors, both analog and digital apply some form of overshoot control to the output filtering section. In most designs, this function is a form of integrated protection clipper working around the final low pass filter to obtain control.

In each case the overshoot component can be calculated as a product of what is known as the 'Gibbs Phenomenon'[1], which states that an overshoot will occur at one-third the cut-off frequency of any low pass filter whenever a non-linear waveform is passed through it. In the case of broadcasting, the non-linear waveform would be that of a clipped waveform.

Knowing that the audio bandwidth used in FM Stereo is 15kHz, overshoot components will begin with any non-linear waveform above 5kHz. In this example, this would affect any signal above 5kHz that was clipped.

Should the slope of the previously described up-sampled interpolation filter appear greater than the slope of the final filter in the audio processor, then output overshoots may result in the sample rate conversion process! Unfortunately, these overshoots are generated after the processing unit. To remove them would require another limiting device.

This does not necessarily indicate that all sample rate converters will cause overshoots. But in most cases the filtering used in the sample rate converter will be of a large magnitude in the bandstop rejection area. In all probability it will be an FIR filter with at least 96dB rejection in the stop band.

Of interest will be the direction of rate conversion. Should the host sampling rate be lower in value, than the transformed rate, chances of overshoot are small. This happens due to the up-sampled filter being set to a broader spectrum than the spectrum of the host signal. Potential problems may arise when transforming a larger sampling value to a lower rate, as described in the example of converting 48kHz to 32kHz sampling. Then the details of the above description apply.

Integrated Limiter: Some digital exciters provide a baseband limiter to eliminate potential overshoot problems. The use of a limiter can have benefits of preventing modulation error caused by components between the audio processor and exciter (including the exciter) but it can also undo the processing done by the audio processor by further changing the audio.

The integrated limiter used in exciters is incorporated as part of the stereo generator. Different types of limiters will add different effects to the audio. A hard limiter or clipper is the least desirable, it induces harmonic distortion and aliasing distortion (a.k.a. digital grunge). A time delay, look-ahead limiter controls peaks without a harmonic induced clipping function. Waveforms are controlled with little or no harmonic distortion (T.H.D.) components, but will produce a larger intermodulation (I.M.) level.

Technically, this style of limiter will operate sufficiently when controlling overshoot peaks, or as an additional limiter to the audio processing. Sonically, this type of limiter will produce a “busier” sound. It will sound more like a limiter that is operating in “heavy” levels of compression. That is the result of the added I.M. In the audio processing realm, adding more I.M. to an already processed signal is usually not desired!

Though not used in any digital exciters to date, the cleanest type of limiting is composite clipping, as long as the limiting algorithm provides linear filtering that will maintain clean spectra in the 19kHz pilot, and SCA regions. The key ingredient is the linear filtering method applied to the algorithm. Composite clipping produces far less audible I.M. products as does a delay limiter, and it will yield cleaner sound for the same amount of limiting/clipping used.

Preemphasis: The exciter has the option of adding the required emphasis. The optimum setup of the transmission system would be where the emphasis is generated once in the audio processor, and that emphasized, processed signal is coupled directly to the exciter.

Broadcast audio processors employ preemphasis within their system architecture. Since emphasized audio must also fit within the imposed modulation limits, the processor employs specialized high frequency control sections that provide both the emphasized boost and control of the high frequency energy. In this manner, efficient high levels of modulation are easily obtained since the processor is designed and set to limit any tradeoffs resulting from preemphasis and high frequency limiting requirements. Basically, these two sections work in concert with one another to allow preemphasis to be employed, and yet control the emphasized energy content.

Interpolation: Before stereo generation, there is one last bit of processing that must be carried out. Just as the analog MPX signal needs a high sampling rate due to its wide bandwidth, the digital L/R path requires a translation to a higher sampling rate (interpolation) before it is passed into the stereo generator because it will come out of the stereo generator as MPX. An interpolator is a digital signal processing function that inserts zero value samples in the audio stream (three per input sample for 4X increase in sampling rate) and then applies a filter to recreate the input signal at the higher sampling rate.

AUDIO PATH IMPROVEMENTS

There is no question that the audio path shown in figure 1 should perform very well, but can it be made to work even better? Yes, it can.

To summarize the preceding issues we see that placing the audio processor close to the exciter is best in order to eliminate sample rate conversions other than that in the digital exciter. We also see that superior performance is achieved if the audio processor does all of the audio processing including filtering, limiting and pre-emphasis because it is the best equipment to use for precise control the modulation depth.

Consider also that the audio signal at the output of the audio processor only requires stereo generation which is a function that is standard in audio processors. The reason for not using the stereo generator is that it requires converting the MPX signal to analog for transfer to the digital exciter where it will then be digitized again. The process of converting to analog and back to digital introduces unwanted noise. To transfer the audio and AES/EBU requires that it pass through a multitude of processing in the digital exciter in order to get it to the stereo generator. All of which can degrade the modulation efficiency, and sonic performance.

To further illustrate the issue of using AES/EBU between the audio processor and exciter, consider the following analysis performed by Omnia. It evaluates the performance of Omnia equipment with a digital exciter whose input sample rate is 32kHz. The tests were run using a several different output sample rates from the audio processor. In addition to analyzing the performance of the AES/EBU connection at various sample rates, the analysis is extended to compare the digital connection to a traditional analog MPX connection.

FM MODULATION ANALYSIS USING THE AES/EBU TRANSMISSION PATH BETWEEN AN AUDIO PROCESSOR AND DIGITAL EXCITER

Introduction: A hot topic over the past few years has been the issue of interfacing audio processing and digital exciters in the FM transmission plant. The issue, as discussed, has been centered around which sampling rate works best for AES/EBU interfacing, and what additional anomalies transpire in this configuration. This analysis reflects some recent observations made in one of the author's test facility regarding FM modulation control when using a digital exciter, and modulated via the AES/EBU path. For comparison's sake, observations were also made using the conventional composite multiplex (MPX) input to the exciter.

Following is an analysis that reveals modulation performance using a popular Digital FM Exciter. There have been quite a few questions from customers who want to understand which interface method is more desirable for broadcast usage…AES/EBU or the conventional MPX input. The findings are based upon tests that were done using an Omnia.6 audio processor, as well as, the Optimod 8400 system. It must be pointed out that these tests were done using the two units for comparative reasons only, *not* to compare the performance of one against the other! The objective was to show the consistency of our findings, and that it didn't matter which processor was used. As it will be revealed, they both perform relatively the same, within 1% of each other.

Test Setup: The test setup was quite basic. The goal was to observe FM modulation when the audio processor was connected directly to the FM Exciter using the AES/EBU input. The Exciter was configured to operate into a dummy load, which also was provided an RF tap for the modulation monitor. The monitor used was the popular Belar Wizard System, which included the PC monitoring function so we could grab histograms for usage in this report.

The test setup was as follows:

1. CD player connected directly to processor. Aggressive program material of rock music chosen.
2. Processor operating with AES output.
3. Aggressive preset <CHR> chosen
4. Main Clipper set to +3.0dB. This is quite extreme!
5. Output sample rates tested at 32kfs, 44.1kfs, and 48kfs.
6. Exciter operating into RF Dummy load.
7. Tap off of RF output connected directly into Belar Wizard Modulation Monitor.
8. Measurements made using Belar Remote PC S/W.

While this setup does not exactly replicate a radio station facility, it does provide the basis to determine if the audio processor and exciter are capable of generating well-controlled peak levels under modulation. To make the tests equal in *rigor*, the exact same segment of program material was used throughout testing. This removes any possible inaccuracies in the results. The audio sample was from the *Talking Heads* CD, "*Stop Making Sense.*" *NOTE:* All testing was performed with the Digital Exciter's internal limiter set to OFF, and the MPX testing was done without using any composite processing in either the Omnia.6 or 8400.

Test Results: The results were revealing, and for a number of reasons.

1. When using the AES/EBU path, there *does* still seem to be some overshoot present. The good news is that it is not at the wild levels that were observed with older generation exciters. The overshoot component appears to be somewhere between 2% **-** 4% of modulation. This was observed on *BOTH* the Omnia.6 and Optimod 8400 processors.

2. The output sampling rate does seem to effect the characteristics of peak control. When operating the audio processors in the 32kfs mode, overshoot appears to occur a bit more than when set to 44.1kfs or 48kfs. This was observed in both processing systems. Of note, is that the overshoot component level did not really change when using the lower rate, it just occurred a bit more often.

3. Observed was a new anomaly that bears discussion. The Digital Exciter itself seems to have a *bobble* in modulation of approximately 1%. This was confirmed when it could not be

set for total modulation to a precise level using a 400Hz tone. The test engineers tried to set the modulation reference via a tone using both Omnia.6 and the Optimod 8400. Neither unit was capable of being set for a precise modulation level of 100% on the Belar Wizard. There was always a *bobble* of a few tenths of a percent.

This last item may help explain the modulation *uncertainty* that exists under certain program conditions. Observed were moments where there is a modulation *spike* of a few percent. This *bobble* that was uncovered would help explain why this happens from time to time. In tests, this was observed on both the Omnia.6 and 8400 processor, which leaves one to believe that there's a systemic issue within the Exciter. Consider that this *bobble* can create an uncertainty of +/-1% that results in an overall 2% modulation *error factor*. When you add this to the minor overshoot anomalies of Sample Rate Conversion, it becomes a bit clearer as to what is happening.

Following are graphics that depict the modulation performance of both the Omnia.6 and Optimod 8400. These display performance at the three popular sampling rates of: 32kfs, 44.1kfs, and 48kfs.

Omnia.6 @ 32kfs Sampling

8400 @ 32kfs Sampling

Omnia.6 @ 44.1kfs Sampling

8400 @ 44.1kfs Sampling

Omnia.6 @ 48kfs Sampling

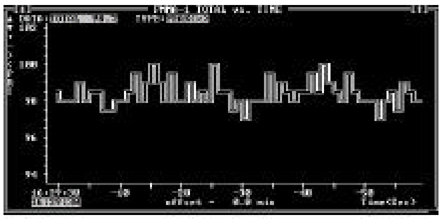
8400 @ 48kfs Sampling

It needs to be pointed out that these screen shots show the PC display of the *Belar Wizard* software after they have been magnified to a considerable extent. Any peak uncertainties are only over a magnitude of a few percent, which does not adversely affect coverage, loudness, or interference due to FM deviation. As the screen shots indicate, the level of overshoot is reduced when the sampling rate is increased.

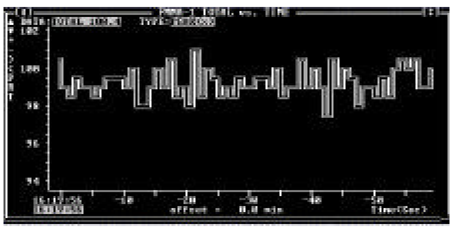
There is an explanation as to why there appears to be a bit more *grass-like spikes* on the Omnia. It has to do with the low pass filtering in the AES/EBU path, which employs a 15kHz low pass filter that does not truncate the passband to a zero stopband level at 16kHz. The rolloff slope of our filter is a bit more gentle, and thus there is a negligible amount of spectrum that exists in the 16kHz to 17.5kHz area. It is this spectra that is causing the slight amount of overshoot in the sample rate conversion filter of the Digital Exciter. The 8400 employs a base sampling rate of 32kfs, and thus must adhere to this tighter filter performance at 16kHz.

The question can be asked why not follow the same criteria? It was a design decision, and judgment, that employing a tighter filter slope of this magnitude will degenerate the audio quality. Research and testing, during the design phase revealed this. Considering that the magnitude of overshoot is not of any *additional* significance as compared to the 8400, it was decided to remain with the existing design. While the amount of modulation uncertainty appears a bit more often in the Omnia.6, it is not, in any way, a deterrent to the modulation performance. As the screen shots indicate, there is modulation uncertainty with *BOTH* processing systems. Had the performance of one system varied considerably over the other, then, there would be a reason for concern. But based upon these findings, the observed differences appear to be moot.

Now to contrast the above results, the same tests were run using the MPX input on the Digital Exciter. Notice that the performance of both systems appears to be relatively the same, *and* there is still some negligible overshoot. It is believed that this is due to the +/-1% modulation *bobble* as described earlier.



Omnia.6 MPX



8400 MPX

It is quite clear that modulation *uncertainty* exists under the present AES/EBU connectivity method. History has shown that this was never the case when the MPX output signal of an audio processor was connected to the MPX input on an analog exciter. Precision peak control that was achieved in the audio processor was passed on through to the exciter, and that would be the performance of the deviation in the modulator section.

The same performance that was achievable with the older generation analog exciter must be capable in the newer digital devices.

ADDING HD RADIO TO THE MIX

The addition of HD Radio creates some interesting propositions. At present, there's the need to employ two separate exciters: one for the conventional (analog) signal, and one for HD Radio. Seems a bit cumbersome to the thinking here.

Now is the time to design a single exciter solution, and at the same time, *finally*, address the modulation overshoot issue that does exist for the conventional channel.

DIGITAL MPX (D-MPX)

In the discussion section about the digital exciter, numerous options were explained about the interfacing possibilities of the audio processor to the exciter. All of them revolve around the usage of the AES/EBU input protocol. In that configuration the audio data arrives in Left/Right format and requires the exciter to perform the MPX generation.

Question: Why can't the digital audio processor, which already has the MPX encoder inside, be able to connect its digitally generated baseband signal directly to the digital modulator of the exciter? This would be analogous to the analog composite input on any exciter.

It is of interest to the authors why any of the digital exciters available today do not provide, or propose an application like this. It provides the best possible coupling to the exciter, and the performance benefits are significant. Imagine having the power of a complete digital processing system and integrated stereo generator that is directly connected to a digital modulator. Now we're talking about super efficient modulation capability. Zero overshoots due to added emphasis, coding, or sample rate converters. That would be real power!

Figure-5

Fortunately, a solution is on the horizon. Nautel limited is adding this feature to their latest FM exciter, the M50. Nautel and Omnia have agreed on a format for passing a digital MPX data stream serially using RJ45/CAT-5. The M50 exciter is Nautels first exciter to support hybrid and all digital HD-R. While the data format is not standard it is based on common serial protocol supported by a variety of CODECs and DSPs. Essentially, the interface is synchronous with the exciter providing a clock to the audio processor and the audio processor providing a data clock, data and frame sync on separate signals. All signals are differential LVDS. The sampling rate of this interface is approximately 372kHz which is almost 7 times the bandwidth of the MPX signal (without SCAs).

Naturally, this type of configuration would require installing the processing at the transmitter facility, since transporting a digital composite signal of this speed and size would be cost prohibitive. Locating the audio processing at the transmitter is not a problem, as all current generation digital processors provide some form of computer control via modem, or network.

CONCLUSION

The total digital transmission path is capable of providing outstanding performance results. To achieve this, audio processing must be inserted at the transmitter site, and a “flat” input should be used on the digital exciter. If a STL system is employed, a linear system would be preferable, but a high bitrate coded system is acceptable as long as the dynamics processing occurs after the coding.

The goal here is to digitally achieve the same technical model as the accepted method done in analog: Interconnection of the mpx signal from an audio processor to the modulator stage of the exciter. History has proven that this method yields superior peak control results, and allows the use of ancillary functions like composite processing.

Additionally, with HD Radio now a reality, there is reason to offer a complete digital excitation system that operates 100% in the digital domain, yet it provides input capabilities for any type of input, be it AES/EBU, or mpx.

As long as the systems design engineer in a broadcast facility is aware of these critical issues, there is no reason why an all digital broadcast facility can not exist today and not provide exceptional quality broadcasting. As technological development continues in a manner where products can provide faster and more powerful digital delivery methods, the concerns shared here will continue to dissipate until they are rendered meaningless.

REFERENCES

[1] Baher, H. Analog & Digital Signal Processing, J. Wiley & Sons, 1990

[2] Foti, F. Critical Issues And Considerations For An All Digital Transmission Path, NAB Convention, 1998