

Figure 6. Theoretical BER Curves For M-ary Biorthogonal Signalling (Reprinted From [12]).

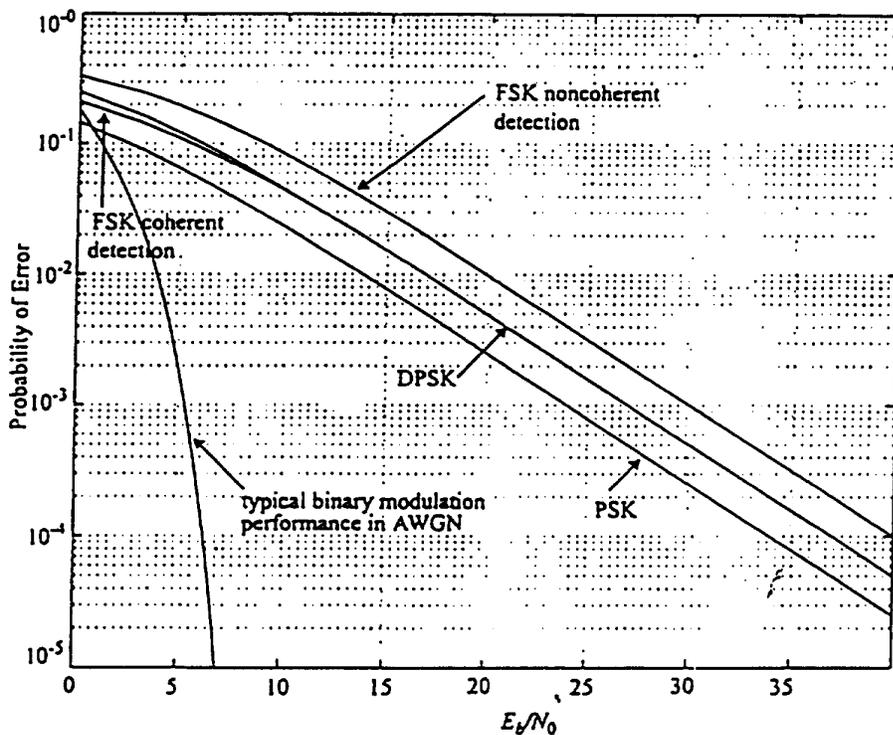
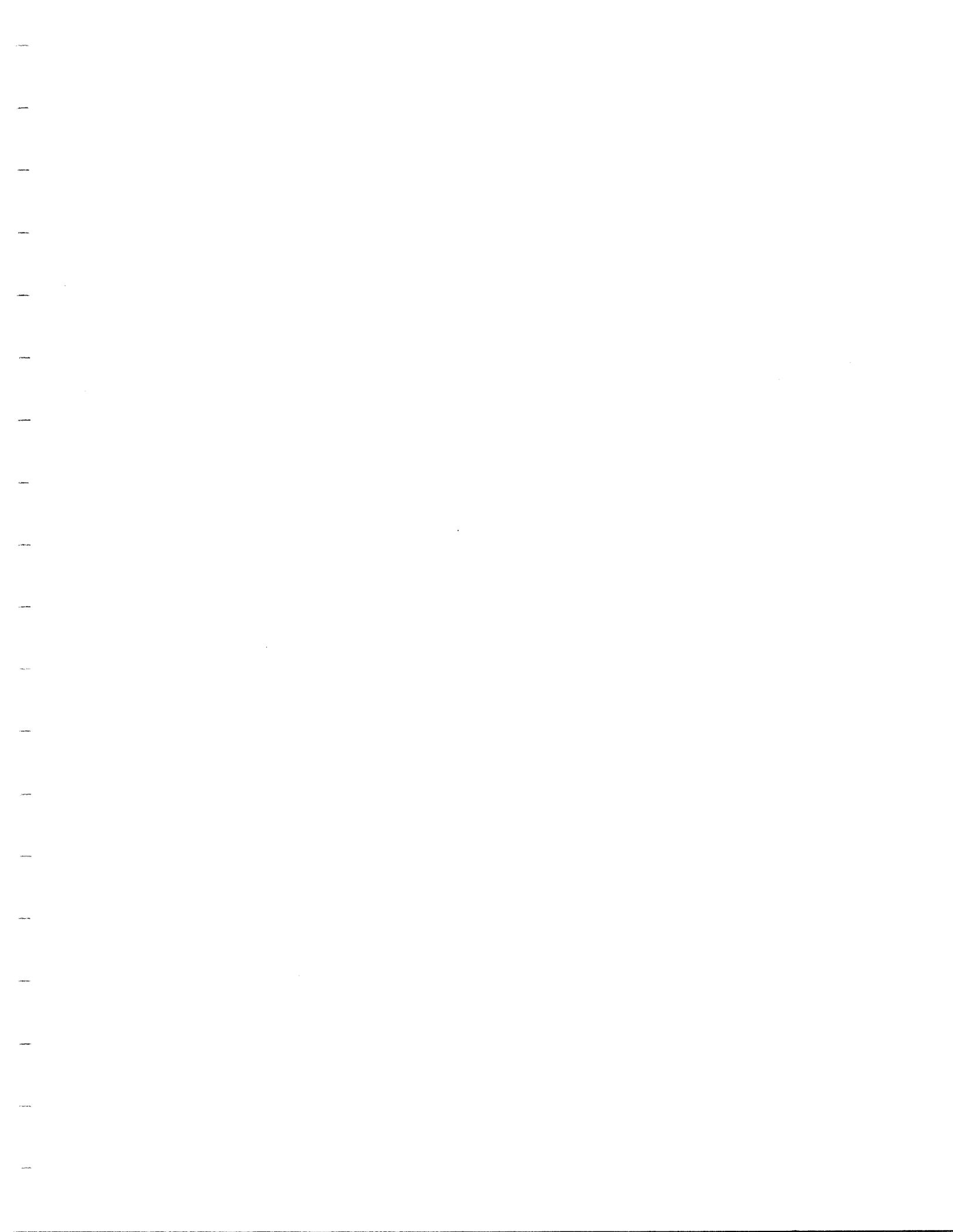
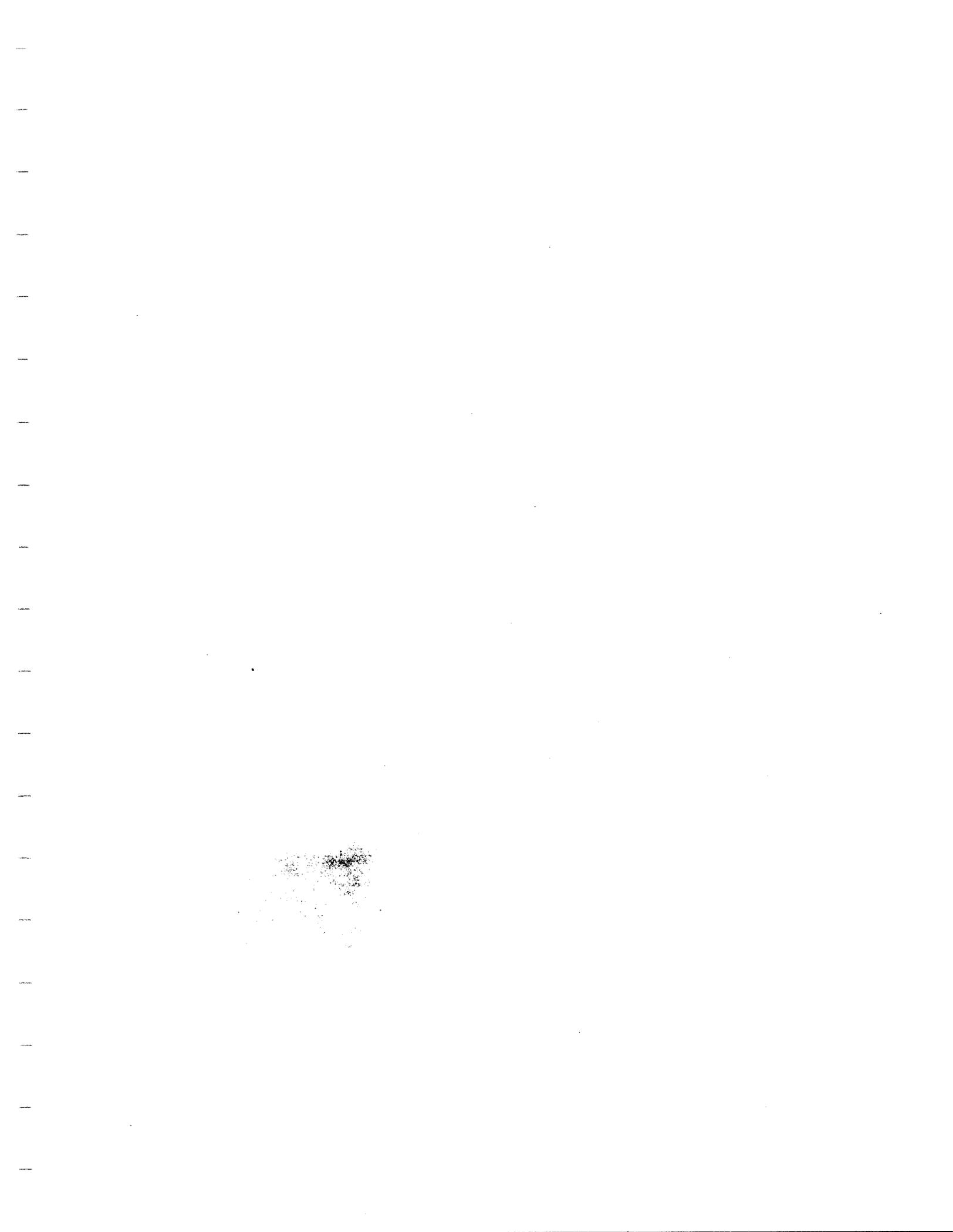


Figure 7. Theoretical BER Curves Comparing Performance In AWGN Vs. A Fading Channel (Reprinted from [11]).





ROBUST IBOC DAB AM AND FM TECHNOLOGY FOR DIGITAL AUDIO BROADCASTING

Brian W. Kroeger, D.Sc.

Paul J. Peyla

Westinghouse Wireless Solutions Co., Linthicum, MD

ABSTRACT

A robust In-Band On-Channel (IBOC) Digital Audio Broadcast (DAB) System for improved performance over existing AM and FM broadcasting is under development by Westinghouse for USA Digital Radio. The solution is both forward and backward compatible without the allocation of additional channel spectrum. Broadcasters can simultaneously transmit both analog and digital signals within the allocated channel mask allowing full compatibility with existing analog receivers. The solution also allows broadcasters to transmit an all-digital signal, replacing the hybrid analog/digital signal. The solution is tolerant of interference from adjacent channels, or interference from the co-channel analog transmission, even in a multiple station, strong-signal urban market. This paper describes spectral occupancy, power ratios, modulation formats and coding as well as the introduction of frequency and time diversity. It also addresses the adoption of a forward compatible all-digital transmission for both AM and FM broadcasting.

I. INTRODUCTION AND BACKGROUND

Digital Audio Broadcasting is a medium for providing digital-quality audio, superior to existing analog broadcasting formats. The advantages of digital transmission for audio include better signal quality with less noise and wider dynamic range than with existing FM and AM radio. The goal of FM DAB is to provide virtual-CD quality stereo audio along with an ancillary data channel with optional capacity up to 64 kbps depending upon a particular station's interference environment. The goal of AM DAB is to provide stereo audio with quality comparable to present analog FM quality and a 2.4 kbps ancillary data channel. The development of new high-quality stereo codec algorithms indicates that virtual-CD stereo quality will soon be practical at rates as low as 96 kbps while stereo audio, startlingly superior in quality to existing AM audio, can be attained at 48 kbps. IBOC requires no new spectral allocations because each DAB signal is simultaneously transmitted within the same spectral mask of an

existing allocation. IBOC DAB is designed, through power level and spectral occupancy, to be transparent to the analog radio listener. IBOC promotes economy of spectrum while enabling broadcasters to supply digital quality audio to their present base of listeners.

An independent technical evaluation conducted by the *Deskin Research Group* in 1996 revealed various weaknesses [1] in the previously proposed FM IBOC systems [5]. These deficiencies included DAB interference to host, first and second adjacent interference, and lack of robustness in multipath fading. These deficiencies were addressed in subsequent development work as reported in [2], and substantially eliminated in the new design which has evolved over the past year since the *Deskin* study.

The design of the AM DAB is continuing as planned at *Xetron*. A very brief overview of the AM IBOC DAB system is also presented in this paper.

II. FM OFDM IBOC SYSTEM DESCRIPTION

A brief description of the IBOC simulation model is presented here. Several modulation techniques were evaluated for the IBOC DAB application, including multicarrier spread spectrum, high-rate single carriers and Orthogonal Frequency Division Multiplexing (OFDM). Tradeoff analyses led to the selection of OFDM. OFDM modulation has been shown to be tolerant of multipath fading when used in conjunction with FEC coding and interleaving. Furthermore, OFDM can be tailored to fit an interference environment that is nonuniform across frequency while also providing flexibility for additional optional subcarriers.

The DAB signal is transmitted on OFDM subcarriers located on either side of the analog spectrum. A spectral mask along with the FM and DAB power spectral densities is presented in Figure 1. Note that the FM spectral mask is defined as peak power measured in a 1 kHz bandwidth over any 5 minute interval. The power spectral density for the FM signal was empirically determined by power-averaging the FM spectrum over 5 minutes. Five stations in the Baltimore/Washington area exhibited the triangular power spectral density with a slope between 0.35 and 0.38 dB/kHz. Interestingly the

stations measured included diverse signals ranging from "heavy metal" music to talk. The average slope of the 5 stations is 0.36 dB/kHz, which is assumed for the modulated spectral plots.

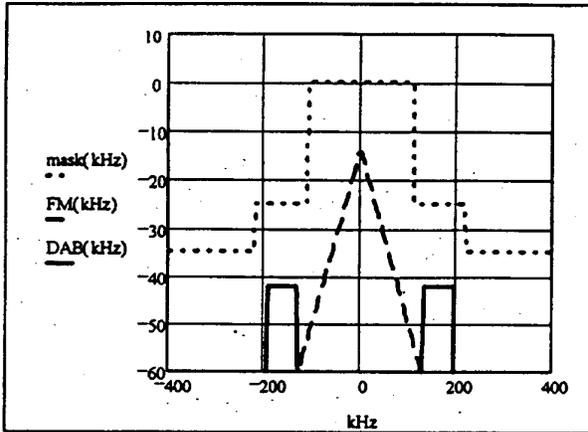


Figure 1. Power spectral densities of FM and DAB signals below FM spectral mask.

The total FM power can be found by integrating the triangular power spectral density.

$$P_{total} = \int_{-\infty}^{\infty} P_{peak} \cdot 10^{-0.36|f|/10} \cdot df = 24.12747 \cdot P_{peak}$$

Then the peak of the FM power spectral density is located 13.8 dB ($10 \cdot \log(24.12747)$) below the total carrier power reference level (0 dB) as shown in Figure 1. The DAB power level on each side of the FM spectrum is placed 25 dB below the total FM power (this value is adjustable by the broadcaster to accommodate special interference situations). The DAB density in a 1 kHz bandwidth can be calculated. The power spectral density of the DAB signal can be very closely approximated by dividing its total power by its effective Nyquist Bandwidth.

$$PSD_{DAB} = \frac{10^{-25/10}}{81 \cdot 0.796875} = 4.9 \cdot 10^{-5}$$

Then the power spectral density of the DAB signal in dB as shown in Figure 1 is computed to be -43 dB/kHz ($10 \cdot \log(4.9 \cdot 10^{-5})$).

The baseline DAB system assumes 81 subcarriers above and 81 below the host FM spectrum. Each DAB subcarrier is QPSK modulated at a symbol rate of 750 Hz. The inphase and quadrature pulse shapes are root raised cosine tapered (excess time=17/16) at the edges to suppress the spectral sidelobes. Although this pulse shape reduces the throughput capacity relative to the rectangular pulse

by 5.88%, performance in multipath is improved and the resulting spectral sidelobes are reduced, lowering interference. This pulse shape results in orthogonal subcarrier frequency spacing of 796.875 Hz. A plot of the pulse shape normalized to 1 unit of time is presented in Figure 2.

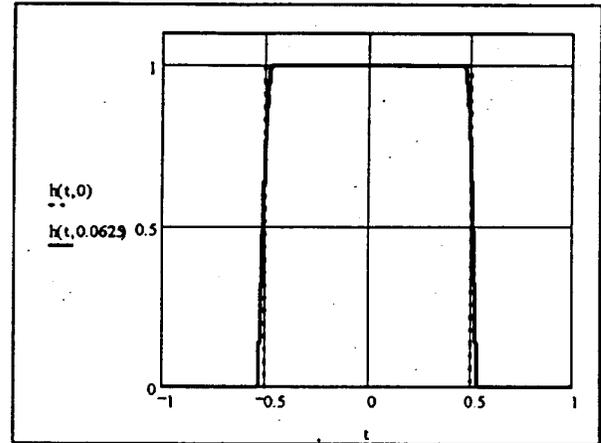


Figure 2. Plot showing rectangular Nyquist pulse (dotted) and the root raised cosine tapered pulse (solid).

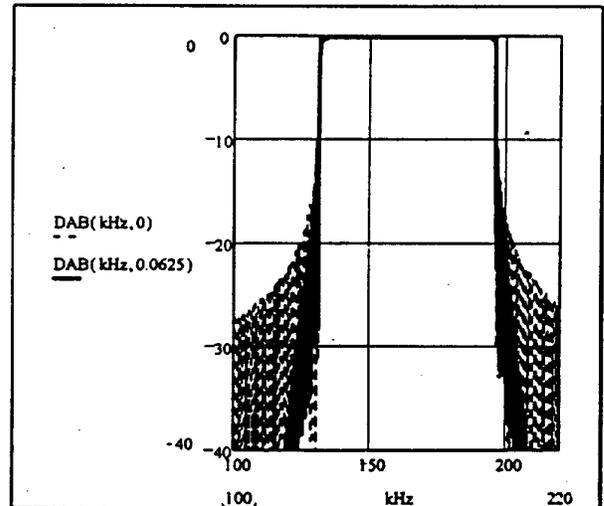


Figure 3. Improved spectral sidelobe suppression of Nyquist root raised cosine tapered pulse (solid) and rectangular Nyquist pulse (dotted).

Figure 3 shows plots of the DAB spectra using both the rectangular and root raised cosine pulse shapes. The 81 subcarriers in this case are Nyquist-spaced at 796.875 Hz. The effective Nyquist pulse time is approximately 1.2549 milliseconds for both cases

yielding an effective Nyquist bandwidth of 796.875 Hz.

Potential subcarrier locations are identified by their offset from the host FM center frequency. For example, a reference subcarrier is placed at 144234.375 Hz above and below the FM center frequency. This reference subcarrier position is number 181 from the FM center frequency ($181 \times 796.875 = 144234.375$ Hz). The reference subcarrier is used to aid in acquisition and tracking of the symbol timing and carrier frequency. A narrowband phase locked loops tracks the reference subcarrier, effectively rendering it immune to dynamic fading conditions. The reference subcarrier also provides the local reference to initiate differential decoding at the receiver.

Subcarriers 182 through 245 carry 96 kbps. Subcarriers 165 through 180 can carrier an additional 24 kbps of FEC coded bits to create an effective code rate of $R=4/5$ on each side of the FM signal. The placement of DAB at ± 15 kHz about 114 kHz is avoided in the baseline system in order to reduce the noise introduced into inadequately filtered receivers. However the broadcaster will have the option to utilize this portion of the spectrum to improve robustness of the digital audio signal and/or to provide additional datacasting capacity.

Although each DAB sideband can be demodulated independently of the other, the intention is to combine the two sidebands, yielding a power gain of 3 dB, plus additional coding gain achieved by a $R=1/2$ code over the $4/5$ coding gain on each independent sideband.

The total capacity of each sideband is 120 kbps (uncoded). After $R=4/5$ rate FEC coding, the coded capacity is 96 kbps for each redundant sideband. This data rate is sufficient for transmission of virtual-CD quality music plus a modest datacasting capacity. Optionally, additional carriers can be added to increase the datacasting capacity. These carriers would be located closer to the host analog FM signal.

III. INTERFERENCE ANALYSIS

The interference to and from the first adjacent channels placed ± 200 kHz from the host signal can be derived from the relationship of the adjacent signals shown in the plot of Figure 4. FM stations are geographically placed such that the nominal received power of an undesired adjacent channel is at least 6 dB below the desired station's power at the edge of its coverage area. Then the D/U (desired to undesired power ratio in dB) is at least 6 dB. Knowledge of the

ratio of each station's DAB signal power to its FM host permits assessment of first adjacent interference to DAB. Similarly the interference of the first adjacent DAB to the host FM signal can be assessed from the relationship.

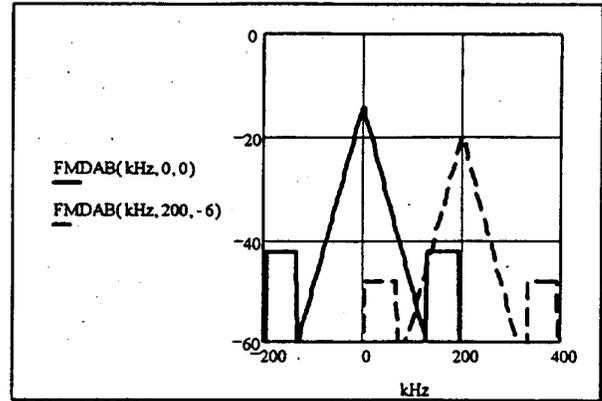


Figure 4. Interference scenario showing first adjacent at -6 dB (worst case edge of coverage).

Figure 5 illustrates the need for DAB spectral sidelobe suppression and bandlimiting due to the second adjacent DAB interference to the host DAB signal. At a station's edge of coverage, a second adjacent's nominal power can be up to 20 dB greater than the host's nominal power.

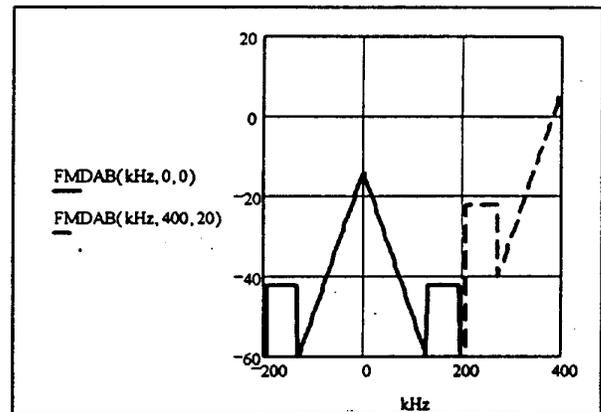


Figure 5. Interference scenario with second adjacent at +20 dB.

The effects of the various interference scenarios illustrated here are quantified through analysis and supported through simulation and testing. Analysis of the DAB to first adjacent interference at the edge of coverage showed that the total DAB signal should be set about -22 dB relative to its FM power.

The solution to the first adjacent interference problem is to place redundant, although not identical,

DAB signals on either side of the carrier. Although the potential capacity is halved with this redundancy, interference problems are substantially reduced and substantial coding gain is achieved after combining both halves. A survey of existing U.S. radio allocations shows that it is very unlikely that both upper and lower adjacent channel interferers are present at their maximum interference levels (-6 dB) at the same geographic location within the host's coverage area. This frequency diversity is especially useful when multipath interference or spectral notches affect one sideband or the other.

A variety of simulations and analyses have characterized performance of the host FM signal in the presence of IBOC DAB. Specifically, main audio channel performance, SCAs, adjacent channels, and stereo subcarrier demodulation were investigated with an IBOC DAB signal appended to the host FM.

Main audio channel performance

Simulations have provided valuable insight into the character of FM post-detection noise in the presence of an IBOC DAB signal. For instance, results indicate that the audio noise level increases with the deviation of the FM signal. In fact, Figure 6 illustrates a significant rise in the post-detection noise power spectral density (PSD) as the FM deviation varies from minimum to maximum in the presence of an IBOC DAB signal placed between 78 kHz and 197 kHz from the FM carrier.

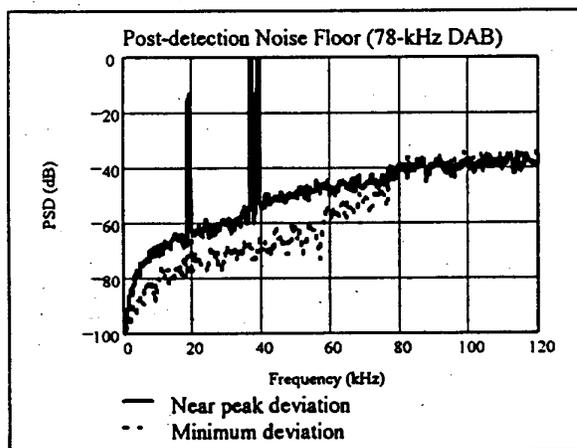


Figure 6. Audio Deviation Effects.

The nonlinear FM detector is responsible for intermodulating overlapping portions of the host FM and DAB spectra. The products are folding back into the post-detection audio band and raising its noise floor. Similar observations and conclusions were

independently reached by the Electronic Industries Association (EIA) during their IBOC DAB testing [4].

Although these results are intriguing, they do not predict a degradation in host FM audio quality due to IBOC DAB. Because the DAB-induced post-detection noise floor increases in proportion to the deviation of the FM signal, the effect is self-masking: audio noise will be lowest during quiet passages, and highest only when the audio is loudest. Simulations have demonstrated this phenomenon.

The absolute level of host FM degradation will depend on the particular configuration of DAB. To determine the relationship between DAB location and audio signal-to-noise ratio (SNR), a number of performance tests were run when DAB noise would be most audible – during quiet passages of minimum FM deviation. Simulations were performed in which the receiver audio dynamic range was measured with only a 10%-deviated, 19-kHz-pilot-modulated FM signal and a DAB signal input to an FM stereo receiver located at the transmitter. The total power of the DAB signal was 22 dB below the power of the FM carrier. In the first four tests, the DAB was modulated using orthogonal frequency-division multiplexing (OFDM) with 4750-symbol-per-second quadrature phase-shift keying (QPSK) subcarriers using rectangular pulse shaping. The fifth test employed DAB with four times the number of OFDM carriers – each occupying one-fourth the bandwidth (1187.5 Hz) – and root-raised-cosine pulse shaping (to reduce spectral sidelobes that interfere with the host FM). In each test, the spectral occupancy of the DAB signal was changed: the start frequency was varied with respect to the FM center frequency, while the stop frequency was fixed at 197 kHz. Table 1 summarizes the results.

Table 1 - Audio Dynamic Range at Transmitter (peak-to-noise-floor SNR)	
DAB start frequency	Audio SNR (dB/15 kHz)
78 kHz	64.7
100 kHz	67.3
124 kHz	68.3
129 kHz	68.8
129 kHz, pulse shaped	77.6

These results indicate that moving the DAB away from the FM carrier, increasing the number of DAB carriers, and pulse shaping the transmitted DAB symbols to reduce spectral sidelobes will significantly improve the performance of the host FM. Modulation and coding characteristics of the DAB signal can be traded for spectral occupancy to meet these goals.

Note that the new DAB baseline employs subcarriers spaced at 796 Hz which would improve performance over the carrier spacings reported here.

Audio simulations have verified that an SNR of 77.6 dB during quiet passages should render DAB-induced audio noise imperceptible to the listener. Furthermore, implementation constraints limit the SNR of typical receivers to around 60 dB. The noise engendered by these receivers will mask any degradation caused by DAB. The -22-dB, 129-kHz pulse-shaped DAB configuration is used as the baseline for the balance of this discussion.

SCA performance

SCAs (Subsidiary Communications Authorization) are optional channels multiplexed onto the baseband stereo spectrum from 53 kHz to 100 kHz. The SCA signal, which can be analog or digital, is transmitted by some FM stations for the use of private subscribers who typically pay for program material. Simulations were used to determine the impact of SCAs on IBOC DAB host FM performance, and to determine the impact of DAB on the performance of SCAs. SCAs with 10% deviation at 67 kHz and 92 kHz were simulated because they represent a large percentage of operational subcarriers.

In the current analog FM system, SCAs generally cause negligible interference to the host FM signal. However, when DAB is present, the addition of SCAs could increase the host FM audio noise floor due to the DAB/FM intermodulation effect described above. Figure 7 illustrates stereo subcarrier sensitivity to 92-kHz SCAs when subject to a pulse-shaped (PS) DAB signal starting at 129 kHz. In this case, the 92-kHz SCA reduces the host FM audio SNR from 77.6 to 69.8 dB; however, this noise level is still too low to produce audible effects. Figure 8 shows that SCAs located at 67 kHz have even less impact on audio performance.

Due to their location at the high end of the baseband spectrum, some SCAs currently operate at low SNRs because the post-detection noise floor increases with the square of the frequency. When DAB is added, the deviation of a wideband host FM signal into its IBOC DAB signal produces intermodulation which increases the post-detection noise floor, particularly in the higher baseband frequencies (since this is nearest the location of the pre-detection DAB). Moreover, the noise masking effect described above does not apply for SCAs, since their audio may be quiet while the main audio channel, at peak deviation, is causing an increase in the SCA noise floor.

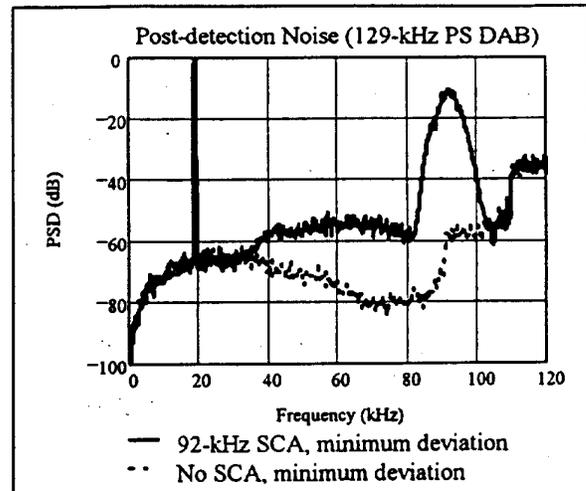


Figure 7. Effects of 92-kHz SCA

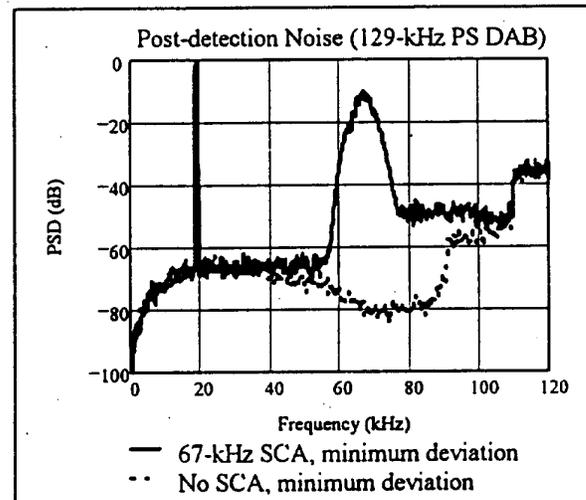


Figure 8. Effects of 67-kHz SCA.

Simulations were performed using SCAs with peak-deviated audio signals in the presence of a -22-dB, 129-kHz pulse-shaped DAB signal. Figure 9 indicates that the SNR of a 67-kHz SCA (in a 10-kHz bandwidth) is 25-30 dB at the transmitter when the main audio channel is near maximum deviation.

For 92-kHz SCAs, the SNR is 20-25 dB, as illustrated in Figure 10.

Without DAB, typical noise floors are roughly 40 dB. The increase in noise floor should not pose a problem for digital SCAs (e.g., Seiko and Radio Broadcast Data System), since they should be robust enough to operate at reasonably low SNRs.

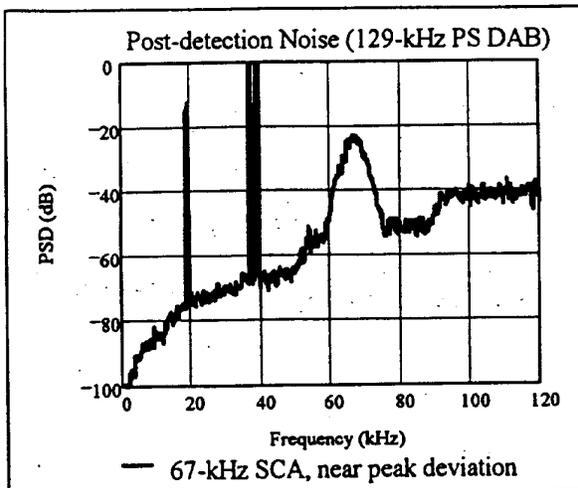


Figure 9. 67-kHz SCA Performance.

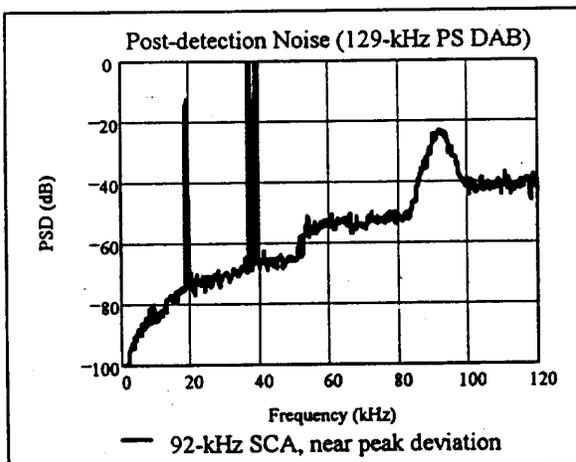


Figure 10. 92-kHz SCA Performance.

Adjacent channel performance

The Federal Communications Commission defines the “edge of coverage” for class B stations as the 54-dBu coverage contour, where 0 dBu is equivalent to 1 microvolt per meter field strength. Class B stations are protected to the 54-dBu contour from 48-dBu interference introduced by first adjacent stations. Thus, at the edge of coverage, the minimum desired-to-undesired signal ratio (D/U) is 6 dB. Simulations have quantified the amount of degradation that would be introduced into the desired IBOC DAB host FM signal when located at the edge of coverage and subject to interference from a -6 dB IBOC DAB first adjacent.

To properly interpret simulation results, it is first necessary to calculate the audio SNR of the simulated receiver at the 54-dBu contour, assuming that the noise contribution is due solely to ambient additive white gaussian noise (AWGN). Assuming a

mid-band carrier frequency of 100 MHz and a half-wave dipole antenna, electric field intensity E (V/m) can be converted to carrier power C (W) at the input to the FM receiver using

$$C = \frac{E^2}{120\pi} A_e$$

where $A_e = 1.177 \text{ m}^2$ is the effective aperture of the half-wave dipole antenna. Using this formula, a 54 dBu field strength corresponds to a -91.1 dBW carrier power.

An ambient noise temperature of 10,000 K is representative of the FM frequency band; in a 15-kHz bandwidth, this temperature produces a noise power of -146.8 dBW. Hence, a carrier power of -91.1 dBW at the receiver antenna terminals would yield a 55.7 dB/15 kHz carrier-to-noise ratio (CNR). The receiver noise characteristic enables one to determine audio SNR given an input CNR. Using the measured noise characteristic of the simulated FM stereo receiver, this input CNR corresponds to an audio SNR of 64.4 dB/15 kHz.

Recall that a -22-dB pulse-shaped IBOC DAB signal starting at 129 kHz yields an audio SNR of 77.6 dB at the transmitter (and at the edge of coverage if ambient noise were ignored). The preceding calculations demonstrate that, at the edge of coverage, the contribution to audio SNR is dominated by ambient noise; the effects of -22-dB, 129-kHz pulse-shaped IBOC DAB are negligible.

In the simulation, a -22-dB pulse-shaped DAB signal starting at 129 kHz was added to both a quiet FM host at the transmitter (10%-deviated 19-kHz pilot, with no audio or SCAs) and a -6 dB, fully modulated first adjacent. A 150-kHz pre-detection filter (300-kHz total 3-dB bandwidth) was used in the simulated FM stereo receiver. The signal at the input to the FM demodulator is shown in Figure 11.

Results indicate that introduction of the adjacent IBOC DAB channel degrades the audio SNR to 50.0 dB. Although the simulation was performed with the desired signal located at its transmitter, it is clear that the introduction of noise associated with translation to the edge of coverage would be negligible. Therefore, when a -6-dB first-adjacent FM/DAB signal impinges on a quiet host FM/DAB signal at the edge of coverage, the SNR degrades from around 64 dB to 50 dB. Figure 12 illustrates this effect.

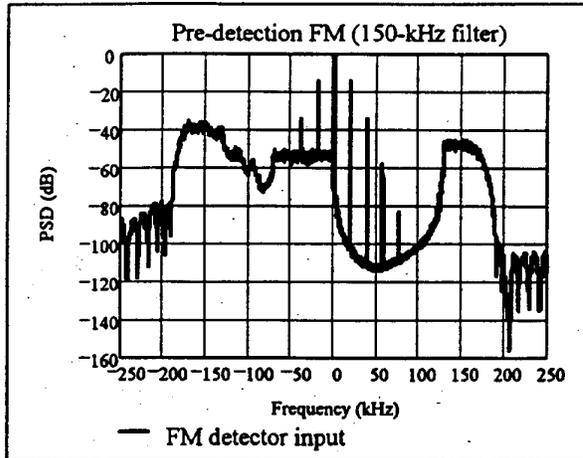


Figure 11. Pre-detection Effect of First Adjacent with 129-kHz PS DAB at edge of coverage.

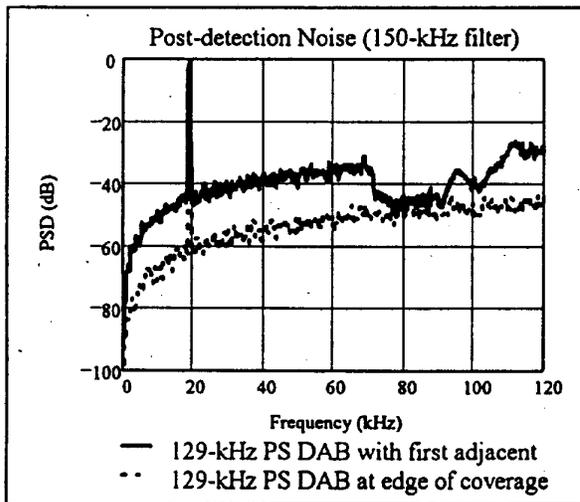


Figure 12. Post-detection Effect of First Adjacent with 129-kHz PS DAB at Edge of Coverage.

While the SNR is diminished, it should be noted that the degradation is highly geographically localized; performance will improve rapidly as the receiver moves farther from the interfering station or closer to the desired station. In addition, due to receiver implementation constraints, actual receivers will experience less than 64-dB SNRs at the edge of coverage. Most automotive receivers are blended to mono at the edge of coverage anyway, mitigating the effects of first adjacent DAB interference. This would improve audio SNR by removing the effects of noise around the stereo subcarrier.

Stereo subcarrier demodulation

During EIA testing of the USADR FM-1 IBOC DAB system, certain inexpensive FM stereo receivers suffered an increase in audio noise when receiving an IBOC DAB FM stereo signal [4]. When the DAB signal was removed from the FM signal, the audio noise disappeared. Investigations revealed that the problem was caused by inadequate filtering of the post-detection baseband stereo multiplex signal. The new baseline DAB waveform has been designed to mitigate this effect.

To recover the stereo information, the 30-kHz-wide, double-sideband amplitude-modulated (DSB) left-minus-right (L-R) signal centered at 38 kHz is demodulated using a 38-kHz local oscillator (LO), and subsequently filtered with a 15-kHz lowpass filter. In most receivers, the 38-kHz LO is simply a square wave, with a 38-kHz fundamental and odd harmonics at 114 kHz, 190 kHz, etc. As a result, in the absence of adequate filtering, not only is the desired L-R signal recovered, but so is any energy in the multiplex signal that lies within ± 15 kHz of 114 kHz and 190 kHz.

In the presence of AWGN only (no DAB), this effect is not pronounced. A well-known property of large-signal FM detection in AWGN indicates that the power spectral density of the post-detection noise is directly proportional to the square of the frequency. Hence, the noise power spectral density at 114 kHz is 9 times that at 38 kHz (9.5 dB), and the noise at 190 kHz has 25 times the power (14.0 dB). High noise levels are mitigated because the amplitude of square wave harmonics decreases with their order: if the 38-kHz fundamental has unit amplitude, the 114-kHz third harmonic has amplitude 1/3 (-9.5 dB), and the 190-kHz fifth harmonic has amplitude 1/5 (-14.0 dB). Therefore, the noise contribution from each harmonic is equal to the noise under the desired signal; this causes a 4.8-dB degradation due to AWGN alone (without DAB) in receivers which do not filter the noise around their LO harmonics.

This decrease in SNR is avoided in well-designed receivers. Some receivers use "Walsh" decoders; others simply filter the baseband multiplex signal prior to DSB demodulation, which effectively eliminates components outside the desired L-R band. Most receivers – even those without such post-detection protection – should ameliorate the effects of the 190-kHz fifth harmonic by pre-detection filtering, since a good design would significantly filter the first adjacent FM signal centered 200-kHz from the desired channel.

Thus, in the presence of AWGN alone, certain inexpensive receivers which employ little or no

post-detection protection experience up to a 3-dB stereo SNR degradation (from their DSB LO third harmonic) when compared to their more carefully designed counterparts. Of course, no significant degradation exists when receiving a monaural signal.

As discovered during FM-1 EIA testing, this 3-dB stereo SNR degradation increases when IBOC DAB is added to the analog FM signal [4]. In order to scope the magnitude of the problem, simulations were performed using a well-designed FM stereo receiver with ample protection from 38-kHz harmonics.

Three simulations were run: the first simulated performance in a well-designed receiver by adding AWGN only to a quiet analog FM signal at a level which produced a 64-dB SNR in the left audio receiver channel. The second added DAB only (from 78 kHz to 197 kHz) to the quiet FM signal, at a level which likewise produced a 64-dB SNR in the left audio channel. As shown in Figure 13, the post-detection noise power in the 0-53-kHz audio band is identical for the two simulations (hence the equal 64-dB SNRs).

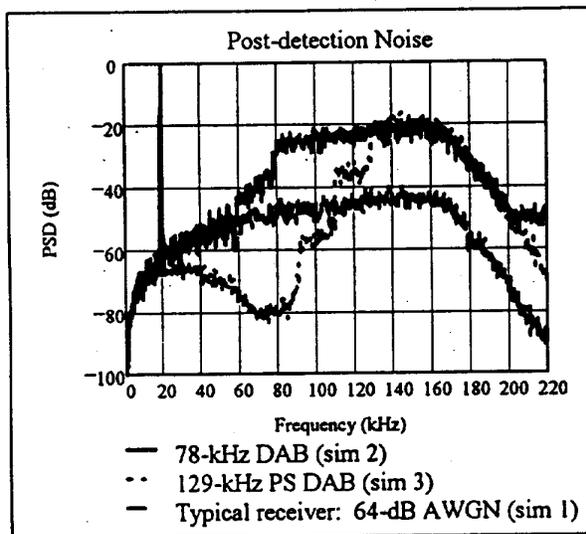


Figure 13. Effect of DAB on 114-kHz Noise Floor.

Note, however, that the noise floors diverge above 60 kHz. In fact, the DAB-induced noise floor is approximately 25 dB higher in the 30-kHz band around 114-kHz. If the simulated receiver did not sufficiently filter the post-detection noise floor, the stereo noise increase would have degraded the audio SNR well below 64 dB in the second simulation.

It has been suggested that simply suppressing DAB energy in the 114±15 kHz band would eliminate the post-detection noise in this region. Due to the non-linear nature of the FM demodulator, this is not entirely the case. Instead, simulations have shown

that such a notching of DAB carriers creates a 12-dB improvement across the 30-kHz band around 114 kHz. Thus, simply limiting DAB bandwidth might still cause stereo SNR degradation in radios with inadequate post-detection filtering.

Significant improvements, however, were observed in the third simulation, in which the DAB signal was moved beyond 129 kHz and pulse shaping was applied. Pulse shaping causes a significant decrease in the noise floor across much of the post-detection band, as illustrated in Figure 13 (129-kHz PS DAB). Figure 14 provides a magnified view of the 30-kHz region around 114 kHz.

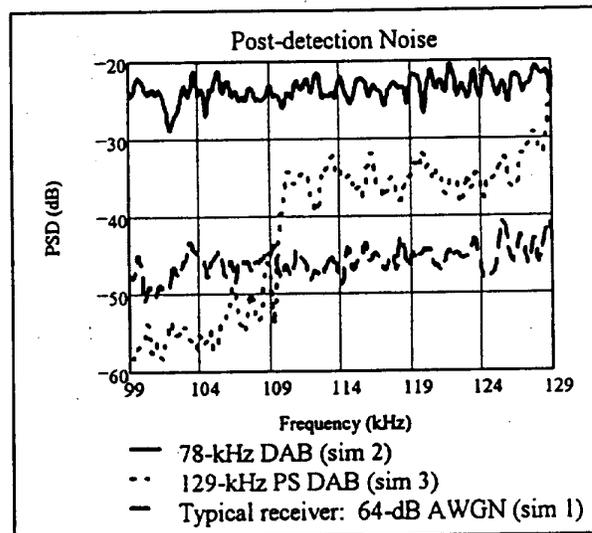


Figure 14. Effect of DAB Placement and Pulse Shaping on 114-kHz Noise Floor.

Note that the noise floor steps up at 110 kHz, due to mixing of the Bessel-weighted 19-kHz pilot harmonics with the DAB signal during FM demodulation. As a result, above 110 kHz, an improvement of around 10 dB is gained over that afforded by 78-kHz (non-pulse-shaped) DAB. Below 110 kHz, however, a 30-dB improvement is observed. Thus, using DAB placement and pulse shaping, the overall stereo noise increase due to the addition of DAB in radios with inadequate filtering can effectively be limited to acceptable levels.

The preceding analysis presents a worst-case bound on the 114-kHz degradation due to DAB; in reality, even the most poorly designed receivers should provide some degree of pre-detection filtering to mitigate the noise level around 114 kHz. Furthermore, in a typical environment, with practical receiver implementations, it is probable that the degradation will be imperceptible to the listener. Note

that none of the car radios employed in the EIA tests exhibited the problem [4].

Interference Summary

Westinghouse has analyzed the impact on performance of the host FM signal in the presence of various IBOC DAB configurations. Simulations and analysis indicate that FM performance is least affected when a pulse-shaped DAB signal is placed between 129 kHz and 197 kHz from the FM carrier. Modulation and coding tradeoffs can be exercised to provide the spectral efficiency required to fit the DAB signal within this bandwidth.

This DAB configuration yields an audio SNR of nearly 78 dB during periods of minimum deviation, with noise during louder passages rendered inaudible to the listener via a masking effect. Even when quiet, noise due to DAB will probably be masked by the noise produced in most typical receivers. SCA interference with the host FM should likewise be inaudible, while the SCAs themselves should perform with SNRs of around 20-30 dB/10 kHz (ample margin for a digital SCA).

When a high-level first adjacent interferer with DAB is present, the audio SNR of the host (with DAB) degrades to 50 dB at the edge of coverage. However, the degradation is highly localized. Most automotive receivers are blended to mono at the edge of coverage anyway, mitigating the effects of first adjacent DAB interference.

Finally, a slight degradation may be observed during stereo subcarrier demodulation in existing inexpensive FM stereo receivers. This degradation, which has not been demonstrated in car radios, may prove imperceptible in typical listening environments.

IV. CHANNEL CODING

Forward error correction and interleaving improve the reliability of the transmitted information. In the presence of adjacent channel interference, the outer OFDM subcarriers are most vulnerable to corruption, and the interference on the upper and lower sidebands is independent. The information, coding and interleaving are specially tailored to deal with this nonuniform interference such that the communication of information is robust. Specifically, this nonuniform interference is the focus here where special coding and error handling results in more robust performance.

The IBOC DAB system will transmit all the digital audio information on each DAB sideband (upper or lower) of the FM carrier. Recall that the baseline system constrains the DAB signal to within

130 kHz to 197 kHz above and below the FM center frequency, as shown in Figure 1. Each sideband can be detected and decoded independently with an FEC coding gain achieved by a rate 4/5 convolutional code on each sideband. This redundancy permits operation on one sideband while the other is corrupted. However, usually both sides are combined to provide additional signal power and coding gain. Furthermore special techniques are employed to demodulate and separate strong first adjacent interferers such that a "recovered" DAB sideband can supplement the opposite sideband to improve coding gain and signal power over any one sideband.

The goal here is to transmit the DAB signal on both the upper and lower sidebands such that the sidebands can be independently detected and decoded, each with some FEC coding gain. Additional coding gain, along with some power gain of course, is desired when both sidebands can be combined. The reason for these requirements is that the interference on each sideband is independent of the other; however, the level of interference across the subcarriers on any one sideband is related to the power spectral density of the adjacent interferer. Therefore the grouping of independently detectable and decodable sidebands is appropriate.

In order to effectively achieve coding gain when the pair of sidebands is combined, the code on each sideband should consist of a subset of a larger (lower rate) code. Each subset can be designed through "complementary" puncturing of the lower rate code.

A simple way of constructing a code for this application is to start with a rate 1/3 convolutional code. This code can be generated as shown in Figure 15.

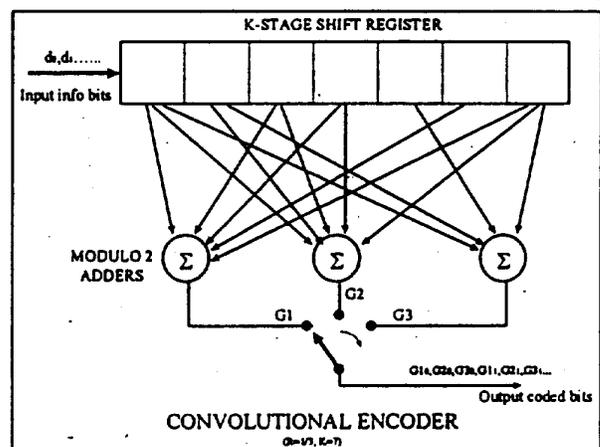


Figure 15. Example of a R=1/3, K=7 convolutional encoder.

The R=1/3 convolutional encoder of Figure 15 can be viewed as producing 3 encoded bit streams (G1, G2 and G3), each at the same bit rate as the input. The combination of these 3 bit streams produces the R=1/3 code output. To create the complementary code pair, for example, a subset of the output code bits is assigned to the lower DAB sideband and a different subset is assigned to the upper sideband. Each subset must contain at least the same rate of bits as the information input rate, plus some additional bits to provide some coding gain. A R=4/5 code on each sideband requires 25% additional bits. One method of allotting bits to the sidebands is presented in Figure 16. The Figure shows the relative spectral locations of the coded bits. These spectral locations are maintained after interleaving by channelizing the interleaver into 8 distinct segments and mapping the segment outputs to the appropriate subcarriers on each sideband.

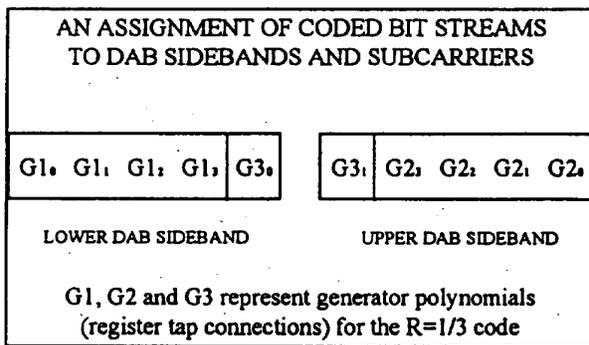


Figure 16. Example of bit partitioning.

V. BLEND WITH TIME DIVERSITY

Perhaps the most effective method for dealing with the nonstationary mobile radio channel is to provide time diversity between two independent transmissions of the same audio source. Both AM and FM IBOC DAB concepts inherently provide this ability by delaying the analog transmission by a fixed time offset relative to the decoded DAB audio transmission. When the DAB transmission is blocked (or corrupted for any reason) for a short time, then the outage at the DAB decoder is heard after the diversity delay. This diversity delay is incurred at the receiver and is comprised of deinterleaving and FEC decoding delay, audio decoding delay, and any additional delay for diversity improvement. The FEC decoder can be used to identify faulty audio frames and, therefore, the exact time of the DAB audio outage can be predicted. If the channel becomes unblocked after the diversity delay, then the analog signal can be demodulated such

that its detected audio output can be blended in while blending out the faulty DAB segment. The listener may detect the temporary degradation in audio quality during the analog blend duration, but will not experience muting or undesirable artifacts.

If the diversity delay is sufficiently large such that the DAB and analog outages are independent, then the probability of an outage after diversity is the square of the probability of outage without diversity. For instance, if the probability of an outage is 1.0%, then the probability of outage after diversity is 0.01%, which is a great improvement. The actual performance can be quantified with knowledge of the autocorrelation function of the channel outage due to severe impairment. This autocorrelation function is expressed as

$$R(\tau) = E\{x(t) \cdot x(t - \tau)\}$$

where $x(t)$ is defined as the stochastic process of the channel loss probability such that a "1" is assigned when the channel is lost and a "0" is assigned when the channel is clear, and τ is the diversity delay time offset. The probability of outage without diversity is $p = E\{x(t)\}$. The autocorrelation function represents the probability of channel outage after diversity improvement as a function of time offset.

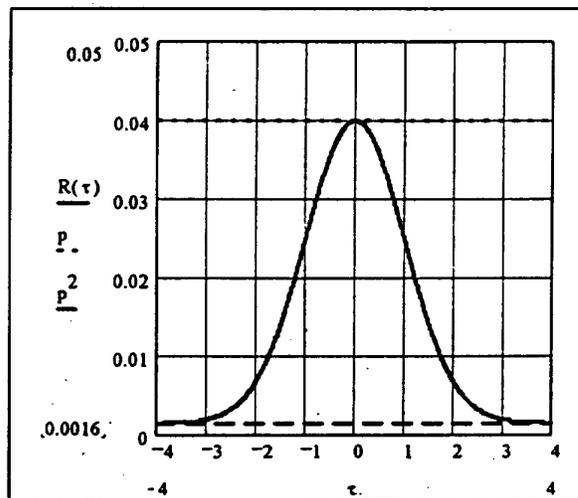


Figure 17. Example Autocorrelation Function of channel loss due to blockage or severe impairment ($p=0.04$).

An example autocorrelation function is shown in Figure 17; however, an actual autocorrelation function depends on distance from the station, terrain, propagation conditions etc. The figure shows that if the analog signal is not delayed relative to the DAB signal (zero time delay), then the outages are correlated and no benefit is gained from blending

since the probability of outage remains the same as without diversity. If the delay is large, then events become uncorrelated and the probability approaches the square of the probability without diversity.

The blend feature also solves the problem of fast tuning time. Without blend, a receiver would incur the diversity delay after tuning to a station before the listener hears the audio. The blend feature will demodulate the analog signal almost instantly, allowing the listener to hear the selection before blending to DAB several seconds later.

VI. ALL-DIGITAL DAB WITH TIME DIVERSITY

The IBOC designs permit evolution to an all-digital DAB format. Without the host analog signal present, the DAB will be transmitted within the primary spectral channel allocation (± 100 kHz). Adjacent channel interference issues are alleviated. The DAB power can be increased by as much as 20 dB, substantially increasing the DAB coverage area. The transmission format will include normal compressed audio plus a more compressed monophonic version of the same signal which is delayed by the diversity time offset. FEC with interleaving is applied to the normal compressed audio. The lower rate audio signal is used for blending during outages in place of the analog signal in the IBOC technique. Furthermore this lower rate audio signal employs FEC coding without interleaving. Therefore the all-DAB signal format facilitates fast tuning and exploits time and frequency diversity with the lower-rate redundant digital audio signal.

VII. AM OFDM IBOC SYSTEM DESCRIPTION

The AM DAB system is only briefly described here. Many of the characteristics of the AM DAB are similar to the FM DAB such as OFDM modulation, audio compression and time diversity. The AM DAB signal is comprised of subcarriers spaced at 500 Hz as illustrated in Figure 18. The symbol rate of each subcarrier is slightly less than 500 Hz due to the guard time between symbol pulses. The carrier complexity starts at BPSK for the 2 carriers closest to the AM carrier, carrying 1 bit per symbol. The subcarriers from 1 kHz to 5 kHz on either side of the AM carrier employ (1,7) modulation carrying 3 bits per symbol while the carriers from 5.5 kHz to 14.5 kHz on either side of the AM carrier employ 32 QAM carrying 5 bits per symbol. Signal processing techniques are

employed to reduce the mutual interference between the AM and DAB signals.

The anticipated digital audio rate is 48 kbps, providing high-quality stereo, which is remarkably superior to standard AM. A modest datacasting rate of 2.4 kbps is also provided. The blend-to-analog feature for time diversity is also employed in the AM DAB system to yield robust performance in adverse conditions.

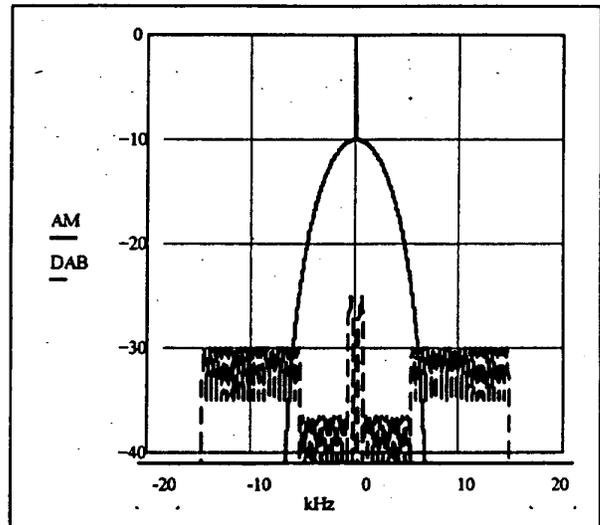


Figure 18. Spectrum of AM plus DAB.

The AM channel is being characterized at 3 frequencies across the AM band. The characterizations consist of transmitting and receiving a flat wideband signal, then analyzing the phase and amplitude characteristics. Channel information is collected only during identified events where the phase and/or amplitude is changing. This data is "played-back" as the channel in a simulator which enables evaluation and refinement of the tracking loops used in channel equalization, AGC, and the phase-locked loop. The statistics of channel events also determine the application and effectiveness of FEC coding and interleaving.

VIII. CONCLUSIONS

AM and FM IBOC DAB systems are being improved and upgraded by *USADR*. Detailed analysis and simulation results support the viability and robustness of these improved systems, of which demonstrations are anticipated in 1997.

Westinghouse has analyzed the impact on performance of the host FM signal in the presence of various IBOC DAB configurations. Simulations and

analysis indicate that FM performance is least affected when a pulse-shaped DAB signal is placed between 129 kHz and 197 kHz from the FM carrier. Modulation and coding tradeoffs can be exercised to provide the spectral efficiency required to fit the DAB signal within this bandwidth.

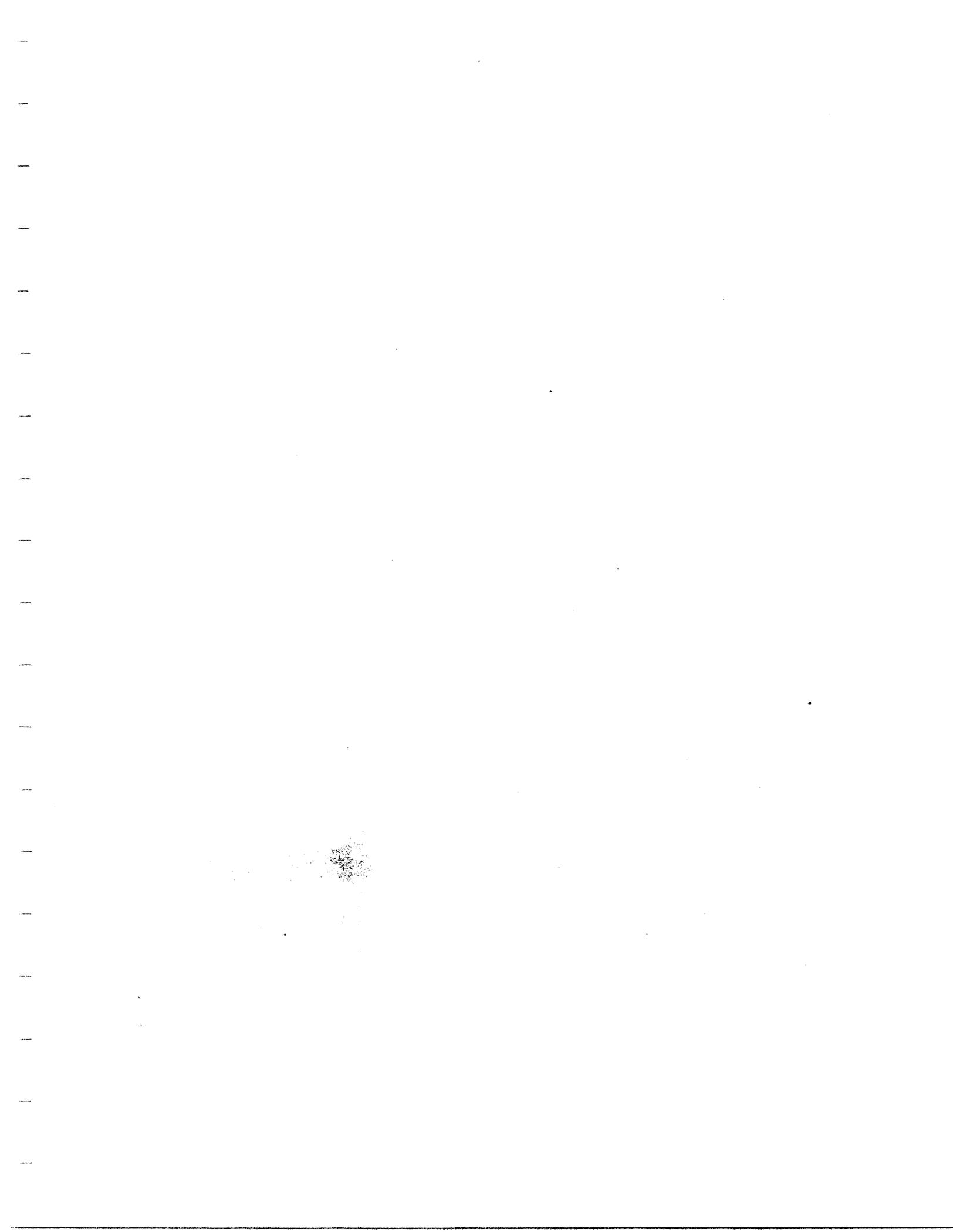
The FM IBOC DAB system will provide virtual-CD quality stereo audio using redundant spectral sidebands to provide frequency diversity and immunity to first adjacent interference. Time diversity is provided through interleaving. A blend-to-analog feature, with time diversity in the order of seconds, permits virtually instant tuning time while filling DAB audio gaps due to blockages or severe impairments. This feature will dramatically improve coverage in areas characterized by intermittent blockages.

AM IBOC DAB will provide stereo audio quality similar to existing FM analog. AM IBOC DAB will exploit interleaving and blending-to-analog with time diversity features similar to FM IBOC DAB.

AM and FM DAB will offer superior DAB coverage through an option to transition, at a future date, to a reduced-quality analog simulcast or to digital only. This option offers an increase in DAB power with the addition of a supplemental DAB transmission consisting of a lower rate compressed audio signal for time diversity reception and nearly instant tuning. This last feature is extremely effective against intermittent blockages and severe impairments, providing performance impossible to otherwise achieve with only frequency diversity, interleaving and FEC.

REFERENCES:

- [1] *USA Digital radio FM-1 Independent Audit Final Report*, D. Grybos, J. Marshall, prepared for USADR by Deskin Research Group, Santa Clara CA, June 1996.
- [2] *Improved IBPC DAB Technology for AM and FM Broadcasting*, B. Kroeger, A. J. Vigil, 1996 NAB SBE conference, Los Angeles, CA, November 1996.
- [3] Robert D. Culver, "Report of the Field Test Task Group; Field Test Data Presentation; Working Group B Testing of the CEMA-DAR Subcommittee; Appendix 6 - FM Modulation Increasing Baseband Noise in the Presence of an IBOC Digital Signal", Lohnes and Culver, Washington, DC, December 1996.
- [4] Robert D. Culver, "Report of the Field Test Task Group; Field Test Data Presentation; Working Group B Testing of the CEMA-DAR Subcommittee; Appendix 9, Analysis of IBOC DAR System Proposals", Lohnes and Culver, Washington, DC, December 1996.
- [5] *Method and System for Simultaneously Broadcasting and Receiving Digital and Analog Signals*, D. Kumar and B. Hunsinger, Patent Application S/N 08/294,140 filed July 1994.



Appendix 13

Summary of DAR Field Testing Procedures

The complete field testing description, procedures, test routes and data are contained in Reference [2], from which this summary is derived.

San Francisco, California was selected as the test venue for its presentation of varied reception conditions, including urban, rural and mountainous terrain, tall buildings, over-water paths, foliage, rural and mountainous areas, industrial and residential areas, and the availability of suitable transmission site(s).

A mobile measurement vehicle was constructed and equipped to make measurements on six test routes specified in Figures 2A-2F of this Appendix.

Measurements were recorded on computer every 1.171 cm traveled and included RF signal level and subjective assessments of received audio quality/impairment (clear, impaired, or failed), distance traveled, and the passage of pre-scripted landmarks en route. Audio recordings were made (on 8-channel DAT) and included the received DAR system audio, up to two analog FM stations co-located with the DAR transmit antenna (where possible, and using two reference car receivers), and vehicle cockpit audio during the measurements. Video tape recordings (S-VHS) included the route traveled, spectrum analyzers plots showing instantaneous RF signal level and RF spectrum near in frequency to the desired DAR transmission, and time-code. All measurements and recordings were synchronized using SMPTE time-code.

Twenty five test audio program segments were selected and transferred to compact disc with a total play time of just over 60 minutes.

The main transmission facilities were located at Mount Beacon, near Sausalito, California. Supplemental coverage of the Eureka-147/DAB system (using L-band frequencies) was also tested using additional transmission facilities. The VOA/JPL S-band satellite system was uplinked from the JPL laboratory in White Sands, New Mexico to a NASA TDRS Satellite in geo-synchronous orbit over Hawaii (with an approximate 23 degree elevation angle from San Francisco). Specific parameters are shown in Tables 1-3, below.

Data presentations in [2] include instantaneous RF signal level, vehicle velocity, audio events, and distance, landmark-to-landmark over the entire test route, along with a summary of total data records and subsets showing percentage of the route with received audio in clear, impaired or muted (failed) conditions. These are presented for all systems tested on all routes.

Additionally, all routes were measured on the AT&T IBAC FM frequency with the DAR system off-air, to examine the presence of potential co-channel interfering signals.

Further potential processing and presentation of measured data can be made. These might include equating RF values to a reference level, such as power at the input of the device under test, voltage at the antenna element output terminals, as field strength or power density, etc. The RF data points can be averaged over a sliding window to filter out short or longer term variations such as those due to modulation, multipath or short blockage fades. The measured RF signal can be analyzed for indications of multipath propagation and a "rating" applied to route sub segments. The various measured and calculated parameters can be compared with each other; such as comparing events to velocity or events to multipath presence and average longer term RF level. For shared VHF band system, the background RF can be compared to the device under test RF to investigate failures due to insufficient C/I ratio.

TABLE 1

**Transmitter parameters for the Eureka 147 DAB SFN installation
in San Francisco, July 1996**

Parameters\Sites	Mount Beacon		Mount San Bruno		Round Top Mountain
Latitude	37° 51' 03" N		37° 41' 15" N		37° 51' 00" N
Longitude	122° 29' 51" W		122° 26' 04" W		122° 11' 30" W
Transmitter DAB RMS Power	200 W		200 W		16 W
Horizontal Beamwidth	40°	40°	40°	40°	14°
Azimuth	35°	135°	150°	5°	45°
Transmission Line Loss	1 dB		1.8 dB		1 dB
Power at Splitter Input	159 W		132 W		---
Power Splitter Loss	0.5 dB	0.5 dB	0.5 dB	0.5 dB	---
Splitter Ratio	1/2	1/2	2/3	1/3	---
Power at Antenna Input	71W	71 W	79 W	39W	12.7 W
Antenna Gain	20.4 dBd	20.4 dBd	20.4 dBd	17.4 dBd	18.9 dBd
Maximum ERP	7.8kW	7.8 kW	8.6 kW	2.15kW	1.0 kW

Table 2

**AT&T - LUCENT VHF DAR (IBAC) FIELD TEST
BROADCAST STATION KE1A - 96.9 MHz, CHANNEL 245
SAUSALITO, CALIFORNIA**

**ENGINEERING SPECIFICATIONS AS INSTALLED
(TAKEN FROM FCC FORM 302 LICENSE APPLICATION)**

A. Transmitter Site and Control Point

Geographic Coordinates (NAD27)	37° 51' 04" N 122° 29' 50" W
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Mt. Beacon, Wolfback Ridge Road, Sausalito, Marin County, California

B. Equipment

Transmitter	CCA FM35,000DAB(custom)	
Transmission line	Andrew, Types LDF5-50A and HJ5-50A	44 m
Tower	Self-supporting (existing)	40 m
Antenna	Shively, Type 6813-3-SS, 3-bay, nondirectional half wave-spaced	

C. Height

Height of site above mean sea level	338 m
Height of tower above site	40 m
Overall height above mean sea level	378 m
Elevation of average terrain above mean sea level	51 m
Effective height of antenna above site	14 m
Effective height of antenna above mean sea level	352 m
Effective height of antenna above average terrain	301 m

D. Operation

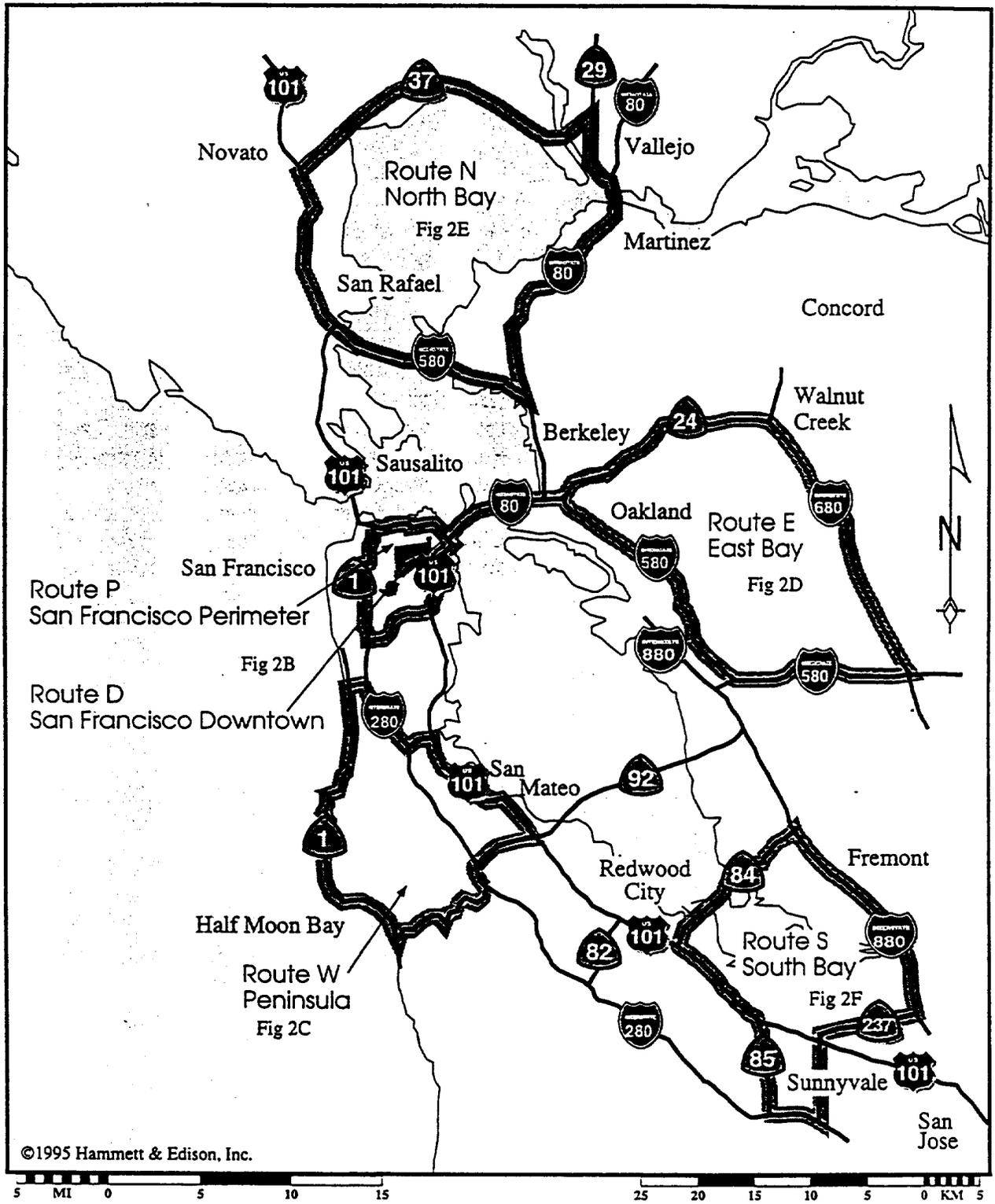
Channel	245
Frequency	96.9 MHz
Transmitter power output (FCC rounding)	6.0 kW
Filter network efficiency	93.3%
Transmission line efficiency	88.5%
Input power to antenna	4.95 kW
Antenna power gain (circularly polarized)	1.01
Effective radiated power (circularly polarized)	5.0 kW

Table 3
LINK BUDGET FOR LINE-OF-SIGHT DIGITAL AUDIO
BROADCASTING RECEPTION AT S-BAND (2.05 GHZ)

AUDIO BIT RATE (Stereo)		160.00	kbps
Satellite transmitter power		7.00	watts
Satellite transmitter power		8.45	dBW
Frequency		2.05	Ghz
Satellite antenna diameter		5.00	m
Satellite antenna gain		38.02	dBi
Satellite antenna beamwidth		2.05	deg
EIRP		46.47	dBW
Satellite Elevation Angle		25.00	deg
Slant Range		39262	km
Free space loss		-190.51	dB
Atmospheric losses		0.25	dB
(Total PFD in 200 kHz BW)		-116.40	dBW/m2
PFD in 4 kHz		-133.39	dBW/m2
Signal at Antenna		-144.29	dBW
Receive Antenna gain		8.00	dBi
Receive Antenna Point Loss		1.00	dB
Received Signal		-137.29	dBW
Antenna Temperature		150	K
Receiver Noise Figure		1.50	dB
Receive System noise temperature		274	K
Receive System G/T (On Antenna Axis)		-16.37	dB/K

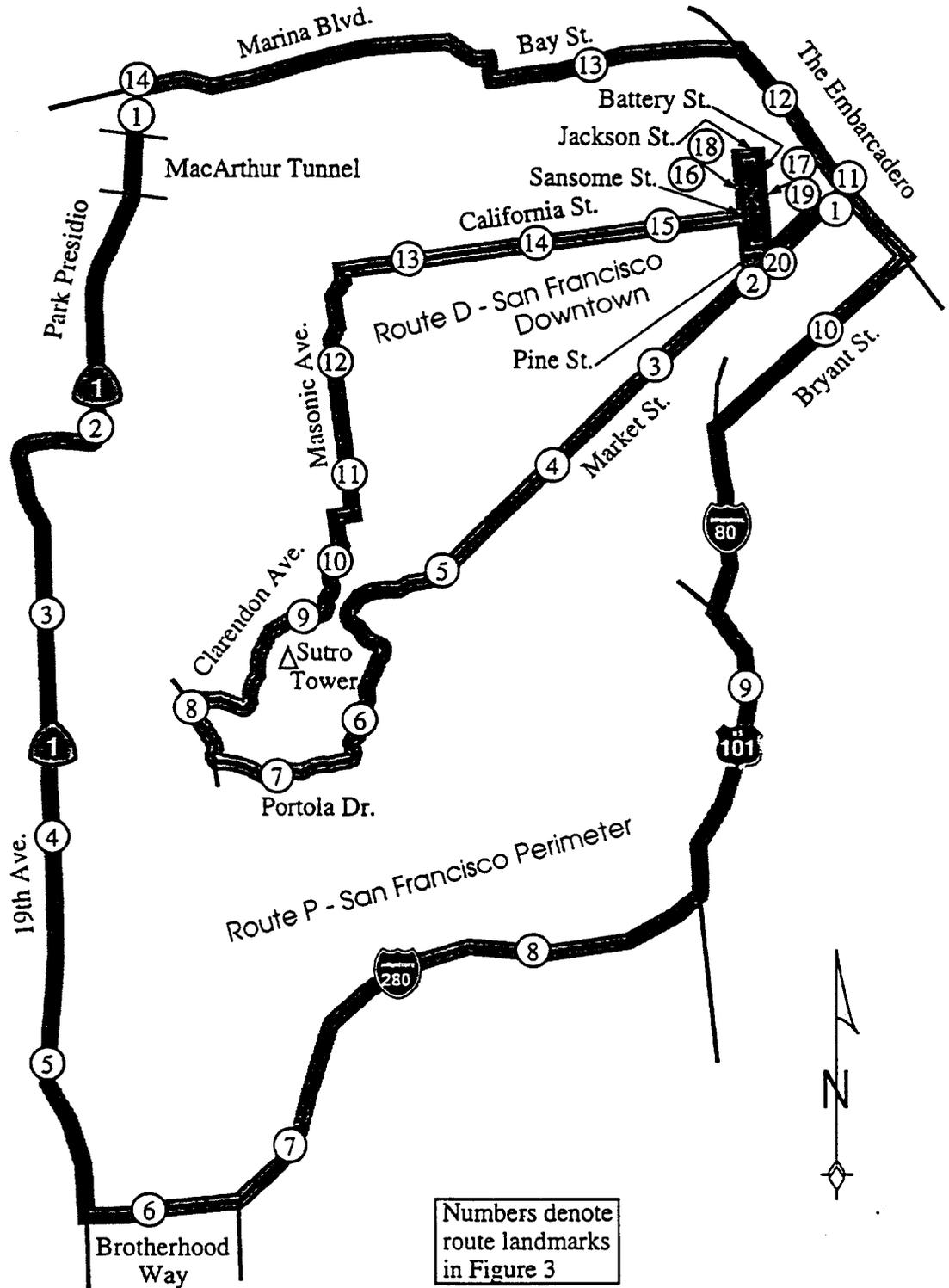
C/No		66.94	dBHz
Bit Rate		52.04	dB
Eb/No Available		14.89	dB
Theoretical Eb/No, B.E.R. = 10E-6		3.50	dB
Receiver implementation loss		1.00	dB
Interference degradation		0.50	dB
Receiver Eb/No Requirement		5.00	dB
LINK MARGIN, Beam Center		9.89	dB
LINK MARGIN, Beam Edge		6.89	dB

Electronic Industries Association
 NRSC DAB Subcommittee • Field Test Task Group
 "Long Path" Test Routes



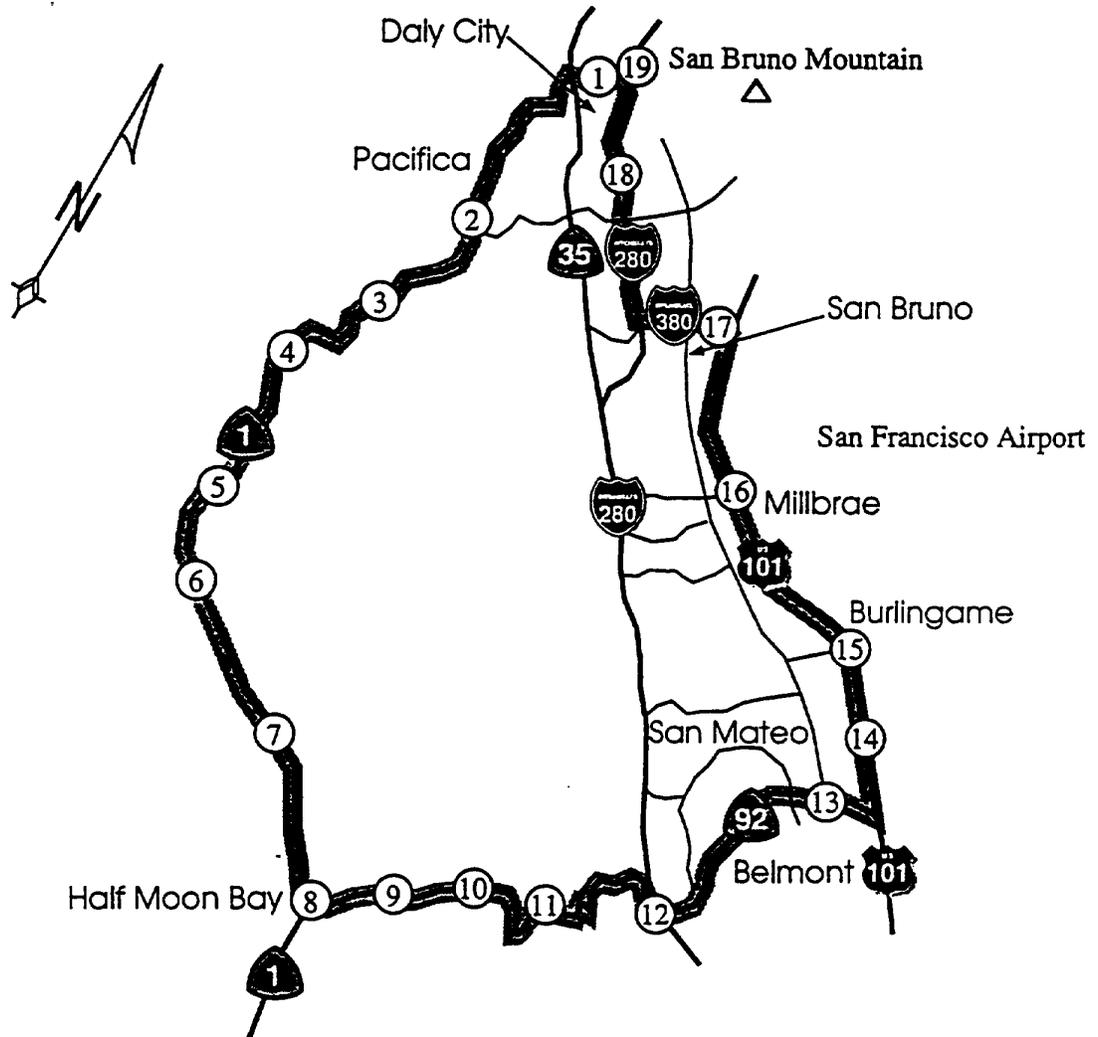
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"Long Path" Test Routes
Routes D & P • San Francisco



“Long Path” Test Routes

Route W • San Francisco Peninsula



Numbers denote
route landmarks
in Figure 3