Appendix B

FM Hybrid IBOC DAB System Description

1. System Overview

IBOC technology provides a means of introducing digital audio broadcasting ("DAB") without the need for new spectrum allocations for the digital signal. USADR's FM hybrid mode allows the station to simultaneously broadcast the same programming in analog and digital. Although the level of the digital signal in the hybrid mode must be limited to accommodate the analog broadcast, the hybrid system will still afford an upgrade over existing analog service by providing enhanced audio fidelity, improved signal robustness, and expanded auxiliary services.

1.1. FM Hybrid IBOC Waveform

The FM hybrid IBOC spectrum is shown in Figure B-1. Low-level digital sidebands are added to each side of the analog signal. The bandwidth is limited to $\pm$200 kHz from the center frequency. USADR has conducted simulations, analyses, laboratory compatibility tests, and field tests which verify that restricting the digital subcarriers to the 70-kHz regions between 129 and 199 kHz from the center frequency on either side of the analog spectrum minimizes interference to the analog host and adjacent channels without exceeding the existing FCC spectral mask. This bandwidth is wide enough to support a robust hybrid IBOC service with virtual CD-quality audio that mirrors the coverage of existing analog radio stations.[1]

The dual-sideband structure enables the use of frequency diversity to further combat the effects of multipath fading and interference. The baseline hybrid system simultaneously transmits 96 kbps of error-protected digital audio information, plus auxiliary services, on each DAB sideband. Each sideband has all the information and thus can stand alone. However, when neither sideband is corrupted, advanced FEC coding techniques allow the combination of both sidebands to provide additional signal power and coding gain.

Figure B-2 shows a scenario in which a desired hybrid IBOC signal and an upper first-adjacent hybrid IBOC interferer can co-exist. The figure shows that there is no interference between the hybrid digital sidebands. However, the analog portion of the first adjacent hybrid signal may interfere with the upper digital sideband of the desired hybrid signal. In an effort to reduce analog interference to the digital sidebands, USADR has developed a technique known as First Adjacent Cancellation ("FAC"). In addition, the use of frequency diversity and advanced FEC coding techniques further improves performance of the desired digital signal under these conditions.[2]

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[1] Refer to the USADR field test results detailed in Appendix H.

[2] Refer to the USADR laboratory performance test results detailed in Appendix C.

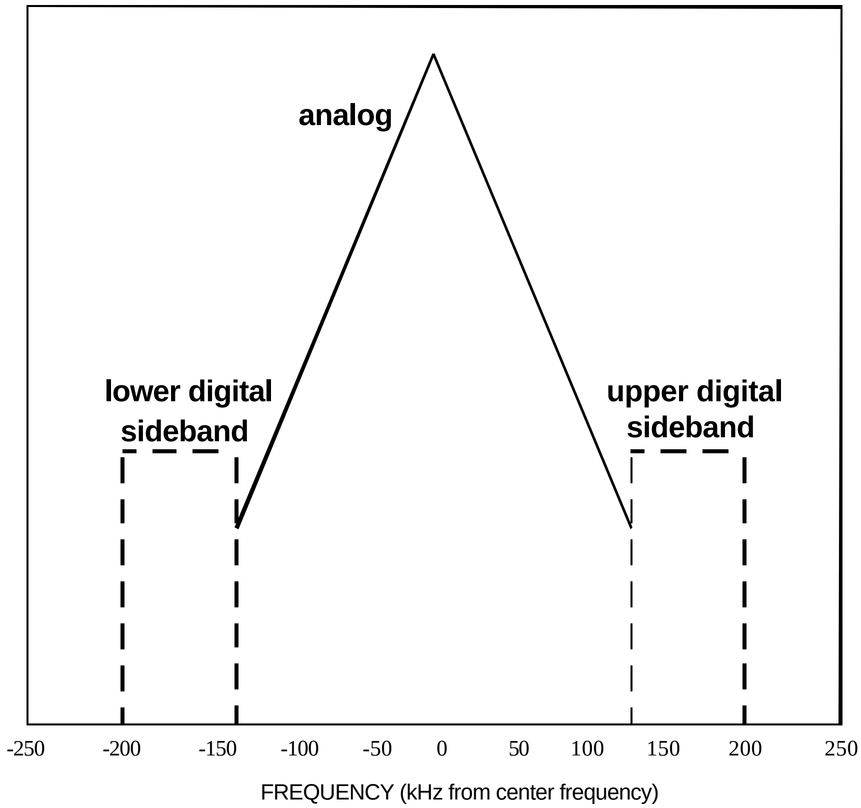

Figure B-1 - FM Hybrid IBOC Power Spectral Density

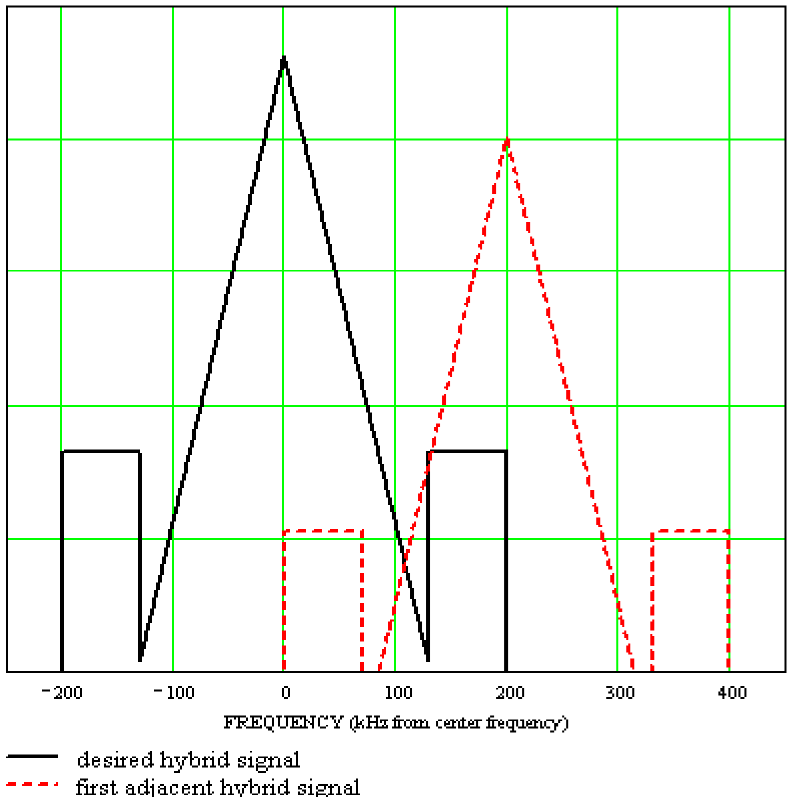

Figure B-2 - Hybrid First-Adjacent Interference Scenario

2. System Design

The USADR FM hybrid IBOC DAB system is comprised of four basic components: the modem, which modulates and demodulates the signal; the codec, which source encodes and decodes the audio source signal; forward error correction ("FEC") coding and interleaving; and blending. All of these core functional areas have been designed and integrated to produce an FM hybrid IBOC DAB system of superior quality.

2.1. Modulation Technique

USADR evaluated several modulation techniques for its FM IBOC DAB system before selecting Quadrature Phase Shift Keying ("QPSK"). QPSK affords robust performance while providing sufficient throughput for virtual CD-quality digital audio. It permits the use of advanced FEC coding techniques which exploit knowledge of the non-uniform interference environment. QPSK is also simpler and more robust than higher-order forms of modulation, especially in a multipath environment.

Since QPSK has a bandwidth efficiency of two bits per second per Hertz, it supports an information bit rate that is sufficient for transmission of virtual CD-quality audio in the bandwidth available.

USADR compared multi-carrier versus single-carrier approaches to transmit the digital signal, and chose a multi-carrier approach called Orthogonal Frequency Division Multiplexing ("OFDM"). OFDM is a scheme in which many QPSK-modulated subcarriers can be frequency-division multiplexed in an orthogonal fashion such that each subcarrier does not interfere with its adjacent subcarriers. OFDM offers a high level of robustness in a multipath channel.

When combined with FEC coding and interleaving, the digital signal's robustness is further enhanced. The OFDM structure naturally supports FEC coding techniques that maximize performance in the non-uniform interference environment. The most important coded bits can modulate OFDM carriers that are located in the most protected regions of the channel.

2.2. Source Coding

CD digital audio has a data rate of 1.4112 Mbps (44,100 16-bit samples per second, for left and right channels). The FM channel bandwidth does not have the capacity to support a sufficiently high data rate to provide uncompressed CD-quality audio. As a result, an audio codec (coder-decoder) compression technique must be employed. The audio codec is a source-encoding device that removes redundant information from a digital audio signal in order to reduce the bit rate, and hence the bandwidth required to transmit the signal. The codec must perform this information rate compression while preventing the generation of perceptible artifacts.

USADR uses the AAC codec in its IBOC DAB system. The AAC codec compresses the CD bit stream to 96 kbps, delivering audio that the listener will perceive to be virtually the same quality as a CD. Use of the AAC codec meets the raw throughput requirements of the modulation and FEC coding techniques. Also, special error concealment techniques employed by the codec help to ensure graceful degradation of the received digital signal for operation in an impaired channel. In addition to its ability to meet the USADR system's audio compression requirements, AAC offers the advantage of being an open system based on the MPEG family of ISO standards.

AAC is the latest MPEG standard on perceptual audio coding and is part of the world-wide MPEG family of audio and video standards.[3] Much of the work on AAC was done by Fraunhofer, AT&T, Dolby Labs, and Sony, all leading experts in audio compression technology. It builds upon the existing MPEG Layer-3 standard by further optimizing coding efficiency.

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[3] MPEG is the Moving Pictures Expert Group, working under the joint direction of the International Standards Organization ("ISO") and the International Electro-Technical Commission ("IEC"). Its main goal is the standardization of audio and video coding schemes. MPEG AAC was standardized as ISO 13818-7 in April 1997.

AAC is a very flexible coding scheme, supporting data rates above 8kbps. It can encode mono and stereo input data, as well as multichannel data (up to 48 channels). It is used for a wide range of applications, from internet audio to multichannel surround sound. The high coding efficiency makes AAC attractive, especially for applications with very high quality demands or very limited transmission bandwidth.[4] Even though the basic structure of AAC is similar to previous audio coding techniques, including the commonly used MPEG Layer 3, AAC contains numerous innovations which are particularly helpful for the implementation of DAB. The crucial differences between MPEG AAC and its predecessor MPEG Layer-3 are as follows:

- *Filter Bank*: In contrast to the hybrid filter bank of MPEG Layer-3, which was chosen for reasons of compatibility but ultimately displayed certain structural weaknesses, MPEG AAC uses a plain Modified Discrete Cosine Transform ("MDCT"). Together with the increased window length (2048 instead of 1152 lines per transformation), the MDCT outperforms the filter banks used in previous coding methods.

- *Temporal Noise Shaping ("TNS")*: A true novelty in the area of time/frequency coding schemes, TNS shapes the distribution of quantization noise in time by prediction in the frequency domain. Voice signals in particular experience considerable improvement through TNS.

- *Prediction*: This is a technique commonly used in the area of speech coding systems. It benefits from the fact that a certain type of audio signal is easy to predict.

- *Quantization*: By allowing finer control of quantization resolution, the given bit rate can be used more efficiently.

- *Bit-Stream Format*: The information to be transmitted undergoes entropy coding in order to keep redundancy as low as possible. The optimization of these coding methods, together with a flexible bit-stream structure, has made further improvement of the coding efficiency possible.

During the standardization process, MPEG performed numerous listening tests to assess the audio quality of AAC. It is difficult to specify audio-coded performance in terms of traditional audio measurement techniques such as frequency response, distortion, and dynamic range; therefore, audio codecs are psychoacoustically compared against a CD reference. In these double blind tests, human testers are given the opportunity to compare compressed and non-compressed segments of the same selection and make judgments as to the quality of the compressed segment. In tests designed to replicate the worst case signals, the AAC codec at 96 kpbs has proven to be almost indistinguishable from the original selection. For the most extreme cases, the difference in the compressed signal is audible, but not considered a major issue for listeners. These tests use what is essentially a short audio clip played over and over to train the listener. In other words, while listening to 96 kbps AAC-encoded

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[4] AAC has been chosen as the audio coding standard for the Japanese HDTV system, which will be introduced in 2000.

audio with high quality headsets, the average listener will not be able to distinguish between it and the original CD unless a short music selection is played over and over from both the CD and AAC (*i.e.,* the listener is trained). Because typical radio listeners will never listen to IBOC DAB under such pristine lab conditions with studio quality headsets and amplifiers, 96 kbps AAC will be perceived by typical listeners as "virtually" the same as a CD.[5]

2.3. FEC Coding and Interleaving

Forward error correction and interleaving greatly improve the reliability of the transmitted information. Advanced FEC coding techniques exploit the non-uniform nature of the interference. Special interleaving techniques spread burst errors over time and frequency to assist the FEC decoder in its decision-making process. The combination of these advanced FEC coding and interleaving techniques, together with superior modem performance, allow the IBOC system to robustly deliver, in a mobile environment, virtual CD-quality audio with coverage comparable to existing analog service.

2.4. Blend

The USADR system employs time diversity between two independent transmissions of the same audio source to provide robust reception during outages typical of a mobile environment. In addition, rather than abruptly muting, the blend function allows graceful degradation of the digital signal as the receiver nears the edge of a station's coverage. The FM hybrid IBOC DAB system provides this capability by delaying the analog transmission by a fixed time offset (several seconds) relative to the digital audio transmission. When the primary digital signal is corrupted, the receiver blends to the backup analog audio which, by virtue of its time diversity with the digital signal, does not experience the outage.

The blend feature also provides a means of quickly acquiring the signal upon tuning or re-acquisition. Without blend, a digital receiver would incur a significant delay after tuning to a station before the listener hears the audio. The blend feature will allow the receiver to instantaneously acquire the analog signal. This allows the listener to hear the selection while the receiver is acquiring the digital signal. After acquisition, the receiver then blends to the digital signal.

Figure B-3 graphically depicts the blend-with-time-diversity process. The top part of the figure shows that, for the transmitted signal, the analog backup portion is delayed relative to the primary digital portion. This delay at the transmitter provides the time diversity as the signal travels through the channel. The top part of the figure also shows an impairment that prevents recovery of sections 6 and 7 of the digital signal. The impairment also interferes with sections 2 and 3 of the analog backup signal.

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[5] Refer to Appendix G for an audio quality assessment of AAC.

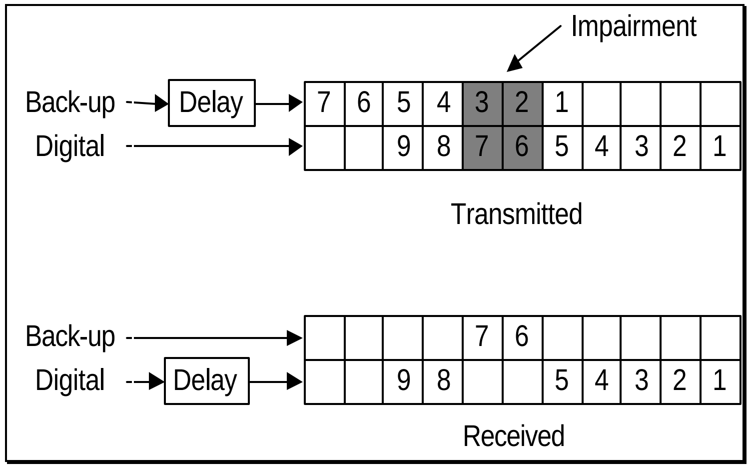

Figure B-3 - Blend with Time Diversity

The bottom part of Figure B-3 shows the received signal. At the receiver, the digital signal is delayed relative to the analog backup, and the signals are time-aligned. Part of the delay is due to the processing required to recover the digital signal, and the remainder of the delay can be implemented directly.

As shown in the bottom part of Figure B-3, sections 1–5 of the digital signal are recovered. Sections 6 and 7 of the digital signal will be marked as non-recoverable by the receiver. However, because of the time-diversity, sections 6 and 7 of the analog signal are not affected by the impairment, and the receiver can smoothly blend to analog, with the blend beginning during section 5. When the digital signal can be recovered again, the receiver can smoothly blend back to digital.

For an illustration of USADR's time-diversity blend, refer to the audio files on the CD submitted as a companion to this appendix.

3. Signal Flow

3.1. FM Transmitter Signal Flow

A functional block diagram of an FM hybrid IBOC transmitter is presented in Figure B-4. The sampled stereo audio source feeds both the analog and digital signal generation paths. A diversity delay is introduced in the analog path for blend purposes. In the power combiner method shown here, the analog audio is processed within the exciter just as an existing analog FM signal would be, prior to amplification by the high power amplifier ("HPA").

The DAB path first source encodes the audio signal in the audio encoder. The audio encoder removes redundant information from the audio signal to reduce the bit rate, and hence the bandwidth required to transmit the signal. To ensure that the communication of information through the fading channel is robust, the compressed bit stream is then passed through the FEC coding and interleaving function. The resulting bit stream is packaged into a modem frame and QPSK and OFDM modulated to produce the DAB baseband signal. This baseband signal is upconverted and amplified before being power combined with the analog signal.[6]

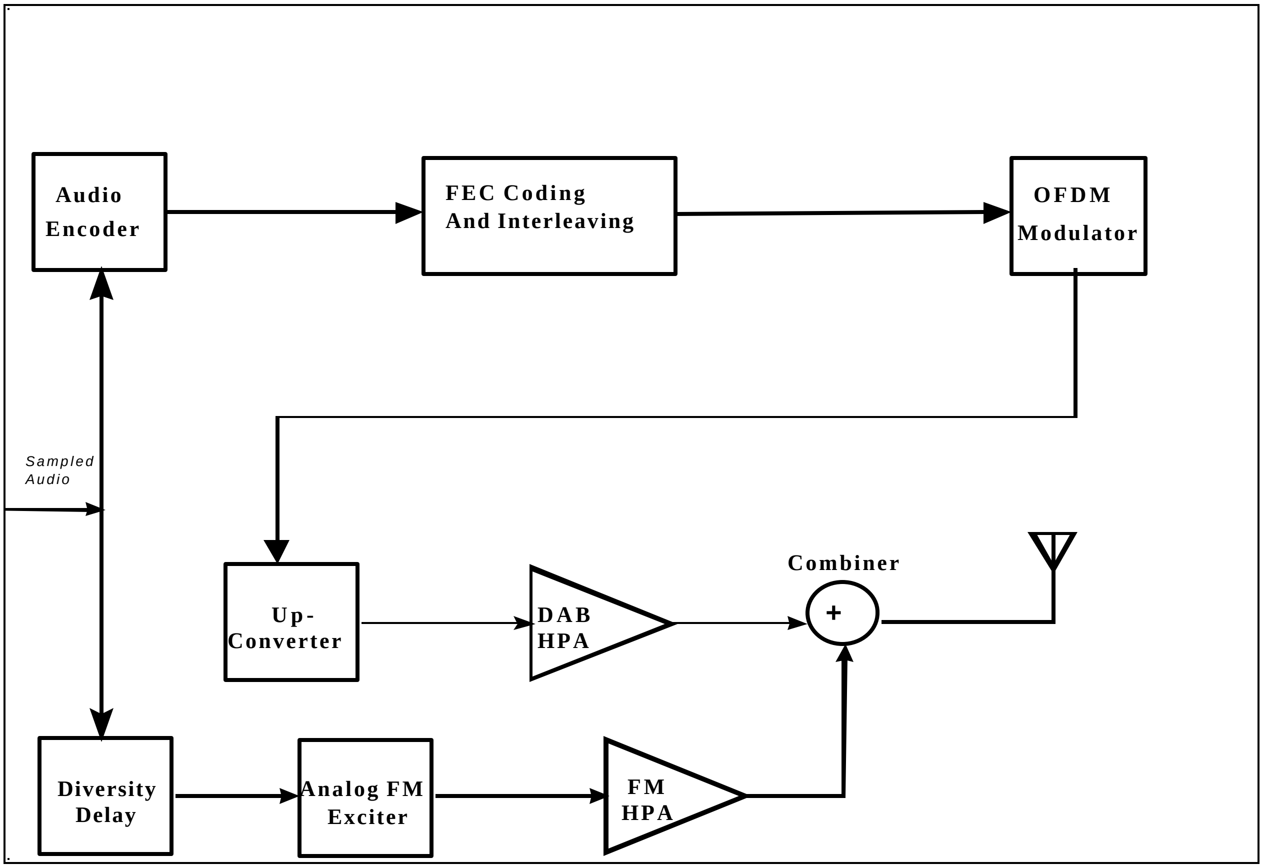

Figure B-4 - FM Hybrid IBOC Transmitter Functional Block Diagram

3.2. FM Receiver Signal Flow

A functional block diagram of an FM hybrid IBOC receiver is presented in Figure B-5. The signal is received by the antenna, passed through an RF front end, and mixed to an intermediate frequency ("IF"), as in existing analog receivers. Unlike typical analog receivers, however, the signal is then digitized at IF, and digitally down-converted to produce in-phase and quadrature baseband components. The hybrid signal is then separated into an analog FM component and a DAB component. The analog FM stereo signal is digitally demodulated and de-multiplexed by the FM receiver to produce a sampled, stereo audio signal.

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6 Details such as data insertion and FM/DAB blend synchronization have been omitted here for simplicity.

The baseband DAB signal is first sent to the modem, where it is processed by the FAC to suppress interference from potential first-adjacent analog FM signals. The signal is then OFDM demodulated, deframed, and passed to the FEC decoding and de-interleaving function. The resulting bit stream is processed by the codec function to decompress the source-encoded digital audio signal. This DAB stereo audio signal is then passed to the blend function.

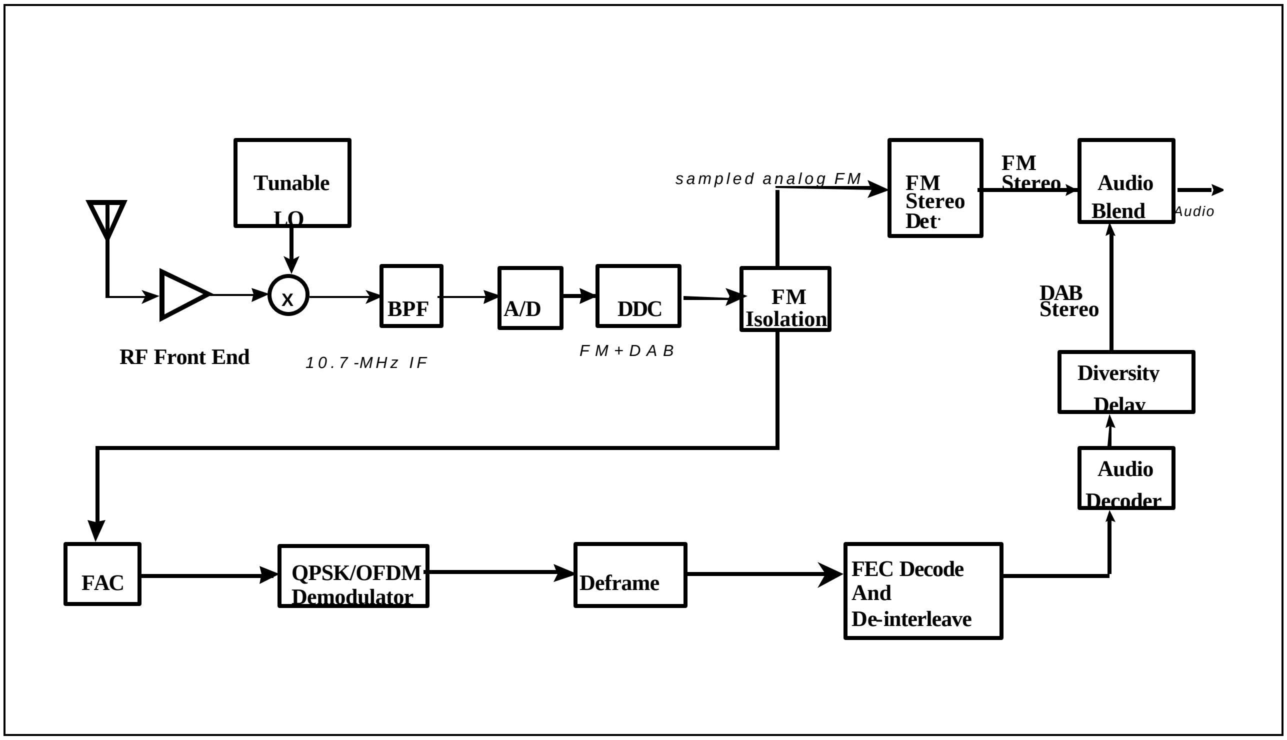

Figure B-5 - FM Hybrid IBOC Receiver Functional Block Diagram

4. Flexibility and Auxiliary Services

The FM IBOC DAB system, as a digital medium, essentially provides a bit stream. Since a given bit in the stream can be devoted to audio, data, or parity, tradeoffs among robustness, data throughput, and audio quality must be performed to produce a system which is optimal for a given application. The design of the USADR hybrid FM IBOC DAB system provides the inherent flexibility required to afford broadcasters the ability to tailor their systems to meet their unique requirements.

In order to provide robust, virtual-CD-quality audio, most of the capacity of the FM hybrid IBOC DAB system is dedicated to the primary audio program. The system does, however, support auxiliary data services that will upgrade existing analog FM Subsidiary Communications Authorizations ("SCAs") by offering much higher availability, reliability and robustness.

The system has a large capacity that could be devoted to auxiliary services, since a throughput of up to 120 kbps can be achieved. The portion of this throughput that is dedicated to data services is selectable by the broadcaster, and must be traded off versus desired audio quality and required digital signal robustness.

The USADR FM hybrid IBOC DAB system incorporates two main types of auxiliary services, which are explained below:

- Ancillary Services: A data rate of up to 120 kbps can be transmitted by reducing the audio quality and FEC parity accordingly. The audio rate could vary over time, based on the programming or desired audio quality. For example, speech may be encoded at a lower rate than music; news and talk formats require less throughput that could be dedicated to data. If data throughput were not required, the extra capacity could be allocated to FEC parity, to increase the robustness of the digital audio.

- Opportunistic Data: Up to 32 kbps can be intermittently multiplexed with the audio on a priority basis, or when spare bandwidth is available, as determined by an entropy analysis from the audio encoder. For example, the opportunistic data rate might be increased during a less complex passage of music, or during a news report following a musical program. Program associated data ("PAD"), which is used to display audio and station information on the receiver, takes a portion of the capacity, since it does not need to be transmitted real-time.