

country. For this case at 500 kHz, the value of F_a is about 12 dB higher than the value at 1 MHz. At 2.5 MHz, the value of F_a is about 14 dB lower than the value at 1 MHz.

Because lightning is the source of atmospheric noise, it exhibits an impulsive characteristic. Lightning pulses have short durations. An average length pulse falls to below 10% of its maximum amplitude in less than 20 μ s. and the longest pulses are not significantly longer than this.²⁶ The impulsive nature of atmospheric noise can be characterized by several parameters. One study gives the standard deviation of the median value of F_a , the ratio of the upper and lower decile values to the median value, the standard deviation of the decile values, and V_d for a 200 Hz bandwidth.²⁷ The values for summer from 2000-2400 local time are given in Table H-3. The impulsive nature of natural noise can also be characterized by APDs. One study notes that measured values of F_a for a location in the southwestern United States are generally higher than those predicted, but it is also noted that this may be in part due to the presence of local thunderstorms.²⁸

Frequency (MHz)	$\sigma_{F_{am}}$ (dB)	Upper Decile (dB)	$\sigma_{Upper Decile}$ (dB)	Lower Decile (dB)	$\sigma_{Lower Decile}$ (dB)	V_d (dB)
0.5	4.6	9	3.1	7.7	2.1	6.2
1	4.8	8.3	2.8	7.2	2	5.8
2.5	4.6	6.4	1.9	6.1	1.8	5

Table H-3 - Variation Parameters for Natural Noise

²⁶ F. C. Breeze, NAB Engineering Handbook, "Lightning protection for broadcast facilities," § 2.2, pp. 101-03 (8th ed. 1992).

²⁷ International Radio Consultative Committee, "Radio Noise," ITU Recommendation ITU-R PI.372-6, Int. Telecommun. Union (Geneva, 1994) (contains a chart for converting V_d to other bandwidths).

²⁸ J.R. Herman, A. A. Giordano, X. A. DeAngelis, K. F. Marzotto, and F. M. Hsu, "Measurement and statistical analysis of wideband MF atmospheric radio noise; 1. Structure and distribution and time variation of noise power," Radio Science, Volume 21, Number 1, pp. 25-46 (Jan.-Feb. 1986).

2.3.3 Noise Mitigation Strategies

In contrast to many communications systems that operate in Gaussian noise environments, both man-made and natural noise have an impulsive nature, which leads to a broadband spectrum. The impulsive nature of the noise results in short-time signal-to-noise ratios that can be much lower than noted above, and can certainly limit the performance of current analog broadcasting. The effect that noise has on analog reception is usually described as bursts of static.

While noise adds directly to a demodulated analog signal, it may not affect the audio in a digital system, as long as the digital signal is received with few errors after forward error correction. Clipping and noise blanking can be used to reduce the effects of noise on the analog signal, enhancing listener satisfaction. In addition to clipping and noise blanking, digital modulation can use techniques such as forward error correction, interleaving, and frequency diversity to overcome the effects of noise.

3.0 Existing AM Analog Performance in the Presence of IBOC DAB

In addition to performing a thorough investigation of the existing environmental in the AM band, USADR has analyzed the effects of IBOC DAB on the existing analog AM signal.

3.1 Impact of Digital Signal on Analog Host AM Performance

USADR has conducted simulations and experiments to determine the level of interference that occurs from the hybrid digital signal to the host analog signal. The system is designed to introduce minimal interference to existing AM analog reception while simultaneously providing a robust digital signal for new IBOC-compatible receivers.

3.1.1 Methods to Prevent Interference from Digital Carriers

The USADR hybrid AM system uses several techniques to dramatically minimize the interference from the digital carriers to the host analog signal. The techniques include a combination of waveform design methods and signal processing at the transmitter.

3.1.1.1 Power Level of Digital Carriers

The levels of the digital carriers that occupy the same spectrum as the analog signal (± 5 kHz from the AM carrier) are set to minimize interference to the analog signal, yet provide robust digital performance. In an ideal coherent AM receiver, the quadrature relationship between the AM analog and the digital components renders the interference negligible. New AM receiver designs would therefore encourage coherent AM detection. However, in a more typical noncoherent (envelope detector) AM receiver, the quadrature relationship between the digital and analog components of the hybrid signal will decrease interference due to the digital carriers to as small as -56 dBc. This corresponds to a dynamic range of 53 dB measuring a tone at 100% modulation relative to the interference (referred herein as the max-to-interference ratio). Present AM reception is protected to no better than about 26 dB D/U ratio from a co-channel interferer translates to about 33 dB signal-to-noise ratio after demodulation by a simple diode detector. Better signal-to-noise ratios can be obtained when interference from adjacent or co-channel stations is less than the protected levels. Therefore, ideally, the introduction of the digital signal introduces a small amount of noise compared to current AM protected interference levels.

3.1.1.2 Quadrature Digital Carriers

Monophonic analog AM receivers respond only to the envelope and not the phase of the received signal. Interference in this type of receiver is minimized when the interference is in quadrature to the AM carrier.

The USADR system minimizes interference from the digital signal to the analog signal by placing a portion of the digital signal in quadrature to the analog carrier. Placing two signals in quadrature is straightforward if the two signals have carriers that are at the same frequency. However, when the digital signal consists of several digital carriers that are not at the same frequency as the analog carrier, as in the OFDM-modulated USADR system, carrier pairs can be used to place the digital signal in quadrature to the analog carrier.

Analysis and simulations have shown that the use of quadrature digital carriers significantly reduces the interference to the host analog signal.

3.1.1.3 Spectral Sidelobe Reduction

Even though the digital carriers in the sidebands are outside of the spectral region of the analog signal, spectral spillage from the sidelobes adds interference to existing analog receivers.²⁹ The USADR system uses pulse shaping techniques at the transmitter to control this source of interference.

3.1.1.4 Pre-compensation

Because the analog and digital signals are known prior to transmission, interference caused by the digital signal when received in an analog receiver can be determined. Therefore, the transmitted signal can be pre-compensated to reduce this source of interference. Additionally, distortions due to transmission systems can lead to increased interference from the digital signal to the host analog signal. Pre-compensation can be used to eliminate this source of interference. A variety of methods are available for implementing pre-compensation, and optimization of the pre-compensation technique has been applied by USADR.

²⁹

See Appendix F.

3.1.2 Limitations from Existing Receivers

Limitations in AM analog receivers result in lower signal-to-noise ratios than would be predicted in an ideal situation. Non-ideal characteristics in the IF filters of real analog AM receivers results in decreased SNR because the complementary subcarriers will not be orthogonal to the AM carrier. Additionally, the signal-to-noise ratio in analog receivers is limited by other factors such as interfering stations and noise.

3.1.2.1 Magnitude and Group Delay Response

Existing analog receivers use IF filters that may have non-symmetric magnitude or group delay about the AM carrier frequency. The effect of non-symmetric magnitude or group delay is that the received digital quadrature carriers will no longer be strictly in quadrature to the AM carrier. This will cause a decrease in the received signal-to-noise ratio. IF filters typically have larger asymmetry farther away from the AM carrier; however, the highest level digital subcarriers are closest to the AM carrier. Therefore, the interference caused by the asymmetry is minimized. Another advantage is that the frequency response of an analog AM receiver typically decreases for higher frequencies, and therefore larger asymmetry at higher frequencies has less effect.

In the absence of all interference and assuming a strong received RF signal strength of -45 dBm, typical AM receivers are capable of providing approximately 56 dB max-to-interference ratio.³⁰ Preliminary experiments with automobile receivers using a previous 30 kHz AM DAB signal, without the benefit of dynamic predistortion, indicate that good SNRs can be obtained with most receivers. However, there are some receivers with significant IF filter asymmetry that are more sensitive to this problem. The present 20kHz AM DAB signal is designed to reduce this interference even further.

3.2 Analog Performance with Co-Channel and Adjacent-Channel IBOC Signals

The AM hybrid IBOC system has been designed to introduce minimal interference to the existing AM analog reception while simultaneously providing a robust digital signal for new IBOC-compatible receivers. This section describes the effects of co-, first, and second adjacent channel interference, from hybrid, and all-digital IBOC signals.

3.2.1 Interference from Co- or First Adjacent Hybrid Signals

The proposed DAB signal limits the analog AM audio bandwidth to within ± 5 kHz. The existing AM transmission is limited to ± 10 kHz. Furthermore, the new digital carriers placed beyond ± 5 kHz out to nearly ± 10 kHz will be transmitted at a comparable or lower power spectral density than present AM transmissions. The overall result is that the interference from co-channel or first adjacent hybrid IBOC interferers is comparable to that which presently exists from AM signals.

3.2.2 Interference from Co-Channel or First Adjacent All-Digital IBOC Signals

The new DAB carriers placed within ± 5 kHz will be transmitted at a comparable or lower power spectral density than present AM transmissions. Furthermore, the new digital carriers

³⁰ Electronic Industries Association, Digital Audio Radio Laboratory Tests, Transmission Quality Failure Characterization and Analog Compatibility (August 11, 1995).

of information bits per block varies with the audio codec rates of 48, 32, or 16 kbps. Block error rate is used as a metric since it provides the most accurate indication of the threshold of audibility (TOA) of the audio codec. TOA is defined as the block error rate above which noticeable impairments may just be detected. For the USADR hybrid IBOC system, the TOA is defined as a 1% block error rate.

4.1.2 Host CNR for Gaussian Noise

The ratio of signal to noise is defined here as the ratio of the power in the main carrier

$C_{main_carrier}$ to the noise power in a single digital carrier bin $N_{digcarrier}$, so that $CNR = \frac{C_{main_carrier}}{N_{digcarrier}}$.

In most of the simulations, the CNR was set to a constant 300 dB, since Gaussian noise is not a dominating factor in the AM channel.

AM channel noise is more impulsive in nature than Gaussian noise. A good receiver design would include a noise blanker or clipper to mitigate the effects of the impulsive noise. In fact, an A/D converter would naturally provide some clipping protection, since an AGC algorithm would tend to keep the pre-sampled signal as high as possible, while allowing occasional clipping. The impulsive noise would then tend to be clipped. Clipping should be performed in a wide bandwidth to effectively combat lightning noise. Conversely, the A/D converter input bandwidth should be limited to prevent other channels in the band from consuming the dynamic range after sampling. A compromise is made to accommodate the competing requirements of effective impulsive noise processing and selectivity. This compromise is best accomplished by placing the noise blanker prior to the presampling filter. In any case, effective impulsive noise processing tends to make the resulting noise more Gaussian-like.

Impulsive noise processing has not been incorporated into the simulation at this time. Instead of injecting unprocessed impulsive noise into the simulated receiver input, an alternative noise source was provided. In order to provide some sort of ambient non-Gaussian noise, the first and second adjacent channel interferers were set no lower than -50 dB in the interference tests.

4.1.3 Interfering Stations

Many simulations were performed in the presence of various combinations of co-channel, first adjacent and second adjacent channel interference. The analog AM host and analog portion of the AM interference signals were modulated with various audio segments which were processed by typical broadcast audio processing equipment. The processed audio was post-filtered to yield a 3 dB point at ± 4.6 kHz. All interferers were mutually uncorrelated. The host AM analog signal was present in all measurements. Interferers were scaled relative to the analog host AM carrier. Thus, an interferer at -20 dB is 20 dB below the host AM carrier power, which is assumed to be 0 dB. All interferers were either hybrid or all-digital signals. Purely analog interferers were not used, since it is assumed that hybrid interferers are worse than analog-only interferers. Except for the Gaussian-noise-only case, no interferer was set below -50 dB; thus, a minimum level of noise (interference) was always present.

4.2 Results of Simulations and Analyses

Simulations and analyses have characterized the performance of the hybrid IBOC digital signal in the presence of Gaussian noise and interference. Results while subject to various combinations of these impairments are presented and interpreted in the following sections.

4.2.1 Hybrid Performance in the Presence of Gaussian Noise

In order to calibrate the simulation and provide a bound to system performance, simulations were performed in Gaussian noise only, in the absence of channel impairments and interference.

The TOA for the 32 kbps audio codec is CNR=56 dB. The TOA for the 48 kbps audio codec is CNR=58 dB. The TOA for the 16 kbps audio codec is CNR=54 dB. These results confirm the expected performance of the DAB detection and FEC decoding so that there is confidence that the system is correctly simulated.

4.2.2 Hybrid Performance in Presence of Interference

Interference is comprised of various combinations of hybrid and all-digital upper and lower first-adjacent, second-adjacent and co-channel signals.

Single Interferers

Simulations were run with a single interferer raised over the -50 dB floor to the level at which TOA was reached. Single interferer scenarios were run for all audio modes for the 20 kHz system. The interference level relative to the Signal Of Interest ("SOI") at its TOA is identified for the following scenarios. Interference levels which are lower than that specified for the TOA should result in negligible audio impairment.

The level at which the TOA is reached for a single co-channel interferer is -19.5 dB relative to the SOI for the 32 kbps audio codec. The level at which the TOA is reached for a single co-channel interferer is -29.5 dB relative to the SOI for the 48 kbps audio codec.

The level at which the TOA is reached for a single first-adjacent interferer is -20.5 dB relative to the SOI for the 32 kbps audio codec. The level at which the TOA is reached for a single first-adjacent interferer is -25.5 dB relative to the SOI for the 48 kbps audio codec.

The level at which the TOA is reached for a single second adjacent interferer is at least +30 dB relative to the SOI for both the 32 kbps and 48 kbps audio codecs. The simulation was stopped at +30 dB since the signal is virtually immune to second adjacent interference.

Pairs of Interferers

The TOA was determined for simultaneous pairs of hybrid interferers. Results of the 48 and 32 kbps audio modes of the 20 kHz system are presented below. The results are shown in graphs that plot the levels of the two interferers against one another. A curve was drawn through the levels of TOA for the pairs of interferers. Interference levels which are lower than that specified for the TOA should result in negligible audio impairment. Since second adjacent channels do not cause significant interference to the 20 kHz AM IBOC signal, combinations of interferers which include second adjacents were not simulated.

The levels at which the TOA is reached for simultaneous co-channel and first-adjacent interferers are plotted in Figure H-2. The axes define the levels of the co-channel and first-adjacent interferers relative to the SOI. The two traces on the plot comprise the results for both the 32-kbps and the 48-kbps audio codecs.

As an example, the TOA is reached when both the co-channel and the first-adjacent interferers are at a level of about -26 dB for the 32 kbps codec, and -31 dB for the 48 kbps codec. Other combinations of interference levels for the TOA can be estimated by following the plot trace. Clearly the asymptotic limit of a single interferer is approached as either of the two interferers is diminished toward -50 dB.

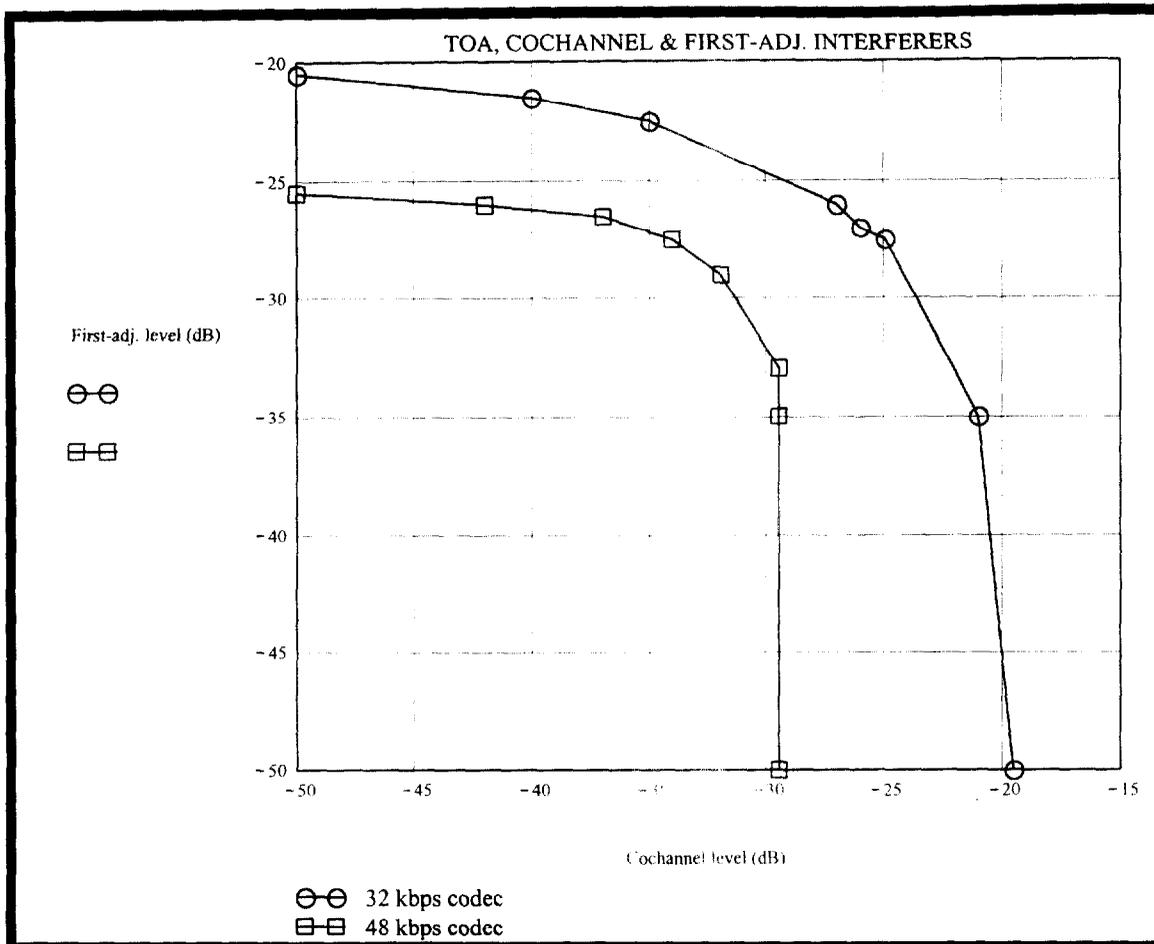


Figure H-2 - TOA Levels of Simultaneous Co-channel and First-Adjacent Interferers.

Simultaneous First Adjacent

The levels at which the TOA is reached for simultaneous dual first-adjacent interferers are plotted in Figure H-3. The axes define the levels of the first-adjacent interferers relative to the SOI. The two traces on the plot comprise the results for both the 32-kbps and the 48-kbps audio codecs.

As an example, the TOA is reached when both first adjacent interferers are at a level of about -25 dB for the 32 kbps codec and -28 dB for the 48 kbps codec. Other combinations of interference levels for the TOA can be estimated by following the plot trace. Clearly the asymptotic limit of a single interferer is approached as either of the two interferers is diminished toward -50 dB.

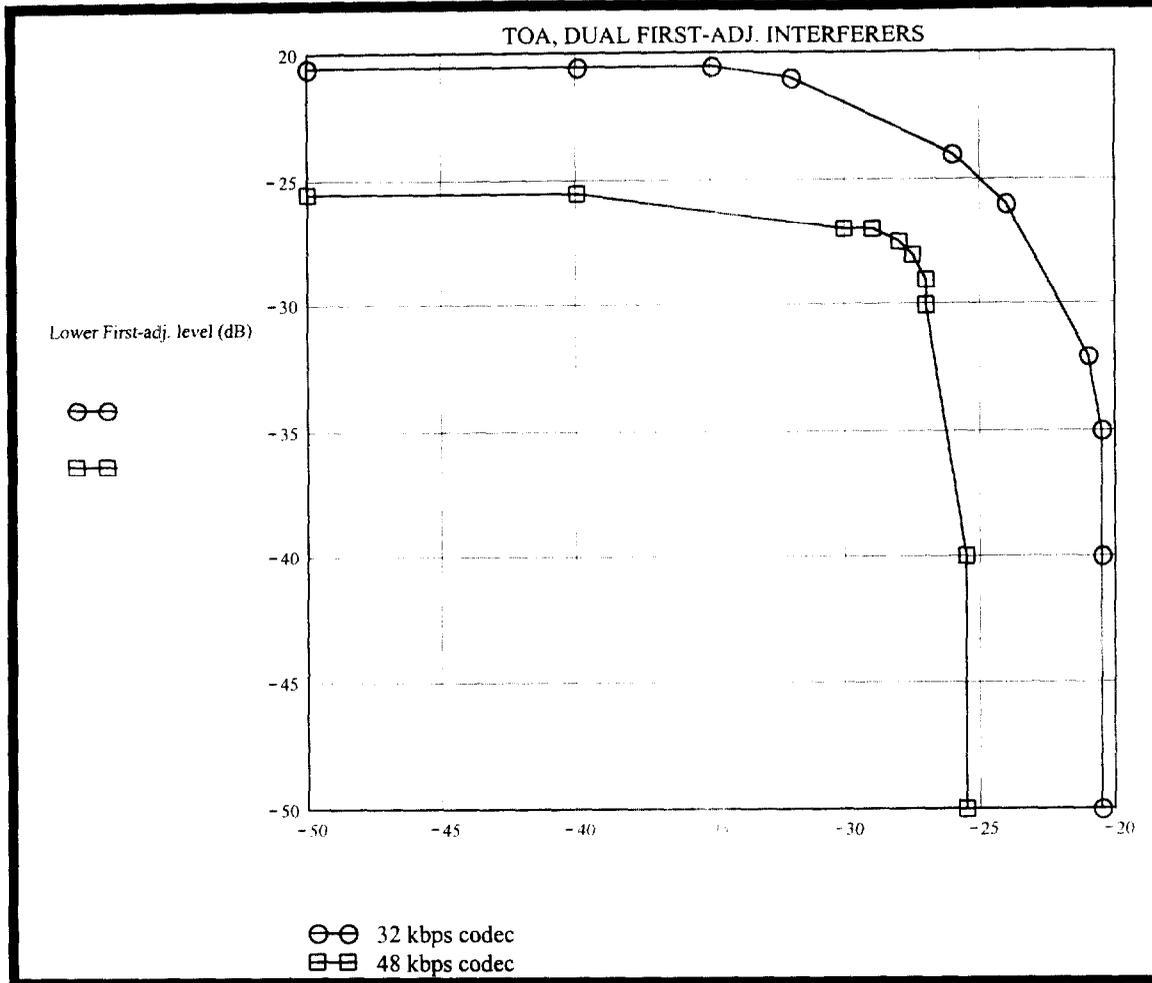


Figure H-3 - TOA Levels for Simultaneous Dual First-Adjacent Interferers

4.2.3 All-Digital System Performance

The level of the carriers in the ± 5 kHz region of the all-digital signal is increased by about 20 dB over its hybrid counterpart, providing an additional 20 dB of margin for these digital carriers. The performance of the all-digital signal will be substantially more robust than its hybrid counterpart.

Hybrid system performance

Simulation and analyses indicate that the IBOC signal will be compatible with both existing and future radio frequency environments. It can be seen that the D/U ranges for determining the TOA above are roughly comparable to the D/U level of 26 dB presently used as

the protection ratio for a co-channel interferer for AM analog reception. In addition, IBOC signals should provide high-quality digital audio coverage over areas where the existing analog AM signal has an SNR of roughly 26 dB or greater

**REPORT OF
TELECOMMUNICATIONS
ASSOCIATES, INC.**

Dated: October 6, 1998

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Declaration

We, Raymond L. Pickholtz, Ph.D., and Branimir R. Vojcic, D.Sc., having been engaged to provide an independent assessment of the proposed USADR In-Band, On-Channel (IBOC) digital audio broadcast (DAB), hereby declare the following:

We have summarized our findings in the reports identified as Modeling and Performance of the IBOC FM/DAB system and Modeling and Performance of the IBOC AM/DAB system.

We have carefully studied the proposed system design and simulation models for the FM Hybrid, and had meetings with the persons who coded the signal processing work system (SPW) tool that was used for the detailed studies. After that, we independently put together the descriptions of the simulation models, verified that they were consistent with the system description given in the USADR report, and checked the validity and reasonableness of the results.

Among the many things we verified in the IBOC FM/DAB system were the details about the placement of the Orthogonal Frequency Division Multiplexing (OFDM) subcarriers, the validity of the time varying fading channel models, the nature of the coding and interleaving that was used, the feasibility of achieving symbol detection, and numerous other proprietary techniques that make USADR IBOC work in the presence of natural impairments and interference.

We conclude, based on bit and block error rates, that IBOC FM/DAB will result in CD-like quality, given what we have been told about the audio compression coder/decoder. When those components become available, subjective tests and field trials will be possible and provide the final verification.

Also, and totally independently, we have done analytical work and some simulation to demonstrate that the proposed placement of the outside sub-carriers for DAB is nearly optimum insofar as having minimal impact on the existing analog FM signal. We included in these analyses, first and second adjacent channel interference and the effects on both the L+R and L-R components of the FM stereo. We also included the effects of harmonics generated in inexpensive receivers that would reflect back interference to the analog signal. We have also verified by some analysis and independent simulation that detection of the QPSK modulated OFDM will work and that it will yield good performance in this channel, especially if the optimum channel state estimators are used to aid the channel decoding.

Importantly, we observe that the interference from the host DAB to the host analog FM and the first adjacent channel FM is minimal in most operational scenarios.

A similar independent evaluation was made of the proposed system design and models used for the IBOC AM/DAB Hybrid. In this model MATLAB was used for the simulation and the details are given in the second report on Modeling and Performance of the IBOC AM/DAB System.

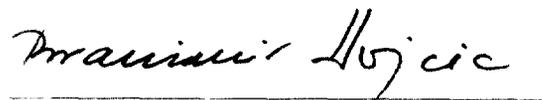
The system is carefully designed to exploit the characteristics of the analog AM signal and interference peculiarities enabling coexistence of analog AM and DAB. It also takes into account the channel impairments that can be exhibited in the AM band and provides appropriate signal processing counter measures. The MATLAB simulation model, that is used to verify the validity of the system, is a rather complete and realistic description of the system. Although all features in the simulation have not been used yet, the initial simulations indicate the feasibility of the engineering design.

The USADR IBOC is a unique system in that it achieves the objective of getting from the present all-analog radio, through a transition where both analog and DAB will coexist, then allow a further transition to all DAB. All this will be achieved without additional spectrum; users with analog receivers will still be able to receive analog signals as long as broadcasters transmit them, and the new technology may be introduced on top of that without causing noticeable interference. The DAB signal will, at the same time, provide a marked improvement in listening quality and have the opportunity of providing auxiliary multicast and broadcast data services as well, if desired.

Both IBOC systems, FM/DAB and AM/DAB, are based on mostly standard digital communication techniques except in cases where specific techniques are invented and incorporated to provide extra protection against interference. This, and the performance evaluation through simulation, provides a high degree of confidence in the feasibility of achieving the system objectives and will enable a smooth transition from engineering design to commercialization.

Dated: October 6, 1998


R. L. Pickholtz


B. R. Vojcic

MODELING AND PERFORMANCE OF THE IBOC FM/DAB SYSTEM

R. L. PICKHOLTZ AND B. R. VOJCIC

INTRODUCTION

The In-Band On-Channel (IBOC) hybrid of analog FM and Digital Audio Broadcasting (DAB), proposed by USADR, represents a novel and unique approach for sharing the radio frequency spectrum. As such, it does not require new frequency allocations to provide digital service with significantly enhanced performance. In addition to compatibility with existing analog FM systems, the hybrid system provides a smooth transition to an all-digital system in the future.

The current technology of digital signal processing (DSP) and of application specific integrated circuits (ASIC) is ideally matched to the requirements of IBOC. This synergy will permit the deployment of reliable and economical transmitters, excitors and consumer receivers. The transmission of DAB will also expedite the multicasting and/or broadcasting of auxiliary data services.

The proposed IBOC FM/DAB is optimized for co-existence of respective analog and digital signals in the same channel. This co-existence could result in a minimal impact on analog FM reception, in most cases, while providing high-fidelity digital reception even under the most adverse channel and interference conditions that can be expected in practice. Our confidence is currently based on extensive simulations incorporating the gamut of expected radio channel impairments.

The IBOC FM/DAB system, including transmitter and receiver, was modeled using the Signal Processing Work-System (SPW) simulation tool set. The simulation model contains the entire signal chain from the source, representing encoded audio, through the reception and decoding of the digital audio signal for production through a loudspeaker. The principal block diagram of the modeled system is shown in Figure 1.

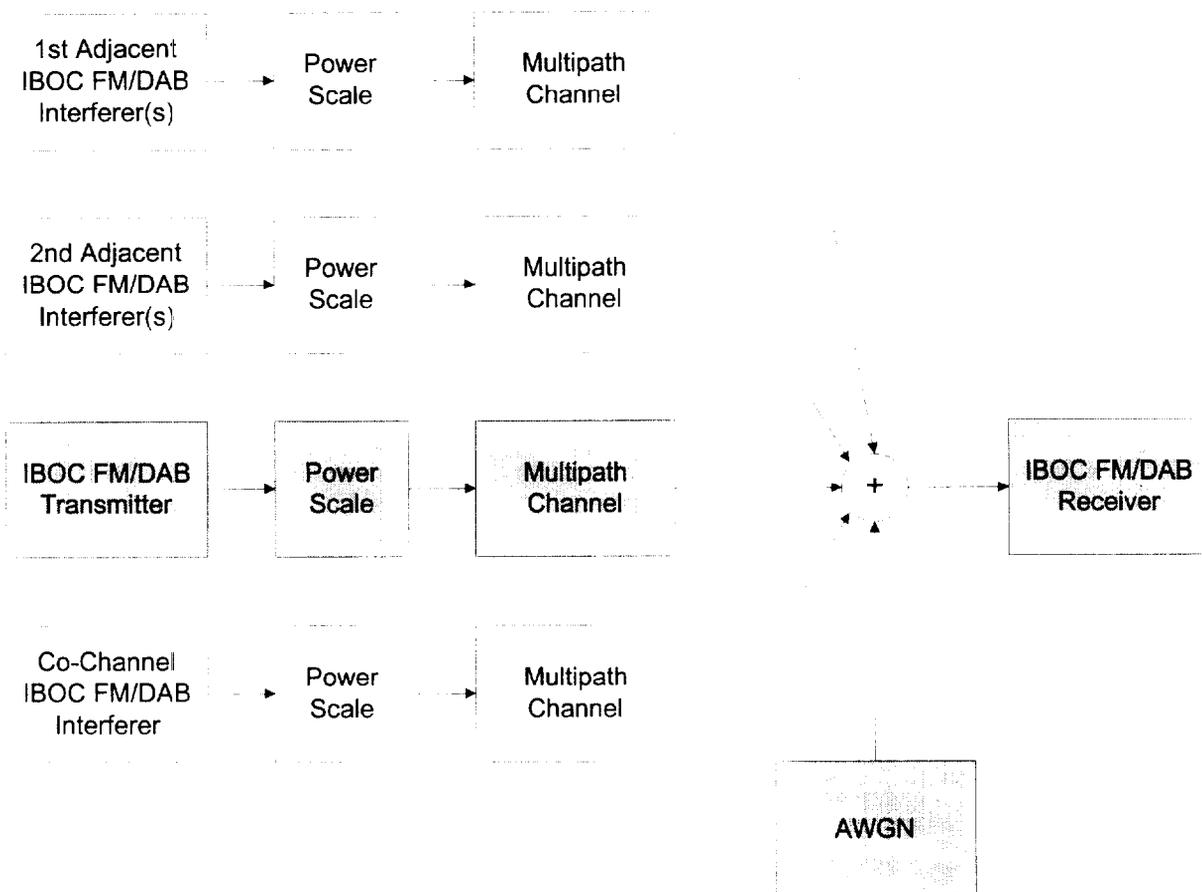


Figure 1. Block diagram of the end-to-end IBOC FM/DAB system.

The main components of the model are IBOC FM/DAB transmitters corresponding to the desired signal, first and second-adjacent channel interference signals, co-channel interference signal, and

the IBOC FM/DAB receiver. To model the system realistically, all signals are passed through independent multipath fading channels. The aggregate of all other sources of noise and interference is modeled by the additive white Gaussian noise (AWGN) block.

In the following sections we will describe each of the blocks in Figure 1, as they are implemented in SPW. Nuances peculiar to efficient SPW implementation will be intentionally omitted to avoid clutter and improve the clarity of presentation. Also, only the principal functions in the communication chain are shown to simplify the presentation and avoid revealing confidential information. The important functions remain and we are confident that what is being described below accurately represents the simulation model. The simulation model is a realistic representation of the proposed system, in that it captures all important functions and critical impairments. Therefore, we believe that the simulation results provide an accurate assessment of performance that can be achieved with the IBOC FM/DAB system.

TRANSMITTER MODELING

The main blocks of the transmitter are shown in Figure 2. The source bit generator outputs a pseudorandom bit stream representing the output of the audio Codec¹. These bits are fed into a powerful forward error correction (FEC) encoder. Powerful FEC, specially designed and optimized for this application, is one of the most crucial elements of the proposed system design. Other important considerations are the placement of DAB OFDM subcarriers and a unique approach for cancellation of first-adjacent FM interference. These elements of the system design are mainly responsible for the remarkable system performance.

¹ As the actual Codec becomes available, this will be replaced with Codec delivered bits.

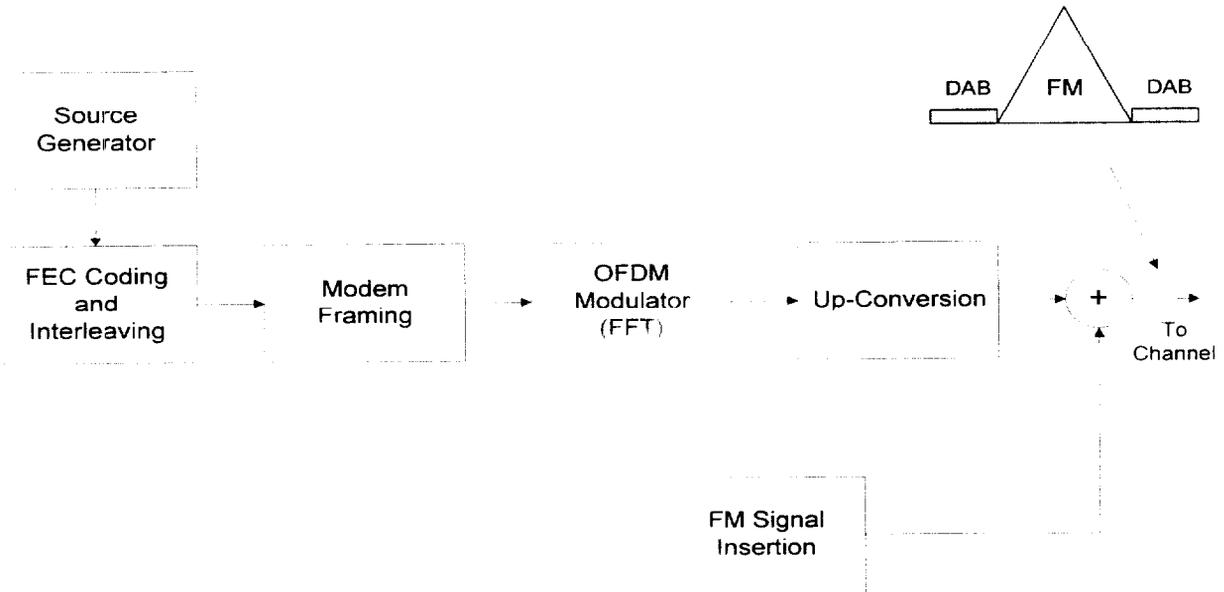


Figure 2. Principal block diagram of IBOC FM/DAB transmitter model.

After interleaving of the coded bits, they are grouped properly in Modem Framing, where complex quaternary symbols are also formed. The framed quaternary sequence is processed by the OFDM modulator via Inverse Fast Fourier Transform (IFFT). The IFFT block size is 2048 (2048 is a power of two so that a simple, readily available radix-2 IFFT can be used), whereby there are two sets of 191 active subcarriers on each side padded with zeros in the middle, to minimize the mutual interference between DAB and FM signals. Also, the total power of the DAB signal is adjusted to be 22 dB below that of the FM signal. A cyclic prefix of 112 samples is added, representing 5.46% of overhead. A cyclic prefix is equivalent to a guard time, and has the function of minimizing/eliminating the adverse impact of multipath induced by the channel.

In addition to adding the cyclic prefix, signal shaping is performed to reduce spectral sidelobes. Finally, the DAB signal is up-converted and combined with the FM signal to obtain the composite hybrid IBOC FM/DAB signal that is transmitted to the channel.

CHANNEL MODELING

To model the system realistically, all the known impairments of the radio channel must be modeled and the composite signal needs to be passed through them. The dominant impairments in the FM band include a randomly fading channel (caused by scattering multipath), motion of the receiver in this environment, interference from other stations, and thermal noise.

The main blocks representing the impact of the channel, shown in Figure 1, are:

- First-Adjacent IBOC FM/DAB signal
- Second-Adjacent IBOC FM/DAB signal
- Co-channel IBOC FM/DAB signal
- Scaling for establishing interference and desired signal levels (Power Scale)
- Doubly selective (in frequency and time) multipath fading channel (Multipath Channel)
- Additive White Gaussian Noise (AWGN)

The co-channel, first-adjacent, and second-adjacent interference signals are incorporated in the simulation model to demonstrate the system robustness under various operational interference scenarios. In order to simplify the modeling and reduce run time, advantage is taken of the decorrelation effects of time displacement; therefore, the interference by these adjacent

hybrid/DAB signals is modeled using a significantly delayed replica of the host DAB. Each of these interfering signals can be independently scaled to set a desired signal-to-interference ratio (SIR).

The channel is modeled as a 9-tap Rayleigh channel. Several EIA test channel models are considered, corresponding to urban, rural, and terrain obstructed fading. The EIA channel models are based on extensive measurements, and provide a description of actual channels that could be experienced in severe environments. The DAB performance in multipath fading channels was analyzed for the above mentioned scenarios. The overall performance in interference conditions was extensively assessed for the urban fast channel model. The Urban Fast² channel is characterized by a Doppler spread of 5.2314 Hz, corresponding to a vehicle speed of about 60km/h at a carrier frequency of 100 MHz. The multipath profile is given in the table below.

Path	1	2	3	4	5	6	7	8	9
Delay (us)	0.0	0.2	0.5	0.9	1.2	1.4	2.0	2.4	3.0
Attenuation (dB)	2	0	3	4	2	0	3	5	10

Table 1. Urban Fast Rayleigh Multipath Channel Model.

² EIA Test Channel Model, Urban Fast, DAR90110.PRO, Profile #2. Digital Audio Radio Laboratory Tests, Transmission Quality Failure Characteristics and Analog Compatibility, August 11, 1995.

Finally, the AWGN block is used to model the aggregate of thermal noise in the receiver and any other noise and interference not accounted for by described interference blocks.

The impairments modeled by the Channel block are characteristic of expected operational scenarios, and will enable a realistic assessment of system performance under a variety of conditions.

RECEIVER MODELING

The basic features of the unique IBOC FM/DAB receiver design are represented in Figure 3. The first step after down-conversion is the separation of the host FM signal, so that only the DAB signal and eventual interference are passed to the subsequent stages of the digital receiver. Figure 3 shows an important feature of the receiver, unique to the USADR system, that allows it to tolerate large first adjacent channel FM signals. This is the FM signal First Adjacent Cancellor (FAC), which suppresses the interfering FM signal to yield a "cleaned" DAB signal. This will be shown to work remarkably well, even in the presence of strong interference.

The "cleaned" DAB signal, after FAC, is OFDM-demodulated, via FFT, to obtain soft estimates of transmitted coded bits. Carrier and timing recovery are performed by the Synchronization Loops, to facilitate proper demodulation. Finally, deinterleaving and FEC decoding are performed to recover the transmitted information sequence. The decoded bit sequence is passed to the bit error rate (BER) performance measurement subsystem for comparison with the true transmitted sequence.

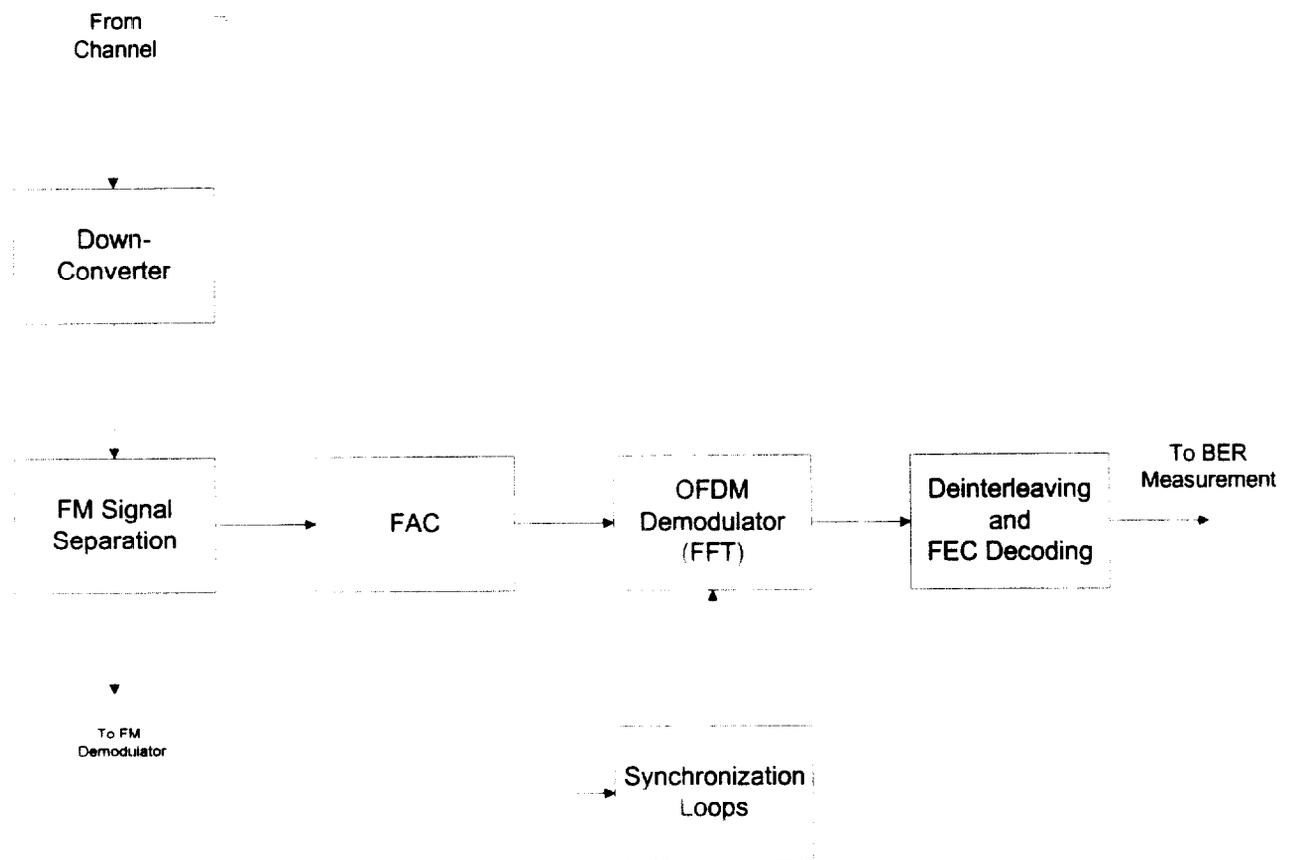


Figure 3. The principal block diagram of the IBOC FM/DAB receiver

PERFORMANCE MEASUREMENT SUBSYSTEM

Provision was made to run up to 36,000,000 information bits for each run and to compare the input bits with those recovered at the end of the receiver chain. For each point in a vector of values for Cd/No^3 and upper and lower first- and second-adjacent channels (five parameters), a single point was obtained for the bit error rate (BER). The following metrics and attributes were recorded:

- Bit errors
- Block errors