AUDIO PROCESSING & HD RADIO

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Hello, HD Radio!

We live in exciting times. Think about it… communications and computing technology have changed the broadcast industry in ways unimaginable just a few short years ago. While the PC and associated networks dramatically transformed how radio programs are produced, new transmission technology resulted, within just five short years, in DSB, HDTV, and now HD Radio. The last time broadcasting saw innovation with similar significance was nearly a half-century ago, when color television and FM Stereo were introduced. We're lucky to have the experience of working with these exciting new technologies – it really is exhilarating!

Is HD Radio what we, as broadcasters, need to take us into the digital era? Some may be skeptical, but in our experience the technology works and is ready for the world. Is it perfect? No. All engineering involves trade-offs, and digital radio is no exception. I'll bet that when FM stereo was introduced, it prompted a bit of nay-saying, too – some justified: audio bandwidth was reduced to 15kHz, and broadcast engineers were soon to encounter the infernal beast known as multipath.

Yet we did our best to learn and understand the medium, and we have improved the performance of the FM Stereo system over time. Certainly the same will be true with HD Radio. We, together, have the opportunity to take this technology and make it sound the best possible. Certainly there are aural limitations imposed by the FM Stereo system but we learned to live with them, and we will certainly have some aural challenges with HD Radio. Nevertheless, there are ways to work around those challenges and unlock the benefits of this system.

Over the past few years, as development and testing of the HD Radio system have progressed, audio processing has been one of the system's key components. Early on it became apparent that dynamics processing would figure in both the aural and technical performance aspects of the transmission system. This has also been the case regarding signal processing for other bit rate-reduced audio services, like Internet audio streaming. Opening up a dialog about audio processing for HD Radio requires more thought and insight than anyone imagined. If you thought 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 important issues that must be thought out for digital radio's benefits to be fully realized.

Fortunately, we at "Omnia HQ" are in a pretty good position to bring some understanding to the new topics that accompany HD Radio. We've been doing FM processing for a decade, and our sister company, Telos Systems, has been doing codecs even longer. We are both heavily involved in leading edge DSP development, and our development teams include many professional broadcasters. Working together, Omnia engineers learned how to optimize audio to deal with the limitations imposed by codecs and other bit-reduced environments – knowledge we found to be essential when tweaking audio processing for maximum HD Radio performance. (As a side note, this cross-pollination works the other way, too – the latest Telos codec product has Omnia audio processing inside.) We also had access to a fully functional HD Radio transmission system, with which we logged a lot of lab time listening and experimenting, so it's possible that we are a bit closer to understanding the processing demands of the new HD Radio system than the pioneers of FM Stereo processing were!

The major principle to keep in mind as we embark on our discussion is that *all transmission mediums are not the same.* Sure, this sounds obvious – but how many people do you know who thought an old FM Stereo limiter would suffice to process a narrow-band Internet audio service, only to find that coding artifacts seemed to be enhanced rather than suppressed? The same principle applies to HD Radio: a processor designed specifically for the new medium will be needed, and thought must be given to the complete transmission system. So, let's get started.

Differences Between HD Radio And FM Analog

Before considering processing for HD Radio, let's consider the technical differences between the two mediums. The most obvious difference is that HD Radio has a wider audio bandwidth, extending to 20kHz, compared to FM Analog's maximum 15kHz.

To achieve HD Radio's 20kHz audio response, your audio processor must support this bandwidth. This means that the processor needs to have a base sampling rate of 44.1kHz or above. (This is according to the Nyquist principle, which states that the sampling frequency must be at least two times the maximum operating frequency.) A processor designed for analog FM utilizing only 32kHz sampling *will not* provide the full audio response. HD

Radio was developed with the goal of providing CD-like audio; why then should we limit audio bandwidth on the digital channel to anything below 20kHz?

Another significant difference is that HD Radio does not use any form of emphasis in the audio path, whereas FM Analog does: 75µs in the USA and 50µs in Europe. The shape of the 75µs HF curve has a 2.2kHz breakpoint and a 17dB boost at 15kHz. As a long-time designer of processor algorithms, I can tell you that this is the number one challenge in producing clean audio from an analog FM channel; current processors must carefully manage the high-frequency gain generated by pre-emphasis. Thankfully, the HD Radio system has a flat response. Without the HF boost, much less of the HF spectrum is driven deep into the final limiters, so familiar FM processing side effects, such as intermodulation and harmonic distortion are greatly reduced. HD Radio processors can sound much smoother and cleaner than their FM counterparts at a given processing level.

Not everyone may perceive this as a benefit. Pre-emphasis boost creates what some industry insiders refer to as the "sounds like radio" effect. Advanced FM Analog processors have a special distortion control function in their final limiter to reduce the worst of the audible effects of HF clipping, and the sonic result is that certain "radio sound" that we are used to hearing. I expect when some program directors first hear HD Radio, they will be unhappy because it won't sound like what they have grown used to. No matter how much processing you use to try to fix this "problem," the aesthetic will remain obstinately different. (I suppose we could implement some kind of HF clipping distortion simulator, without actually doing the clipping, but let's agree not to go *there*…)

Differences Between HD Radio And AM Analog

The biggest difference between HD Radio and AM Analog is frequency response. To implement HD Radio for AM, the audio bandwidth of the analog channel must be further restricted to 5kHz, as compared to the 10kHz NRSC spectrum that has been in effect. This is necessary to eliminate any interference from the analog channel into the digital spectra. This does further reduce the fidelity for the analog channel. The HD Radio system is designed to provide a 12 kHz response on the AM band.

At the time of this writing, development of HD Radio for AM is still ongoing, so for now we'll concentrate on the FM system.

Processing For HD Radio: The Landscape Has Changed!

The challenges for an HD Radio audio processor are much different than for an FM Analog processor. In the FM Analog system, the processor must:

- Provide precision peak control to guard against over-modulation,
- Manage the pre-emphasis boost to avoid audible distortion generated by the processing,
- 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. An 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 pre-emphasis boost. The processor needs to be thought of as a partner with the audio encoder, analogous to the way an FM Analog processor works together with the stereo multiplex encoder.

We also need to understand what needs to be done to manage 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, and dynamic algorithms in the processor can offset these conditions and, in many cases, remove unwanted artifacts, especially at higher bit rates (such as the 96kbps rate used for HD Radio). When properly designed and implemented, 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, an aggressive clipping function is not required – and actually becomes a problem for the codec process.

The End Of Clipping?

Precision peak control can be achieved using numerous methods. Probably the most common is the hard limiter, or peak clipper. Most audio processors designed over the last twenty years also employ some form of distortion masking as a tool to suppress the Total Harmonic Distortion (THD) that is created by the clipper. This makes it possible to utilize more of the clipping function, which translates directly to more perceived loudness.

Employing a clipper as a peak limiter in a HD Radio system will work, but there are sonic penalties to be paid. Any clipping process yields harmonics of the fundamental source signal, and even with distortion masking some second order harmonic content will remain. This adds to the audio spectrum and aggravates the encoder, which in turn spawns additional sonic artifacts. This is particularly noticeable in the high-frequency range – where most codec artifacts exist – and is very noticeable with certain program material; therefore, another form of peak limiter is needed.

Enter the Look-Ahead Limiter

There is another form of peak limiter that is the perfect companion for the HD Radio application: the Look-Ahead Limiter. This limiter provides excellent peak control, and it does so with very little – or no – added harmonic content.

Here's how a look-ahead processor operates: the processor calculates the peak level of a signal over a specified period of time and the audio is delayed by a like amount. Then, as the control signal is applied to the audio gain function, the peak is reduced at the precise time that the control signal reaches the maximum control level so that the crest of the peak is reduced without truncation. This is how hard clipping is avoided. Figure 1 illustrates this process:

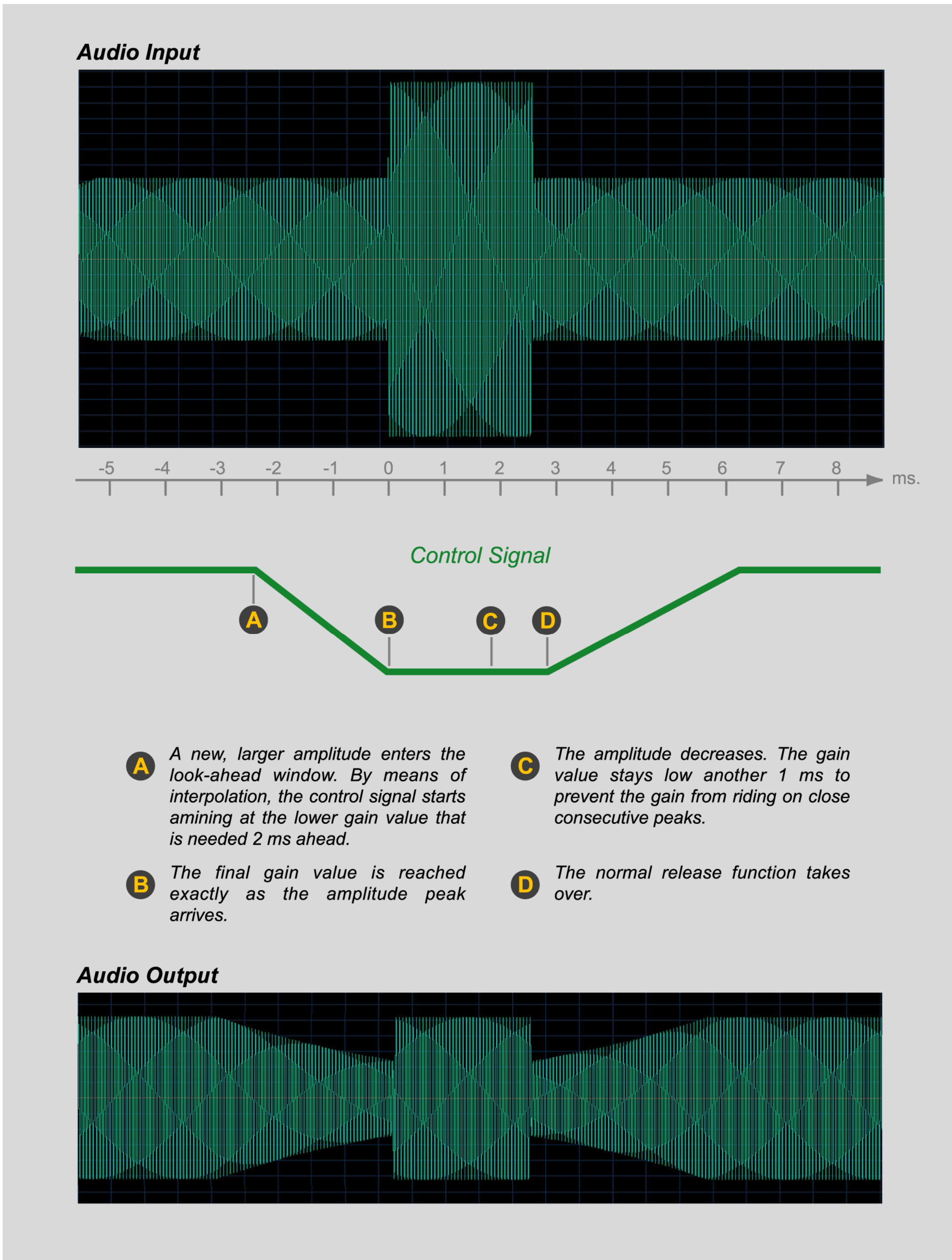

Figure 1: Operation of a Look-Ahead processor (simplified)

As the illustration shows, peak control is achieved without creating any harmonic distortion. If the diagrams were expanded to show detailed sine waves, you would see no peak truncation during the processing period.

Unfortunately, there is no free lunch. A simple wide-band 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" or "mushy" quality that can be just as annoying as harmonic distortion.

Innovative Algorithm Design

THD is very hard to eliminate and will cause audio to contain out of spectrum components (although some of those can be removed through precise filtering.) But all is not lost, because in the case of IMD we can design our processor to "know" what signals will cause those distortions and when they will occur. Then we can take advantage of the look-ahead calculation time to add ancillary control signals that monitor and remove IMD. We can have our cake and eat it: precise peak control and very little THD and IMD distortion.

Great care must also be taken in designing the attack and release time constants. Even more so than with an FM Analog processor, these constants must be optimized to provide a transient feel, for the most natural sound. We've discovered that the best results are achieved when attack and release times increase as frequency decreases. In the case of a look-ahead processing system, this means we will also require different processing delays for each audio band.

Transmission System Considerations: Blend-To-Analog Accuracy

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 new items that need consideration.

First, there is the important 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 HD Radio system has designed time-diversity into the transmit/receive path (so the audio from both the digital and analog signals arrive at the same time), audio spectrum and phase relationships must be similar on both the digital and analog transmission paths.

Should there exist significant phase differences across the audio spectrums of the digital and analog signals, the blend-to-analog action will not be heard as a smooth transition, and the audible jump is likely to be jarring to listeners. 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.

One way to ensure that the phase relationships are maintained between the digital and analog channels is to apply a single processor that provides outputs for each of the required channels. A system of this nature would integrate dedicated final limiting functions for the HD Radio and FM Analog channels. Figure 2 illustrates such a system.

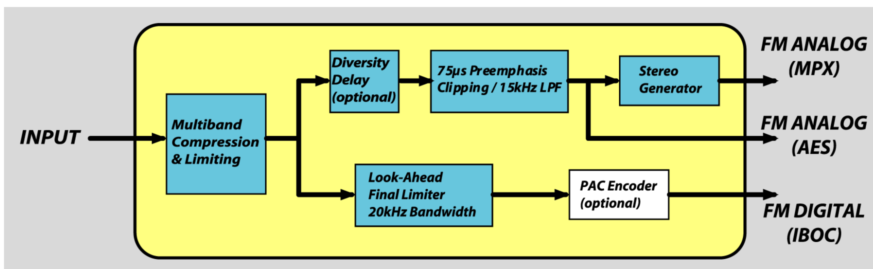

Figure 2: Combination FM Analog and HD Radio Processor

An important consequence of the required integration is that the processor will probably need to be located at the transmitter facility; the dedicated outputs of two transmission channels would require double the STL occupancy in order to be studio-located. Modern processors provide excellent remote operation capabilities, so remote location should not be as much of a concern.

Level Normalization Between The HD Radio and FM Analog Channels

A critical component to the HD Radio transmission system is the reference audio levels of the digital and analog signals in the receiver. We want to create audio levels that are perceived to be relatively the same in average volume when compared to each other, since the listener experience could be adversely affected if the audio level abruptly changes when the receiver switches between the digital and analog signals. How can this be *normalized,* and what are the operating levels required to accomplish this?

The FM/AM Analog path transmits a maximum peak deviation of +/-75kHz for FM, and double the RF carrier level for AM. These levels correlate to 100% modulation. It is accepted that 100% modulation will create a specified level within a receiver. For the sake of discussion, let's assume that 100% modulation will create a peak level of 0dBu in a receiver. For this given modulation level, an average level will be derived based upon the amount of processing employed. The RMS level will average a larger value when more processing is used.

Most FM and AM stations employ enough processing that the RMS level is usually within a few dB, whether or not the processing is set for light or aggressive operation. Present generations of audio processors do a very fine job of maintaining a normalized RMS level. Basically, the sonic difference between light and aggressive processing is the perceived density – anywhere from "packed up and thick" to "open and airy." For this discussion, our concern is not the aural texture but the perceived average level of the analog path.

One of the advantages to the digital channel is the ability to offer wider dynamic range, which allows you to use less processing. This would reduce the RMS average level and sound perceptibly quieter compared to the FM/AM Analog path. So, again, the dilemma concerns what level the HD Radio signal should be set at to sound comparable to the more heavily processed analog signal.

To help determine the optimum operating level for the HD Radio processing system, the following subjective test was derived. Two dedicated processors, one for FM/AM Analog and one for HD Radio, were set side by side and aurally compared. The FM/AM-Analog processor was set for a relative reference level of 0dBu. Think of this as the given 100% modulation level that would feed a transmitter. Since all pre-emphasis/de-emphasis and peak control are handled in the processor, the output level can be set to a known reference. Likewise the HD Radio processor can be set up in the same fashion. Since both units will provide absolute, precise peak control, output levels can be set to maximize the dynamic range of their respective transmission channels.

The FM/AM Analog processor was set for heavy, aggressive processing, whereas the HD Radio system was set for light processing. Once set up, the processing parameters were not further adjusted during the testing. Fourteen audio clips including voice, voice-over and music were recorded with the digital processor (serving as the reference) held at 0dB and the output of the analog processor recorded in 1 dB steps from 0 to -9dB.

Thirty-one subjects from iBiquity's offices in Maryland and New Jersey and from Omnia's office in Ohio participated in the test. The result of this evaluation determined that the perceptual difference between a heavily processed analog and lightly processed digital transmission is 3.57 dB.

In the transmission system there are two parameters that can't be changed: first, the FM/AM-Analog is set to 100% peak modulation, and second, the HD Radio channel is limited to 0dBfs peak operation. Therefore, to insure smooth blending between the analog and digital sources, the offset to normalize these levels will have to be incorporated in the receiver.

Based upon the results of the subjective test, it is recommended that to normalize the audio levels between the HD Radio and FM/AM Analog signal paths, a 5.0 dB relational difference in level needs to be implemented. This number, for FM operation, is derived from the 3.57 dB determined through subjective evaluation and approximate 1.5 dB of pad to allow broadcaster flexibility. To ensure proper blending, every HD Radio receiver, independent of manufacture, will be required to have the same relative offset.

Automotive receiver manufacturers currently match the levels of various sources, including CD players, AM & FM radio and DVDs so that there are minimal level disparities between the devices. For HD Radio the offset can be done in one of two ways – either the receiver manufacturer can choose to increase the level of the digital by 4.5 dB or decrease the level of the analog by 4.5 dB.

STL System

HD Radio offers broader audio bandwidth of 20kHz, requiring an STL link capable of supporting this response with a sampling rate of 44.1kHz or 48kHz. The HD Radio system operates at 44.1kHz, but 48kHz is the professional audio standard. The importance of linear phase relationships between the HD Radio and FM Analog systems has already been mentioned – and this is best accomplished by using audio processors that employ a common design and sampling rate! Even though time alignment between two different sample-rate systems is possible, it adds needless complication and potential trouble to the overall system.

So again, since just about every facility has an STL link that can carry only one linear stereo audio pair, the most logical location for audio processing is the transmitter site. We need two specialized audio signals for transmission: The 15kHz emphasized signal for the FM Analog channel (which can in the form of discrete Left/Right or MPX), and the 20kHz/flat signal for HD Radio. The single STL link delivers 20kHz audio to the transmitter site, where it can be routed to a combination processor as described earlier, or to multiple units.

If your desire is to locate the processing at the studio, you will need two linear stereo audio channels in your link. This would be possible via a multi-channel T-1 STL, but remember that the link must be linear and uncompressed.

Latency Issues

Throughput delay is a given with the HD Radio system. The latency of the system, end-to-end, is almost seven seconds. The days of monitoring off-the-air are over! This will require alternative methods for talent monitors, as well as IFB feeds for remote broadcasts. A simple method for creating an air-like monitor for talent would be to insert an older processor into the control room monitor path. This will allow disc jockeys to maintain the same processed sonic texture in their headphones.

The bigger challenge lies in providing remote talent cuing for live events such as remotes, traffic reports, etc, since off-air feeds are no longer feasible. Possible alternatives are the use of the SCA channel, RPU, or ISDN/POTS codecs with low IFB delay.

System Layout

Presently, there are two methods of implementing the RF plant for HD Radio: low-level and high-level.

Low-level refers to the method of mixing or diplexing the FM Analog and HD Radio RF signals together at the lower power level, which can be found at the output of the exciters. This requires the use of a single transmitter, but it must have sufficient RF response and group delay specifications to pass both the FM Analog and HD Radio signals without degradation. High-level is the method of diplexing the output of two dedicated transmitters together and passing those onto the antenna. This method puts less stringent requirements on the FM Analog transmitter, but it will consume more power, as there will be an insertion loss in RF power to the FM Analog transmitter. If you choose to use your existing FM Analog transmitter, be certain that it contains enough RF power headroom to replenish the insertion loss that will occur in the diplexer.

With regards to audio processing, it doesn't matter whether you employ the high-level or low-level method; all of the data and timing issues are the same.

The intended goal for successful HD Radio deployment should be that the existing FM Analog channel is allowed to operate as it presently does, while the HD Radio signal is seamlessly added into the transmission path.

Many FM transmission systems that operate using one of two connection methods to the input of the FM exciter: composite multiplex (MPX) or AES digital, where the stereo encoding is performed in the exciter. This capability should not be altered with the addition of the HD Radio path. It is understood that there are critical issues concerning data timing and rate conversion, as well as synchronization of the diversity delay between the FM Analog and HD Radio audio channels. Insuring that these signals are properly deployed is key to the operation of the entire HD Radio system.

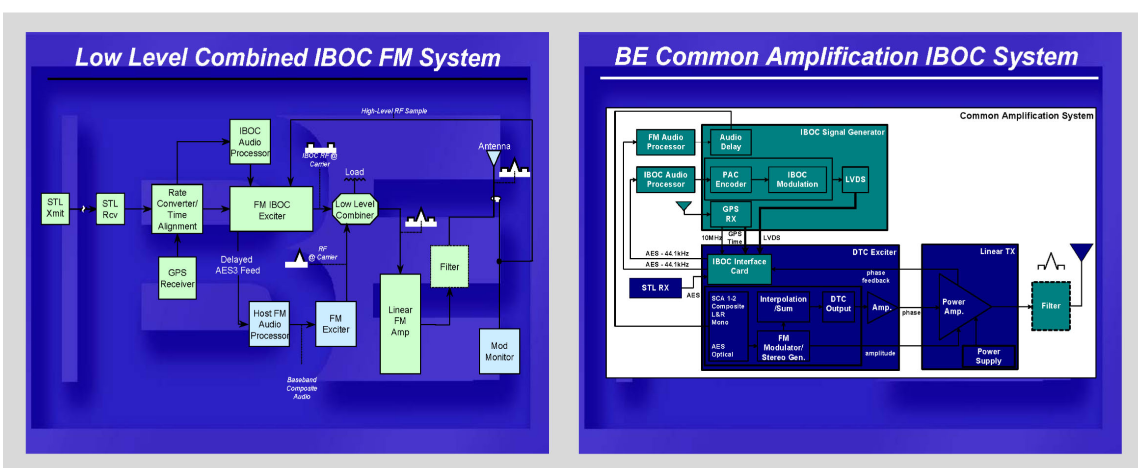

Figure 3.1: Sample schematics for Low-Level (left) and High-Level Combining Systems (courtesy of Broadcast Electronics)

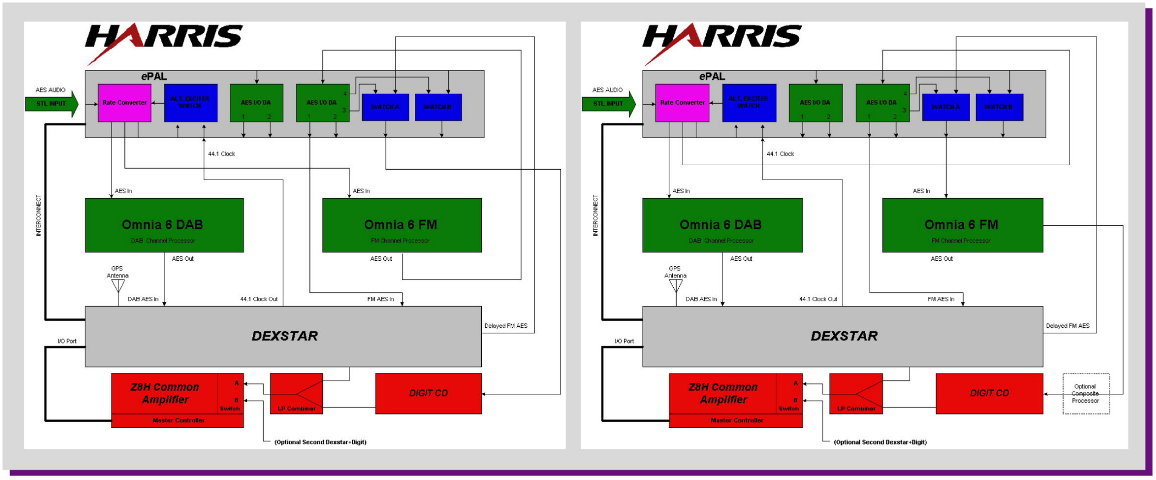

Figure 3.2: Another example of Low-Level (left) and High-Level (right) Combining Systems (courtesy of Harris Broadcast)

Please note that the system diagrams presented here are provided as examples only; since the rollout of HD Radio, both Harris and Broadcast Electronics have further refined their designs. Please contact your transmitter manufacturer of choice for further information regarding system design, implementation, and setup. As this system continues to evolve, further enhancements are sure to occur.

Processing As A Tool, Not A Weapon!

All too often, processing is pushed to a level where it's thought of as a "weapon." HD Radio will require us to think of it more as a subtle tool. Due to the processor's ability to enhance or improve the efficiency of the audio encoder, it will act as more of a partner to the transmission system. Processing will actually improve the intelligibility of the perceived audio. (Listening to a low bit rate Internet stream that has a well-tuned audio processor applied to it will verify this.)

Processing for effect is still possible – make no mistake! Creating the appearance of that larger-than-life, big "phat" sound is most certainly possible. But that synthetic "smash-mouth" sound characteristic of many current FM stations will be far less possible.

OK…Let's Jam!

Now that we've taken care of the preliminary info, we can discuss the processing in more detail. As we saw, processing for HD Radio can be done in one of two ways: using a combined processor, where a single system will manage both the HD Radio and FM Analog signals, or using a dedicated processor for each channel. If the latter is chosen, make sure that both units "agree" with one another regarding phase relationships. You don't want problems with the blend-to-analog mode.

The combined processing system will create a similar sound on both channels. This system will employ its multiband sections as common functions to the HD and Analog channels. The only difference is in the final limiter sections. Decisions about processing setup will doubtless be dominated by the analog channel until more HD radios have filled the market. Thus, the recommendation is to set your processing for the analog channel, and then fine tune the HD channel by setting the final mixer and limiter for the desired amount of density.

Since processing for the analog channel will be set up to take preemphasis into account, the final spectral mix will be representative for that of an emphasized signal. Therefore, it will be important to perform a separate final mix for the HD channel, so that the spectrum is properly balanced with respect to the analog channel.

With the combined system, this is where differential adjustments can be made for the HD channel. Be careful to avoid widely different EQ among the bands, as this can cause the frequency range that contains the most gain to dominate the amount of look-ahead limiting.

For example, if the output of Band 2 is set so that it has a +3dB gain, as compared to all other bands, than it will dominate the action of the final limiter due to its louder level. Also, be careful not to over-emphasize the high frequencies, as they can agitate the audio encoder and be heard as HF coding artifacts.

The last adjustment will be setting the relative offset in HD output level so that the two channels appear normalized. This is done with the output level adjustment.

Using separate processors for the two channels provides a range of possibilities. Again, the processors must be consistent with each other regarding phase relations, especially in the respective crossover networks. It would not be a good idea to retain that +20 year old two-band processor for the analog channel and install a new multiband unit for the HD side. That type of arrangement will surely cause massive phase errors whenever the blend-to-analog function occurs!

In this type of configuration, the analog channel can be set up as usual and the HD channel can be adjusted for whatever texture is desired. If the processing goal for the HD channel is to sound considerably more "open" and "airy" as compared to the analog channel, this is the method to use. Care should be taken to avoid over-enhancing high frequencies.

If you need to jam on the HD channel, you can, but with a few noticeable differences. Since the final limiters will not contain clippers but look-ahead limiters, there will be a textural difference in how low frequencies will appear. Clippers add a "phatness" to the audio due to the harmonics they generate. Most processors offer some type of Bass-EQ or bass enhancement. You may need to adjust those functions a bit differently for the HD channel.

Also, while the HD processor does not contain a clipper, the final look-ahead limiter will create its own sonic artifacts if overdriven. When a look-ahead limiter is driven too hard, you will get added IMD, rather than THD. As with anything processing-related, a little goes a long way.

Digital Loudness Wars?

To understand the nature of future loudness wars, we need to look deeper into how the HD Radio receiver operates. When selecting an HD Radio station, an internal buffer inside the receiver must fill before the digital audio is routed to the speakers. Depending upon the receiver design, this buffer may take a few seconds. So, when switching between stations, the analog signal will be the first to be heard and the digital audio will follow, when the buffer has been filled in the receiver.

Because of the length of time it takes for the digital signal to be heard after a station swtich, it will be quite difficult for a listener to remember the loudness level of one HD Radio station compared to the next. The ear normally has a retention memory for only a few milliseconds. So, if loudness is still the key desire for your station, it will still be possible to create that "button-push" illusion of being louder than the next guy, but once the HD channel switches in, you can relax the processing. Create the loudness illusion using the analog channel, and take advantage of the fidelity possible with the HD channel.

Also note – you can't gain extra loudness via over-modulation anymore. This will not be technically possible because there is simply no more to be had after you've reached digital full-scale. So the days of over-modulation will cease with digital transmission; trying to exceed full-scale will only result in drastic and annoying distortion.

Further On The Horizon…

HD Radio continues to evolve, and there will be much more to learn. Nevertheless, we are confident that today's Omnia processing and the HD Radio system will work in concert to provide an experience for listeners that will keep radio interesting and relevant in a time when the competition for people's attention is overwhelming.

Please visit www.omniaaudio.com *for updates and more information.*