NATIONAL RADIO SYSTEMS COMMITTEE

NRSC-R58 Digital Audio Radio IBOC Laboratory Tests Transmission Quality Failure Characterization and Analog Compatibility August 11, 1995

Part II – Appendices A through E



REPORT

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NRSC-R58

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NRSC-R58

FOREWORD

NRSC-R58, *Digital Audio Radio IBOC Laboratory Tests – Transmission Quality Failure Characterization and Analog Compatibility*, documents the first comprehensive testing of in-band/on-channel digital radio systems. This report was prepared for Working Group B and the Combined EIA DAR and NRSC DAB Subcommittees.

The NRSC is jointly sponsored by the Consumer Electronics Association and the National Association of Broadcasters. It serves as an industry-wide standards-setting body for technical aspects of terrestrial overthe-air radio broadcasting systems in the United States.

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APPENDIX A

System Descriptions

AT&T

In-Band, Adjacent Channel System In-Band, Reserved Channel System

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THE AT&T DAR SYSTEM UPDATE

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Abstract

This paper describes a system developed at AT&T Bell Laboratories for broadcasting digital 20 kHz stereo over a 200 kHz channel in the terrestrial FM radio band. This *In-Band Adjacent Channel* system for digital audio broadcasting, or digital audio radio (DAR) uses the Perceptual Audio Coder (PAC) for compressing CD-stereo to bit rates on the order of 128 to 160 kbps, a threelevel strategy for transmission error protection, a 4-phase modem, and adaptive algorithms for synchronization and channel equalization. This paper defines primary as well as secondary modes of the AT&T-DAR system.

1. Introduction

We describe a system for digital audio broadcasting developed at AT&T Bell Laboratories. The system is initially designed to operate in the so-called In-band Adjacent Channel (IBAC) or In-Band Reserved Channel (IBRC) modes in the 88-108 MHz terrestrial FM radio band. The RF bandwidth needed by the IBAC system is 200 kHz. Digital audio coding is provided by the AT&T Perceptual Audio Coder (PAC) which provides CD-quality stereo at bit rates on the order of 128 to 160 kilobits per second (kbps). A robust 4phase modem, an adaptive channel equalizer and a three-layer method of error protection ensure that the audio quality is maintained in the presence of transmission imperfections such as multipath fading and Doppler frequency shifts at high vehicle speeds. The IBAC or IBRC paradigms are designed for interference and coverage characteristics that are fundamentally better than those for

the alternative of In-Band On-Channel (IBOC) transmission. The AT&T – IBAC solution is readily adaptable to satellite transmission and to AM-band solutions for DAR.

2. In-band DAR Systems

Figure 1 depicts various possible paradigms for simulating analog and digital versions of the same audio program in the FM terrestrial band. Without loss of generality, the same paradigms also apply to digital broadcasting of a separate audio program. The analog FM transmissions are in 200 kHz slots in the 88-108 MHz band. The in-band DAR solutions use vacant slots in one of three ways: (a) two 100 kHz slots on either side of the analog channel (Row 1), or (b) a single 200 kHz slot (Rows 2. 3, and 4), or (c) a single 100 kHz slot (Row 5). The arrangements in the last four rows of the figure are termed Inband Adjacent-Channel (IBAC) solutions. The simulcasting (in general, the broadcasting) efficiency of the 100 kHz IBAC solution is clearly superior to that of the 200 kHz solution. For example, simulcasting is provided for both Stations 2 and 3 in Row 5 of the figure, but only for Station 2 in Row 3, and only for Station 3 in Row 4. However, in the current state of audio compression and radio modem technology, it is not possible to provide CD-quality stereo combined with transmission-robustness in a 100 kHz solution. On the other hand, the better frequency granularity of the 100 kHz system makes a milder assumption about available RF space, and enables some other paradigms, such as a single side-lobe version of the IBOC system in Row 1. This is

indeed a secondary mode of the AT&T-A:nati IBOC system [1].

The remainder of the paper focuses on a 200 kii. IBAC solution for DAR, as in Rows 2, 3 and 4 of Figure 1. In the expected transitional period between analog FM and DAR, the IBAC solution will depend on the availability of vacant 200 kHz slots in the RF spectrum at a given geographical location. If such a vacancy exists, the IBAC solution provides interference and coverage properties that are inherently superior to those of the In Band On Channel (IBOC) alternative (which is constrained to operate at a transmit power level significantly below that of the on-channel FM broadcast). The IBAC solution thus provides a realization of the best possible performance of a DAR system, provided sourceand channel-coding algorithms are optimized and well-matched to each other. The IBAC solution is also readily extensible to DAR solutions in the terrestrial AM-Radio and satellite bands.

The remainder of this paper is organized as follows. Section 3 describes the PAC audio codec as used in the DAR system, Section 4 describes the three-level strategies for transmission error protection, and Section 5 summarizes the modem and RF transceiver, including techniques for synchronization and adaptive channel equalization. Section 6 defines the secondary modes of the AT&T-DAR system. Section 7 talks about possibilities for future enhancements.

3. The PAC Audio Codec

The concept of perceptual audio coding is derived from the notion of *distortion-masking* in the human auditory system, the phenomenon whereby one signal can completely mask a sufficiently weaker signal in its frequency- or timevicinity. In the context of low bit rate coding, the desired audio is the masking signal and the coding distortion is the weaker, masked signal. The ratio of masking-to-masked signal can be quite modest (and in general, substantially smaller than the 96 dB SNR used in 16-bit PCM coding as in the CD or DAT) [2].

Masking in the frequency-domain (also called

simu!'aneous masking) is illustrated in Figure 2, where the staircase function specifies the justnoticeable-distortion (JND) or masking threshold as a function of frequency for the given input power spectrum (a trumpet signal). The twentyfive treads of the JND function are the so-called critical bands in human hearing. The JND is derived from an empirical interpolation between results for tone-masking-noise and noise-maskingtone, and by the incorporation of a model for the spread of masking into adjacent critical bands. The JND is re-evaluated in real time for every new sample of incoming audio spectrum, and thus provides a continuously changing masking threshold. The use of the JND permits optimum coding in that incoming segments are neither overcoded nor undercoded. The JND paradigm provides in principle a framework for variablerate, constant-quality (transparent) coding using a block-transform algorithm, as suggested in Figure 2. The PAC coder is adapted to a constant bit rate, almost-constant-quality mode by means of real-time iterative techniques for bit allocation and noiseless coding [3]. This monophonic threshold model is an advanced version of previously described models [2] [4].

Masking in the time-domain (also called nonsimultaneous masking) is illustrated in Figure 3, where a strong signal in a given time block can mask an adequately lower distortion in a previous or future block (backward or forward masking). Although less understood than simultaneous masking, temporal masking is implicitly used in block-transform coders such as PAC. In these coders, a long block length (for example, 512 samples) is used whenever possible to provide fine frequency analysis, and consequently better JND thresholding and coding gain. However, during occurrences of sharp transients in the audio, a long block length (with a possible audio onset late in the block) can cause a pre-echo (distortion lasting for a large or full fraction of the block length) that precedes the audio itself. In such cases, a shorter block length maximizes the possibility of temporal masking, in spite of the decaying skirts in Figure 3. An important feature of PAC is a window-switching algorithm (Figure 4) that can change the window length from 512 samples to.

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say. 128 samples, with the switching based on a proprietary perceptual algorithm.

The joint-stereo coding system in PAC is based on a proprietary algorithm that uses either Left-Right coding (Coding of L, R signals) or Sum-Difference coding (coding of L + R and L - Rsignals). in an adaptive fashion in time and frequency. Judicious switching between these modes minimizes the occurrence of a phenomenon called noise-unmasking in the context of stereo listening.

In internal tests at AT&T Bell Laboratories, PAC has demonstrated CD-quality compression of a very broad class of inputs at stereo coding rates in the range of 128 to 160 kbps.

The performance of PAC at bit rates in the range of 96 to 160 kbps (for stereo) makes it a powerful candidate for DAR applications in the AM. FM and satellite frequency bands. In the primary mode of the AT&T DAR system, the bit rate for audio coding is 152 kbps.

The PAC decoder has been implemented on a single general purpose DSP processor. The PAC encoder is implemented on a parallel processing platform, also based on general purpose processors. With a sufficiently powerful processor such as the Intel i860, the number of processors for PAC encoding can be as few as two.

4. Strategies for Robust Transmission

The goal of a DAR system is to maintain an excellent level of audio quality over a wide range of transmission environments that include signal fading due to multipath, vehicle speed effects (such as Doppler at high speeds, deep fades and nulls at slow speeds and stoplight stalls), and fringe-area reception. Briefly, the idea is to maintain CD quality over a significant majority of transmission conditions, and graceful degradation until the point of total failure.

The AT&T DAR systems provides adaptive equalization to alleviate frequency-selective channel fading (Section 5), time-interleaving to randomize bursty errors due to slow fading, and three levels of protection against residual bit error effects: (i) an outer code: protection of a small number of very critical bits in the transmitted bit stream (using an 8 kbps overhead out of a total of 160 kbps).

(ii) an *inner* code: protection of the audio-plusauxiliary data bitstream by a shortened Reed-Solomon code of rate in the neighborhood of onehalf, and

(iii) error-concealment: a proprietary algorithm for the concealment of residual errors at the PAC decoder input, in response to a block-error flag provided by the Reed-Solomon decoder when it fails to decode a codeword. The concealment algorithm uses audio signal redundancies, including left-right correlations in stereo, to provide a reconstruction quality that is significantly better than that due to muting. Muting is resorted to only when there is a burst of several consecutive block-error flags.

A soft-decision capability in the Reed-Solomon (R-S) coder can minimize the flag probability, and a simple arrangement ensures that flags occur more frequently than needed, rather than less frequently than needed.

Table 1 shows the mean time between flags (secs) as a function of channel signal-to-noise ratio (dB) and interleaving depth (ms). Results are for a flat Rayleigh-fading channel, and for three (R-S) codes. The results assume differential encoding of data at the encoder and differential demodulation at the receiver. Modulation is by four-level phase shift keying. It is seen that for a given frequency of error concealment, the channel quality and the interleaving time can be traded off against each other. However, the interleaving depth controls the latency in digital channel-switching, and for that reason, the interleaver cannot be arbitrarily long.

In Table 1 the (R-S) codeword is 32 symbols or 256 bits long. The (32, k) code can correct up to all 0.5 (32 - k) errors (and many of higher weight), and has information and redundancy symbol numbers of k and (32 - k). The audio block length is 1700 bits. The vehicle speed is 20 mph. The channel signal-to-noise ratio is CSNR.

For CSNR = 17 dB and a rate-1/2 (or 32, 16) code, the block error rate is about 2×10^{-3} , and

CSNR (dB)	Delay (ms)	Mean Time between Failures (sec)				Mean Time	
		(32, 20)	/0)	(32, 16)	(32, 14)	(32, 12)	
17	425	-	.02	.07	0.21	0.75	
17 -	725	.03	.08	.30	1.2	4.70	
20	425	-	5		_	-	
20	725	45	150	-		-	

Table 1: AT&T DAR System: Channel Coding

the bit error rate is on the order of $5 * 10^{-4}$ (for a 1700-bit block). The bit error rate without the rate-1/2 code would be on the order of $2 * 10^{-2}$ (Figure 6). For CSNR values of 18 and 19 dB, the bit error rate with a rate-1/2 code is about $4 * 10^{-5}$ and $9 * 10^{-7}$ respectively, and decreases very rapidly for higher values of CSNR.

5. DAR Modem and Transceiver

Modulation is based on four-phase signalling with coherent detection. The choice of 4ϕ -PSK provides a good compromise between the robustness of 2ϕ -PSK and the efficiency (in bits per second per Hertz) of 8ϕ -PSK (Figure 7). The use of differential PSK is a good match to the environment of a mobile receiver, an important and challenging segment of the DAR market.

The input to the 4- ϕ modulator is a 360 kbps bit stream (composed of 340 kbps of multiplexed audio and data and an overhead of 20 kbps for synchronization and channel equalization). The 4- ϕ modulator provides an ideal clficiency of 2 bits/sec/Hz, and an actual rate of 1.8 bits/sec/Hz in packing the 360 kbps data into \approx 200 kHz FM channel. This is shown in Figure 5 which also depicts the staircase FCC mask for FM radio operation. The RF spectrum of the DAR system includes a pilot tone (at the + 100 Hz part of the baseband spectrum) to aid in efficient carrier recovery.

The channel equalizer is a separable, non-crosscoupled passband equalizer using a fractionalspacing algorithm of resolution T/3 where T is the symbol interval (1/360 kbps = 3μ s). Equalization is periodically adaptive, and occurs once for every block of 1700 bits. The training sequence is a pseudo-noise sequence of length 100 bits, centered on the 1700-bit block. The overhead is 100/1800 (20 kbps/360 kbps). The PN sequence is used for estimating the channel impulse response, and also for bit synchronization.

The basic transceiver functions of the DAR system are summarized in Figure 8.

6. The DAR System: Primary and Secondary Modes

The overall block diagram of the AT&T-DAR system is shown in Figure 9.

The DAR system has provisions for two sources of auxiliary data transmission: asynchronous data made possible by the inherent variability in the (nominally 152 kbps) PAC encoder, and synchronous data that gets multiplexed with the PAC output prior to inner error protection coding. The effective data throughput is expected to be in the range of 10 to 20 kbps. The bit error rate and block-failure rate of the data are identical to those for audio in the arrangement shown in the diagram. Not detailed are essential details of circuitry such as the derivation of various subsystem clocks.

The block diagram permits several models. The primary mode uses a PAC bit rate of 160 kbps (including outer error protection), a rate 1/2 inner code (170 kbps/340 kbps), and an interleaver delay of 640 ms. Secondary modes result due to other combinations of PAC rate (such as 128 or 192 kbps), corresponding Reed-Solomon rates (for a total of 340 kbps), and other interleaver designs (such as the shorter latency included in Table 1).

The use of antenna diversity is an option. This option can complement the time-diversity provided by the combination of the channel coder and interleaver. A large degree of frequency diversity is not available in In-Band Narrow-Band systems, as a class. However, frequency diversity, if available, is implicitly exploited by the equalizer to improve receiver performance.

7. Future Enhancements of the DAR System

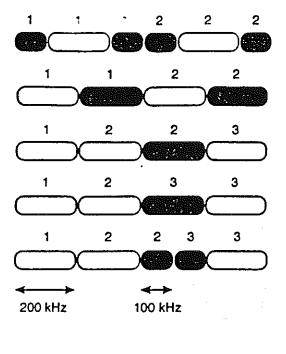
The system of Figure 9 is an initial proposal for an IBAC-DAR system. Anticipated future enhancements will be a result of more sophisticated implementations in the near term as well as improvements in audio. modem and radio technologies in the longer term. The following is a short list of these enhancements:

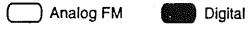
- increased capacity for auxiliary data
- the provision of more than two audio channels
- better granularity in terms of demands on available RF space
- extended service range because of the elastic nature of the PAC algorithm.

References

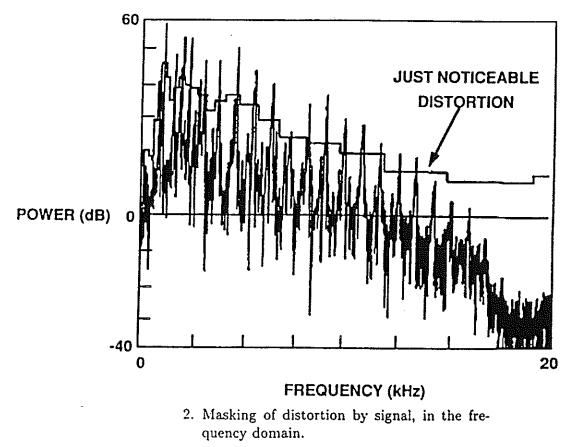
- J. M. Cioffi, "Amati DAR Transmission", *Radio '92* (National Association of Broadcasters), New Orleans, LA, September 12, 1992.
- [2] J. D. Johnston, "Transform Coding of Audio Signals Using Perceptual Noise Criteria". *IEEE J. Sel. Areas in Commun.*, pp. 314-323. Feb. 1988.
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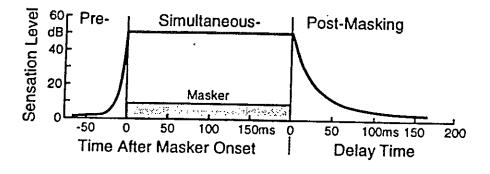




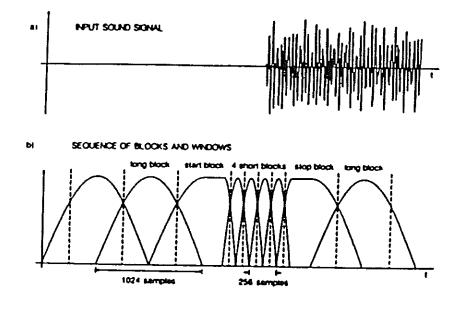
1. Alternative paradigms for analog and digital audio broadcasting.



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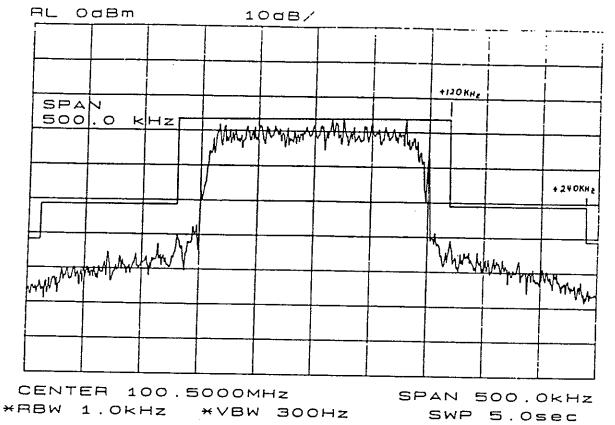


3. Masking of distortion by signal, in the timedomain.

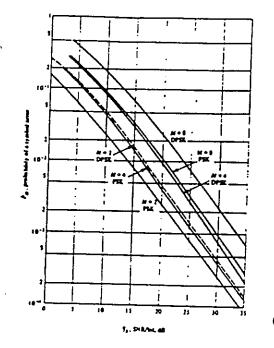


4. Adaptation of window length in transform coding.

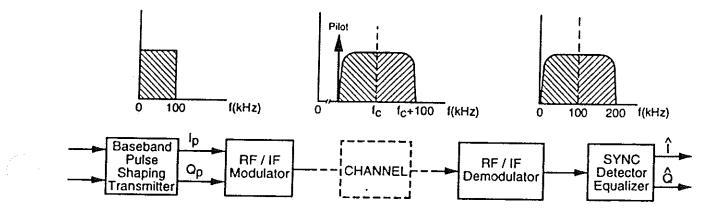
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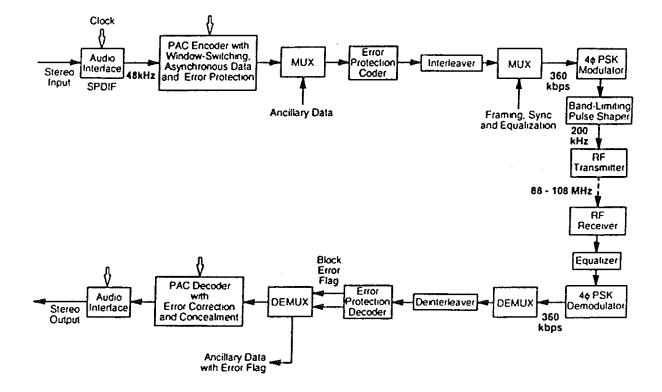
5. Transmitted power spectrum in the AT&T-DAR system, and the FCC-mask for broadcasting in the FM radio band.



6. Error rate versus channel quality for 2-, 4and 8-level (differential) phase modulation.



7. Schematic block diagram of the AT&T-DAR Transceiver.



8. Block-diagram of the AT&T-DAR system.

AT&T/Amati

In-Band, On-Channel System

AT&T/AMATI DAR SYSTEM: AN UPDATE

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ABSTRACT

US broadcasters have expressed a strong preference for an In-Band On-Channel (IBOC) digital audio radio system: that is, one that can be superimposed on the existing VHF FM system. Two groups of US companies are developing systems to meet these very difficult requirements, and the EIA will begin testing the candidates in January 1994. The results will be reported to the NAB to help them in their decisions on DAR deployment in the US.

This paper describes the RF environment in the USA and the resultant problems for IBOC DAR. It then describes the solution proposed by a partnership of AT&T (audio codec) and Amati (transceiver)

INTRODUCTION

Digital Audio Radio, a general term for the broadcasting of high (i.e., CD) -quality audio in compressed digital form, is a subject of considerable interest throughout the world. Ideally this quality should be achieved with the small antennas that are suitable for stylish installation on automobiles, and it should be maintained in all commonlyencountered multipath fading environments

This paper, which is a follow-up to presentations to the NAB in April 1993 [1] and the SBE in October 1993 [2], describes a system that has been developed by AT&T and Amati in response to the needs of US broadcasters. The system and four others (including one other IBOC system) will be tested by the EIA beginning in January 1994. The NAB is liaising with the EIA through the National Radio Systems Committee in order to monitor the tests of the IBOC systems only.

It has previously been assumed that the US needs are unique, and that therefore the solution for them must also be, but it begins to appear that they are not so unique. It is widely believed that the ideal frequency band for <u>any</u> DAR system is Band II (the present FM band), so we have suggested to the international community [3] that it should explore ways in which the two extremes of a wideband, single-frequency system (i.e., Eureka 147) and a narrowband, single-transmitter system (e.g., the ATT/Amati system) might attain at least a minimum level of manufacturing compatibility.

Section 1 of this paper describes the In-Band¹ environment, and the special problems of digital broadcasting in it. Section 2 very briefly recapitulates some of the reasons for the choice of multicarrier as the modulation method, and describes the overall system in some detail. Section 3 describes the use of an auxiliary overhead channel (AOC) to control the options in the present system, and discusses the possible generalization of the method; Section 4 gives some preliminary results of laboratory and field tests.

IN-BAND DAR

The Environment

FM carrier frequencies in the US are separated by 200 kHz, but it is rare for two stations in a geographical area to be separated by less than 400 kHz. The long-term, average Power Spectral Density (PSD) of a transmitted signal is limited by a mask defined by the FCC, which is shown in Figure 1(a). Also shown are an approximation to an actual PSD, and the permitted levels of a first-adjacent signal (carrier removed by -200 kHz) and a second-adjacent signal (carrier removed by +400 kHz) at the edge of a station's "Normally Protected Contour".

The IBOC Problem

The four requirements for any IB system that dictate the design approach are:

1. Under conditions of multipath fading the DAR receiver must operate much better than a good FM receiver receiving a conventional FM signal. This is the basic reason for interest in DAR.

2. The DAR signal should not interfere in any way with its host FM, and, similarly, should not be affected by it.

3. Both DAR and FM receivers must operate under the conditions of (a) the more common second-adjacent-

² "In-Band" may be confusing because the obvious question is "In what band?" The meaning here is that in the USA: the FM band of 88 to 110 MHz.

that is interence, and (b) the less common first-adjacent channel inteference.

4. The composite DAR/FM signal in any IBOC system should conform to the FCC PSD masks (25 dB below carrier at 120 to 240 kHz from carrier and 35 dB beyond 240 kHz).

In order to define the problem in more detail, we must decide on the method of separating the DAR signal from its host FM. The method that is simplest in implementation, and most assured of agreement between theory, simulation and practice is to assign the FM and DAR signals to different frequency bands, and separate them by filters.

The spectrum outside $f_c \pm 100$ kHz is not needed for highquality FM reception, so it is possible to place a DAR signal--at a PSD that conforms to the FCC mask--in one or both of the sidelobes, as shown in Figure 1(b). Such placement of the DAR signal is called On-Channel (IBOC), and a broadcaster would not need a new license--a very important consideration. If the first- or second-adjacent channels are vacant, then it is technically feasible to place much higher powered DAR signals there; whether such transmission could be licensed, and what PSDs would be permitted are much more difficult questions.

The most common "bifurcated" arrangement of pass-bands and PSDs that satisfy requirements 3(a) and 4 are shown in Figure 1(a). It can seen that the maximum total bandwidth available to the DAR signal in this double-sidelobe mode is $2(f_1 - f_2)$, which is about 140 kHz. If there is a potentially interfering first-adjacent channel (requirement 3(b)), as shown in Figure 1(b), the bandwidth of the single usable sidelobe must be increased somewhat; the maximum available is about 80 kHz.

These calculations are based on the assumption of an FM signal that includes only audio. A 67 kHz SCA does not significantly increase the bandwidth of the FM, but the more recently-installed 92 kHz SCA does. Interference of the DAR signal with and from such a composite FM signal could, of course, be reduced by moving the digital sidelobes away from the center, but this increases the interference with and from adjacent-channel DAR signals. Whether there is a compromise placement of the sidelobes that will simultaneously satisfy all requirements, or whether the placement will have to be adjusted to balance the broadcasters' need for these wideband SCAs against the presence of adjacent channels will need careful study. Eventually the preferred arrangement would be for the data capability of the SCAs to be carried, much more efficiently, by the auxiliary data channel.

The audio compression and encoding problem, therefore, is to achieve CD-quality with a data rate that can be reliably transmitted in 140 kHz, and near-CD-quality in 80 kHz; the proposed solution to this problem using a Perceptual Audio Coder (PAC) is described in a companion paper [3]. The data rates chosen were 160 and 128 kbit/s. The transceiver problem is to transmit and receive those data rates in the appropriate bands under conditions of adjacent-channel interference and multipath fading.

MULTICARRIER MODULATION

The Choice of Multicarrier Modulation

Multipath propagation of a radio signal has two possible effects. If the product of the delay spread and the bandwidth is greater than about 0.5 the attenuation and delay responses of the channel will be strongly frequencydependent, and inter-symbol interference (ISI) may result. On the other hand, if the product is less than about 0.25 the responses will be fairly constant across the whole band, and the signals from the separate paths will either reinforce or cause wideband fades.

The problem of ISI caused by large delay spreads could perhaps be solved with single-carrier modulation by equalization of the received signal, but the computation required to make the equalization adapt fast enough to track a moving receiver² is a very challenging one. Another solution is to use multicarrier modulation [4] with a guard period (cyclic prefix) whose length is greater than the largest delay spread. This is the solution adopted in the Eureka 147 system where multicarrier modulation is called Coded Orthogonal FDM (COFDM). Because of the large delay spreads encountered in receiving from the many transmitters in an SFN the guard period in the Eureka system is much longer than is needed in an IBOC system.

The opposite problem of small delay spreads, which cause "wideband" fades, is harder to solve. The best technical solution--space diversity through the use of two antennas-has been judged impractical by automobile manufacturers. The next best solution, which is implemented in the Eureka system, is to achieve frequency diversity by a combination of frequency interleaving, trellis coding and Forward Error Correction (FEC). In the much narrower band available to IBOC systems (less than 400 kHz, compared to at least 1.5 MHz for Eureka 147), however, not much frequency diversity can be achieved, and the only solution to the notorious "deep stop-light fade" problem is to wait for a green light!

The choice of multicarrier over single-carrier modulation for the DAR system was not, however, based on the relative performance merits of the two methods, which are complicated, controversial and beyond the scope of this paper, but in the much greater flexibility of data rates and frequency bands that multicarrier provides.

 2 At 30 mph a vehicle will travel through one wavelength of an FM carrier in approximately 200 ms.

The Amati Discrete Multitone (DMT) Solution

The Transmitter

The audio data signals of 160 $\{128\}^3$ kbit/s are augmented with a (32,20) $\{(24,16)\}$ Reed-Solomon FEC code to generate aggregate data rates of 256 $\{192\}$ kbit/s respectively. An auxiliary data channel can be provided, but the method of multiplexing it has not been decided. The output rate of a PAC encoder depends on the source material, and an "opportunistic" data channel (i.e., available only when the material is predictable and therefore easily encoded) with an <u>average</u> data rate of about 15 kbit/s can be made available with no degradation of audio quality or increase of transmitted data rate or bandwidth. Such an auxiliary channel would have to be buffered and flowcontroled; whether such an arrangement would be acceptable remains to be seen.

DMT is Amati's implementation of generic multicarrier modulation. It uses a sub-carrier spacing of approximately 4 kHZ, and the transmitter can be configured to use any combination of sub-carriers needed for the three IBOC and two IBAC modes. The symbol duration is 250 μ s, and the cyclic prefix is 14.5 μ s: more than enough to cover all delay spreads encountered in a single-transmitter system. 32 {18} sub-carriers are used in the double {single} -sidelobe mode using a mixture of differential 4-phase and 8-phase.

The spectrum of a conventional multicarrier signal (DMT or COFDM) that uses only a sub-set (or sets) of the available sub-carriers falls off fairly slowly at the edges of the nominal band(s). In order to prevent such a signal from interfering with its host FM, it would have to be strongly filtered. A better method of bandlimiting the signal is to shape the envelope of the cyclic prefix; with a raisedcosine shaping an extra 25 dB of suppression of the DAR signal can be achieved across most of the FM band, and a transmit filter is not needed.

The design of trellis codes for a fading environment, the best relationship between them and FEC codes, the best method of decoding, and the individual and aggregate gains that can be obtained from the two codes are subjects that are not adequately understood; small improvements are still being made

One sub-carrier in each sidelobe is not used for data; it serves as a pilot, which can be used in the receiver to help in synchronization, and to transmit the slow Auxiliary Data Channel (see Section 3).

The Receiver

The present system uses differential demodulation, but

the ratio of the symbol rate (4 kHz) to the rate of change of a channel (< 10 Hz) is probably such that coherent demodulation--requiring continual learning of the subcarrier phases--would be feasible (though computationally intensive). The improvement in performance when the demodulation is embedded in an interleaved system with FEC is very difficult to predict; it will certainly be less than the text-book figure of 2.3 dB.

With hard decoding the FEC decoder can correct only 6 $\{4\}$ byte errors in each block. In the Amati system, however, the outputs of the receive FFT (the demodulated sub-carriers) are processed to yield a measure of the confidence level of each signal, and if the aggregate confidence level of an FEC block is below a threshold, then the block is tagged for erasure. Use of the erasure signal enables the decoder to correct almost twice as many (i.e., 12 (8)) byte errors in each block. The efficiency of this method depends on the criterion for erasing, which is continually being refined.

Similarly, the quality of the decoded audio signal can be improved if the modem receiver outputs a flag when the FEC decoder is unable to correct all the errors in a block. Then the PAC decoder implements a concealment algorithm, which is described in more detail in [5].

THE AUXILIARY OVERHEAD CHANNEL (AOC)

With the present system a broadcaster may choose one of only four options (one double-sidelobe mode, two singlesidelobe modes, and, perhaps eventually, one pure IB DAR mode) depending on the potential interference from other adjacent-channel stations. Therefore the AOC need transmit only two bits, and it can do this at a 1 kbit/s rate by lightly modulating the two pilot sub-carriers without significantly reducing their synchronization capability. With this rate for the AOC a receiver could configure itself to match the broadcast signal within a few milliseconds.

As proposed in [1], however, the AOC could carry much more information, and this could be used to indicate any one of a large set of in-band multicarrier configurations. This set might include individually-tunable single-channel signals and wide-band multi-channel signals--both types with or without associated FM.

The present system was developed to be used with a single transmitter; the symbol and guard periods are not long enough to deal with the larger delay spreads encountered in a multi-transmitter system⁵. Amati is studying the

⁵ These are also called "single-frequency" systems, but their important characteristic is the simultaneous transmission from multiple transmitters, not the number of "channels" contained within the signal,

⁴ The numbers in braces are those for the single-sidelobe mode.

transmission problems of multicarrier systems with very long symbol periods (very small sub-carrier frequency separations and very large FFT sizes.)

PRELIMINARY RESULTS

As previously demonstrated by AT&T, a back-to-back PAC encoder and decoder reproduce CD-quality audio with no impairment for most tested signals at stereo rates in therange of 128 to 160 kbit/s. Furthermore, a connection of encoder, transmitter, receiver, and decoder with no channel impairments produces near-perfect FM and DAR signals; it is clear that neither is being interfered with by the other.

The composite DAR/FM signal has also been transmitted through simulated RF multipaths with delay differences up to 10 μ s. With the larger spreads the FM was severely (i.e., annoyingly) distorted, but the DAR was completely unperturbed. The R-S error corrector, which in the tested system could correct up to 6 error bytes in a block, was never exercised to its limit.

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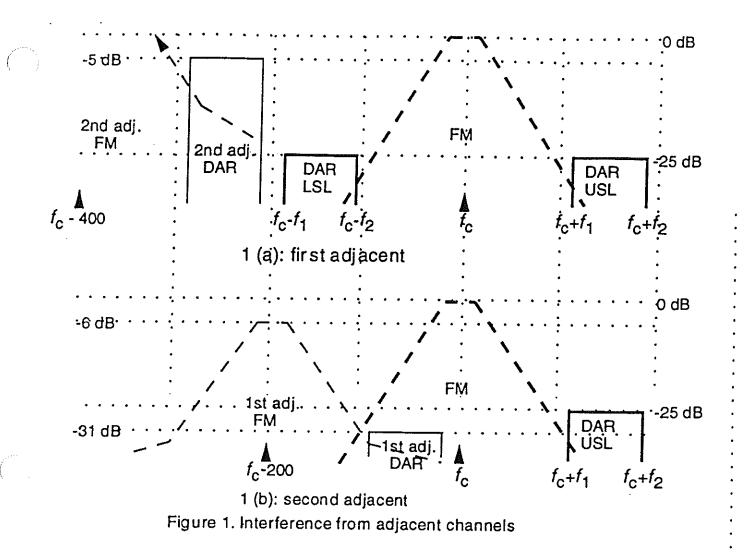
[1] J.M. Cioffi & J.A.C. Bingham, "Digital Sound Broadcast with Auxiliary Overhead Control", Proceedings 47th Annual NAB Broadcast Engineering Conference, p. 243, April 1993.

[2] J.A.C. Bingham, "AT&T/Amati's Proposal for DAR in the USA", p. 87, Proceedings SBE Engineering Conference, Oct. 1993.

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[4] J.A.C. Bingham "Multicarrier Modulation: an idea whose time has come." IEEE Communications Magazine, pp. 5-14, vol. 28, No. 5, May 1990.

[5] N.S.Jayant, "The AT&T IBAC DAR System:an Update", Proceedings 48th Annual NAB Broadcast Engineering Conference, March, 1994



AT&T-DAR Systems

Ed Chen and Nikil Jayant

NAB Conference and Exposition, April 1995

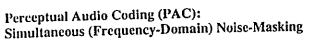
Perceptual Audio Coding (PAC)

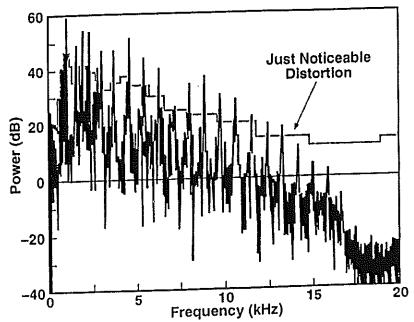
Refined psychoacoustic models for:

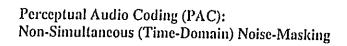
Frequency-domain and time-domain noise-masking Dynamic switching between short and long analysis windows Joint coding of multiple channels (in 2-and 5-channel audio) Transmission error concealment

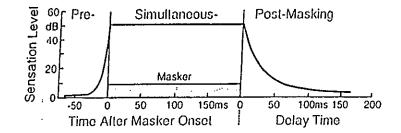
Inexpensive single-chip decoder

Leading candidate in NBC-contest for MPEG-2 standard Ideal match to the needs of DAR technology









Subjective Assessment of AT&T-Audio Technology

Swedish-Radio Tests for MPEG-AUDIO (1991) Layer 3: Best Stereo coder at 256, 192 and 128 kbps

AT&T-Internal Results (1992-93) PAC: CD-Quality Stereo at 128-192 kbps

BBC and Deutsche Telekom (1994) MPAC: Best 5-channel coder at 320 kbps

Comparison of 5-channel audio codecs at 320 kbps Number of Signals (out of 10) that are transparently coded*

		Testing Laboratory		
		Deutsche Telekom	BBC	
Philips	(MPEG-Layer 2)	1	0	
Dolby	(AC3)	2	1	
AT&T	(MPAC)	6	5	

*Average quality loss of less than 0.5 on a 5-point scale Based on the opinions of 45 expert listeners

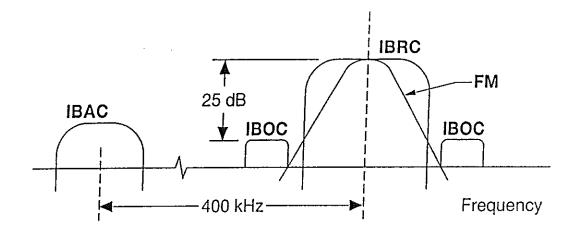
Excerpted from MPEG data NSJ 3.28.94

DAR Systems Using PAC

The perceptual audio coder (PAC) developed at AT&T Bell Laboratories is being used in conjunction with various transmission technologies for Digital Audio Radio. The following DAR proposals are based on PAC operating at bit rates in the range of 128 to 160 kbps.

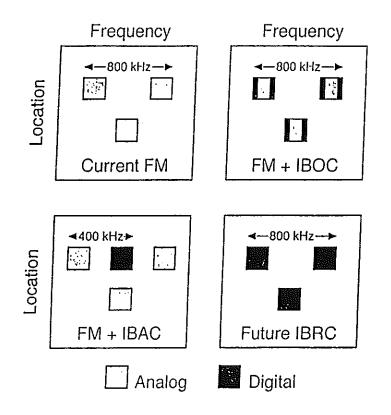
- AT&T (In-Band Adjacent-Channel System)
- AT&T-AMATI (In-Band On-Channel System)
- VOA/NASA/JPL (Satellite System)
- CD-Radio (Satellite System)

Audio Broadcasting in the FM-Band



MH4P866Y03

Audio Broadcasting in the FM-Band



MH4P866Y02

The AT&T-DAR Systems: Primary Characteristics

Based on PAC, the quality leader in low bit rate audio

Leading-edge technology for robust transmission

Three levels of transmission-error protection Proprietary technologies for equalization and multicarrier modulation

Synchronous and asynchronous channels for ancillary data

The AT&T-DAR Systems: Potential for Enhancement

Increased capacity for audio channels and ancillary data Finer granularity in terms of demands on continguous RF space Extended coverage due to elastic response of audio coder Merging of IBOC, IBAC and IBRC technologies

AT&T-IBAC and IBRC Systems: General Description

Based on the Perceptual Audio Coder (PAC) CD-quality stereo at rate of 160 kbps In-band off-channel operation in the 88-108 MHz band Digital transmission rate of 360 kbps on a 200 KHz channel Three layers of error protection Advanced algorithms for channel equalization Synchronous and asynchronous data channels (up to 15 kbps) Latency of 320-640 msec for full audio quality after a station switch

AT&T-Amati IBOC System: General Description

Based on the perceptual audio coder (PAC) CD-quality stereo at rates of 128 to 160 kbps In-band on-channel operation in the 88-108 MHz band Primary mode utilizing two RF sidelobes Secondary mode utilizing one RF sidelobe Auxiliary Overhead Control (AOC) of transmission mode Total transmission rates of 216 and 264 kbps in the two modes DSP algorithms for multicarrier modulation

Field Testing of AT&T-DAR and AT&T-Audio Technology

WPRB-FM, Princeton 103.3 (Jan-Feb 1994) IBAC and IBOC Systems, Point-to-Point and Mobile

CD-RADIO, Washington, DC (1993-94) Satellite-DAR with 128 kbps PAC

NASA-JPL (1994) Satellite-DAR with 160 kbps PAC: DAB-Symposium, Toronto; TDRS Experiment

EIA-NRSC (1995) San Francisco: IBAC, IBRC and IBOC Tests

Trenton, New Jersey (1995) IBRC and IBOC Tests at 89.1 MHz

AT&T-DAR System: Over-the-Air Field Test (Princeton, 1994)

103.3 MHz, WPRB class B station (licensed with 14 KW) Princeton, NJ

Non-co-location 1st adjacent channel broadcasting test at 103.3 MHz with 103.5 MHz being the 1st adjacent channel at New York City (46.6 miles away)

:

Several DAR transmission power levels were used (6.6 W to 6.7 KW ERP)

For stationary receiving tests, DAR transmission power levels were varied between 6.6 W and 6.7 KW

For mobile receiving tests, 666 W and 6.7 KW were the two DAR transmission power levels used

Trenton, NJ Field Test for AT&T IBRC and AT&T-Amati IBOC Systems

Over-the-air field tests for IBRC and IBOC systems Broadcast frequency of 89.1 MHz Field tests to be carried out between the months of March and of May 1995 Demonstration of stationary as well as mobile performance of DAB

Linear Power Amplifier

1

Procured from CCA Electronics, Inc. High reliability and low noise Continuous power rating of 30 KW Intermodulation Distortion: -60 dB AM Noise: -55 dB

EIA/NRSC Field Test for AT&T-Amati IBOC System

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KABL Channel 251 (98.1 MHz) at Mt. Beacon, San FranciscoSimultaneous operation of FM and DAB channelsComposite power (host FM + DAB) of 20 KW ERP (current FM level: 82 KW)

EIA/NRSC Field Test for AT&T IBAC System

AT&T IBAC will be tested as a second adjacent channel in field test Channel 245 (96.9 MHz) will be used at Mt. Beacon, San Francisco

The two co-channels are KSEG of Sacramento and KWAV of Monterey Channel 245 is a second adjacent channel to KRQR (Channel 247, 97.3 MHz) IBAC transmitter will be co-located with KRQR transmitter The proposed IBAC transmitter power is 5.0 KW ERP Co-channel interference will be minimum, as per Cleveland laboratory tests

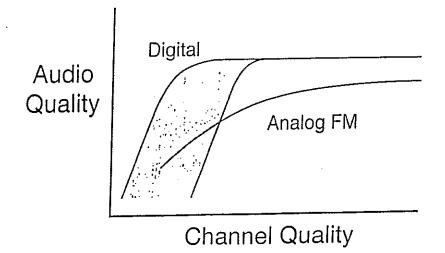
EIA/NRSC Field Test for AT&T IBRC System

AT&T IBRC will be tested as an FM replacement channel

KABL Channel 251 (98.1 MHz) at Mt. Beacon is proposed as test station During IBRC DAB field test, the FM channel will be off the airIBRC transmitter power will be 20 KW ERP (current FM level: 82 KW)

There should not be any channel interference from IBRC

Audio Broadcasting: Qualitative Description of Service Quality



Status Report on PAC and MPAC: Perceptual Audio Coders from AT&T

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Abstract

The perceptual audio coder (PAC) is a powerful psychoacoustic algorithm that provides high-quality CD stereo at compression ratios exceeding 10:1. This capability of the PAC algorithm is critically needed for providing high-quality audio services in bandwidth-limited applications such as ISDN music delivery, digital audio broadcasting and multichannel sound for advanced television. Currently, the PAC algorithm addresses these applications with respective transmission rates of 64 kbps (stereo), 128 kbps (stereo) and 320 kbps (five channels). This paper provides a status report on PAC and its current applications.

Introduction

Following its contribution to the MPEG-1 Audio Standard [1,2], in particular to its lowest bit rate version (Layer 3), AT&T Bell Laboratories proceeded with the creation of a new coding algorithm that was particularly suited for compression ratios on the order of 10:1 or higher – for example, coding of the 1411 kbps CD-stereo signal at rates on the order of 128 kbps. The fundamental needs of such a design, as dictated by considerations of signal processing psychoacoustics and coding, caused an inevitable divergence from the MPEG-1 audio format.

Recently, following the successful implementation and application of the stereo PAC algorithm in DAB experiments in the United States, the PAC algorithm was brought back into the MPEG standardization process as a non-backward-compatible (NBC) system in the MPEG-2 process for the coding of 5-channel audio. A rigorous subjective test of various backward-compatible (BC) and NBC systems was undertaken as part of the MPEG-2 work. In this test, the multichannel PAC algorithm (MPAC) emerged as the leading contender

for a new MPEG-2 standard. This paper is a status report on the stereo PAC coder (as applied to the DAB standard process in the United States), and the MPAC coder (as used in the ongoing MPEG2 process).

Section 1 describes the stereo PAC algorithm. Section 2 summarizes its rate in the DAB process. Section 3 reviews the still-evolving MPAC algorithm. Section 4 comments on future work on PAC, MPAC and their extensions.

1 The Stereo PAC Algorithm

The PAC algorithm [3,4] is based on transform coding of audio signals using perceptual noise criteria. a technique that was pioneered at AT&T Bell Laboratories [5]. The perceptual audio coder is a psychoacoustically driven system based on empirical, but well-calibrated rules for utilizing the phenomenon of noise-masking. The principle of *simultaneous* or *frequency-domain* masking defines a just-*noticeable*-distortion (JND) profile (Figure 1) below which quantization noise (say, due to compression) cannot be perceived. The JND profile is a reflection of the fact that a signal can mask a weaker signal in its *frequency vicinity*, even when the difference between the levels of the two signals is not substantial. The principle of *non-simultaneous*, or *time-domain* masking (Figure 2a) utilizes the masking of the weaker signal in the *time-vicinity* of the stronger signal. All psychoacoustic coders attempt to utilize the above phenomenon, but the effective use of masking depends on the accuracy of the psychoacoustic model and on how well the signal-analysis framework facilitates the application of that model for coding.

The JND model in the PAC algorithm is currently based on an input-dependent interpolation between well-known models for *noise-masking-tone* and *tone-masking-noise*, combined with additional, masking terms which reflect the spread of masking beyond the critical band (staircase tread in Figure 1) that contains the masker.

The phenomenon of temporal masking is maximized in PAC by means of input-dependent switching between long and short blocklengths for frequency-analysis (Figure 2b). Transitional segments tend to be analyzed with a shorter blocklength in the MDCT (modified discrete cosine transform). As mentioned, block switching is input-adaptive, and it is based on a carefully designed psychoacoustic criterion.

Another unique feature of PAC is the method used for the joint-coding of the left (L) and right (R) channels in a stereo pair. The PAC algorithm provides both for the independent coding of these channels (L and R) and for composite coding that uses the sum and difference signals (L+R and L-R) as coder inputs. The decision of stereo-coding mode is flexible, time- and frequency-dependent, and based on psychoacoustic principles that avoid psychoacoustic artifacts such as *noise-unmasking*.

The PAC algorithm finally includes an adaptive entropy coder that further reduces the total bit rate. Entropy coding and psychoacoustic quantization are jointly performed in an iterative operation.

A block diagram of the stereo PAC coder appears in Figure 3. Although the stereo encoder is fairly sophisticated, its design is guided by the need for robust implementation in current signal processing technology. The stereo decoder is quite simple, and it is currently implemented on a single general-purpose microprocessor.

2 The Application of PAC to DAB Technology

The United States has begun the process for defining standards for digital audio broadcasting (DAB), also referred to as digital audio radio (DAR). The process includes testing of the wideband *Eureka* system, an S-band satellite system, and a number of *In-Band* systems that are matched to the basic 200 kHz subdivision in terrestrial FM broadcasting. The In-Band systems are classified into the categories of *On-Channel* (IBOC), *Adjacent Channel* (IBAC) and *Reserved-Channel* (IBRC). Figure 4 provides simplified descriptions of In-Band DAR and FM spectra, and Figure 5 provides an illustration of how the In-Band technologies may evolve in a system where the space-frequency plan is currently based on fairly well-separated FM stations.

The performance of the stereo PAC coder at compression ratios on the order of 10:1 makes it an ideal candidate for the audio subsystem of DAR technology. In the USA-DAR contest, the AT&T systems for IBAC and IBRC broadcasting use PAC at a rate of 160 kbps, while the AT&T-Amati system for IBOC broadcasting uses PAC at two alternative bit rates: 160 kbps for the double-sidelobe operation (as in Figure 4), and 128 kbps for single-sidelobe operation. The satellite system being developed by the Voice of America – NASA – JPL consortium uses PAC at 160 kbps. Outside of the contest, an experiment satellite system developed by CD-Radio uses PAC at 128 kbps.

The MPEG-Layer 2 coder is also being tested in the DAR contest, as part of the systems offered by Eureka and USA-Digital Radio. This coder operates at higher bit rates, up to 256 kbps for the stereo pair.

3

The USA-DAR contest is being administered jointly by the Electronics Industries Association (EIA) and the National Radio Systems Committee (NRSC). Laboratory tests at the NASA-Lewis Research Center, with subjective tests at the CRC (Communications Research Centre, Canada) are expected to last through the end of 1994. Field testing of candidate DAR systems are planued for 1995.

The low bit rate capability of the stereo PAC coder is extremely well-matched to the needs of DAR technology for two fundamental reasons: it permits the use of a grater part of the 200 kHz capacity for transmission error protection, and it permits the use of a significant portion of the capacity for the transmission of additional data services.

In the AT&T-IBAC and AT&T-IBRC systems, the 200 kHz channel carries 360 kbps, permitting a view powerful rate-1/2 code for protecting the PAC bit stream. The DAR system actually has three levels of error-protection: an initial protection of a few very critical bits in the initial PAC bit stream, the rate-1/2 protection of the final PAC bit "ream, and a proprietary error-concealment procedure at the receiver. The concealment "sorithm addresses occasional block-error failures (audio mutes) which are caused when the transmission channel is poor enough to defeat the combined capability of the error protection-interleaving-channel equalization system.

The additional data capacity in the DAR system is on the order of 10 to 20 kbps. This includes synchronous data that are multiplexed to the PAC bit stream as well as asynchronous data that can be added when the (constant-quality, variable-rate) PAC algorithm does not need the allocated constant coding rate (say 160 kbps) for providing high-quality reproduction of an audio segment (which is typically about 10 ms long).

3 The Multichannel Perceptual Audio Coder (MPAC)

The 5-channel MPAC coder at 320 kbps is a natural extension of the 2-channel PAC algorithm at bit rates on the order of 128 kbps.

In a simple version of MPAC, the signal-dependent composite coding algorithm in the stereo PAC coder is repeatedly applied to pairwise combinations of the five channels (L, R, C, LS, and RS) at the input of the MPAC algorithm. This results in various sets of JND thresholds which are either specific to an individual channel or to a channel-pair. Simple subalgorithms provide coding of 3-channels (L, R, C) or of stereo (L, R).

The MPAC decoder is designed for simple implementation, and the 5-channel decoder for MPEG-2 testing in 1993 has been implemented on a single microprocessor.

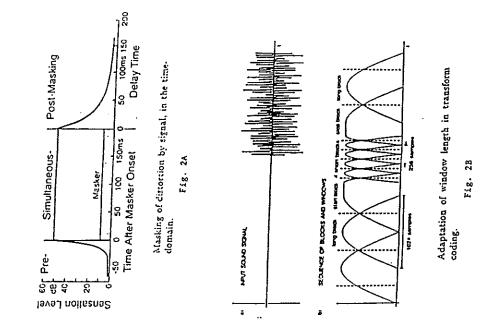
Table 1 is an excerpt from the subjective tests conducted by Deutsche Telekom and the BBC in support of an initial phase of the MPEG-NBC process. The purpose of this test was to demonstrate the need for an NBC part of the MPEG-2 process. The results of Table 1 indeed demonstrated such a need, and this has led to the decision to begin a formal contest for an NBC standard for multichannel audio. A second result, also clear from Table 1, is that the MPAC coder provided the overall best performance at 320 kbps, with a significant margin over the second best system tested. One of the detailed results in the test, not apparent in Table 1, is that the MPAC system performed conspicuously poorly on one of the 10 audio stimulii tested (*fountain-music*). This has been addressed subsequent to the test, and the current version of the MPAC coder has been observed to be - more robust across different stimulii. Recent modifications of the MPAC algorithm have in fact resulted in an even simpler algorithm for encoding and decoding.

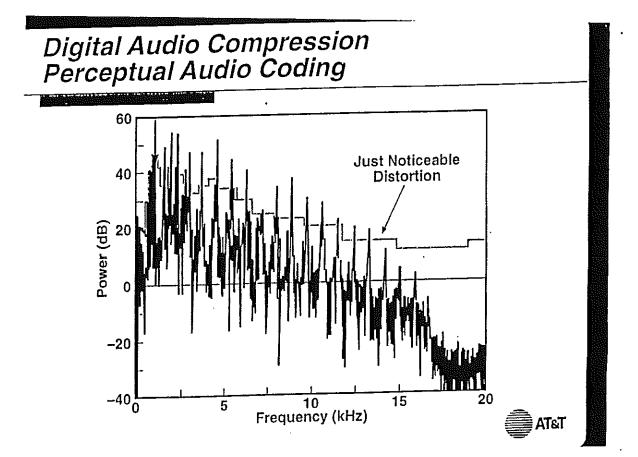
4 Future Work

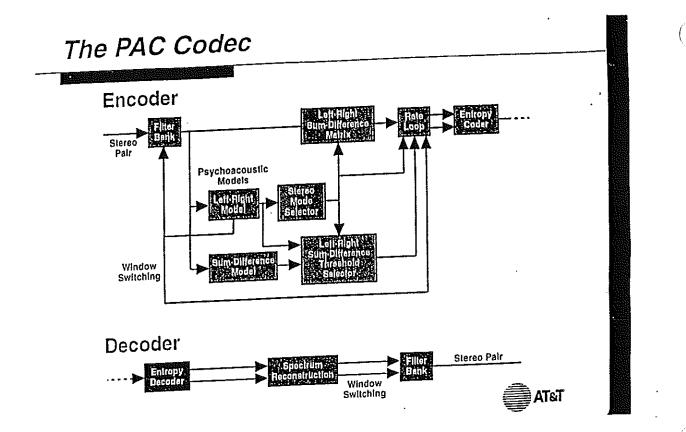
Recent work has shown that PAC and PAC-like algorithms degrade extremely gracefully as the compression ratio is increased beyond the level of about 10:1 discussed in this paper. In particular, compression ratios on the order of 20:1 will be very significant for emerging applications such as MPEG4 audio and the next generation of FM-band and AM-band broadcasting. Work on these extensions of PAC are in progress and will be reported in due course. At that time, we also plan to provide reports on real-time implementations of the MPAC codec.

5 References

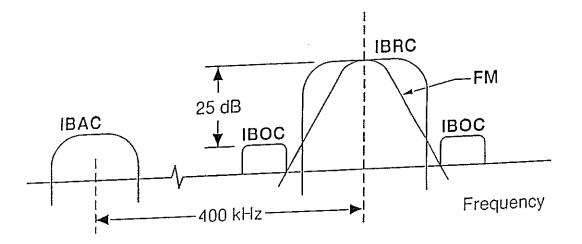
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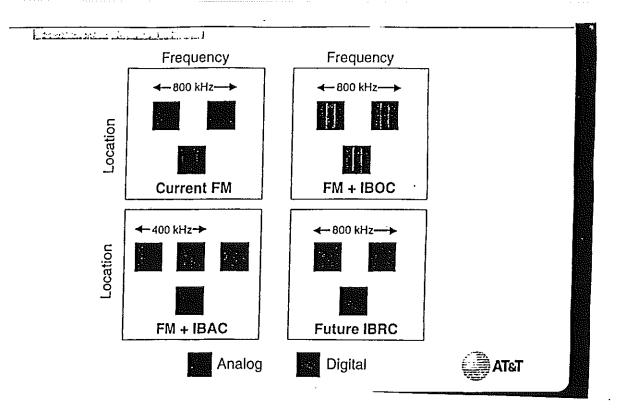






Audio Broadcasting in the FM-Band





MPEG2 Results (5-Channel Audio)

Comparison of 5-channel audio codecs at 320kbps Number of Signals (out of 10) that are transparently coded*

		Testing Laboratory	
		Deutsche Telekom	BBC
Philips	(MPEG-Layer 2)	1	0
Dolby	(AC3)	2	1
AT&T	(MPAC)	6	5

*Average quality loss of less than 0.5 on a 5-point scale . Based on the opinions of 45 expert listeners

Excerpted from MPEG data NSJ 3.28.94

AT&T

Eureka 147

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L-Band System

EUREKA 147/DAB SYSTEM DESCRIPTION

1. Introduction

The Eureka 147 DAB System is designed to provide high-quality, multi-service digital radio broadcasting for reception by vehicular, portable and fixed receivers. It is designed to operate at any frequency up to 3000 MHz for terrestrial, satellite, hybrid (satellite and terrestrial), and cable broadcast delivery. The System is also designed as a flexible, general-purpose Integrated Services Digital Broadcasting (ISDB) system which can support a wide range of source and channel coding options, sound-programme associated data and independent data services, in conformity with the flexible and broad-ranging service and system requirements given in CCIR Recommendations 789 and 774, supported by Reports 1203-2 and 955-3.

The system is a rugged, yet highly spectrum and power-efficient sound and data broadcasting system. It uses advanced digital techniques to remove redundancy and perceptually irrelevant information from the audio source signal, then it applies closely-controlled redundancy to the transmitted signal for error correction. The transmitted information is then spread in both the frequency and time domains so a high quality signal is obtained in the DAB receivers, even when working in conditions of severe multipath propagation, whether stationary or mobile. Efficient spectrum utilization is achieved by interleaving multiple programme signals and a special feature of frequency re-use permits broadcasting networks to be extended, virtually vithout limit, UGFN using additional transmitters all operating on the same radiated frequency.

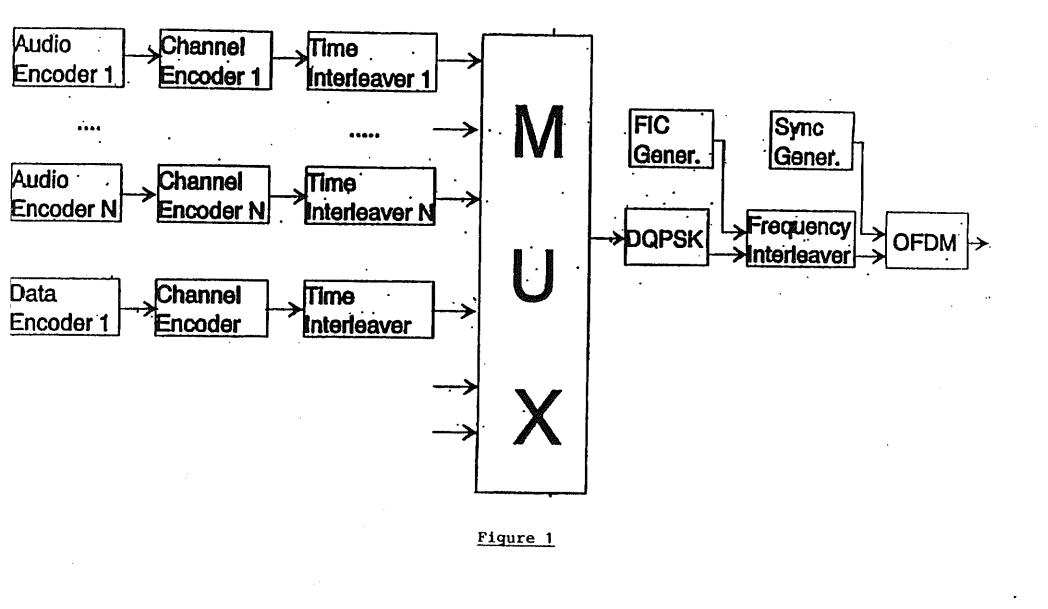
A conceptual block diagram of the emission part of the System is shown in <u>Fig. 1</u>.

2. Use of a layered model

The System is capable of complying with the ISO Open System Interconnection (OSI) basic reference model described in ISO 7498 (1984). The use of this model is recommended in draft new Recommendation 11/67 and Report 1207, and a suitable interpretation for use with layered broadcasting systems is given in the Recommendation. In accordance with this guidance, the System vill be described in relation to the layers of the model.

Descriptions of many of the techniques involved are most easily given in relation to the operation of the equipment at the transmitter, or at the central point of a distribution network in the case of a network of transmitters.

The fundamental purpose of the System is to provide sound programmes to the radio listener, so the order of sections in the following description will start from the application layer (use of the broadcast information), and proceed downwards to the physical layer (the means for radio transmission).



:

3. Application layer

This layer concerns the use of the System at the application level. It considers the facilities and audio quality which the System provides and which broadcasters can offer to their listeners, and the different transmission modes.

3.1 Facilities offered by the System

The System provides a signal which carries a multiplex of digital data, and this conveys several programmes at the same time. The multiplex contains audio programme data, and ancillary data comprising Programme-Associated Data (PAD), Multiplex Configuration Information (MCI) and Service Information (SI). The multiplex may also carry general data services which may not be related to the transmission of sound programmes.

In particular, the following facilities are made available to users of the System:

- a) the audio signal (i.e. the programme) being provided by the selected programme service (i.e. the broadcaster's channel, carrying a succession of programmes)
- b) the optional application of receiver functions, for example dynamic range control, which may use ancillary data carried with the programme,
- c) a text display of selected information carried in the SI. This may be information about the selected programme, or about others which are available
- d) options which are available for selecting other programmes, other receiver
- e) one or more general data services, for example a Traffic Message Channel

The System includes facilities for conditional access, and a receiver can be equipped with digital outputs for audio and data signals.

3.2 Audio quality

Within the capacity of the multiplex, the number of programme services and, for each, the presentation format (e.g. stereo, mono, surround-sound, etc.), the audio quality and the degree of error protection (and hence ruggedness) can be chosen to meet the needs of the broadcasters.

The following range of options is available for the audio quality:

- a) very high quality, sufficient for audio post-processing,
- b) fully transparent quality, sufficient for the highest quality broadcasting, c) high quality, equivalent to good FM service quality,
- d) medium quality, equivalent to good AM service quality, e) speech-only quality.

The System provides full quality reception within the limits of transmitter coverage; beyond these limits reception degrades in a subjectively graceful manner.

3.3 Transmission modes

The System has 3 optional transmission modes which allow the use of a wide range of transmitting frequencies up to 3 GHz. These transmission modes have been designed to cope with Doppler spread and delay spread, for mobile reception in presence of multipath echoes.

The following table gives the constructive each delay and nominal frequency range for mobile reception. The noise degradation at the highest frequency and in the most critical multipath condition occuring infrequently in practice is equal to 1 dB at 60 mph.

Parameter	Mode I	Mode II	Mode III
Guard interval duration Constructive echo delay up to: Nominal frequency range	250 μs 300 μs	62.5 μs 75 μs	31.25 µs 37.5 µs
(for mobile reception) up to:	375 HEZ (,8 m)	1.5 GHz (.2 m)	3 GHz

From this table, it can be seen that the use of higher frequencies imposes a greater limitation on the maximum echo delay. Mode I is most suitable for a terrestrial single-frequency network (SFN), because it allows the greatest transmitter separations. Mode II is most suitable for local radio applications requiring one terrestrial transmitter, and hybrid satellite/terrestrial transmission. However, Mode II can also be used for a medium-to-large scale SFN (e.g. at 1.5 GHz) by inserting artificial delays at the transmitters and by using directive transmitting antennas. Mode III is most appropriate for cable, satellite and complementary terrestrial transmission, since it is able to operate at all frequencies up to 3 GHz and for mobile reception.

4. <u>Presentation layer</u>

This layer concerns the conversion and presentation of the broadcast information.

4.1 <u>Audio source encoding</u>

The audio source encoding method used by the System is ISO/IEC MPEG-Audio Layer II, given in the ISO Standard 11172-3. This sub-band coding compression system is also known as the MUSICAM system.

The System accepts a number of PCM audio signals at a sampling rate of 48 kHz with programme-associated data (PAD). The number of possible audio sources depends on the bit rate and the error protection profile. The audio encoder can work at 32, 48, 56, 64, 80, 96, 112, 128, 160 or 192 kbit/s per monophonic channel. In stereophonic or dual channel mode, the encoder produces twice the bit rate of a mono channel.

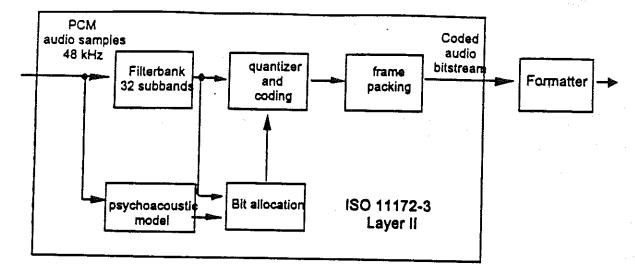
The different bit-rate options can be exploited by broadcasters depending on the intrinsic quality required and/or the number of sound programmes to be provided. For example, the use of bit-rates greater than or equal to 128 kbit/s for mono, or greater than or equal to 256 kbit/s for a stereo programme, provides not only very high quality, but also some processing margin, sufficient for further multiple encoding/decoding processes, including audio post-processing. For high-quality broadcasting purposes, a bit-rate of 128 kbit/s for mono or 256 kbit/s for stereo is preferred, giving fully transparent audio quality. Even the bit-rate of 192 kbit/s per stereo programme generally fulfils the EBU requirements for digital audio bit-rate reduction 48 kbit/s can provide roughly the same quality as normal AM broadcasts. For some speech-only programmes, a bit-rate of 32 kbit/s may be sufficient where the greatest number of services is required within the DAB multiplex.

A block diagram of the functional units in the audio encoder is given in <u>Fig. 2</u>. The input PCM audio samples are fed into the audio encoder. One encoder is capable of processing both channels of a stereo signal, although it may, optionally, be presented with a mono signal. A polyphase filter bank divides the digital audio signal into 32 sub-band signals, and creates a filtered and sub-sampled representation of the input audio signal. The filtered samples are called sub-band samples. A perceptual model of the human ear creates a set of data to control the quantizer and coding. These data can be different, depending on the actual implementation of the encoder. One possibility is to use an estimation of the masking threshold to obtain these into blocks, then, in each block, the maximum amplitude attained by each sub-band signal is determined and indicated by a scale factor. The quantizer and coding unit creates a set of coding words from the sub-band samples. These the Network layer.

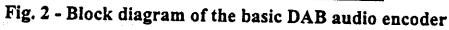
4.2 Audio decoding

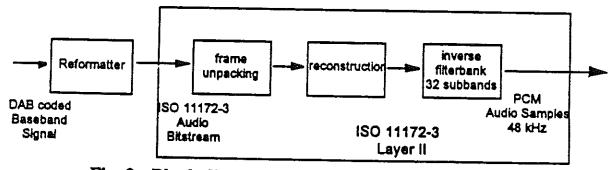
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Decoding in the receiver is straightforward and economical using a simple signal processing technique, requiring only de-multiplexing, expanding and inverse-filtering operations. A block diagram of the functional units in the decoder is given in <u>Fig. 3</u>.



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The ISO audio frame is fed into the ISO/HPEG-Audio Layer II decoder, which unpacks the data of the frame to recover the various elements of information. The reconstruction unit reconstructs the quantized sub-band samples, and an inverse filter bank transforms the sub-band samples back to produce digital uniform PCM audio signals at 48 kHz sampling rate.

4.3 Audio presentation

Audio signals may be presented monophonically or stereophonically, or audio channels may be grouped for surround-sound. Programmes may be linked to provide the same programme simultaneously in a number of different languages. In order to satisfy listeners in both Hi-Fi and noisy environments, the broadcaster can optionally transmit a Dynamic Range Control (DRC) signal in the receiver in the noisy environments to compress the dynamic range of the reproduced audio signal. Note that this technique can also be beneficial to listeners with impaired hearing.

4.4 Presentation of service information

With each programme transmitted by the System, the following elements of SI can be made available for display on a receiver:

- basic programme label (i.e. the name of the programme)
- time and date
- cross-reference to the same, or similar programme (e.g. in another language) being transmitted in another DAB signal or being simulcast by an AM or FM service
- extended service label for programme-related services
- programme label (e.g. the names of performers)

- programme type (e.g. news, sport, music, etc.)
- transmitter identifier
- Traffic Message Channel (THC, which may use a speech synthesiser in the receiver).

Transmitter network data can also be included for internal use by broadcasters.

5. <u>Session</u> layer

This layer concerns the selection of, and access to, broadcast information.

5.1 <u>Programme selection</u>

In order that a receiver can gain access to any or all of the individual services with a minimum overall delay, information about the current and future content of the multiplex is carried by the Fast Information Channel (FIC). This Information is the MCI, which is machine-readable data. Data in the FIC are not time-interleaved, so the MCI does not suffer the delay inherent in the time-interleaving process applied to audio and general data services. However, these data are repeated frequently to ensure their ruggedness. When the multiplex configuration is about to change, the new information, together with the timing of the change, is sent in advance in the MCI.

⁻ language

The user of a receiver can select programmes on the basis of textual information carried in the SI, using the programme service identity, the programme type identity or the language. The selection is then implemented in the receiver using the corresponding elements of the MCI.

If alternative sources of a chosen programme service are available and an original digital service becomes untenable, then linking data carried in the SI (i.e. the 'cross reference') may be used to identify an alternative (e.g. on FM service) and switch to it. However, in such a case, the receiver will switch back to the original service as soon as reception is possible.

5.2 <u>Conditional access</u>

Provision is made for both synchronization and control of conditional access using information contained in the PAD.

6. Transport layer

This layer concerns the identification of groups of data as programme services, the multiplexing of data for those services and the association of elements of the multiplexed data.

6.1 Programme services

A programme service is the group of one or more programmes which are being broadcast by a service provider (i.e. a broadcaster) at any given time. The whole capacity of the multiplex may be devoted to one programme service (e.g. broadcasting five or six high-quality sound programmes), or it may be divided amongst several programme services.

6.2 <u>Hain service multiplex</u>

With reference to <u>Fig. 1</u>, the data representing each of the programmes being broadcast (digital audio data with some ancillary data, and maybe also general data) are subjected to convolutional encoding (see Section 9.1) and time-interleaving, both for error protection. Time-interleaving improves the ruggedness of data transmission in a changing environment (e.g. reception by a moving vehicular receiver) and imposes a predictable transmission delay. The interleaved and encoded data are then fed to the main service multiplexer where, each 24 ms, the data are gathered in sequence into the multiplex frame. The combined bit-stream output from the multiplexer is known as the Main Service Channel (MSC) which has a gross capacity of 2.3 Mbit/s. Depending on the choosen code rate (which can be different from one application to another), this gives a net bit rate ranging from approximately 0.8 to 1.7 Mbit/s, through a 1.5 MHz bandwidth. The main service multiplexer is the point at which synchronized data from all of the programme services using the multiplex are brought together.

General data may be sent in the MSC as an unstructured stream or organized as a packet multiplex where several sources are combined. The data rate may be any multiple of 8 kbit/s, synchronized to the System multiplex, subject to sufficient total multiplex capacity, taking into account the demand for audio services.

The Fast Information Channel (FIC) is external to the MSC and is not time-interleaved.

6.3 Ancillary data

There are three areas where ancillary data may be carried within the System multiplex: "Fast Information Channell"

- a) the FIC, which has limited capacity, depending on the amount of essential MCI included, "Program Associated Data"
- b) there is special provision for a moderate amount of PAD to be carried within each audio channel.
- c) all remaining ancillary data are treated as a separate service within the MSC. The presence of this information is signalled in the MCI. "Main service channel" "Maltiples Configuration Information"

6.4 Association of data

A precise description of the current and future content of the MSC is provided by the MCI, which is carried by the FIC. Essential items of SI which concern the content of the MSC (i.e. for programme selection) must also be carried in the FIC. More extensive text, such as a list of all the day's programmes, must be carried separately as a general data service. Thus, the HCI and SI contain contributions from all of the programme services using the multiplex, describing all of the programmes being broadcast.

The PAD, carried within each audio channel, comprises mainly the information which is intimately linked to the sound programme and therefore cannot be sent in a different data channel which may be subject to a different transmission delay.

7. Network layer

This layer concerns the identification of groups of data as programmes.

7.1 ISO audio frames

The processes in the audio source encoder are carried out during ISO audio frames of 24 ms duration. The bit allocation, which varies from frame to frame, and the scale factors are coded and multiplexed with the sub-band samples in each ISO audio frame. The frame packing unit (see Fig. 2) assembles the actual bit stream from the output data of the quantizer and coding unit, and adds other information, such as header information CRC words for error detection, and PAD, which travel along with the coded audio signal. Each audio channel contains a PAD channel, having a variable capacity (generally at least 2 kbit/s), which can be used to convey information which is intimately linked to the sound programme. Typical examples are lyrics, speech/music indication and DRC information.

The resulting audio frame carries data representing 24 ms duration of stereo (or mono) audio, plus the PAD, for a single programme and complies with the ISO 11172-3 Layer II format, so it can be called an ISO frame. This allows the use of an ISO/MPEG-Audio Layer II decoder in the receiver.

8. Data link layer

This layer provides the means for receiver synchronization.

8.1 The transmission frame

In order to facilitate receiver synchronization, the transmitted signal is built up with a frame structure having a fixed sequence of symbols (see Section 9.2). Each transmission frame begins with a null symbol for coarse synchronization (when no RF signal is transmitted), followed by a fixed reference symbol to provide fine synchronization, AGC, AFC and phase reference functions in the receiver. The next symbols are reserved for the FIC, and the remaining symbols provide the MSC. The total frame duration T, is either 96 ms or 24 ms, depending on the transmission mode as given in Table below:

	Mode I	Mode II	Mode III
	(375 мнг)	(1.5 GHz)	(3 GHz)
Troll Troll Ts ts t guard ts ts ts ts ts ts ts ts ts ts ts ts ts	96 ms 1 ms 1.25 ms 1 ms 1536 25° μκς	24 ms 250 us 312.5 us 250 us 384 625 pxc	24 ms 250 μs 156.25 μs 125 μs 192 31,25 μs

The following notation is used:

T_{NULL} is the null symbol duration

t, is the useful symbol duration

N is the number of radiated carriers

Each audio service within the MSC is allotted a fixed time slot in the frame.

9. The physical layer

This layer concerns the means for radio transmission; the modulation scheme and the associated error protection.

9.1 Convolutional encoding

Convolutional encoding is applied to each of the data sources feeding the multiplex to facilitate error correction in receivers. The encoding process involves adding deliberate redundancy to the source data bursts using a constraint length of 7. This gives 'gross' data bursts. In the case of an audio signal, greater protection is given to some source-encoded bits than others, following a preselected pattern known as the Unequal Error Protection (UEP) profile. The average code rate, defined as the ratio between the number of source-encoded bits and the number of encoded bits after convolutional encoding, may take a value from 1/3 (the highest protection level) to 3/4 (the lowest portection level). Different average code rates can be applied to different audio sources, subject to the protection level required and the bit-rate of the source-encoded data. For example, the protection level of audio services carried by cable networks may be lower than that of services transmitted in radio-frequency channels. j.

General data services are convolutionally encoded using one of a selection of uniform rates. Data in the FIC are encoded at a constant 1/3 rate.

9.2 <u>Modulation by COFDM</u>

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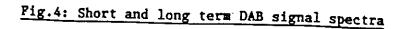
The System uses a modulation scheme known as Coded Orthogonal Frequency Division Multiplex (COFDM). This scheme meets the exacting requirements of high bit-rate digital broadcasting to mobile, portable and fixed receivers, especially in multipath environments.

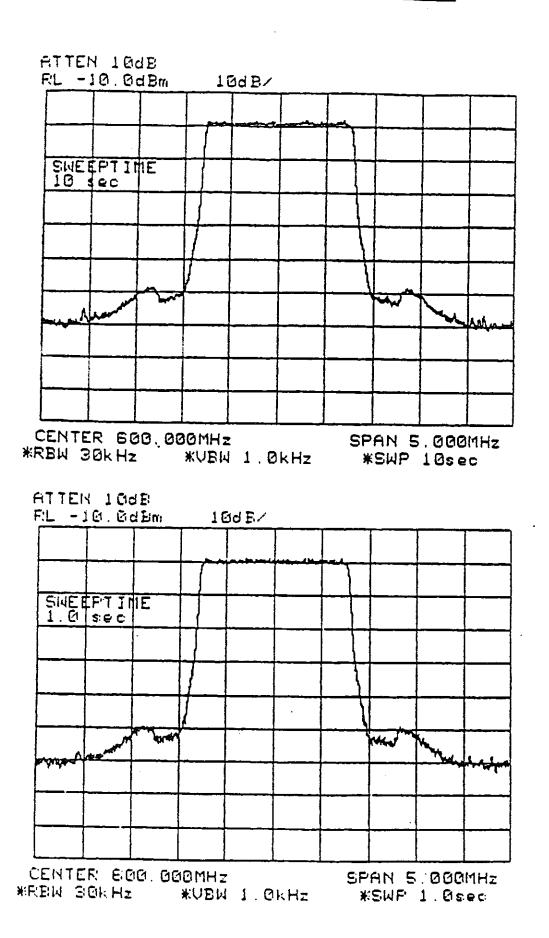
The basic principle consists of dividing the information to be transmitted into a large number of bit-streams, having low bit-rates individually, which are then used to modulate individual sub-carriers. The corresponding symbol duration becomes larger than the delay spread of the transmission channel. Therefore, provided that a temporal guard interval is inserted between successive symbols, channel selectivity and multipath propagation will not cause inter-symbol interference. The large number of sub-carriers is known, collectively, as an ensemble.

In the presence of multipath propagation, some of the carriers are enhanced by constructive signals, while others suffer destructive interference (frequency selective fading). Therefore, the System includes a redistribution of the elements of the digital bit stream, such that successive source samples are affected by independent fades. When the receiver is stationary, the diversity in the frequency domain is the only means to ensure successful reception; the time diversity provided by time-interleaving does not assist a static receiver. For the System, multipath propagation is a form of space-diversity and is considered to be a significant advantage, in stark contrast to conventional FM or narrow-band digital systems where multipath propagation can completely destroy a service.

In any system able to benefit from multipath, the larger the transmission channel bandwidth, the more rugged the system. In the system, an ensemble bandwidth of 1.5 MHz is used to secure the advantages of the wideband technique as well as to allow planning flexibility. The table above also indicates the number of COFDM carriers within this bandwidth for each transmission mode. The long and short power spectra are shown in the <u>Fig. 4</u>.

Further benefit of using COFDM is that high spectrum and power efficiency can be obtained with single frequency networks for large area coverage and also for city area dense networks. Any number of transmitters providing the same programmes may be operated on the same frequency which also results in an overall reduction in required operating powers. As a further consequence, the frequency reuse distances between different service areas are .





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Another benefit is that all types of receiver (e.g. portable, home, vehicular) may utilize a simple, non directional antenna.

9.3 Energy dispersal

In order to ensure appropriate energy dispersal in the transmitted signal, the individual sources feeding the multiplex are scrambled.

10. System implementation plan

The System has been intensively tested and demonstrated, jointly with the EBU, in America and in Europe, since 1988. The design of the integrated circuits for the receiver began in 1990, and the first service to the public is planned to begin in Summer 1995 in Europe. Experimenal broadcast signals are regularly on air in the VHF and UHF broadcasting bands in several countries in Europe. The System is currently being standardized by the European Telecommunications Standards Institute.

11. Specific frequency for testing

As indicated in Section 3.3, the System has 3 transmission modes which allow the use of a wide range of transmitting frequencies up to 3 GHz for mobile reception. Higher frequencies may be used for fixed or portable reception.

For EIA testing, operation in the 1452-1492 MHz is desired.

However, the equipment can operate in VHF and UHF bands as previously communicated to the EIA and test results are already available.

12. Data rate and structure

The hardware submitted to EIA testing will operate with a sample set of the many possible multiplex configurations. The multiplex configuration considered is the following:

Source	Net bit rate	Channel code rate	Gross bit rate	kHz/channel
Stereo prog. 1	256 kb/s	0.57	450 kb/s	300
Stereo prog. 2	224 kb/s	0.50	450 kb/s	300
Stereo prog. 3	224 kb/s	0.60	370 kb/s	250
Stereo prog. 4	192 kb/s	0.51	370 kb/s	250
Stereo prog. 5	192 kb/s	0.62	310 kb/s	200
Hono prog. 6	64 kb/s	0.5	128 kb/s	85
Data service 1	64 kb/s	0.5	128 kb/s	85
Data service 2	24 kb/s	0.37	64 kb/s	43

The source bit rate of each audio signal permits about 2 kbit/s Programme Associated Data capacity for each programme, as well as specific error protection data. TABLE 2: LIST OF SPECIFIC INFORMATION NEEDED FOR DAR SYSTEM TESTING

1. Pover measurement

The transmitted signal is shaped in such a way that the difference between the peak power and the average power is approximately 8 dB.

This is achieved by digital processing in the transmitter. The average power should be measured by a calorimetric method, whereas the peak power can be deduced from the peak-voltage measurements at IF using a fast storage oscilloscope; this is well known to be an accurate method.

2. <u>Hultipath and coverage improvements methods</u>

The method to minimize multipath interference is described in Section 9 of the previous system description.

Additional information is provided in the following:

Some types of diversity techniques are vell-known in the field of digital communication in the frequency-selective Rayleigh channel, such as the mobile radio channel. These are mainly time, frequency and space diversity techniques.

Table 1 indicates the relation between the type of channel and the allability of such diversities used by the system.

Table 2 recalls the system features allowing for the use of these diversity techniques.

Diversity Channel type	Time diversity diversi	Frequency ity	- •	Space diversity at the receiver
Rural area	available when receiver in motion	moderate when delay spread x Bw is small (flat fading)	/	available optionally with multiple antennas receiver
Urban area	available when receiver in motion	generally available, especially when using gap filling	available when using gap filling	available optionally with multiple antennas receiver
Single frequency network	available	always available where necessary	always available especially where necessary	available optionally with multiple antennas receiver

TABLE 1

TABLE 2

Type of diversity	Associated feature		
Time diversity	- convolutional coding + time interleaving		
Frequency diversity and space diversity	- guard interval - convolutional coding + frequency interleaving		

It can be seen from Table 1, that, as soon as gap-filling or single-frequency networking are used, the space diversity becomes available on the network side. This means that spacially distributed transmitters contribute by power adding to the received signal. As generally the position of these transmitters is not concentrated in only one direction, this feature is very useful to avoid a complete shadowing when a obstacle (building, hill) is masking the signal in a given direction of the horizontal plane.

On the other hand, when the receiver is close to a given transmitter of the network, this space diversity is not needed because the vicinity of the transmitter implies high signal-to-noise ratio.

The simultaneous contribution of these various diversity techniques makes the system very robust and efficient for digital audio broadcasting of high-quality sound with a continuity of service inside the coverage area. This feature was predicted by simulation studies, and confirmed by numerous on-air experiments in Europe and in America.

As far as echoes can be resolved (separated) in relation to the system bandwidth (1.5 MHz), their powers add fully constructively as long as their delays are shorter than the guard interval duration; when an echo, natural or artificial, is delayed by more than the guard interval duration, a part of its power remains useful while the complementary part becomes interfering, with white gaussian noise characteristics.

3. Prequency band and bandwidth

The signal for testing should be included in the band 1452-1492 MHz. The signal bandwidth is 1.5 MHz.

The system is rugged against co-channel interferers, especially narrowband or narrow-band-like interferers, as well as impulsive noise.

As far as adjacent channel interferers are concerned, the filtering capability of the system is the conjunction between analog filtering and rectangular filtering (1.5 HHz) of the FFT in the receiver. This means that guard bands are theoretically required only when the adjacent interferer is received at a much higher level than the wanted signal.

4. Co-channel/adjacent channel testing

Efforts will be done to provide possibly a second transmitter for DAR to protection ratio measurements.

However, it has already been proven that the characteristics of an uncorrelated DAR interferer are those of a Gaussian noise, white over 1.5 MHz bandwidth.

5. System signal acquisition response

When switching on a receiver, the delay to obtain an audio response is less than 0.5 second. This delay includes the time and frequency synchronization of the receiver and the memory delay for time deinterleaving.

When a receiver is already synchronized and the average C/N falls below the limit of audio signal availability, the synchronization will still remain available up to a value approximately 10 dB below this threshold, and there will be no extra-delay for resynchronization when the received power is back to the threshold of availability of the audio signals.

6. Program content restrictions

This point does not apply.

7. RF system interaction with DAR system

The hardware provided for testing will include a receiving antenna, matched to the L-Band range; this antenna is a "lambda over 4" antenna including a band-pass filter and an active preamplifier.

It is also intended to supply transmitting RF hardware and antenna, as appropriate for EIA test requirements.

8. Test system physical parameters

Transmitter equipment:

 Source encoders rack: 19" - 6 U analog and EBU/AES inputs 	15 kg
- Channel encoder : 19" - 6 U - Analog modulator : 19" - 3 U coaxial RF output 50 ohms; - 5 dBm	10 kg 6 kg
- Synthesizer for transposion: 19" - 3 U - Power amplifier and antenna: dependent on te	15 kg est requirements
Total consumption: 220 Volts 400 Watts max. (Without power amplifier)	
Receiving equipment: 1 rack + cable + antenna	
Size: Height: 31 cm, Width: 23 cm, Depth: 40 c	±

Weigth: 12 kg, consumption 12 to 28 Volts DC, 70 Watts

9. Service range and system response

The protection given by the convolutional code (see Section 9 of the system description) is optimized in such a way that greater protection is given to some audio bits, more sensitive to errors, following a preselected pattern known as an unequal error protection profile. This results, in conjunction with adequate audio concealment, in a subjective graceful degradation of the audio signal, at the limit of the coverage area.

This coverage limit depends on the choice (made by the broadcaster) of the redundancy added to the source data by the convolutional encoder. The code rate ranges between 1/3 and 3/4, but for the available hardware, not all possibilities are implemented (see System description). The performance of the system is dependent on the averaged value (over approx. 1/3 second) of the C/N+I ratios, where:

C is the total useful power of all echoes (as defined in point 2) N is the noise power I is the total interfering power of the echoes exceeding the guard interval.

Furthermore, in the multiplex control data included in the FIC (see system description), provision is made for data linking the DAR programmes to alternative FM frequencies where the same programmes might be available.

10. System application limitations

Section 3.3 of the system description indicates as a function of the transmission mode used, the constructive echo delay range and the nominal transmission frequency range for mobile reception.

Parameter	Mode I	Mode II	Mode III
Constructive echo delay up to: Nominal frequency for mobile	300 µs	75 µs	37 .5 µs
reception up to:	375 MHz	1.5 GHz	3 GHz

At the maximum frequency, the noise degradation in the most critical multipath conditions, which rarely occurs, is equal to 1 dB at approximately 60 mph.

It should be noted that in special receiving conditions, as in aircraft reception, the multipath situation is often not critical in the sense that the Doppler effect is not frequency dispersive, so that the above mentioned restrictions in upper frequency and speed do not apply.

When time interleaving is becoming useless due to very low speed, the frequency diversity obtained through the use of a 1.5 MHz bandwidth is used, thanks to frequency interleaving, to combat wide and deep fades.

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A.J. VIGIL, PH.D

Engineering Manager

July 12, 1995

Dave Wilson Staff Engineer Science and Technology Department National Association of Broadcasters 1771 N Street NW Washington, DC 20036

Dear Mr. Wilson,

In response to your recent request for system descriptions for the three USADR IBOC-DAB systems undergoing testing by the EIA, thank you for this opportunity to showcase what USA Digital Radio believes to be the technology of choice for the implementation of DAB in the U.S. as well as worldwide. Please find, enclosed, copies of the most recent versions of the three system descriptions requested. Copies of these documents have also been forwarded to Ralph Justus of the EIA.

Please keep in mind that this material is copyrighted. You are entitled to reprint this information, as necessary for distribution in support of documentation related to the forthcoming NRSC mobile field testing, with the following credit line:

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Please let me know if may assist in any other matter.

Sincerely A.J. Vigil

Engineering Manager

enclosures

USA Digital Radio

AM Band In-Band, On-Channel System

AM IBOC DAB SYSTEM DESCRIPTION

USA DIGITAL RADIO

332 South Michigan Avenue Suite 605 Chicago, Illinois 60604

1.0 INTRODUCTION

USA Digital Radio was founded by broadcasters for the purpose of developing cost-effective backwards-compatible systems for the deliver of in-band on-channel (IBOC) digital audio broadcasting (DAB) in order to introduce compact disc (CD) quality broadcast radio while preserving the infrastructure and investment of the broadcast industry as well as its heritage. The infrastructure and heritage that help constitute broadcasting today is most apparent in the industry's AM segment.

AM offers DAB the often overlooked advantage of favorable and extensive radio propagation characteristics. AM also offers DAB a readily available network of broadcasting facilities, including directional and nondirectional antenna towers and arrays, in place and operational at this time. The potential for delivering IBOC DAB in the AM band is too often dismissed due to potential technical challenges such as limited allocation bandwidth, interference due to market saturation and antenna pattern bandwidth considerations. However, USA Digital Radio has taken the position that both the infrastructure and the heritage of the AM broadcast industry are of a value that justifies meeting the technical challenges necessary to make AM IBOC DAB practical and realizable.

This paper describes the USA Digital Radio AM IBOC DAB system. The system is described in terms of its source encoding, ancillary data capability, forward error correction and its modulation. These are illustrated in the system block diagram shown in Fig. 1. This block diagram shows both the analog signal path, above, and the digital signal path, below.

2.0 SYSTEM DESCRIPTION

2.1 AUDIO SOURCE ENCODING AND ANCILLARY DATA CHANNEL

Key to the realization of IBOC DAB in limited AM band allocations is a powerful source compression algorithm. The audio source encoding scheme for the USADR AM IBOC-DAB system is based on *MUSICAM*^R which is in turn based on the ISO/MPEG Audio Layer II (ISO 11172-3) standard for sub-band encoding. The standard has been advanced through the development of the psychoacoustic model to the point where music may be transcoded at a rate of 96 kbps in order to reproduce 16 bit stereo at a 15 kHz audio bandwidth. The resulting 96 kbps bit stream includes, in addition to compressed music, a 2.4 kbps ancillary data stream. The compression of music to 96 kbps enables broadcasting of DAB over the narrow bandwidth available to the AM-DAB allocation.

2.2 FORWARD ERROR CORRECTION

Forward error correction is realized at rate 3/4. The addition of 32 kbps per second of forward error correction overhead brings the modulation data rate up to 128 kbps. The bit error rate profile in the mobile environment has been found to be bursty in nature. An interleaver of 480 ms duration is used to more evenly distribute error bursts in time.

Data presented in each MUSICAM frame is not protected uniformly. A small degree of coarse prioritization is applied to each MUSICAM compressed data frame. As a consequence, the highest degree of error protection is applied to the MUSICAM header. Successively lower degrees of error protection are applied to the sub-band sample exponents and mantissas. The average rate of error protection is rate 3/4.

2.3 MODULATION

One of the most significant challenges in the implementation of AM IBOC DAB is the transmission of 128 kbps through the limited bandwidth AM allocation. The USADR AM IBOC DAB modulation spectrum is illustrated in Fig. 2. Fig. 2 shows the spectrum of the host analog as well as that of the digital against the FCC spectral allocation mask in the background. Complicating the challenge of high spectral efficiency, as shown in Fig. 2, is the challenge of coexisting with the host analog in the interest of remaining backwards compatible.

Most of the signal power transmitted in the host analog is confined near the AM carrier. In the interest of not interfering with the analog, modulation near the carrier, near the center of the allocation is transmitted in quadrature with the AM carrier. As the frequency becomes removed from

the carrier, the DAB signal transitions from a quadrature signal only into an in-phase and quadrature complex signal. This transition takes place after the point where host analog power has attenuated to a sufficiently low level.

Modulation order is also a function of spectral position. As the DAB signal frequency is removed from the carrier, the effects of interference from the host analog diminish as a consequence of the fact that the host analog power diminishes. As a result, DAB modulation order is increased as the DAB spectrum is removed from the carrier. The highest order DAB modulation is present farthest away from the carrier, in the portions of the allocation where typical AM stations transmit little or no power.

2.4 TRANSMISSION

The USADR AM DAB system uses a commercially available transmitter to combine the IBOC DAB digital signal with the host analog AM. Over the air tests have been conducted through both nondirectional and directional antenna systems.

3.0 CONCLUSION

The USADR AM IBOC DAB system is being developed in the interest of preserving and enhancing the value, the functionality and the heritage of today's existing AM broadcasting infrastructure. The challenges of limited bandwidth availability are being met through powerful source compression algorithms as well as through novel modulation techniques. USADR AM DAB system has been tested through both nondirectional as well as directional antenna systems using a conventional AM transmitter.

The USADR AM IBOC DAB system is under continuing development with the objective of improving mobile performance as well as interference limited performance. However, all indications to date show that AM IBOC DAB is realizable and can be made practical and cost effective.

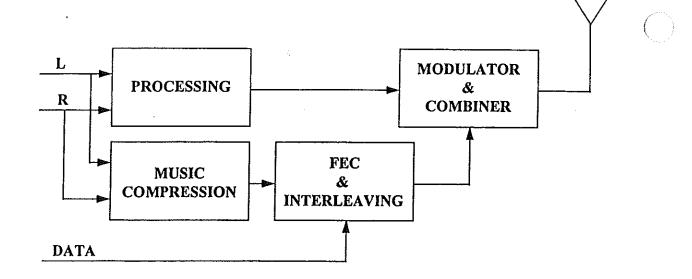


Fig. 1. USADR AM IBOC DAB System Block Diagram.

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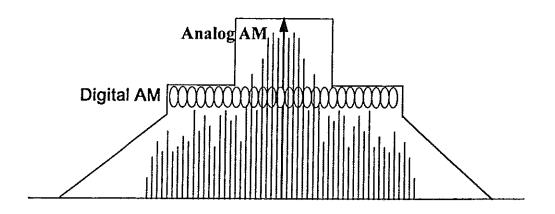


Fig. 2. USADR AM IBOC DAB Modulation Spectrum.



DIGITAL RADIO

USA Digital Radio

FM Band In-Band, On-Channel System (FM-1)

FM-1 SYSTEM DESCRIPTION

USA DIGITAL RADIO

332 South Michigan Avenue Suite 605 Chicago, Illinois 60604

1.0 INTRODUCTION

USA Digital Radio (USADR) has embarked on an aggressive program to develop practical systems for the delivery of digital music and data services which leverage existing broadcaster capital assets. In-Band On-Channel (IBOC) Digital Audio Broadcasting (DAB) is the only approach to DAB endorsed by the US broadcasters. Only IBOC meets the economic, political and regulatory needs of U.S. broadcasters. IBOC allows for simultaneous transmission of new digital with existing analog within a broadcaster's existing spectral allocation.

The fundamental design requirements of the FM-1 system include the introduction of new digital music quality. Audio delivered by FM-1 is equivalent in quality to that delivered by compact disc (CD) which is 16 bit stereo with 20 kHz bandwidth per channel. FM-1 allows for expanded digital data services, with an aggregate data channel of up to 64 kbps. FM-1 employs new technology in the interest of mitigating VHF multipath which is so commonly found in the mobile environment. The deployment of FM-1 transmission systems will be affordable due to the fact that existing broadcasting assets will remain in place as integral parts of the system. The spectral efficiency inherent in the dual use of each allocation greatly simplifies regulatory considerations. Finally, backwards compatibility allows for the smoothest possible transition from analog broadcast radio to digital broadcast radio.

This paper describes the USA Digital Radio FM-1 system. Special focus is given by highlighting those system details in which new technology is applied which yield significant performance advantages. The system is described in terms of its source encoding, ancillary data capability, forward error correction and its modulation. These are illustrated in the system block diagram shown in Fig. 1. This block diagram shows both the analog path, above, and the digital path, below.

2.0 SYSTEM DESCRIPTION

2.1 AUDIO SOURCE ENCODING AND ANCILLARY DATA CHANNEL

The audio source encoding scheme for the USADR FM-1 IBOC-DAB system is based on *MUSICAM*^{*} which is in turn based on the ISO/MPEG Audio Layer II (ISO 11172-3) standard for sub-band encoding. The standard has been advanced through the development of variable rate music coding based on program demand. The standard has also been advance through the implementation of a soft failure mechanism for graceful audio degradation under severely impaired channel conditions. Source coding preserves 16 bit stereo quality over 20 kHz of audio bandwidth.

The audio coding rate for FM-1 IBOC DAB is variable on a frame-by-frame basis as a function of the audio program material and of the ancillary data requirements. The purpose of a continuously variable audio coding rate is to optimally utilize the channel throughput, or the modulation data rate, as a function of the audio encoding requirements, the ancillary data requirements, and the intent of the broadcaster. Data throughput is allocated to audio source coding on an as-needed basis, continuously limiting the coding rate on a frame-by-frame basis, based on the program material, to that needed to reproduce CD quality audio. Continuous conservation of the audio coding rate allows for increased availability of data for ancillary data as well as for forward error correction.

An illustration of variable data rate music coding is shown in Fig. 2. Fig. 2-a shows the spectrum of a complex frame of music. The abundance of music components demands a high audio coding rate. In contrast, Fig. 2-b illustrates a very simple passage of music which demands a much lower data rate to encode its reduced music information content. FM-1 audio source coding rates vary, on a frame-by-frame basis from a minimum of 128 kbps to a maximum of 256 kbps. Ancillary data, which is buffered in and whose transmission rate is also varied on a frame-by-frame basis, may be transmitted at an average rate of up to 64 kbps.

2.2 FORWARD ERROR CORRECTION

Conventional digital audio transmission systems are typically characterized by a very sharply defined failure mode. This "cliff effect" is in contrast to the familiar gradual degradation which takes place in conventional broadcast radio, where a transmission may gradually lose quality while retaining intelligibility. The blending mechanisms incorporated into many mobile FM receivers extend this gradual failure characteristic of analog FM by concealing the effects of multipath-impaired propagation channels on the FM transmission.

The USADR FM-1 IBOC DAB system incorporates a soft failure characteristic for "graceful degradation " that resembles the gradual degradation familiar to analog FM listeners. The use of hierarchical forward error correction and advanced error concealment methods enable graceful degradation to be realized in a digital system.

Hierarchical forward error correction is based on the concept that the information in the coded audio data stream takes on varying degrees of importance. For example, frame header information is critically important because it establishes the location of all of the audio information in the frame. Furthermore, the most significant bits (MSB's) in each audio sample are much more important than the least significant bits (LSB's). The exponent portion of sample values in exponent-mantissa format are even more important than the MSB's of the mantissa. Also, low frequency audio components are usually more important to audio reproduction than higher frequency audio components.

The process of applying hierarchical forward error correction to coded audio involves recognizing the relative importance of the coded information and the subsequent application of commensurate error protection. Following the previous example, the header information in a *MUSICAM*^R frame is assigned the highest level of error protection, followed by the exponents of audio samples and the more significant bits. The less significant bits in the audio samples are assigned lower degrees of error protection. Additionally, low frequency components of audio are protected to a higher degree than the higher frequencies.

The advantage of hierarchical forward error correction in producing a graceful mode of audio degradation is shown in Fig. 3. Fig. 3-a shows the audio spectrum of a clean passage of music. Fig. 3-b shows what may happen in the audio spectrum in the case of constant error protection. Bit errors under impaired channel conditions may affect high frequency components, low frequency components, MSB's, LSB's, audio sample exponents or even header information with equal probability. The result is a catastrophic mode of failure, which results in a randomly scattered loss of audio information and a possible misrepresentation of header information. The listener is subject to either unpleasant artifacts or sudden muting.

In contrast, hierarchical forward error correction employed by the USADR FM-1 IBOC-DAB system yields audio which degrades gradually in the presence of severe channel impairments. The result, shown in Fig. 3-c, is a gradual loss of dynamic range as well as a fold-back in the audio response of the receiver as bit errors are detected in variously relatively lightly protected portions of the coded audio. Graceful degradation is found by the listener to be a much less objectionable failure mode than catastrophic failure.

Hierarchical forward error correction is provided by a variable rate concatenated code. Forward error correction also includes interleaving of 480 ms duration.

2.3 MODULATION

The modulation used in the USADR FM-1 IBOC DAB was expressly designed with multipath in mind. The system designers realized early on that FM-band multipath was significant in both urban and rural terrain-featured environments, particularly in the mobile case. Intersymbol interference (ISI) and frequency selective fading caused by multipath become severe sources of impairment for digital demodulators. The USADR FM-1 IBOC DAB system applies multichannel modulation to the problem of ISI and spread spectrum based techniques to the problem of frequency selective fading.

Multichannel modulation is often used to alleviate ISI caused by multipath. Multiple subchannels allow the baud interval to be increased to an interval much longer than the longest multipath delay. This effectively eliminates ISI at the expense of baud rate. The decrease in baud rate is compensated by allowing for multiple subchannels. The USADR FM-1 IBOC DAB modulation system employs 48 data subchannels, each at a subchannel data rate of 8 kbps, for a total channel data rate of 384 kbps. The baud interval is 125 microseconds. In addition to the 48 data subchannel, a 49th subchannel is transmitted simultaneously as a reference sounding waveform for multipath equalization.

USADR FM-1 IBOC DAB modulation subchannels are not narrowband, as is the case for OFDM, but are wideband, as is the case for spread spectrum. Spread spectrum modulation techniques have been used for some years by the military and are now gaining acceptance in commercial communication systems. Consumer telecommunications applications which employ spread spectrum include cellular telephone, global positioning system (GPS) and wireless local area networks (LAN's). The advantage of spread spectrum techniques for IBOC-DAB modulation is their resistance to multipath. USADR FM-1 IBOC DAB modulation improves upon conventional direct sequence spread spectrum (DS-SS) by introducing a unique signaling waveform design which combines the multipath resistance of DS-SS with a spectral efficiency which approaches two bits per second per Hertz.

The spectrum of FM and USADR FM-1 IBOC DAB signals are shown in Fig. 4. Fig. 4-a depicts a typical conventional analog FM signal spectrum. Fig. 4-b illustrates various USADR FM-1 IBOC DAB subchannels. The subchannel modulation waveforms were designed using a unique method which ensures each subchannel to be wideband, noiselike and orthogonal to every other subchannel. Additionally, a void is maintained in the center of the spectrum of each subchannel waveform to allow FM to survive undisturbed. When observed individually, each of the DAB carrier signals has a spectrum which spans the entire 460 kHz DAB bandwidth with a 220 kHz void centered about the analog FM carrier. Channel modulation data is applied to each FM-1 DAB subchannel using bipolar keying. The spectrum of the DAB composite is shown in Fig. 4-c. The spectrum of a typical combined FM plus DAB signal is shown in Fig. 4-d.

When compared to conventional code division multiple access (CDMA) DS-SS techniques, the USADR FM-1 IBOC DAB signaling set is found to have the unique property that waveform filtering is intrinsically incorporated into the design of the subchannel modulation symbols. Further filtering is unnecessary, avoiding undesirable corruption, due to filter distortion, of signal set properties such as subchannel orthogonality. Mutual orthogonality of subchannel waveforms minimizes, both in clear propagation channels and in multipath, interference due to cross-talk between channels

The wideband nature of the subchannel modulation symbols offers a clear performance advantage in multipath propagation channels. The advantage of wideband data subchannels in multipath over narrowband data subchannels, as is the case in OFDM, is illustrated in Fig. 5. Fig. 5-a illustrates demodulation of a tone or narrowband data subchannel in the case of a clear propagation channel. Demodulation of tone subchannels involves a frequency shift, as shown in Fig. 5-a. Fig. 5-b illustrates the despreading operation involved in demodulating a wideband subchannel. Although the modulation subchannel waveforms and the demodulation processes are different, the same subchannel baseband data stream is recovered in both processes.

Fig. 5-c illustrates demodulation of a tone carrier in a possible multipath environment. In the case of OFDM through multipath, individual subcarriers may be subject to severe attenuation due to spectral nulls in the propagation response introduced by multipath. In the case illustrated in Fig. 5-c, subchannel demodulation effectiveness is limited by multipath-induced localized attenuation in the frequency response of the channel. In contrast, Fig. 5-d shows how wideband subchannel demodulation is affected to a lesser degree due to the fact that any localized attenuation in the frequency response due to multipath only attenuates a small fraction of any wideband subchannel's power.

In addition to wideband data modulation subchannels, a wideband reference waveform is transmitted as part of the USADR FM-1 IBOC DAB modulation waveform. The reference waveform serves a dual purpose. The first is to serve as a sounding pulse for multipath equalization. The reference waveform is recovered at the receiver as a wideband reference used to train the multipath equalizer at the receiver. The second purpose is to serve as a pilot tone for modulation frame synchronization.

The USADR FM-1 IBOC DAB modulation is spectrally efficient. The modulation waveform spectrum is contained well within the guidelines established by rule 73.317 of the FCC code. The USADR FM-1 DAB spectrum consists of two sidebands within the authorized spectral mask. These sidebands begin (-3 dB points) at a 120 kHz offset from either side of the unmodulated analog FM carrier and extend to 220 kHz from the carrier. The amplitude of these sidebands is attenuated by 38 dB with respect to an unmodulated analog carrier in a 1 kHz resolution bandwidth. The total DAB half-power modulation bandwidth is 200 kHz. Total DAB power is -15 dBc, or 1/32 of analog FM power.

3.0 CONCLUSION

The submission of USADR FM-1 for testing to the EIA represents an important step in USA Digital Radio's development of practical IBOC-DAB. Based on the results of FM-1 testing, it is being shown that IBOC-DAB can be used to realize reliable delivery of digital music and data services while leveraging existing broadcast industry infrastructure at a reasonable cost to the broadcaster. Recent developments in the areas of audio source coding, forward error correction and wideband multichannel modulation make IBOC-DAB capable of providing CD quality music through mobile multipath channels.

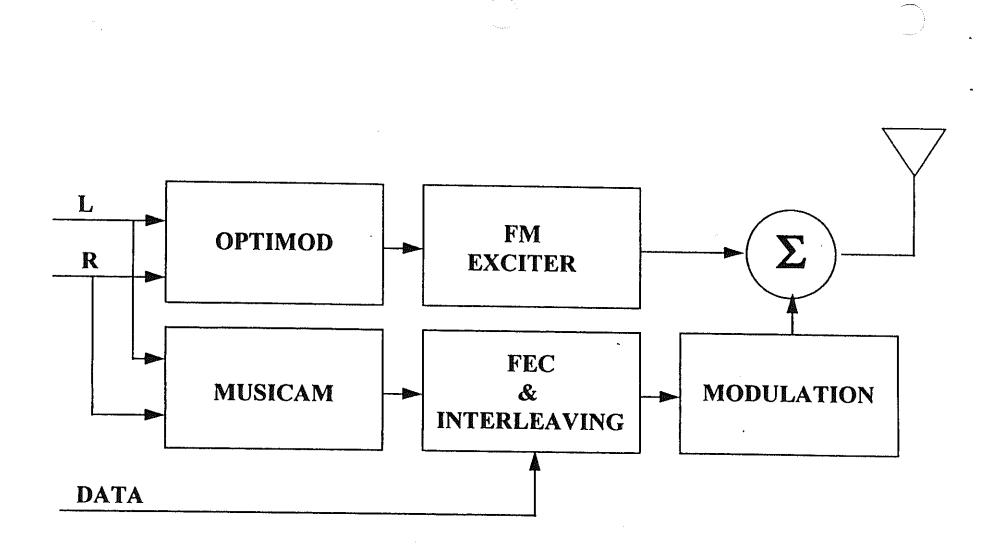
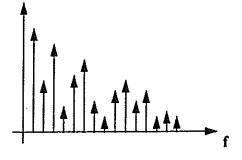


Fig. 1. USADR FM-1 IBOC DAB System Block Diagram.



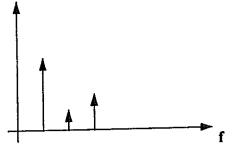


a) High Data Rate Audio Coding

Fig. 2. Variable Data Rate Music Coding.

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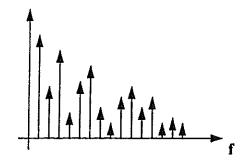




b) Low Data Rate Audio Coding

Fig. 2. Variable Data Rate Music Coding.

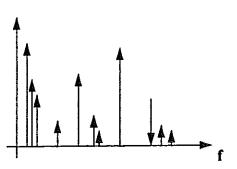




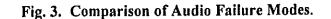
a) Accurate Reproduction

Fig. 3. Comparison of Audio Failure Modes.

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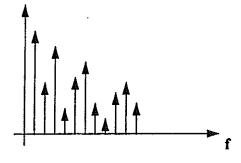


b) Catastrophic Failure





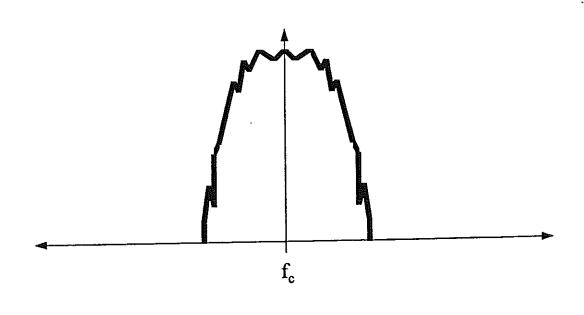
DIGITAL RADIO



c) Graceful Degradation

Fig. 3. Comparison of Audio Failure Modes.

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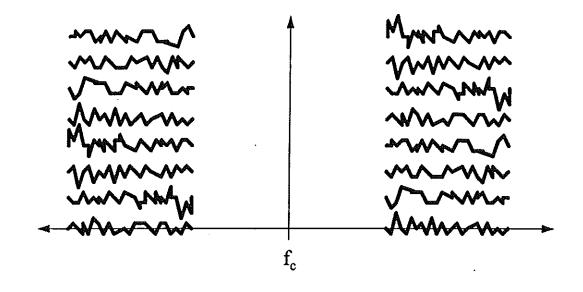


a) FM Spectrum

Fig. 4. USADR FM-1 IBOC DAB Modulation Waveform.



USA DIGITAL RADIO



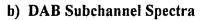
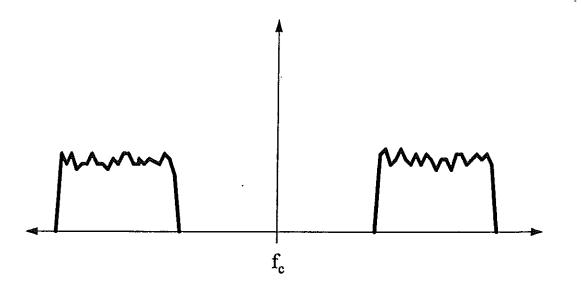


Fig. 4. USADR FM-1 IBOC DAB Modulation Waveform.

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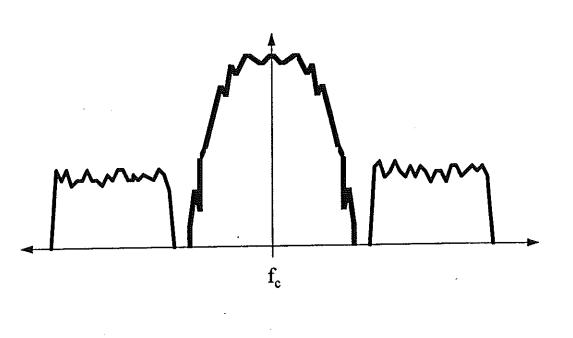


c) DAB Composite Spectrum

Fig. 4. USADR FM-1 IBOC DAB Modulation Waveform.



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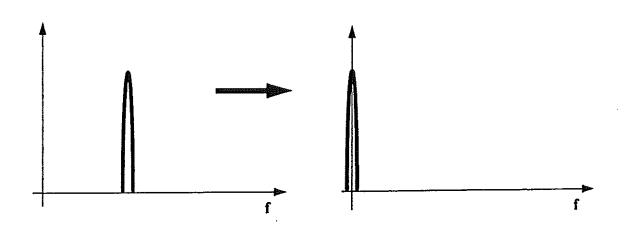


d) Combined FM + DAB Spectrum

Fig. 4. USADR FM-1 IBOC DAB Modulation Waveform.



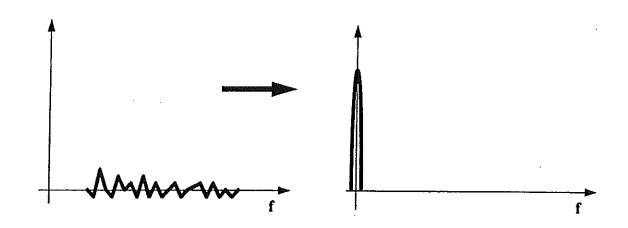
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a) Clear Propagation Channel, Narrowband Data Subchannel

Fig. 5. Demodulation of Narrow and Wideband Data Subchannels in Clear and Multipath Propagation Channels.

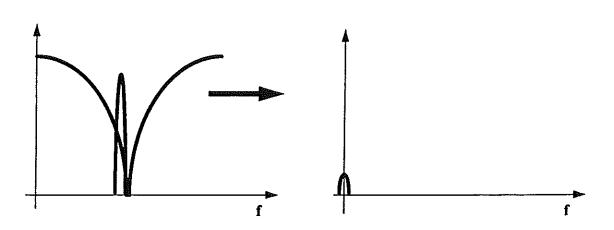




b) Clear Propagation Channel, Wideband Data Subchannel

Fig. 5. Demodulation of Narrow and Wideband Data Subchannels in Clear and Multipath Propagation Channels.

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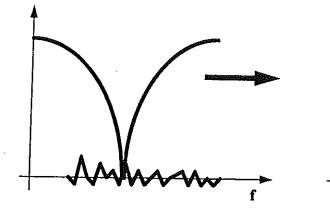


c) Multipath Propagation Channel, Narrowband Data Subchannel

Fig. 5. Demodulation of Narrow and Wideband Data Subchannels in Clear and Multipath Propagation Channels.



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d) Multipath Propagation Channel, Wideband Data Subchannel

Fig. 5. Demodulation of Narrow and Wideband Data Subchannels in Clear and Multipath Propagation Channels.

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f

USA Digital Radio

FM Band In-Band, On-Channel System (FM-2)

FM-2 SYSTEM DESCRIPTION

USA DIGITAL RADIO

332 South Michigan Avenue Suite 605 Chicago, Illinois 60604

1.0 INTRODUCTION

USA Digital Radio (USADR) has considered many different system options and alternatives in the ongoing development and design of In-Band On-Channel (IBOC) Digital Audio Broadcasting (DAB) systems. Due to its excellent performance through multipath and its backwards compatibility with analog, USA Digital Radio has selected its FM-1 as the primary focus of her continuing development of FM-IBOC DAB.

This being the case, the USADR FM-2 system has some very unique characteristics which are of great value to FM-IBOC DAB system. For example, FM-2 IBOC DAB modulation was designed to be completely orthogonal to the host analog FM. The use of a frequency shifting technique maintains this mutual orthogonality while minimizing total bandwidth. The result is complete orthogonality between analog and digital waveforms while minimizing transmission bandwidth. FM-2 also achieves a reduction in bandwidth by incorporating multiple level modulation.

FM-2 is not completely different from FM-1. The modulation techniques of the two systems do address different performance tradeoff objectives. However, the source encoding and forward error correction systems are based on common design objectives.

FM-2 does not offer the high degree of multipath resistance that FM-1 offers. However, the unique characteristics and advantages of FM-2 suggest studying FM-2 and considering its features and advantages. It is possible that at some point in the future, USA Digital Radio may incorporate some of the more favorable features of the FM-2 into her primary FM-1 system.

This paper describes the USA Digital Radio FM-2 system. Special focus is given by highlighting those system details in which new technology is applied to which yield significant performance advantages. The system is described in terms of its source encoding, ancillary data capability, forward error correction and its modulation. These are illustrated in the system block diagram shown in Fig. 1. This block diagram shows both the analog path, above, and the digital path, below.

2.0 SYSTEM DESCRIPTION

2.1 AUDIO SOURCE ENCODING AND ANCILLARY DATA CHANNEL

The audio source encoding scheme for the USADR FM-2 IBOC-DAB system is based on $MUSICAM^R$ which is in turn based on the ISO/MPEG Audio Layer II (ISO 11172-3) standard for sub-band encoding. The standard has been advanced through the development of variable rate music coding based on program demand. The standard has also been advance through the implementation of a soft failure mechanism for graceful audio degradation under severely impaired channel conditions. Source coding preserves 16 bit stereo quality over 20 kHz of audio bandwidth.

The audio coding rate for FM-2 IBOC DAB is variable on a frame-by-frame basis as a function of the audio program material and of the ancillary data requirements. The purpose of a continuously variable audio coding rate is to optimally utilize the channel throughput, or the modulation data rate, as a function of the audio encoding requirements, the ancillary data requirements, and the intent of the broadcaster. Data throughput is allocated to audio source coding on an as-needed basis, continuously limiting the coding rate on a frame-by-frame basis, based on the program material, to that needed to reproduce CD quality audio. Continuous conservation of the audio coding rate allows for increased availability of data for ancillary data as well as for forward error correction.

An illustration of variable data rate music coding is shown in Fig. 2. Fig. 2-a shows the spectrum of a complex frame of music. The abundance of music components demands a high audio coding rate. In contrast, Fig. 2-b illustrates a very simple passage of music which demands a much lower data rate to encode its reduced music information content. FM-2 audio source coding rates vary, on a frame-by-frame basis from a minimum of 128 kbps to a maximum of 256 kbps. Ancillary data, which is buffered in and whose transmission rate is also varied on a frame-by-frame basis, may be transmitted at an average rate of up to 64 kbps.

2.2 FORWARD ERROR CORRECTION

Conventional digital audio transmission systems are typically characterized by a very sharply defined failure mode. This "cliff effect" is in contrast to the familiar gradual degradation which takes place in conventional broadcast radio, where a transmission may gradually lose quality while retaining intelligibility. The blending mechanisms incorporated into many mobile FM receivers extend this gradual failure characteristic of analog FM by concealing the effects of multipath-impaired propagation channels on the FM transmission.

The USADR FM-2 IBOC DAB system incorporates a soft failure characteristic for "graceful degradation " that resembles the gradual degradation familiar to analog FM listeners. The use of hierarchical forward error correction and advanced error concealment methods enable graceful degradation to be realized in a digital system.

Hierarchical forward error correction is based on the concept that the information in the coded audio data stream takes on varying degrees of importance. For example, frame header information is critically important because it establishes the location of all of the audio information in the frame. Furthermore, the most significant bits (MSB's) in each audio sample are much more important than the least significant bits (LSB's). The exponent portion of sample values in exponent-mantissa format are even more important than the MSB's of the mantissa. Also, low frequency audio components are usually more important to audio reproduction than higher frequency audio components.

The process of applying hierarchical forward error correction to coded audio involves recognizing the relative importance of the coded information and the subsequent application of commensurate error protection. Following the previous example, the header information in a *MUSICAM^R* frame is assigned the highest level of error protection, followed by the exponents of audio samples and the more significant bits. The less significant bits in the audio samples are assigned lower degrees of error protection. Additionally, low frequency components of audio are protected to a higher degree than the higher frequencies.

The advantage of hierarchical forward error correction in producing a graceful mode of audio degradation is shown in Fig. 3. Fig. 3-a shows the audio spectrum of a clean passage of music. Fig. 3-b shows what may happen in the audio spectrum in the case of constant error protection. Bit errors under impaired channel conditions may affect high frequency components, low frequency components, MSB's, LSB's, audio sample exponents or even header information with equal probability. The result is a catastrophic mode of failure, which results in a randomly scattered loss of audio information and a possible misrepresentation of header information. The listener is subject to either unpleasant artifacts or sudden muting.

In contrast, hierarchical forward error correction employed by the USADR FM-2 IBOC-DAB system yields audio which degrades gradually in the presence of severe channel impairments. The result, shown in Fig. 3-c, is a gradual loss of dynamic range as well as a fold-back in the audio response of the receiver as bit errors are detected in variously relatively lightly protected portions of the coded audio. Graceful degradation is found by the listener to be a much less objectionable failure mode than catastrophic failure.

Hierarchical forward error correction is provided by a variable rate concatenated code. Forward error correction also includes interleaving of 480 ms duration.

2.3 MODULATION

The objective of preserving existing analog signal quality was first and foremost in the development of the modulation used in the USADR FM-2 IBOC DAB system. The most carefully planned feature of USADR FM-2 IBOC DAB modulation is the carefully planned, constant invariable orthogonality between the analog and the digital waveforms. The second important objective of the FM-2 system is spectral containment. The third objective is multipath resistance.

Mutual orthogonality between analog and digital is achieved through a frequency slide technique where the center frequency of the digital modulation is continuously adjusted to follow the instantaneous frequency of the host FM waveform. While the spectra of the analog and digital waveforms overlap, the signals never occupy the same instantaneous frequency. As the center frequency of the digital modulation waveform is swept, the resulting signal spectrum becomes spread in relation to the host FM frequency deviation. This manner is similar to the manner in which an FM signal spectrum is spread in spite of the fact that the FM signal is a CW sinusoid with only one instantaneous frequency at any given instant in time.

Although the spectrum of the digital modulation waveform is spread as a result of its varying instantaneous frequency, spectral containment is achieved due to the close proximity with which the digital modulation is placed to the instantaneous frequency of the host analog. If frequency deviation were removed from the host analog signal, there would be no discernible frequency separation between the resulting analog and digital signal spectra.

Multipath resistance is achieved through the use of multiple wideband subchannels. Multichannel modulation is often used to alleviate ISI caused by multipath. Multiple subchannels allow the baud interval to be increased to an interval much longer than the longest multipath delay. This effectively eliminates ISI at the expense of baud rate. The decrease in baud rate is compensated by allowing for multiple subchannels. The USADR FM-2 IBOC DAB modulation system employs 64 data subchannels, each at a subchannel baud rate of 2 kilobaud per second, for a total symbol rate of 2 kilobaud per second. The baud interval is 500 microseconds. Data is applied using 8 level amplitude shift key modulation for a modulation rate of 3 bits per symbol per subchannel or a total data modulation rate of 384 kbps.

Like USADR FM-1, USADR FM-2 IBOC DAB modulation subchannels are not narrowband, as is the case for OFDM, but are wideband, as is the case for spread spectrum. USADR FM-2 IBOC DAB modulation applies direct sequence spread spectrum (DS-SS) through the use of bipolar pseudonoise sequence subchannel modulation waveforms. Spread spectrum techniques such as DS-SS are recognized for their advantages in multipath propagation channels. The USADR FM-2 IBOC DAB subchannels consist of 64 mutually orthogonal bipolar pseudonoise (PN) sequences. The FM-2 modulation composite is filtered to an absolute baseband bandwidth of 125 kHz.

The spectrum and composition of FM and of USADR FM-2 IBOC DAB signals are illustrated in Fig. 4. Fig. 4-a depicts a typical conventional analog FM signal spectrum. Fig. 4-b gives an "instantaneous frequency representation" of an FM signal. If an FM waveform is frozen in time, it consists of a sinusoid at a single instantaneous frequency at that moment. That instantaneous frequency can be increasing, decreasing or remaining unchanged at that moment, depending on the FM composite program at that moment.

Fig. 4-c illustrates the spectrum of the USADR FM-2 IBOC DAB modulation composite. The USADR FM-2 IBOC DAB modulation composite consists of the sum of 64 orthogonal PN sequences, each modulated to 8 levels at a 2 kHz rate, the sum being filtered to an absolute bandwidth of plus and minus 125 kHz. In Fig. 4-c, the USADR FM-2 IBOC-DAB modulation composite is shown centered at a stationary center frequency. It should be noted that of the 64 orthogonal PN sequences comprising the modulation set, none have a DC component. As a result, although the spectrum of the USADR FM-2 IBOC-DAB modulation composite is shown to be continuous, in actuality there is no power at the precise center frequency.

Fig 4-d gives an instantaneous frequency representation of the USADR FM-2 IBOC DAB modulation composite subject to frequency shifting. When the FM-2 digital modulation center frequency is shifted to align with that of the host analog, a "snapshot" in the frequency domain consists of the FM-2 digital modulation composite centered at the instantaneous frequency of the host analog at that moment. Its instantaneous frequency may be rising, falling or remaining stationary. Fig. 4-e shows the same instantaneous frequency shifting, but with the host analog added. Fig. 4-e shows how FM-2 digital is shifted in center frequency so as to follow the instantaneous frequency of the host analog FM. Over any measurable length of time, the analog FM composite program causes both the host analog FM and the FM-2 digital to shift in time, spreading their power over a wider spectrum, as shown in Fig. 4-f. Fig. 4-f illustrates a typical FM-2 FM plus digital modulation spectrum.

3.0 CONCLUSION

The USADR FM-2 IBOC DAB system was designed to provide a spectrally efficient means of broadcasting CD quality audio and ancillary data with some degree of multipath suppression while remaining constantly and invariably orthogonal to the host analog FM within its existing spectral allocation. A novel frequency sliding technique is used to cause the center frequency of the digital FM-2 modulation waveform to follow the instantaneous frequency of the host analog in order to maintain orthogonality mutual orthogonality between the FM-2 digital and the host analog. Spectral confinement is achieved by applying 8 level digital modulation as well as by filtering the PN sequence based digital modulation. Further spectral confinement is achieved through the frequency shifting technique allows the digital to be placed in very close proximity to the analog.

The USADR FM-2 IBOC DAB system offers some multipath suppression capability due to the application of DS-SS techniques in the design of the multichannel PN sequence modulation. However, the primary attribute of the USADR FM-2 IBOC DAB system is the degree to which backwards compatibility and noninterference are ensured through the mutual orthogonality of analog and digital waveforms.

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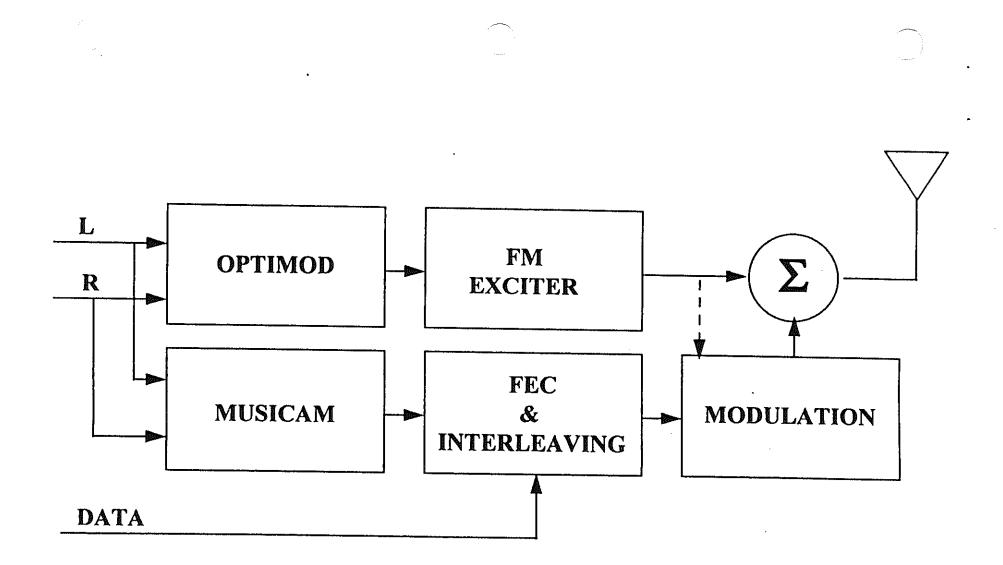
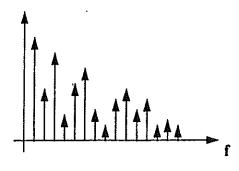


Fig. 1. USADR FM-2 IBOC DAB System Block Diagram.



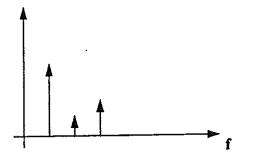


a) High Data Rate Audio Coding

Fig. 2. Variable Data Rate Music Coding.

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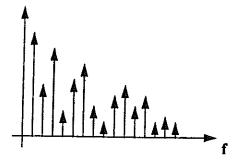




b) Low Data Rate Audio Coding

Fig. 2. Variable Data Rate Music Coding.



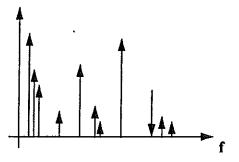


a) Accurate Reproduction

Fig. 3. Comparison of Audio Failure Modes.

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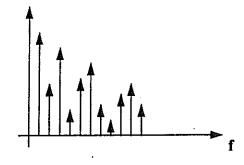




b) Catastrophic Failure

Fig. 3. Comparison of Audio Failure Modes.

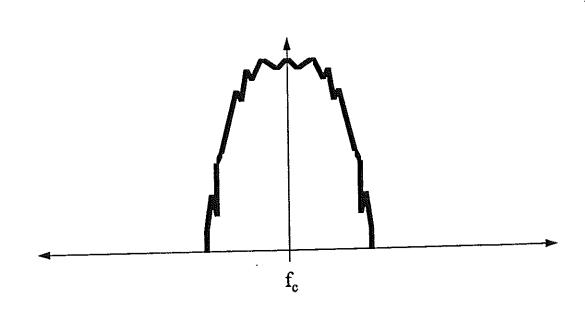




c) Graceful Degradation

Fig. 3. Comparison of Audio Failure Modes.

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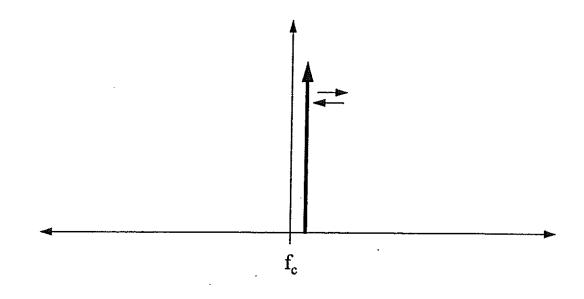


a) FM, Spectral Representation

Fig. 4. USADR FM-2 IBOC DAB Modulation Waveform.



USA DIGITAL RADIO

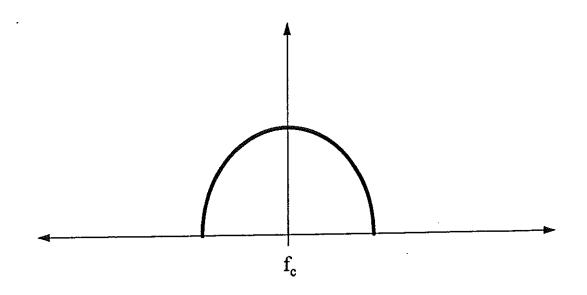


b) FM, Instantaneous Frequency Representation

Fig. 4. USADR FM-2 IBOC DAB Modulation Waveform.

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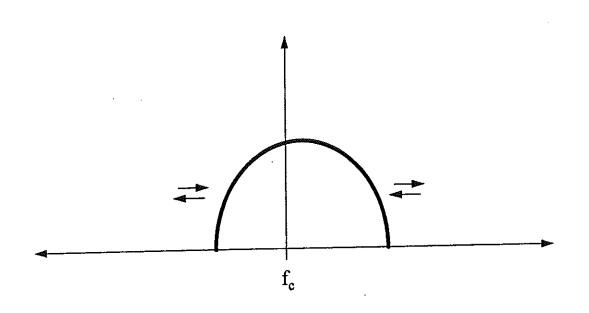




c) Stationary DAB, Spectral Representation

Fig. 4. USADR FM-2 IBOC DAB Modulation Waveform.

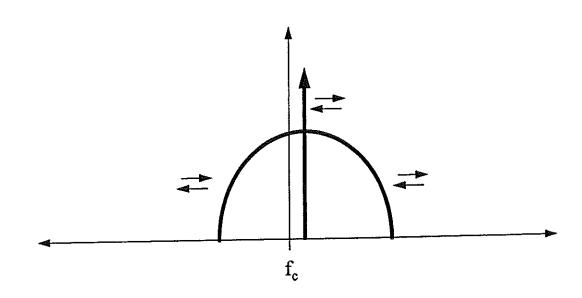




d) DAB, Instantaneous Frequency Representation

Fig. 4. USADR FM-2 IBOC DAB Modulation Waveform.

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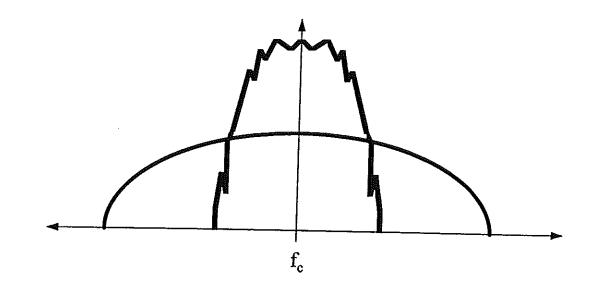


e) FM + DAB, Instantaneous Frequency Representation

Fig. 4. USADR FM-2 IBOC DAB Modulation Waveform.



DIO



f) FM + DAB, Spectral Representation

Fig. 4. USADR FM-2 IBOC DAB Modulation Waveform.



Voice of America / Jet Propulsion Laboratory

S-Band System

July 17, 1995

ANNEX

DIGITAL SYSTEM B

Introduction

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Digital Sound Broadcasting System B is a flexible, bandwidth and power-efficient system for providing digital audio and data broadcasting, for reception by indoor/outdoor, fixed and portable, and mobile receivers. System B is designed for satellite, terrestrial, as well as hybrid broadcasting systems and is suitable for use in any broadcasting band.

System B allows a flexible multiplex of digitized audio and data sources to be modulated onto each carrier. This, together with a range of possible transmission rates, results in an efficient match between service provider requirements and transmitter power and bandwidth resources.

The System B receiver design is modular. A standard core receiver design provides the necessary capability for fixed and portable reception. This design is based on standard, well proven signal processing techniques for which low cost integrated circuits have been developed. Mitigation techniques, which are generally needed for mobile reception, are implemented as add-on processing functions.

In satellite broadcasting, the main impairment is signal blockage by buildings, trees, and other obstacles. Signal blockage produces very deep signal fades and it is generally not possible to completely compensate for it through link margin. Several mitigation techniques were developed or adapted during the design of the System B receiver. The System B receiver can support each of the following:

- Time Diversity (Data Retransmission) A delayed version of the data stream is multiplexed together with the original data and transmitter on the same carrier
- Reception Diversity (Antenna/Receiver Diversity) Two physically separated antennas/receivers receive and process the same signal
- Transmission Diversity (Satellite/Transmitter Diversity) The same data stream is transmitted by two physically separate transmitters on separate frequencies, are received by the same antenna, then processed independently
- On-Channel Boosters (Single Frequency Network) The same data stream is transmitted by two or more physically separate transmitters on the same frequency, then the composite received signal is processed by an equalizer

In a terrestrial system with several on-channel transmitters, as well as in a satellite system with terrestrial on-channel boosters, System B will use equalization in the receiver. This is the only time the core receiver configuration is impacted. If a receiver does not perform equalization, it must have the capability to recognize and discard the training symbols which have been inserted into the data stream.

2 System Overview

An overview of the System B design can be best obtained by examining the functional block diagram of the receiver (starting at the IF) presented in Figure 1. Core receiver functions are shown as solid blocks, while the optional functions for performing mitigation of propagation problems are shown as dashed blocks.

After the desired carrier is selected by the receiver tuning section, the signal is translated down to a fixed IF frequency.

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In the core receiver, carrier reconstruction takes place in a QPSK Costas loop, and symbols are detected by a matched filter with timing provided by a symbol tracking loop. After frame sync is established, the recovered symbols are decoded and demultiplexed. The Reed Solomon decoder performs the additional function of marking data blocks which were not successfully decoded. This information is used by the audio decoder and can be used by the time or signal diversity combiner, if implemented in the receiver.

The selected digital audio source data is provided to the audio decoder while other digital data is provided to the appropriate data interfaces. Each audio encoder will have the capability of multiplexing asynchronous, program related data, with the audio data stream as shown in the figure.

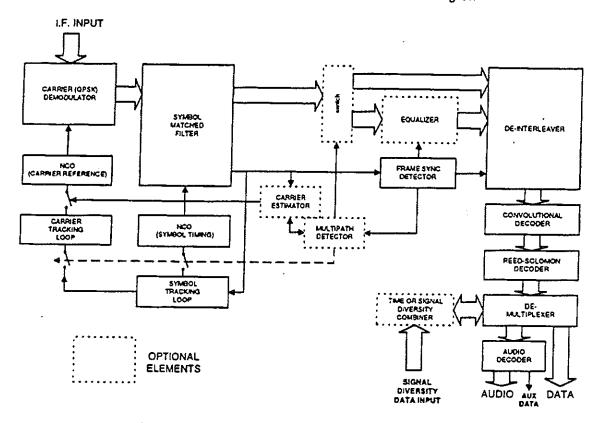


Figure 1. Receiver Functional Block Diagram

In a receiver equipped with an equalizer, the equalization can be disabled in the absence of multipath because the equalizer will introduce a nominal amount of performance degradation.

The presence of multipath can be detected automatically or the equalizer can be switched in manually if the receiver is to be operated in an area served by terrestrial transmitters. When the equalizer is operating, the carrier and symbol tracking loops are opened.

Time diversity is implemented by transmitting a delayed version of a data stream multiplexed together with the original. In the receiver, these two data streams are demultiplexed and time realigned. The data stream with the fewest errors is selected for output.

Signal diversity requires the independent processing of the signal, or of different frequency signals, up to the diversity combiner. The diversity combiner then performs the functions of time alignment and selection of the most error free data stream.

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3 System Description

The processing layers of the System B transmitter and receiver are described block by block, referenced to the diagram of Figure 2. Specifications are defined for each block as appropriate.

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3.1 Transmitter

The transmitter performs all the processing functions needed to generate a single RF carrier. The process includes multiplexing all analog audio and digital data sources to be combined onto one carrier, forward error correction encoding, and QPSK modulation.

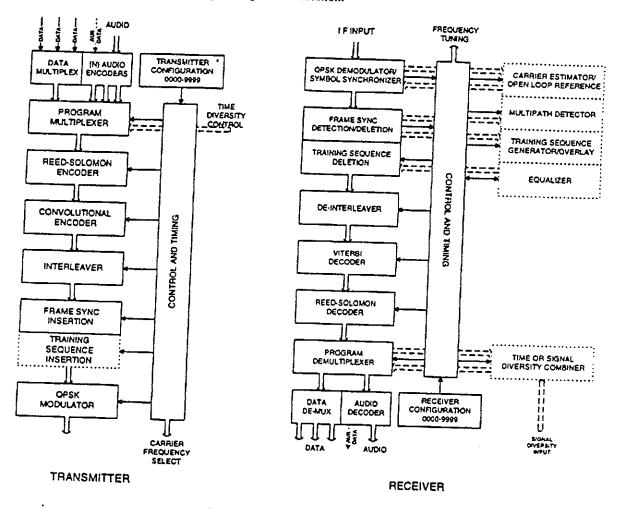


Figure 2. System B Block Diagram

3.1.1 Input Interfaces

The transmitter accepts a set of sampled analog audio signals, a set of asynchronous data sources associated with each audio source, and a set of independent synchronous data sources.

3.1.2 Audio Encoding

A number of audio encoders are provided to handle the required number of limited bandwidth monaural, limited and full bandwidth stereo, and full bandwidth five channel surround sound channels.

Each encoder also accepts an asynchronous data channel, which is multiplexed with the audio data stream. The data rate of these channels varies dynamically according to the unused capacity of the audio channel.

The output of each audio encoder is a synchronous data stream with a data rate proportional to the audio bandwidth and quality. The rate ranges from a minimum of 16 kbps for limited bandwidth monaural, to approximately 320 kbps for five channel (exact rate to be determined by MPEG committee). Audio encoder data rates are limited to multiples of 16 kbps.

3.1.3 Program Multiplexing

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All digitized audio channels and data channels are multiplexed into a composite serial data stream. The output data rate will range from a minimum of 32 kbps to a maximum determined by the transmitting system bandwidth and power resources. This maximum is anticipated to be in the range of 1 Mbps to 10 Mbps.

Each allowed multiplex combination of audio sources and their rates, as well as data sources and their rates, will be assigned a unique transmission identifier number. This number will be used by the receiver to set up the data rate and demultiplexing configuration.

3.1.4 Error Correction Encoding

Error correction encoding of the composite data stream consists of rate 1/2, k=7 convolutional encoding, followed by rate 140/160 Reed Solomon encoding.

3.1.5 Interleaving

A block interleaver is used to time interleave the composite data stream. The interleaver block length will be proportional to the composite data rate to provide an interleaver frame time on the order of 200 milliseconds at any data rate.

3.1.6 Frame Synchronization

A PN code word is inserted at the beginning of each interleaver frame. The interleaver frame sync will also have a unique relationship with the program multiplexer frame.

3.1.7 Training Sequence Insertion

If the broadcast is to be received in an environment with on-channel repeaters, a known training symbol sequence will be inserted, with a training symbol placed every n data symbols, where n can range from 2 to 4. The presence of training symbols and their frequency will be also identified by the unique transmission identifier number.

3.1.7 Modulation

The final step in the process is QPSK modulation at an IF frequency. Pulse shaping will be used to constrain the bandwidth of the signal. From this point the modulated IF signal is translated to the appropriate carrier frequency for transmission. In a frequency division multiplex (FDM) approach, additional carriers are generated by duplicating the transmitter described above.

3.2 Receiver

After tuning to the desired carrier and translating the signal down to a fixed IF frequency, the receiver will perform the demodulation, decoding, and demultiplexing functions, as well as the digital to analog conversion of the selected audio signal.

4

The receiver data rate and program demultiplex configuration will be set up by inserting the unique transmission identifier number. The core receiver will be able to perform all required receive functions in a fixed or portable reception environment, where there is a stable signal with sufficient signal to noise ratio.

In mobile reception environments, where there are sufficient problems with signal blockage, the receiver will include the enhancements needed to accommodate time or signal diversity, or equalization if boosters are used.

3.2.1 Demodulation

Normally carrier demodulation takes place in a phase locked coherent QPSK demodulator, and symbols are detected by a matched filter with timing provided by a symbol tracking loop.

When equalization is used in the presence of echoes, the carrier and symbol tracking loops are opened. A FFT frequency estimator is used to set a fixed carrier demodulation reference. The symbol matched filter is sampled at twice the symbol rate and these samples are provided to the equalizer.

3.2.2 Frame Synchronization

Interleaver frame synchronization is established through cross-correlation detection of the unique frame sync word. This process also removes the ambiguity produced by QPSK modulation.

3.2.3 Equalization

In the presence of echoes, there will be several closely spaced correlation peaks in the frame sync detector output. This information can be used to automatically switch in the equalizer. The equalizer uses a locally generated training sequence whose start is based on an estimate of the position of frame sync word. A comparison of the timing of the locally generated frame sync word and the frame sync detector output allows the equalizer to adjust for any timing error between the incoming symbols and locally generated symbol timing reference.

System B uses a Lattice Predictive Decision Feedback Equalizer (Lattice PDFE) design. The leeway allowed in the time spread of all the echoes is a function of the length of the equalizer. System B performance testing employed an equalizer with 22 forward taps and 4 feed back taps. The equalizer will acquire within 100 symbol times. Equalizer length can be increased if it is necessary to compensate for greater signal delay spread.

3.2.4 Training Sequence Deletion

At the output of the equalizer, the training sequence symbols are discarded. If a receiver without an equalizer works with a signal that contains training symbols, it also must discard these symbols. This is a simple process since the position of the training symbols is known in relation to the frame sync word.

3.2.5 De-Interleaving

The deinterleaver reestablishes the original time sequence of the detected symbols, as it existed in the transmitter prior to interleaving.

3.2.6 Error Correction Decoding

A Viterbi decoder, followed by a Reed Solomon decoder, reduces the detected symbol error rate and converts the symbols back into data bits. If the Reed Solomon decoder is unable to remove all the errors in a data block, it marks the data block as bad. This indication can later be used by the diversity combiner to select the better signal, as well as by the audio decoder to control audio output squelching. .

3.2.6 Program Demultiplexing

At this point the composite data stream is demultiplexed into separate digital data streams and the desired audio data stream is selected and routed to the audio decoder.

If time diversity is used, the program demultiplexer separates the real time and delayed version of the data stream, and sends them to the diversity combiner for selection of the least corrupted data.

If an independent receiver is used for diversity reception, this is the point where the more robust output data is selected.

3.2.7 Audio Decoding

The audio decoder converts the selected digital audio channel to analog. It also demultiplexes the auxiliary data channel and provides the data to the appropriate output interface.

3.2.8 Output Interfaces

Output interfaces consist of the selected audio channel and selected data channels. The data channels can drive displays in the receiver, or be routed to special purpose displays in data casting applications. Since more than one audio channel may exist in a transmission multiplex, the channels not selected for listening can be recorded for later playback.

4 Performance

The performance of System B is referenced to a set of standardized channel models: an additive white Gaussian noise (AWGN) channel; a satellite model for a single satellite signal; and a multiple (single frequency) signal model which can represent a satellite signal with terrestrial boosters or a purely terrestrial network.

4.1 AWGN Channel

A clear line-of-sight satellite link can be approximated with a AWGN channel. There is very little multipath (Rician k factors generally below 10 dB) at satellite elevation angles above 20 degrees. The measured performance of a System B receiver over a AWGN channel is shown in Figure 3. Also shown are some comparisons between theory, simulation, and measurement results.

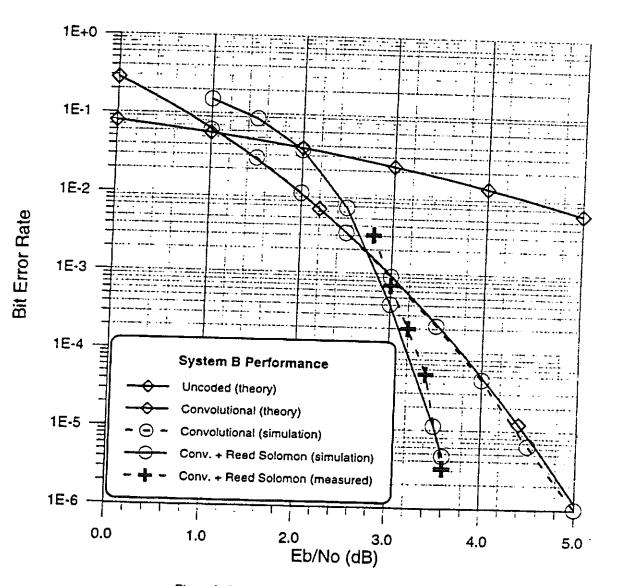


Figure 3. System B Performance in AWGN

Since System B can use several independent carriers in a FDM mode, carrier spacing is of interest. Figure 4 shows the measured performance degradation as a function of adjacent carrier spacing. Spacing is given as a ratio of carrier separation in Hz, to transmitted symbol rate in symbols per second. In System B the symbol rate is equal to the data rate times the Reed Solomon overhead of 160/140, times the training symbol overhead.

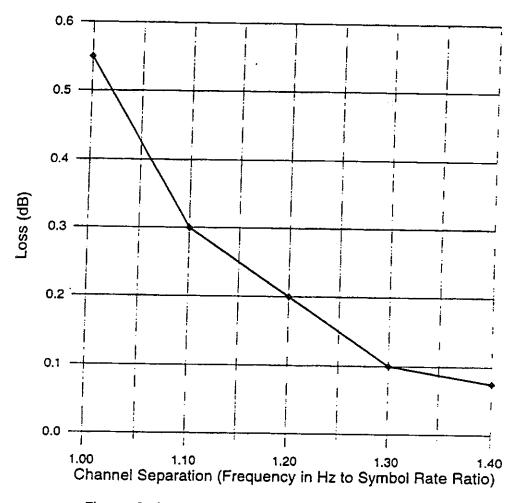


Figure 4. Performance Degradation as a function of Carrier Spacing

4.2 Satellite Channel

The satellite channel changes for mobile reception because the satellite signal is randomly blocked by buildings, trees, and other obstacles. In order to evaluate System B performance under mobile reception conditions, a model was established through a satellite signal measurement over a specific test course in the Pasadena, California area. The test course takes 45 minutes to cover and includes a variety of reception conditions, including open, moderately shadowed, and severely shadowed segments. The satellite signal measurement was a narrow band measurement which yielded a dynamic range of over 35 dB. A time plot of the model is shown in Figure 5. Figure 6 summarizes the statistics of the signal measurement.

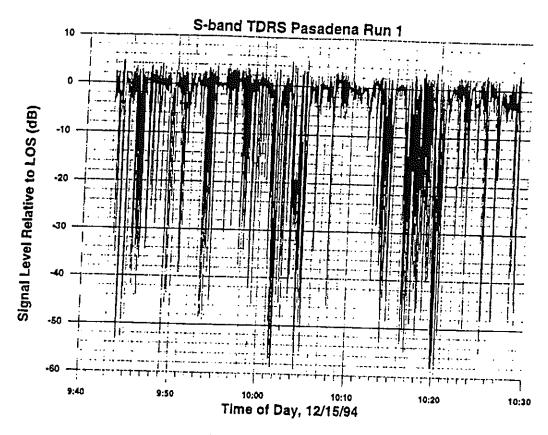


Figure 5. Satellite Channel Model

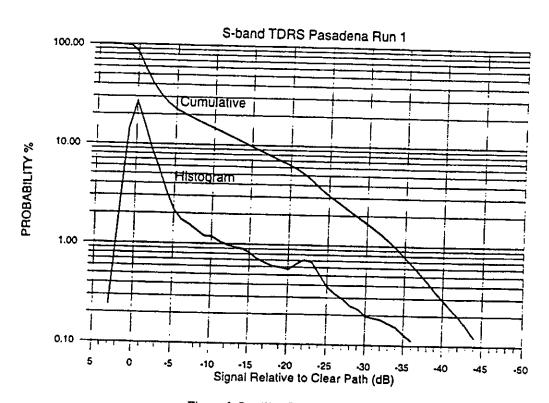
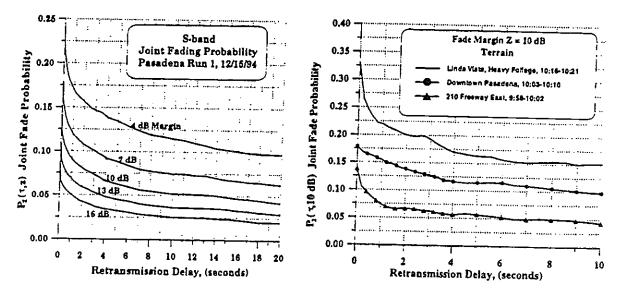


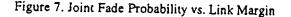
Figure 6. Satellite Channel Model Statistics

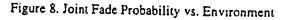
4.2.1 Time Diversity

If only a single satellite signal is available, an effective mitigation technique is time diversity. A delayed version of a data stream is multiplexed with the original data stream, with the expectation that at least one version will not be blocked. The receiver realigns the two data streams in time and selects the one with the fewest errors. This can be done on the basis of the Reed Solomon decoder error indication.

Retransmission of the data stream adds a 3 dB penalty to the system, however it can be shown that this is more effective than a 3 dB increase in link margin. Figures 7 and 8 show the effectiveness of time diversity, using the Pasadena channel model. Figure 7 shows the joint probability of a fade exceeding a range of link margins, averaged over all the model reception conditions. Note that most of the improvement occurs within about 4 seconds of delay. Figure 8 shows the joint fade probabilities, for a fixed 10 dB margin, separated by different reception conditions.







4.2.2 Satellite Diversity

More than one satellite can be used to broadcast the same data stream, using separate frequencies and separate receivers for each signal. The expectation with this technique is that at least one of the signals will not be blocked because of the difference in direction from the receiver to the satellites.

The effectiveness of satellite diversity, as with time diversity, depends on the local geometry of the obstacles producing the signal blockage. Photogrammetric techniques have recently been applied to obtain the statistics on the effectiveness of satellite diversity. These techniques involve taking photographic images with a fish eye lens camera pointed at zenith, then analyzing them to determine the percentage of sky that is clear, shadowed, or blocked. Satellite position can be overlaid on these images to give an assessment of diversity gain over a specific location or path.

4.3 Single Frequency Network

A method for getting a satellite signal into very difficult reception areas is to use a network of onchannel terrestrial retransmitters. System B uses equalization to work in this signal environment. The only restriction in the use of equalization is that each signal is delayed at least one half symbol from every other. There is no restriction as to how close boosters are to each other if different delays are incorporated in each one The maximum delay between boosters will be set by the number of stages incorporated into the equalizer.

4.3.1 Channel Models

Two signal models were set up to evaluate the performance of the System B equalizer. In addition, the effectiveness of signal reception diversity was evaluated.

The first is a Rician model, with one half the power in a direct signal component, and one quarter of the power in each of two Rayleigh components. The Doppler spread on the Rayleigh components was set to +/-213 Hz, which corresponds to a vehicle speed of 100 km/hr, at a carrier frequency of 2.3 GHz. The transmission rate is 300,000 symbols per second. Eb/No is defined on the basis of total signal power and includes the effect of the training sequence overhead.

The second is a Rayleigh model, with three equal power Rayleigh signal components.

4.3.2 Equalizer Performance

Initial trade-offs and performance evaluation was accomplished using a "short-cut" simulation approach that assumed signal time separation in integral symbols times and perfect symbol timing recovery. The results are shown in Figure 9. The bit error rate is uncoded error rate before the Minute in the second state is uncoded error rate.



Solomon decoding. An uncoded error rate of 1 in 10E2 will be reduced to 1 in 10E6 by the decoding process.

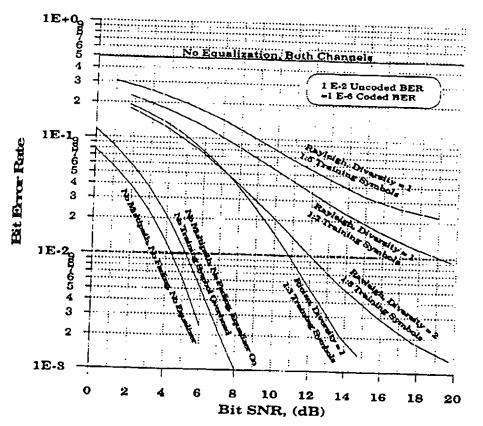


Figure 9. System B Ideal Equalizer Performance

Figure 10 shows performance obtained with full scale simulation, including open loop operation of the carrier demodulation and symbol timing loops.

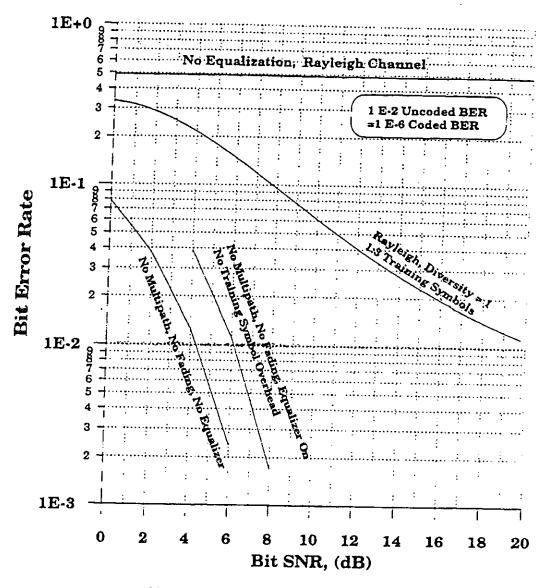


Figure 10. System B Equalizer Performance

APPENDIX B

Unified DAR Laboratory Test Procedures

DAR Laboratory Test Procedures

JULY 1995

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The Laboratory RF Transmission Tests

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EVALUATION

Α.	Calibration			
	1 Daily Check Systom Dover			
	2. Daily System Spectrum Plot	all least		
	3. Daily POF, TUA w Gauges Topsnew	and - And		
	2. Daily System Spectrum Plot 3. Daily POF, TOA & Gaussian No.5(+ sign of 4. Daily audio recording 5. Weekly analog channel proof	1. Al		
	5. Weekly analog channel proof	Host	IBOC	
	6. Reference analog TX proof			
	7. Weekly calibrate modulation monitors			
	8. Proponent self check			
	9. Monthly calibrate test bed			
в.	Signal Failure Characterization			
	1. Noise	- Subjective	- Lab EO&C	
	2. Co-Channel			
	3. Multipath and Noise		**	
	3a. AM -> DAR first	**	**	
~				
с.	DAR Performance with Impairment	15. E		
	1. Impulse Noise 2. CW	- Lab EO&C		
		2		
	3. Airplane Flutter	1 11		
	4. Weak Signal Failure 5. Delay Spread/doppler			
	6. Additional Multipath with Noise			
	7. Environmental Noise (AM band)			
	(AM Dand)			
D.	DAR -> DAR with no other Impairments:	Teb Boss		7
	Co-Channel, First, and Second Adjacent	- Lab EO&C		
1 1	and becond Adjacent			Compatibility
E.	DAR -> DAR with Multipath: Co-Channel,	- Lab EO&C		
	First, and Second Adjacent	Tap FowC		
1				
F.	DAR -> Analog no other Impairment:	- Objective		
	Co-Channel, First, and Second Adjacent	- Subjective ((analog) EOSC	
1			(unalog) Eode	
G.	DAR -> Analog with Multipath on FM:	- Objective		
	Co-Channel, First, and Second Adjacent	- Subjective ((analog) EO&C	
н.				
п.	Analog -> DAR no other Impairment:	- EO&C in Lab	/	
	Co-Channel, First, and Second Adjacent		1	
Ι.	Analog -> DAR with Multipathe de channel			
	Analog -> DAR with Multipath: Co-Channel, First, and Second Adjacent	- EO&C in Lab		
~	List, and becond Adjacent			
J.	Reacquisition (Hysteresis)			
	1. Failure due to simulated weak signal	- EO&C in Lab		
	2. Failure due to multipath	" LOQC IN LAD		
К.	Transmission Quality			
	1. Test materials selection	- Subjective		
	2. Transmission quality	- Subjective -	Lab EO&C	
-		2		
$\mathbf{L}_{\mathbf{r}}$	IBOC -> Host Analog			
	1. Proof of Analog channel	- Objective	TBO	C
	2. Interference to host Analog		11	
	3. Interference to host Analog	- Subjective -	Lab EO&C Text	4
	4. Interference with multipath	**	"	
м.	Host Analog -> IBOC			
	1. Host Analog to DAR	Tab Boas		
	2. Host Analog to DAR with multipath	- Lab EO&C		
	a second of the second se			
N	Multiple Spurious		-I -RAI	
	1. DAR + $FM \rightarrow FM$	- Lab EO&C	TOAC	
		Las Bout] IBAC Test	
0.	Outline of DAR/Subcarrier Compatibility		100	
	Tests			

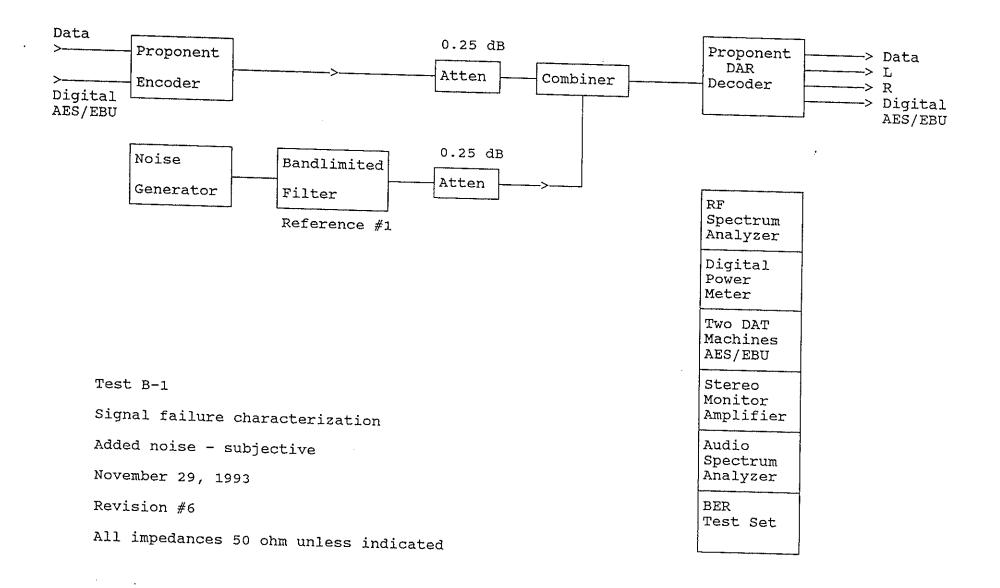
Test Group	V #9A Septer Test &	OAN IED DAR LABORATORY	ESTS					<u></u>		
 A	Impairment 1 Power	TEST PROCEDURE Note: The audio impairment test material will be used for the POF test.	Type of Eval	Sig Lev dBf	NB	IBAC	AM IBOC	FM IBOC	DBS	Test Results Data to
Calibration	(Daily)	 The average and peak power will be recorded for each system at the proponent test bed input. IBOC analog and digital power will be read separately. (For systems where the digital signal cannot be transmitted separately, a spectrum analyzer will be used for monitoring the power.) For the USADR AM DAR power measurement, the composite signal power will be measured with and without analog audio applied. For the AM DAR system USASI noise will be used for the analog audio input. 	Objective	NA	x	x	X	x	x	be Recorded Power level
	2 Spectrum (Daily)	 Daily, a photo or X-Y plot of the system RF spectrum will be taken. Weekly, a spectrum plot will be made. For AM the spectrum analyzer will be set up in accordance with FCC 73.44. Working Group B will establish spectrum analyzer settings for the FM band. For IBOC systems any variations from FCC 73.317 for FM and 73.44 for AM will be noted. 	Objective	M	x	x	x	x	×,	Spectrum record in lab log
ľ	3 POF (Daily)	Gaussian noise will be added to the signal in 0.25 dB steps until POF occurs. (For accurate performance monitoring, TOA and C-4 weak signal tests were added.)	EO&C	м	x	x	x	x	x	POF level
	4 Audio recording (Daily)	 A daily audio recording will be made of all of the proponent audio channels. The test lab specialist will listen to the audio on head phones for any obvious impairments. 	EO&C	м	x	x	x	x	x	Digital audio recording for the
	5 Proof IBOC (Weekly)	During the analog compatibility tests, an automated proof of performance will be conducted weekly on the analog portion of the IBOC systems. The test will include the analog system performance with and without subcarrier groups A and B operating. A high quality demodulator will be used for the test. (With systems where the wide band analog demodulator detects the digital signal during the analog proof, the digital transmitter will be turned off by the lab staff. In these cases a stereo consumer receiver that is not sensitive to the digital signal will be used to confirm the analog channel performance for these systems.)	Objective	Varying	NA	NA	x	x	NA	record in lab log Record of frequency response, separation, and distortion in lab log.
	6 Reference analog TX total proof	A proof of performance test will be conducted on the AM and FM reference transmitters, with and without subcarriers, prior to the compatibility test and at four week intervals during these tests. Both subcarrier groups will also be tested.	Objective	NA	NA	x	x	x	NA	Objective test records.
-	7 Monitor calibration (Weekly)	The AM and FM analog modulation monitors will be calibrated on a weekly basis. The AM analog modulation monitor will measure both peak and average modulation,	Objective	NA	NA	NA	NA	NA	NA	Calibration record in lab log
	8 Proponent self check	This test will use the proponent self certification routine to determine if the DAR system is operating within specified limits.	Objective	System	x	X	x	x	x	Note in lab log
	9 Test bed calibration (Monthly)	All of the critical components in the test bed including the multipath simulator, attenuators, combiners, filters, generators, and measuring instruments will be calibrated on a monthly schedule.	Objective	need NA	NA	NA	NA	NA	NA	Calibration record in lab log

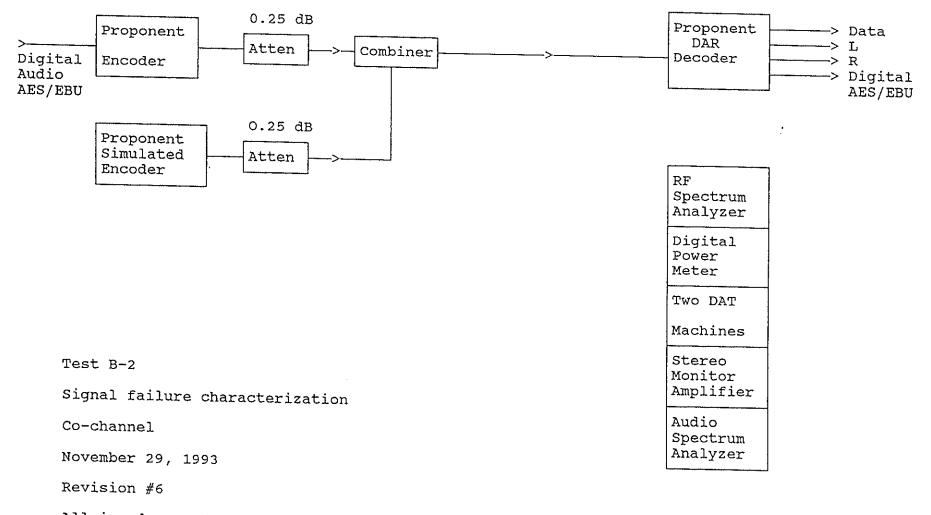
	V #9A Septer	UNIFIED DAR LABORATORY TES	TS							
Test Group	Test & Impairment	 TEST PROCEDURE Note: Impairment audio test material will be used for the digital audio signal. Processed audio will be used for the IBOC analog audio signal. If audio processing is an integral part of the proponent coder, unprocessed audio will be used. At least three audio segments will be used for each impairment test. The detailed procedure for noise measurements will be supplied. 	Type of Eval	Sig Lev dBf	NB	IBAC	AM IBOC	FM IBOC	DBS	Test Results Data to be Recorded
B Impairment tests for character- ization of DAR signal failure	1 Noise	 Gaussian noise will be increased to POF (0.25 dB steps) and the level logged. From the POF the noise will be reduced in 0.5 dB steps until the noise is 1.5 dB below the TOA. Each .5 dB step will be digitally recorded for expert subjective assessment. (To minimize possible measurement variations caused by hysteresis, the noise will be increased rather than decreased prior to recording each 0.5 dB step. Steps #1 & #2 will be repeated for each of the three impairment audio segments. The data channel BER will be measured. 	Subj & EO&C in Lab	м	x	x	x	X	x	Noise Level at TOA & POF
failure	2 Co- channel	 The co-channel interference will be increased to TOA and POF (0.25 dB steps) and the level logged. From the POF, the co-channel interference will be reduced in 0.5 dB steps until the interference is 1.5 dB below the TOA. Each .5 dB step will be digitally recorded for expert subjective assessment. (To minimize possible measurement variations caused by hysteresis, the co-channel signal will be increased rather than decreased prior to recording each 0.5 dB step. Steps # 1 & #2 will be repeated for each of the three impairment audio segments. 	Subj & EO&C in Lab	М	x	x	x	×	x	D/U at TOA and POF
	3 Multipath with noise	 This test will be conducted four times, each with different multipath scenarios. The multipath parameters will be specified by the channel characterization sub-group of Working Group B. A special multipath signal will be used for AM IBOC. Without noise added, each of the multipath signal parameters will be assessed in the transmission laboratory for impairments. If impairments are heard, the signal will be recorded for further assessment. For those multipath tests where no impairment is heard, noise will be added to the signal in 0.5dB steps until the <u>TOA</u> and POF are found. (<i>For those systems that require noise to be added to hear multipath, seven digital audio recordings will be made at the following noise levels: 1 dB below TOA, 0.5 dB below TOA, 0.5 dB above TOA, at wo equal points between TOA and POF, and at POF. These digital recordings are for expert subjective assessment.</i> The noise level and the multipath parameters will be recorded in the laboratory log. 	Subj & EO&C in Lab	М	x	x	NA .	x	x	Noise Level with Multipath at TOA & POF
	3a AM -> DAR first	1. AM -> first adjacent DAR, replaces B-3 for AM system only. B-2 procedure will be used for this test. In step $#1$ first adjacent will be substituted for co-channel. 2. The data channel BER will be measured.	Subj & EO&C	м	NA	NA	x	NA	NA	D/U at TOA and POF

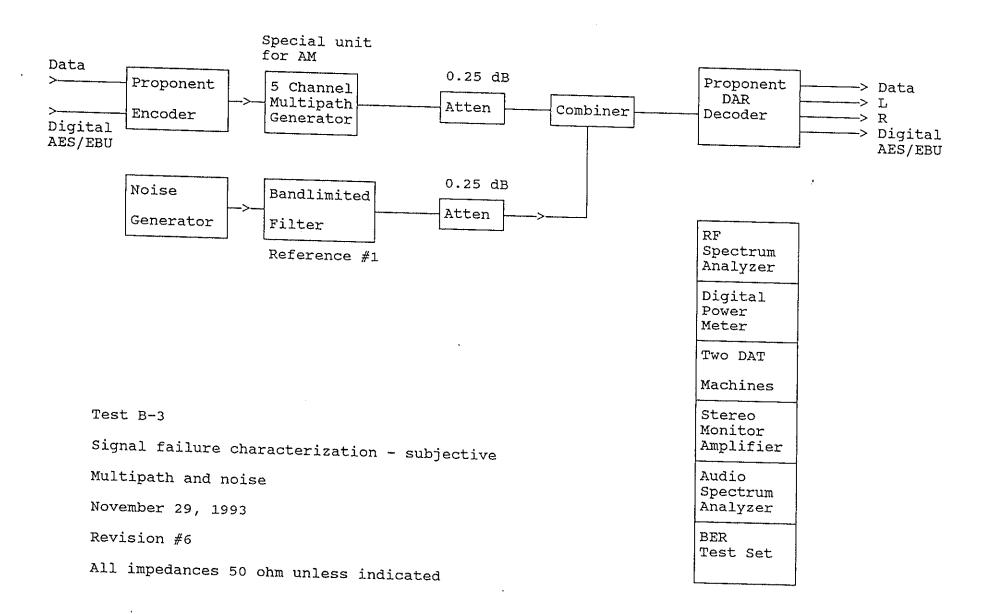
Note: The test procedure changes of September 2, 1994 for tests B.1.2, B.2.2, B.3.5, and B.3.6 are shown in Italics.

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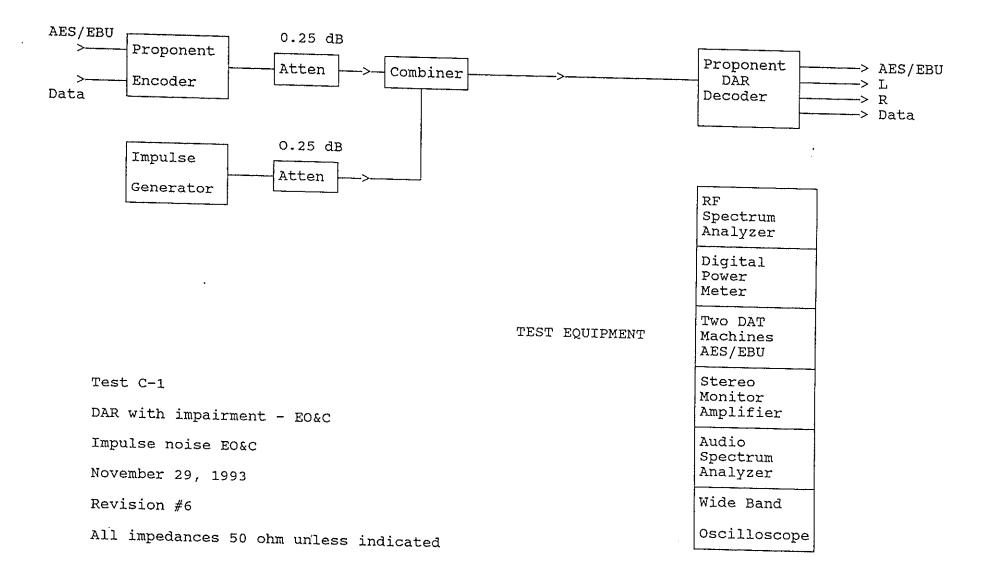


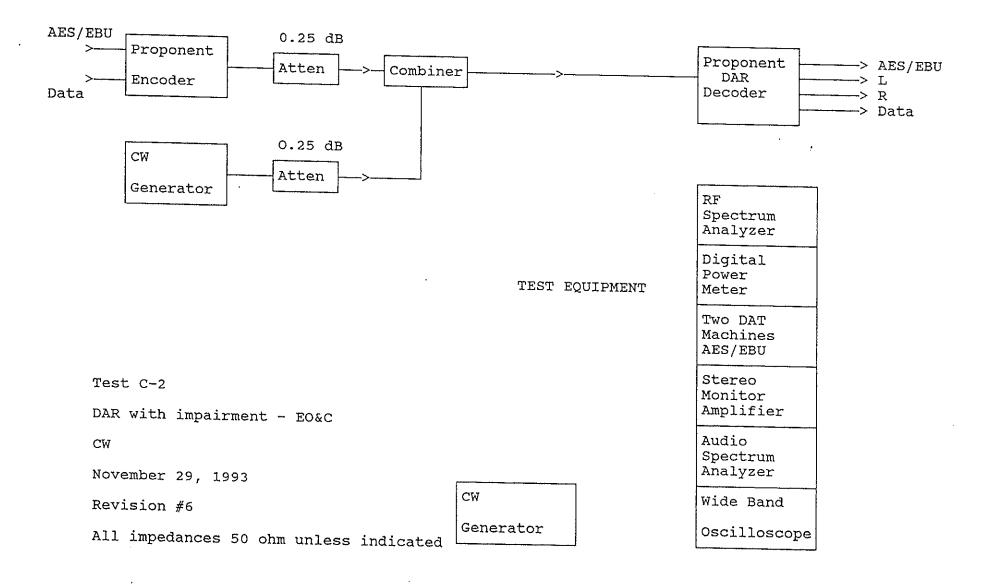
Test Group	V #9A Septen Test &	CHITED DAR LABORATOR	Y TESTS							<u> </u>
	Impairment 1 Impulse	TEST PROCEDURE Note: 1. The DAR audio impairment test material will be used for these tests,	Type of Eval	Sig Lev dBf	NB	IBAC	AM IBOC	FM IBOC	DBS	Test Results Data to be
DAR with pecial mpairment	noise	 A generator capable of generating 10 nanosecond wide pulses with a rise and decay time of 3 to 4 nanoseconds will be used for the test. The pulse rate will be slowly sweep from 100 Hz to 1000 Hz. (Because these pulses do not produce energy in the L and S bands, these tests were conducted only in the medium wave and the VHF bands.) The pulse generator output will be mixed with the DAR signal. The amplitude of the pulses will be increased until the TOA and POF is heard by the laboratory specialist. 	EO&C	М	x	x	X	X	х	Recorded Pulse amplitude in Volts P-P at TOA & POF BER at TOA
	2 CW	 The test will start with a slow RF sweep covering the DAR channel. Starting at a low RF level, the sweep amplitude will be increased in 2 dB steps until the POF is heard. The sweep will be frozen at the POF frequency. The CW signal will then be manually swept across the band while increasing the CW level and noting any further sensitive frequencies. 	EO&C	M & S (Strong was dcleted)	x	x	x	x	x	TOA & POF at sensitive levels across the DAR channel
	3 Airplane flutter	 Tests will be conducted with two simulated aircraft speeds of less than 250 MPH. The simulated reflected signal will be increased until the TOA or POF is heard by the lab specialist. Note: Simulated aircraft speeds to be set by the RF channel characterization task force. 	EO&C	₩& M	x	x	NA	х	NA	Multipath parameters at TOA & POF
	4 Weak signal	 Starting with a medium signal level, the signal will be reduced to TOA & POF (0.25 dB steps) (Steps were changed to 1 dB.) A single audio impairment recording will be used for this test. 	EO&C	Varying	x	x	x	x	x	Signal level at TOA & POF
	5 Delay spread/ doppler	Systems will be tested with four simulated multipath and motion extremes: 1. Flat or short multipath with slow and fast motion. 2. Long multipath with slow and fast motion. Note: The final multipath scenarios for this test will be set by the RF channel characterization subgroup.	EO&C	м	x	x	NA	x	x	Signal level at TOA & POF
1	6 Additional multipath with noise test	 This test will be conducted with four additional multipath scenarios not used in test B-3. For those multipath tests where no impairment is heard, noise will be added to the signal in .5dB steps until the POF is heard. From POF the noise will be reduced in 0.5 dB to find the TOA. 	EO&C in lab	М	x	x	NA	x	NA	Signal level at TOA & POF
	7 Environmental noise (AM band)	This test will compare AM and DAR reception with the following environmental interferences: 1. Variable gated RF noise.	EO&C	W & M	NA	NA	X	NA	NA	Interference amplitude a TOA & POF

Note: Test procedure changes of September 2, 1994 for tests, C.1.1, C.2 Signal Level, and C.4.1 are shown in Italies

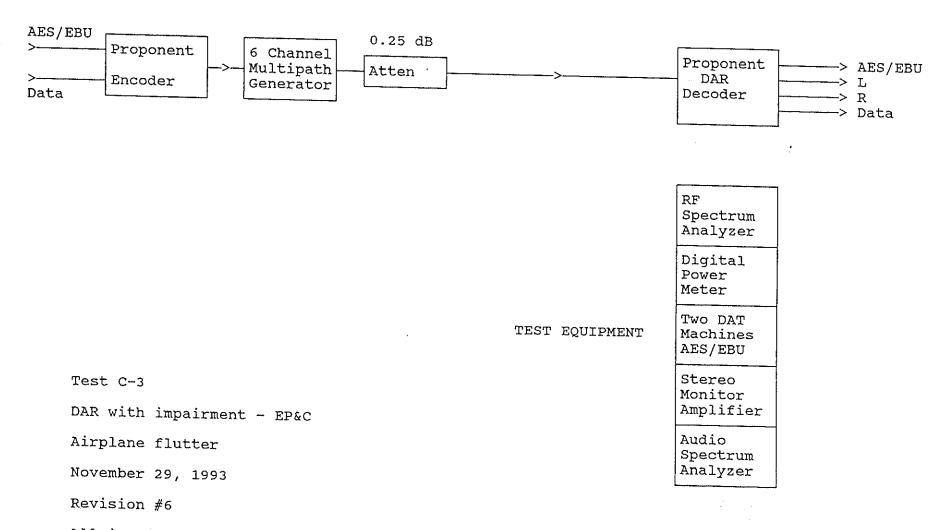
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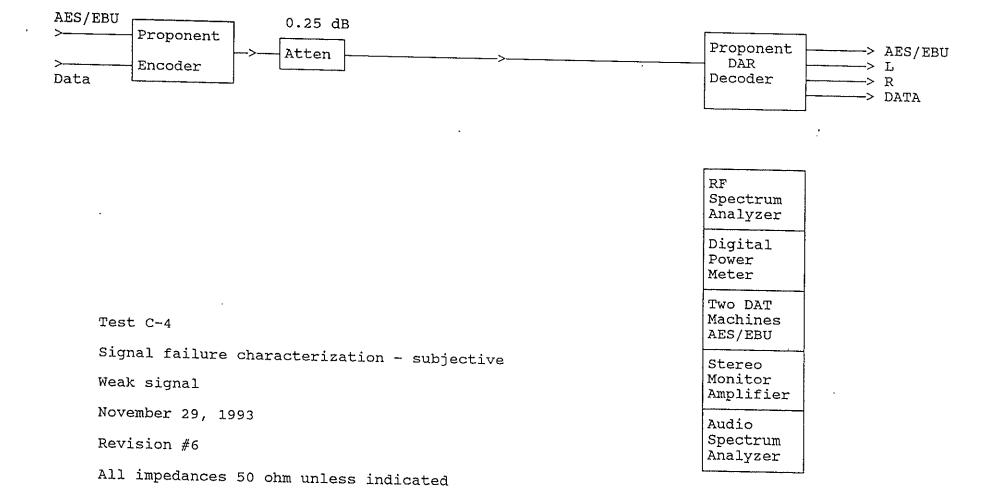
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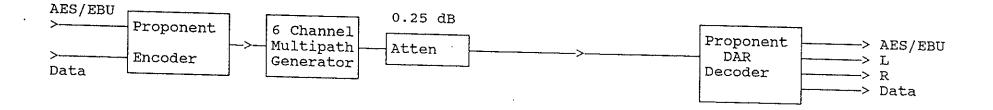
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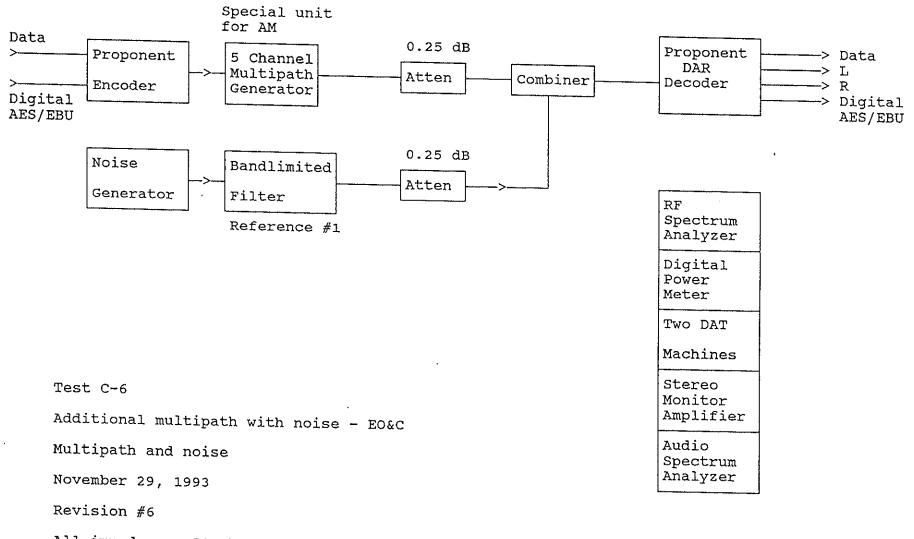
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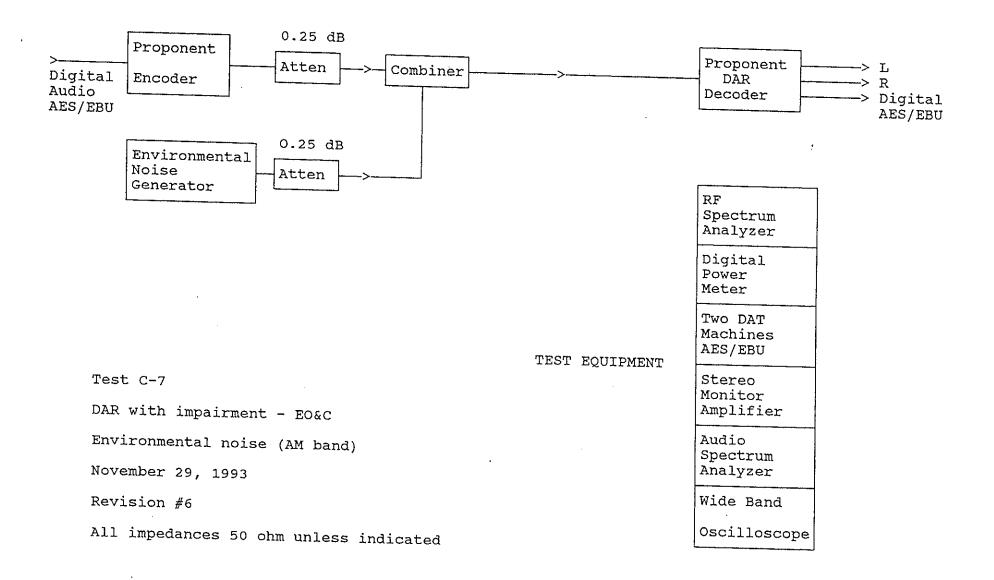


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		RF Spectrum Analyzer
		Digital Power Meter
	TEST EQUIPMENT	Two DAT Machines AES/EBU
Test C-5		Stereo Monitor
DAR with impairment - EP&C		Amplifier
Delay spread/doppler		Audio
November 29, 1993		Spectrum Analyzer
Revision #6		

All impedances 50 ohm unless indicated

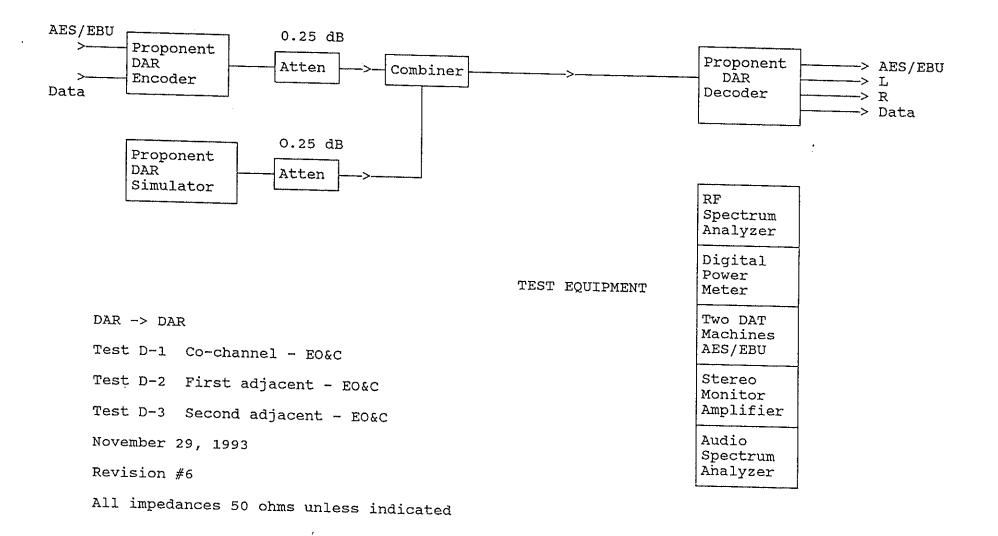




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REVISION #	⁴⁹ November	29, 1993 UNIFIED DAR LABORATOR	RY TESTS	<u></u>	<u></u>	<u></u>				<u></u>
Test Group	Test Number and Impairment	TEST PROCEDURE Note: 1. The undesired DAR signal will be generated with a simulator supplied by the each proponent. 2. The desired DAB signal will be modulated with unprocessed impairment test audio sequences.	Type of Eval	Sig Lev dBf	NB	IBAC	AM IBOC	FM IBOC	DBS	Test Results & Data to be Recorded
D DAR-> DAR) Co-channel	 The undesired co-channel DAR signal will be increased until the TOA and POF are heard by the lab specialist (0.25 dB resolution). A digital audio recording will be made of the desired signal with the interference set at TOA. (note this is a duplication of test B-2) 	EO&C in Lab	м	x	x	x	x	x	D/U & Levels at TOA & POF Data BER at POF -1 dB
	2 First adj	 The undesired first adjacent DAR signal will be increased until the TOA and POF are heard by the lab specialist (0.25 dB resolution). A digital audio recording will be made of the desired signal with the interference set at TOA. Note: This test will be conducted on both the upper and lower first adjacent channels and in both modes for the Amati/AT&T system. 	EO&C in Lab	м	x	x	x	x	x	D/U & Levels at TOA & POF Data BER at POF -1 dB
	3 Second adj	 The undesired second adjacent DAR signal will be increased until the TOA and POF are heard by the lab specialist (0.25 dB resolution). A digital audio recording will be made of the desired signal with the interference set at TOA. Note: This test will be conducted on both the upper and lower second adjacent channels and in both modes for the Amati/AT&T system 	EO&C in Lab	М	NA	x	x	x	NA	D/U & Levels at TOA & POF Data BER at POF -i dB

Note: For the IBOC system tests, Expert Observation and Commentary (EO&C) will be made by the laboratory specialist on the performance of the analog audio channel.

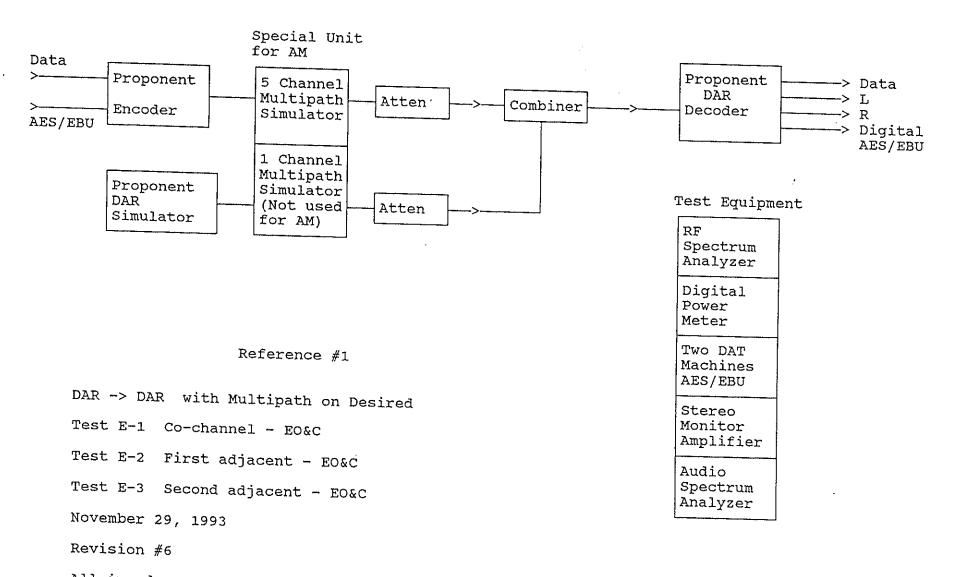


	9 November	29, 1993 UNIFIED DAR LABORATOR	Y TESTS					<u> </u>		
Test Group	Test Number and Impairment	TEST PROCEDURE Note: 1. The undesired DAR signal will be generated with a simulator supplied by the each proponent. 2. The desired DAB signal will be modulated with unprocessed impairment test audio sequences.	Type of Eval	Sig Lev dBf	NB	IBAC	AM IBOC	FM : IBOC	DBS	Test Results & Data to be Recorded
E DAR -> DAR with multipath	1 Co-channel	 Note: This test will be conducted in conjunction with test B-2. 1. This test will be conducted four times, each with different multipath scenarios. The multipath scenarios will be specified by the channel characterization subgroup of Working Group B. 2. Without co-channel added, each of the multipath scenarios will be observed by the transmission laboratory specialist. 3. The test in step #1, conducted four times with a different multipath on the desired signal, will now be conducted four times with multipath on the desired signal. 4. For those multipath tests where no impairment is heard, co-channel interference will be added to the signal until TOA and POF are heard. 5. Listening with the interference set at TOA, the multipath will be removed. Any change in the impairments will be noted. Also, any changes in BER will be logged. 6. For the AM test the desired and undesired signal will have simultaneous amplitude fluctuations to simulate the AM band. 	EO&C in Lab	M	×	x	x	x	NA	TOA and POF levels for each undesired signal and multipath scenarios
	2 First adjacent	Same as Co-channel Test, E-1. Note: This test will be conducted on both the upper and lower first adjacent channels and in both modes for the Amati/AT&T system.	EO&C in Lab	м	x	x	x	x	NA	TOA and POF levels for each undesired signal and multipath scenarios
	3 Second adjacent	Same as Co-channel Test, E-1. Note: This test will be conducted on both the upper and lower second adjacent channels and in both modes for the Amati/AT&T system.	EO&C in Lab	M	NA	x	x	x	NA	TOA and POF levels for each undesired signal and multipath scenarios

Note: For the IBOC system tests, Expert Observation and Commentary (EO&C) will be made by the laboratory specialist on the performance of the analog audio channel.

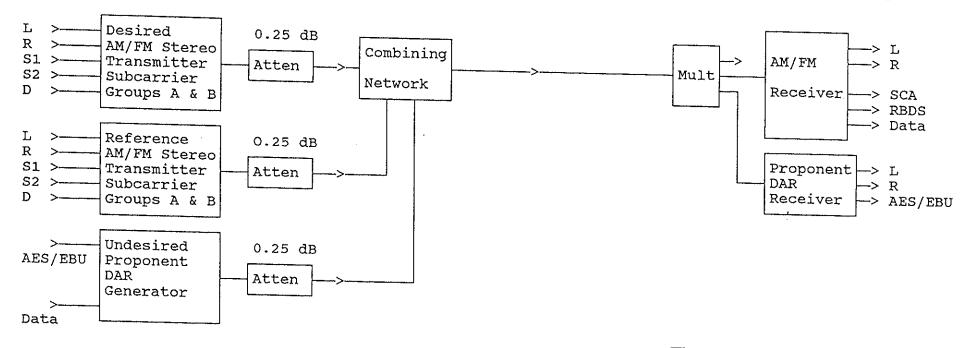
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REVISION #		UNIFIED DAR LABORATORY	TESTS							
	Test	Test Description Note: 1. These tests will compare the analog ->analog and DAR -> analog interference. 2. The undesired DAR audio will be processed rock music.	Type of Eval	Desired Sig Lev dBf	NB	IBAC	AM IBOC	FM IBOC	DBS	Test Results & Data to be Recorde
F DAR -> analog Interference to an analog receiver with no other impairments	1 Co-channel objective	 The five FM stereo, one mono, one with subcarrier group A, and one subcarrier group B receivers selected by the working group will be used for the DAR tests. Two stereo and 2 mono AM receivers will be used for the AM tests. The AM transmitter will be set for 100% modulation with a 400 Hz tone. The FM transmitters will be set for 75 kHz deviation with 400 Hz tone (left channel only), pilot, and subcarrier groups A or B. The subcarrier groups will be tested alternately. The interference will be the audio signal voltage (noise) measured in dB below the 400 Hz level set in step #2. The CCIR recommendation 412-4 weighting filter will be used. With the undesired signal increased until the resulting audio signal/noise ratios are 35 and 50 dB, the D/U will be measured for the interference combinations: noise -> analog, analog -> analog, and the DAR -> analog. For the AM IBOC interference tests, the signal/noise will be set at 25 and 40 dB. 	Objective	W & M	NA	X .	x	×	NA	D/U with auc S/N specified RBDS errors for all conditions SCA noise for all conditions Test digital su
	2 First adjacent 3 Second adj	The first and second adjacent channel tests are the same as Co-channel Test, F-1. The first and second adjacent channel measurements will be made both above and below the desired signal frequency, and the results averaged.	Objective	М	NA	x	x	x	NA	Same as F-1
			Objective	м	NA	x	×	x	NA	Same as F-1
	Co-channel subjective	 After test F-1 is completed, a panel from WG B will review the objective data and select four receivers for FM and two receivers for AM that are the most suitable for this test. The analog transmitters will be setup using the procedure in test F-1 step #2. The DAR transmitter will be gated on and off at 3 second intervals for this test. The desired audio signal will be a moderately processed FM stereo signal. Classical music, rock music, silence, and spoken voice will be used for the audio. This test will be conducted with analog -> analog and DAR -> analog interference that produced S/N ratios of 35 and 50 dB as in test F-1. For the AM IBOC interference is the reference. Step #6 will be recorded on digital audio tape. 	Subjective EO&C	М	NA	x	x	x	NA	NA
	5 First adj	Same as Co-Channel Test, F-4. Note: For step #6 the data will be taken from F-2.	Subjective EO&C	м	NA	x	x	x	NA	NA
	6 Second adj	Same as Co-Channel Test, F-4, Note: For step #6 the data will be taken from F-3.	Subjective EO&C	М	NA	x	x	x	NA	NA

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DAR -> Analog TEST EQUIPMENT F-1 Co-channel - Objective F-2 First adjacent - Objective F-3 Second adjacent - Objective RF F-4 Co-Channel - EO&C & Subjective Spectrum Analyzer F-5 First adjacent - EO&C & Subjective Audio F-6 Second adjacent - EO&C & Subjective Analyzer November 29, 1993 RBDS Revision #6 FM Test Set

Digital Power Meter Three DAT Machines AES/EBU Stereo Monitor Amplifier Audio Spectrum Analyzer FM Modulation Monitor

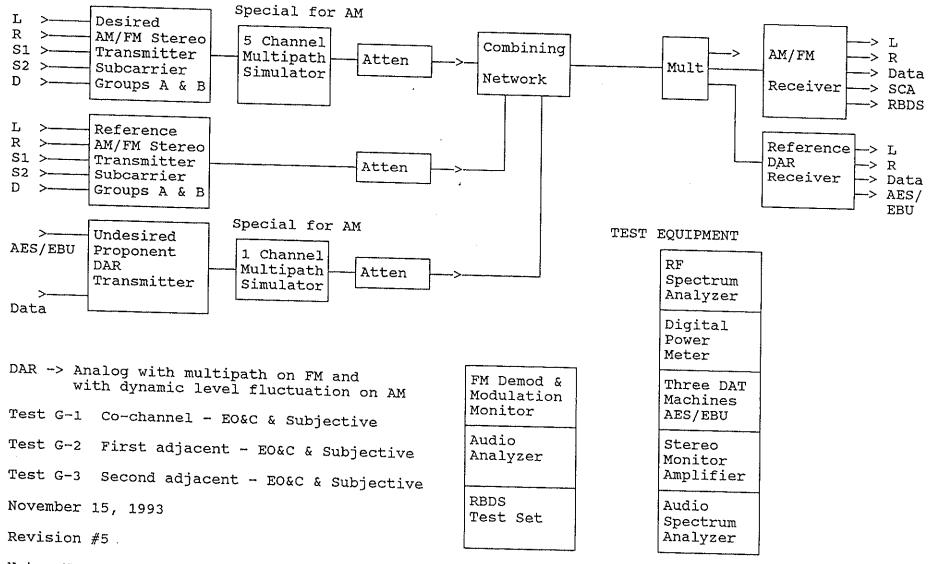
Note: No subcarriers on AM

REVISION #9		UNITED DAR LABORATOR	Y TESTS						<u> </u>	
	Test	 Test Description Note: 1. These tests are intended to subjectively compare analog -> analog interference with DAR -> analog interference with multipath. 2. These tests are not intended to measure the interference to an IBOC host station. 3. The undesired DAR audio signals will be processed rock music. 	Type of Eval	Desired Signal Level in dBf	NB	IBAC	AM IBOC	FM , IBOC	DBS	Test Results & Data to be Recorded
G DAR -> analog with multipath Interference to an analog receiver with multipath on the desired and un- desired signals	1 Co-channel subjective	 This test will be conducted four times, each with different multipath scenarios. The multipath scenarios will be specified by the channel characterization sub- group of Working Group B. The AM test will be conducted with amplitude signal level fluctuations. The one mono and five FM stereo receivers selected by the working group will be used. Two FM receivers, each with one of the subcarrier groups A or B, will be used. The digital subcarrier will be tested for BER using the time varying multipath on a single receiver. Two stereo and two mono AM receivers will be used for the AM tests. At least one of the FM stereo receivers should have poor AM rejection. The desired audio signal will be a moderately processed AM or FM stereo signal. Classical music, rock music, silence, and spoken voice will be used for the audio. The desired FM channel will be set for 75 kHz deviation with 400 Hz tone (left channel only), pilot, and subcarrier groups A or B. The desired AM channel will be set for 100% modulation with a 400 Hz tone. This test will be conducted with analog -> analog and DAR -> analog interference parameters that produced signal/noise ratios of 35 and 50 dB interference in test F-1. For the AM IBOC interference tests, the signal/noise will be set at 25 and 40 dB. Step #7 will be recorded on digital audio tape for subjective evaluation. 	Subjective & EO&C in Lab	м	NA	x	x	X	NA	NA
	2 First adjacent	First and second adjacent channel tests are the same as Co-channel Test, G-1 The first and second adjacent channel measurements will be made both about	Subjective & EO&C in Lab	м	NA	x	x	x	NA	NA
	3 Second adjacent	and below the desired signal and averaged. The Amati/AT&T system will be tested in modes 1 and 2.	Subjective & EO&C in Lab	м	NA	х	х	x	NA	NA

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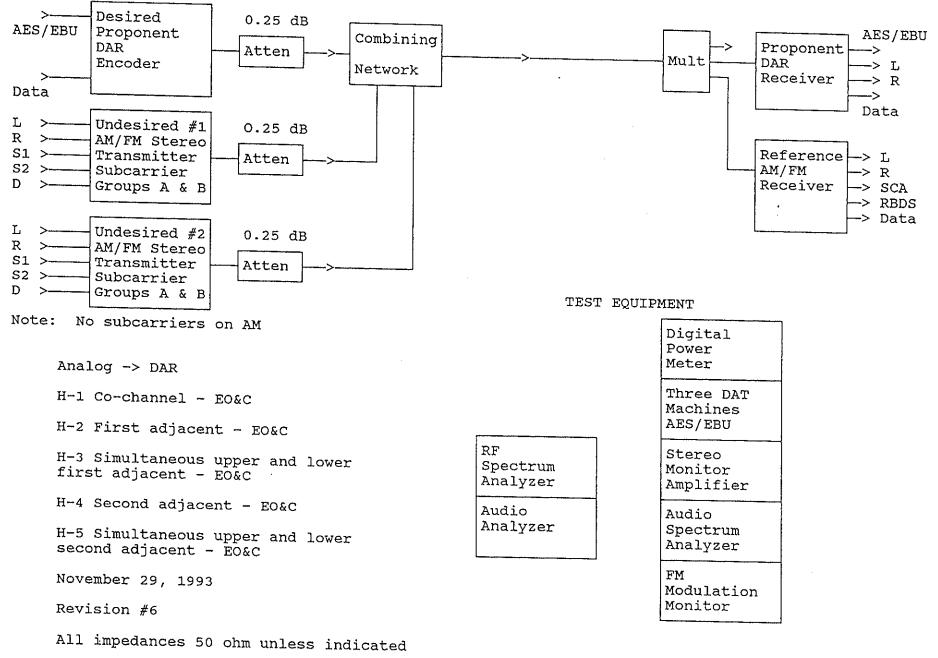
Note: No subcarriers on AM All impedances 50 ohms unless indicated

REVISION #9	November 2	29, 1993 UNIFIED DAR LABORATOR	Y TESTS	<u> </u>						
Test Group	Test Number and Impairment	Test Description Note: 1. The analog signal will be modulated with processed rock stereo. The FM transmitters will use subcarrier group A and subcarrier group B. 2. The DAB signal will be modulated with the impairment test audio.	Type of Eval	Sig Lev dBf	NB	1BAC	AM IBOC	FM IBOC	DBS	Test Results & Data to be Recorded
H Analog -> DAR no other	1 Co-channel	 The undesired analog signal (FM with subcarrier group A) will be increased until the TOA and POF is heard by the lab specialist (0.25 dB steps). For-the FM test step #1 will be repeated with subcarrier group B. The TOA will be recorded on digital tape for the record. 	EO&C in Lab	м	NA	x	x	x	NA	D/U at POA & POF
impairments	2 First adj	 The undesired analog signal (FM with subcarrier group A) will be increased until the TOA and POF is heard by the lab specialist (0.25 dB steps). For the FM test step #1 will be repeated with subcarrier group B. The TOA will be recorded on digital tape for the record. Note: This test will be conducted on both upper and lower first adjacent channels and in both modes for the Amati/AT&T system. 	EO&C in Lab	м	NA	x	x	x	NA	D/U at POA & POF
	3 Simultaneous upper and lower first adjacent	 Two undesired upper and lower first adjacent analog signals (FM with subcarrier group A) will be increased until the TOA and POF is heard by the lab specialist (0.25 dB steps). For the FM test step #1 will be repeated with subcarrier group B. The TOA will be recorded on digital tape for the record. 	EO&C in Lab	М	NA	x	x	NA	NA	D/U at POA & POF
	4 Second adj	Same as first adjacent channel test, H-1. Note: This test will be conducted on both upper and lower second adjacent channels and in both modes for the Amati/AT&T system.	EO&C in Lab	м	NA	x	x	x	NA	D/U at POA & POF
	S Simultaneous upper and lower second adjacent	Same as simultaneous upper and lower first adjacent test, H-3.	EO&C in Lab	м	NA	x	x	x	NA	D/U at POA & POF

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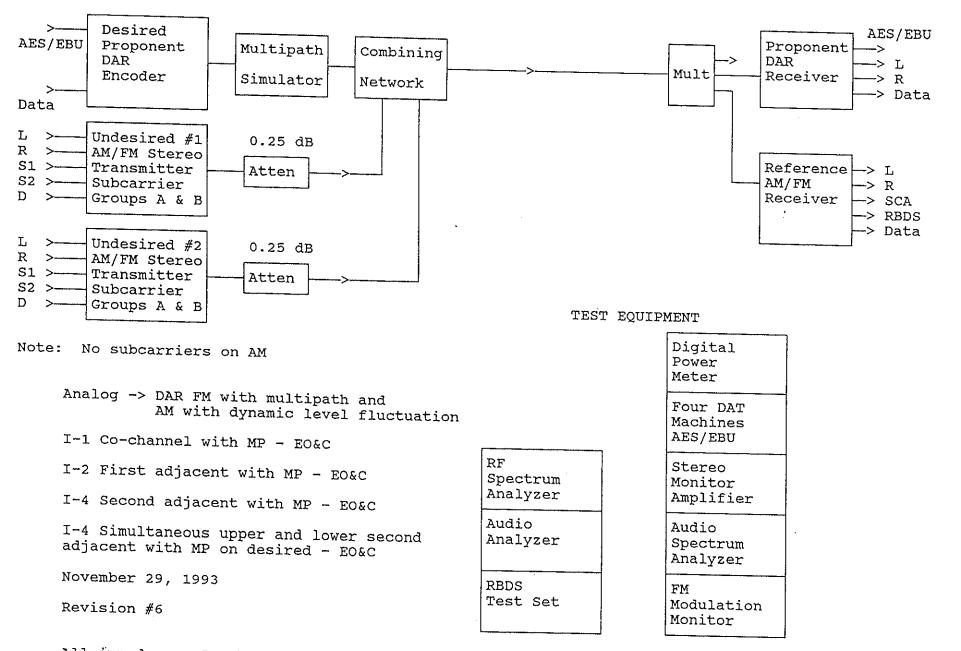
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REVISION #9		, 1993 UNIFIED DAR LABORATOR	Y TESTS	- <u>*** ,, , , ** (*** , * * *</u> .)		<u> </u>	* <u></u> *		· · · · · · · · · · · · · · · · · · ·	
Test Group	Test Number and Impairment	TEST PROCEDURE Note: 1. The FM signal will be modulated with processed rock stereo, subcarrier group A, and subcarrier group B. The AM audio signal will be processed rock music. 2. The DAB signal will be modulated with the impairment test audio. 3. The AM tests will be conducted with amplitude signal variations.	Type of Eval	Sig Lev	NB	IBAC	AM IBOC	FM IBOC	DBS	Test Results & Data to be Recorded
I Analog -> DAR with multipath	1 Co-channel with multipath	 This test will be conducted four times, each with different multipath scenario. The multipath scenarios will be specified by the channel characterization sub- group of Working Group B. The undesired signal will be increased to TOA and POF (0.25 dB steps). The undesired signal will be reduced to the TOA, and then the multipath will be added to the signal. If with multipath additional impairments are heard, the undesired signal will be reduced to a new TOA. 	EO&C in Lab	W&M	NA	x	x	x	NA	D/U at TOA & POF with multipath and any change without multipath
	2 First adj with multipath	Same as co-channel test, I-1. Note: This test will be conducted on both upper and lower first adjacent channels and in both modes for the Amati/AT&T system.	EO&C in Lab	W&M	NA	x	x	x	NA	Same as I-1
	3 Simultaneous upper and lower first adjacent with multipath	 This test will be conducted four times, each with different multipath scenario specified by the channel characterization sub-group of Working Group B. The undesired signal will be increased to TOA and POF (0.25 dB steps). The undesired signal will then be reduced to the TOA and multipath will then be added. If with multipath additional impairments are heard, the undesired signal will be reduced to a new TOA. 	EO&C in Lab	W&M	NA	x	x	NA	NA	Same as I-1
	4 Second adj with multipath	Same as co-channel test I-1 with two multipath tests.	EO&C in Lab	W&M	NA	x	x	x	NA	Same as 1-1
	5 Simultaneous upper and lower second adjacent with multipath	Same as simultaneous first adjacent channel test, I-3. Note: This test will be conducted on both the upper and lower second adjacent channels and in both modes for the Amati/AT&T system.	EO&C in Lab	W&M	NA	x	x	x	NA	Same as I-1

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REVISION #	9 November 2	29, 1993 UNIFIED DAR LABORATORY TES	rs							
Test Group	Test Number and Impairment	TEST PROCEDURE Note: 1. Continuous music will be used for the DAR audio modulation. 2. Hysteresis is the difference in dB between the signal level at POF with a loss of lock or acquisition and the signal level with usable music (complete acquisition).	Type of Eval	Sig Lev dBf	NB	IBAC	AM IBOC	FM IBOC	DBS	Test Results & Data to be Recorded
J DAR Acquisition and reacquisition tests	1 Simulated weak signal failure and acquisition	 Noise will be added to the signal in 0.25 dB steps until POF. The POF level will be recorded. The DAR transmitter will be disconnected from the receiver to assure loss of lock. Three tests will be conducted with the noise reduced in 2dB, 4dB, & 6 dB below POF for each test. The signal will be reconnected to the DAR receiver and acquisition time recorded for each noise level. Acquisition is the reproduction of usable music. EO&C comments will be recorded by the laboratory specialists. 	EO&C in Lab	М	×	x	x	x	x	1. Level at POF 2. Acquisition time at each noise level 3. Hysteresis
	2 Simulated acquisition with multipath and noise	 This test will be conducted four times, each with different multipath scenario. The multipath parameters will be specified by the channel characterization sub-group of Working Group B. Noise will be added until the signal fails. The DAR transmitter will be disconnected from the receiver to assure loss of lock. A different scenario will be selected. For each of the multipath scenarios, three tests will be conducted with the noise reduced to 2dB, 4dB, & 6 dB below POF for each test. The signal will be reconnected to the DAR receiver and acquisition time recorded for each of the test parameters in step #5. Acquisition is the reproduction of usable music. For IBOC only, EO&C comments will be made on the quality of the analog signal. 	EO&C in Lab	м	x	х	NA	x	x	Acquisition time for each multipath and noise scenario.

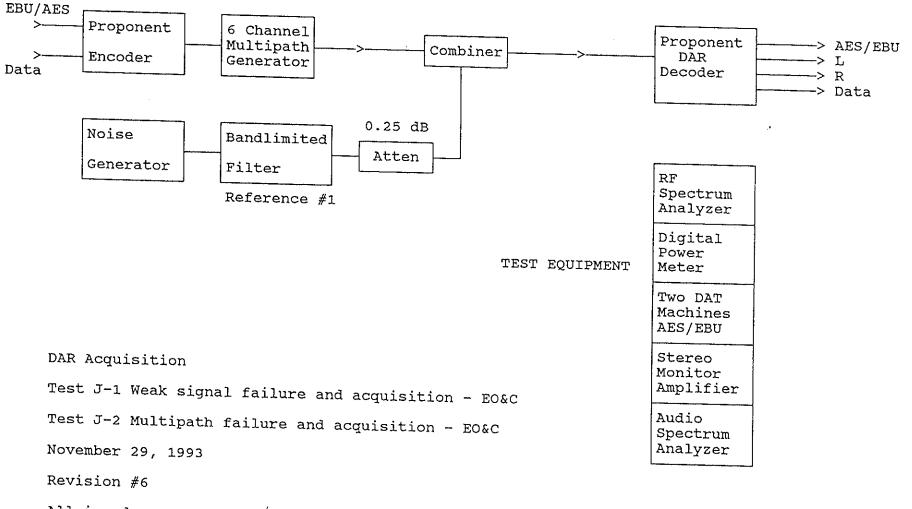
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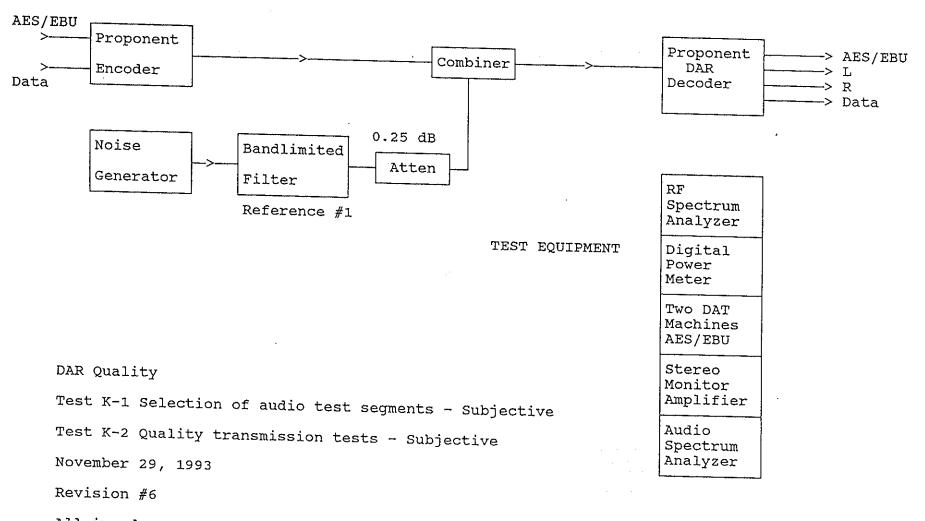
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Note: The detailed procedure for noise measurements is supplied in a separate document.

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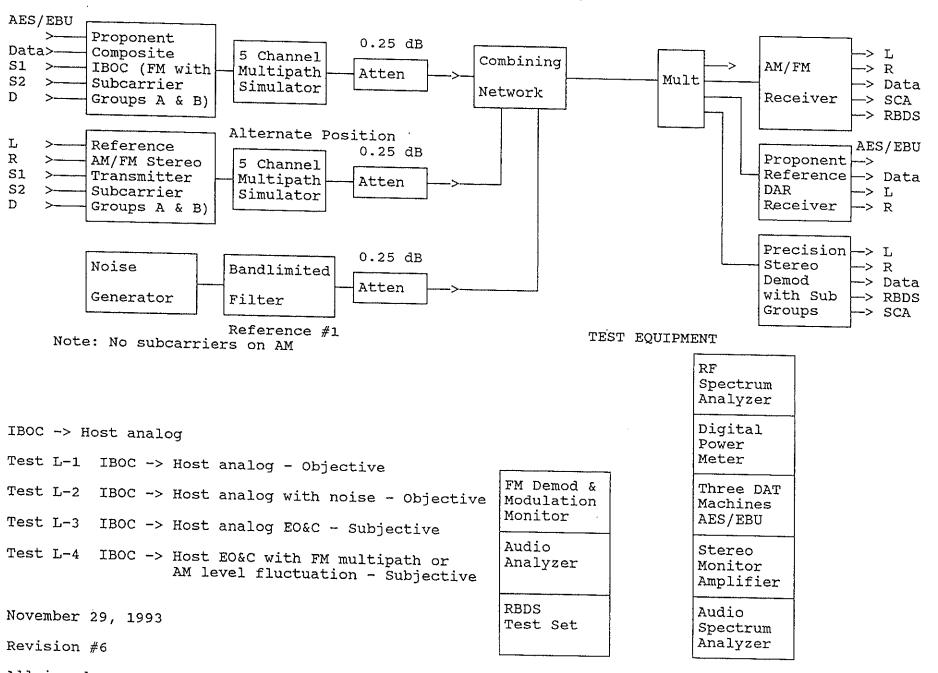


	#9 November	29, 1993 UNIFIED DAR LABORATOF	Y TESTS							
Test Group	Test Number and Impairment		Type of Eval	Sig Lev dBf	NB	IBAC	AM IBOC	FM IBOC	DBS	Test Results & Data to be
K DAR quality	1. Selection of audio test segments a.Quality b.Impairment c.Critical	 Digital test recordings will be submitted by members of Working Group B or any party with an interest in DAR. Each of the selected quality test segment, shall not exceed 30 seconds. The impairment segments will be at least 1 minute. At least 100 segments shall be considered by the working group. Pre-processed digital audio tapes will be included. Each of the original segments will be transmitted through each DAR system, with and without impairments. The impairments will be noise and noise with multipath. Each segment will be monitored by the specialist in the transmission laboratory. Segments that are considered critical by the specialist for quality or impairment testing will be recorded for further approval. The transmission test laboratory will use the following criteria for selecting the audio test materials: Select at least 3 segments that are capable of evaluating system quality. Select at least 2 segments that are considered critical of each proponent encoding system. The final certification of selected materials will be made in a CCIR type listening room by experts approved by WG-B. If at least 2 critical segments cannot be identified for each proponent, additional segments will be obtained. 	Subjective	M	x	X	X	x	x	NA
	2 Quality transmission test	 The quality test materials selected in test K-1 will be transmitted through each DAR system and recorded digitally. For subjective assessment, each recorded segment will then be sent to the subjective assessment laboratory. 	Subjective EO&C in Lab	м	x	x	x	х	x	Level at POF Noise at POF



REVISION #9		0, 1993 UNIFIED DAR LABORATORY TESTS Se	ptember 9.	1993						
Test Group	Test	TEST PROCEDURE Note: 1. The signal level for the composite IBOC is for both digital and host FM in dBf. 2. IBOC digital always includes the host analog signal. 3. The IBOC digital signal will be heavily modulated with processed audio. 4. Amati/AT&T will be tested in both modes.	Type of Eval	Desired Sig Lev dBf	NB	IBAC	AM IBOC	FM IBOC	DBS	Test Results & Data to be Recorded
L DAR-> analog	1 Host analog quality 2	 The host FM channel will have the stereo separation, distortion, frequency response, and noise measurements conducted with a precision demodulator (broadcast proof). A monophonic proof will also be conducted on the analog AM. 	Objective	м	NA	NA	x	x	NA	Performance with and without subs.
IBOC to host analog	2 IBOC to host analog	 A precision FM demodulator, one mono, five FM stereo, and two receivers with subcarrier groups A and B will be used for these FM tests. Two mono and two stereo AM receivers will be used for the AM tests. The host and reference FM transmitters will be set for a total 75 kHz deviation with 400Hz tone(left channel only), pilot, and with subcarrier group A or subcarrier group B. The AM transmitters will be set for 100% modulation with a 400 Hz tone. A separate analog transmitter will be used for the reference. For each test receiver, a plot of the input/output characteristics of the IBOC analog signal and the analog reference will be made. The performance of both subcarrier groups will be objectively measured. 	Objective	Varying W to S	NA	NA	x	x	NA	Plot input/output audio (distortion & freq. response) Subcarrier performance both digital &
	3 IBOC to host analog	 The same receivers used for test L-2 will be used for this test. The desired audio signal will be moderately processed. Classical music, rock music, silence, and spoken voice will be used for the audio. The host and reference FM channels will be set for a total 75 kHz deviation with 400 Hz tone (left channel only), pilot, and with both subcarrier groups. The separate reference AM transmitter will be set for 100% modulation. For each test receiver, a digital audio recording will be made of the IBOC analog audio signal and each receiver with the analog reference. Both subcarrier groups will be monitored for quality. 	Subjective EO&C	W&S	NA	NA	x	x	NA	analog Subjective
	4 IBOC to host analog with multipath	 The same receivers used for test L-2 will be used for this test. The desired audio signal will be a moderately processed. Classical music, rock music, silence, and spoken voice will be used for the audio. The procedure outlined in L-3 step #4 will be used to setup the transmitters. Both subcarrier groups will be used for this test. Four multipath scenarios selected by the RF channel characterization subgroup of WG B will be used. Noise will be added to the multipath to bring the multipath up to the TOA on the analog receivers. Dynamic signal amplitude variations will be substituted for the AM test. At the two signal levels and for each test receiver, an EO&C report will compare the IBOC analog signal quality and the analog reference signal quality. 	EO&C	W & S	NA	NA	x	x		RBDS and digital errors

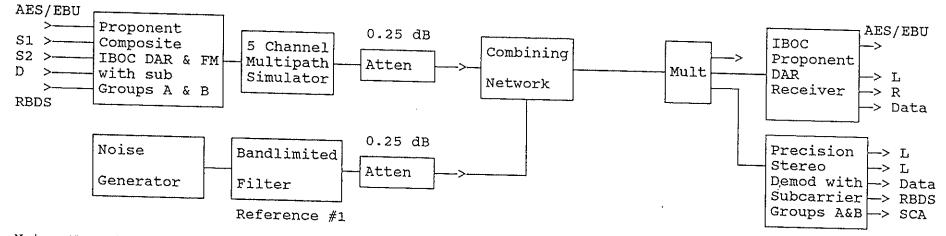
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REVISION #9	T	9, 1993 UNIFIED DAR LABORATORY	TESTS							
Test Group	Test	TEST PROCEDURE Note: 1. The signal level for the composite IBOC is for both digital and host FM in dBf. 2. IBOC digital always includes the host analog signal. 3. The analog signal will be heavily modulated with processed stereo rock music. The FM signal will include both subcarrier modes. 4. The DAR signal will be modulated with the impairment audio test material.	Type of Eval	Desired Sig Lev dB(NB	IBAC	AM IBOC	FM IBOC	DBS	Test Results & Data to be Recorded
M Analog -> DAR Analog to host IBOC	1 Host analog to IBOC digital with no other impairments	 The IBOC analog modulation will be alternately switched on and off while listening to the DAR audio for impairments. Both FM subcarrier modes (A & B) will be switched on and off while listening to the DAR audio for changes in impairments. The test results will be recorded on digital audio tape. 	EO&C in Lab	W&M	NA	NA	x	x	NA	EO&C
	2 Host analog to IBOC digital with multipath	 Four multipath scenarios selected by the RF channel characterization subgroup of WG B will be used for this test. Amplitude varying signals will be used for the AM test. The IBOC analog modulation will be alternately switched on and off while listening to the DAR audio for impairments. Both FM subcarrier modes (A & B) will be switched on and off while listening to the DAR audio for changes in impairments The test results will be recorded on digital audio tape. 	EO&C in Lab	W & M	NA	NA	x	x	NA	EO&C

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Note: No subcarriers on AM

TEST EQUIPMENT

	<u> </u>
	RF Spectrum Analyzer
	Digital Power Meter
FM Demod & Modulation Monitor	Two DAT Machines AES/EBU
Audio Analyzer	Stereo Monitor Amplifier
RBDS Test Set	Audio Spectrum Analyzer

IBOC Host FM -> DAR

Test M-1 Host FM -> DAR - EO&C

Test M-2 Host FM -> DAR with multipath - EO&C

November 29, 1993

Revision #6

All impedances 50 ohm unless indicated

	T	UNIFIED DAR LABORATORY TESTS				<u> </u>			····
Test Group	Test Number and Impairment	TEST PROCEDURE Note: 1. This test is IBOC specific. 2. In addition to co-channel and adjacent channel minimum separations, Part 73.207 of the FCC rules specifies minimum distance separation requirements for FM stations operating at 10.6 MHz and 10.8 MHz (10.7 MHz IF) above and below the operating channel. Using the receiver generated interference caused by two FM stations operating with the 10.7 MHz separation as the reference, Test N will compare the two FM interference (reference) with the interference caused by an IBAC and FM station operating with the same RF power level.	Type of Eval	Sig Lev dBf	NB	IBAC	IBOC	DBS	Test Results & Data to be Recorded
N Multiple spurious (10.6 MHz & 10.8 MHz)	I. Reference	The following test frequencies and procedures will be used to characterize the reference receiver. RF GEN 1 = 94.1 MHz Receiver = 99.95 MHz RF GEN 2 = 104.8 MHz RF1 = the RF level from a single generator will be set to give 30 dB S/N at the tuning frequency (99.95 MHz). RF2 = the RF level of both interfering generators will be set to give 30 dB S/N.	EO&C in Lab	NA	NA	x	NA	NA	30 dB Audic S/N
	2. Test	Using the analog test receiver the proponent IBAC system will replace RF GEN 1 (94.1 MHz). The average power of the IBAC transmitter will be set for the same power level as the signal generator it replaces. Any difference in subjective interference will be noted in the EO&C.	EO&C in Lab	NA	×	x	NA	NA	Changes in Subjective Interference

July 17, 1995

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"O" Outline of DAR/Subcarrier Compatibility Tests REVISION #9A August 9, 1995

Subcarrier Test Gro	oup A	Subcarrier Tes	t Group B	Subcarr (Undesi	ter Test Group red)	C Subcarrier Test Group D
RBDS	_3X	REOS	10%	67 kHz	10%	
66.5 kHz Digital	8.5%	67 kHz	10%	92 Khz	10%	
92 kHz	8,5%					92 kHz HS Digital Date 10%

	SUBCARR DAR -> ANALOG &	IER TESTS ind ANALOG ~>	DAR					
Description	Test	67 kHz SCA Analog Quelity	92 kHz SCA Analog Quality	RBDS 2.5% Test	RBDS 10% Test	Digital Sub 66.5 kHz BER		
F	F-1 Co-channel objective	x	x	x	x	x		
DAR -> Analog	F-2 First adjacent objective	x	x	x	x	x		
	F-3 Second adjacent objective	x	x	x	x	x		
Interference to analog	F-4 Co-channel subjective	× X	x	NA	NA	HA		
receiver with no	F-5 First adjacent subjective	X	x	NA	NA	HA		
other impairments	F-6 Second adjacent subjective	X	x	NA	NA	NA		
G	G-1 Co-channel subjective	x	×	x	x	x		
DAR ->	G-2 First adjacent subjective	x	x	x	x	x		
analog with multipath	G-3 Second adjacent subjective	x	x	x	x	X		
K	K-1 Co-channel	FH SIGNAL SUBCARRIER TO DAR COMPATIBILITY TEST						
Analog ->	R-2 First adjacent "							
DAR no other impairments	H-3 Simultaneous upper & lower first "							
EO&C	K-4 Second adjacent "							
toa & pof	R-5 Simultaneous upper & lower second "							
I	1-1 Co-channel with multipath	FM SIGNAL SUBCARRIER TO DAR COMPATIBILITY TEST			EST			
Analog ->	1-2 First ed] with multipath	11						
DAR with multipath	1-3 Simulteneous up & low with multipath "							
EO&C TOA & POF	1-4 Second adj with multipath	11						
	1-5 Simultaneous second up & low with MP "							
L	L-1 IBOC host analog quality objective with and without subcarriers	FM subcarrier -> storeo program audio objective test Subcarrier groups A and B will be independently used						
DAR -> enalog IBOC to host enalog	L-2 IBOC -> host analog objective	x	x	×	×	X		
	L-3 IBOC -> host analog subjective	x	x	NA	NA	NA		
	1-4 IBOC -> host analog with MP subjective	X	X	X	x	x		
	M-1 Host analog -> IBOC digital FM SIGNAL SUBCARRIER TO DAR COMPATIBILITY TEST							
Host Analog to IBOC DAR EO&C in Lab	M-2 Host analog -> IBOC digitai multipath	FH SIGNAL SUBCARRIER TO DAR COMPATIBILITY TEST						

FM SIGNAL SUBCARRIER TO DAR COMPATIBILITY TEST: By independently switching on and off subcarrier group A and B, the error rate will be monitored at TOA. If an increase is noted with a subcarrier group on, the FM signal will be reduced until a new TOA is logged.

APPENDIX C

CRC Digital Subjective Test Report and Procedures



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EIA-DAR LISTENING TESTS

QUALITY AND IMPAIRMENT TESTS PROCEDURES

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Updated version (1 December 1993) of a document submitted to EIA-DAR Working Group B on July 23rd, 1993

1. Introduction

This document contains a brief description of the test procedures that have been considered by the EIA-DAR Working Group B for the implementation of both the Quality and Impairment subjective tests of the DAR systems.

2. Facilities

The physical facilities available at the CRC are described in detail in [2], [3] and [4]. The same facilities would be used for both the Quality and the Impairment subjective tests. They include a reference, calibrated listening room, professional quality monitor loudspeakers, headphones and electronic equipment. A custom made disk-based playback system, which incorporates several unique and important features for testing high-quality audio systems, would also be used [5]. All audio materials to be used in the tests would be recorded on the PC hard disk via an AES/EBU interface. Playback from the hard disk would also be done via the AES/EBU interface. Listeners would control the presentation of the audio materials with the help of a mouse and a video monitor screen shown in Figure 1. The audio materials to be compared are typically 20-30 seconds long and are repeated indefinitely when listeners activate the *infinite loop* button on the screen. Seamless switching between the "A", "B" and "C" versions of the triple-stimulus is possible either a) by pressing the left, centre or right button of a three-button mouse or, b) by clicking with the mouse on the "A", "B" or "C" button displayed on the screen. A "zoom" tool allows the subjects to listen to a smaller portion of any audio material to be compared and assessed. The start and end time of the portion can be set anywhere within an audio material. Scores for both "B" and "C" versions (a hidden reference and a coded version) are entered directly into the PC with the help of the mouse and a special scoring screen which displays the CCIR 5-grade impairment scale in the form of two adjustable sliding potentiometers.

3. Quality Subjective Tests

The method briefly outlined here is in compliance with the procedure recommended by the CCIR for testing low bit-rate audio coding systems with small impairments and described in [1]. This method was discussed at the EIA-DAR Working Group B meeting held at the CRC on March 11, 1993. It is described in greater detail in [2], [3] and [4]. Copies of [2] and [3] were distributed previously to members of the Working Group B, and were also available at the March 1993 meeting.

3.1 Selection of critical audio materials

As recommended by CCIR [1], only critical audio materials should be used. This should maximize the ability of the subjective tests to discriminate among the DAR systems under tests. Since there is no universally critical audio materials that apply to any system under any condition, the selection of critical materials must be tailored to the particular set of DAR systems under test. One cannot simply take the segments used in previous experiments and expect them to be suitable in a different experiment, especially if new systems are being tested in the latter. In other words, the "criticality" of an audio material is not inherent in the material, but comes about through the interaction between a material and a system. This is clearly outlined in section 2 ("Audio Materials") of reference [2].

The responsibility of selecting the critical materials is to be delegated to a group of skilled expert listeners (minimum 3) who shall work by consensus. Ideally, some of these listeners should understand how the audio codecs of the DAR systems work so they can identify what type of audio materials are likely to be

critical. The starting point should be a very broad range of compact disk materials. This range can be extended by dedicated recordings. Any stimuli that can be considered as potential broadcast material shall be allowed. Synthetic signals deliberately designed to break a specific system should not be included. The artistic or intellectual content of an audio material should be neither so attractive nor so disagreeable or wearisome that the subject is distracted from his main task.

At least two materials that are critical for each DAR system are needed for the quality tests. It is likely that a given suitable audio item will turn out to be critical for more than one DAR system. This means that, perhaps, a total of 8 to 12 different materials will turn out to meet the requirement of two critical materials per system for the 6 DAR systems presently under consideration.

Due to the complexity of the task, the selection panel should have access to: a) the hardware implementation of the DAR systems under test, b) suitable listening facilities (listening room, high quality loudspeakers and headphones) and c) library of compact disks. A successful selection of critical materials can only be achieved if sufficient time and resources are available. Past experience (i.e. ISO/MPEG and CCIR subjective tests) suggests that it may take up to a month to uncover suitable materials for half a dozen coding systems.

3.2 Subjective test procedures

The subjective tests per se can start as soon as the selection of critical materials is completed. As recommended by the CCIR [1], only expert listeners should be used for testing audio systems with small impairments. Good data from at least 12 listeners are needed for reliable evaluations of the DAR systems. It is possible that some of the listeners used might turn out to be insufficiently sensitive to the quality variations in the materials over the DAR systems. Accordingly, some 20 or more listeners may need to participate in the tests to ensure sufficient acceptable data.

Assuming 6 DAR systems and 10 different audio materials, a total of 60 trials would be needed for the complete evaluation. These would be divided in two blocks of 30 trials, with one block of trials being assessed in the first day by three expert listeners and the second block in the next day by the same listeners. The actual tests, then, would require two days per listener to complete.

The morning of each day would be devoted to training on the 30 items that will be rated in the afternoon. Listeners will train three at a time in the same listening room where rating will take place. The training phase would be quite informal and listeners would be encouraged to interact and to help each other in detecting impairments. The disk-based A-B-C switching system would be used during training. "A" would be the known reference and "B" would explicitly be known to be a version processed by a DAR system during training. The identity of DAR systems would not be known to the listeners. "C" is irrelevant during the training phase. Three sets of Stax Lambda Pro headphones would be provided and could be used by the listeners at their discretion, instead of or in addition to the loudspeakers. Listeners would be given a maximum of three hours each day to train on a block of 30 items. Past experiences have shown that most of the listeners took 2 to 2½ hours to train on 30 items.

The 30 trials used for training in the morning would be broken down into three separate sessions of 10 trials each for the afternoon's **blind rating** sessions. Rating would be performed individually by each listener. The effects of session order and time would be factored out of listeners ratings by using a rotation scheme among the three listeners which is applied over the course of the entire experiment to a pre-aranged,

pseudo-randomly ordered item sequences. Each listener would rest during the two rating sessions of the other listeners which intervene between his or her next session, to minimise fatigue.

The A-B-C triple-stimulus presentation method would be used during blind rating. "A" would always be known to the listener as the reference (i.e. the uncoded CD original), while one of "B" or "C" would be a hidden reference identical to "A" and the other would be a version processed through a DAR system. The listener would not know and will not be able to predict what the B-C assignment is on any trial. The task will be to rate both "B" and "C" relative to "A". Any difference will be graded with the help of the CCIR continuous 5-grade impairment scale. This scale is shown in Table 1. The disk-based playback system, which allows seamless switching between "A", "B" and "C" for fine and detailed comparison, would be used. Listeners would be allowed to take as much time as they need on each trial, switching as often as they like, until satisfied with the numerical rating they have assigned to both "B" and "C". They would also be free to use either the loudspeakers or headphones to make a judgement, whichever they felt was the most critical transducer for any trial.

5.0	 Imperceptible	
4.0	 Perceptible but not annoying	
3.0	 Slightly annoying	A REAL PROPERTY AND A REAL
2.0	 Annoying	in the second
1.0	 Very annoying	and the second

Table 1 CCIR continuous 5-grade impairment scale

4. Impairment Tests

The impairment test procedure described herein is the result of discussions held within the EIA-DAR Working Group B.

4.1 Selection of critical audio materials

A minimum of three critical audio materials will be needed for the each impairment tests. These materials should be carefully chosen in a suitable listening environment by expert listeners by consensus. Materials should be selected on the basis of their sensitivity to revealing coding artifacts produced by the various channel impairments tested. It has been proposed that each segment be at least 1 minute long. The set of audio materials selected for the impairment tests is likely to be different from the set used in the quality test. In the absence of previous experience, it is not known how long it will take to select these audio materials.

Nonetheless, the identification of critical materials for the impairment subjective tests should be straightforward: those materials with the largest C/N (or D/U) ratio at POF should be the most critical ones because they reach the POF at a lower bit error rate (BER).

4.2 Impairment levels

As proposed in [6], each audio material selected for the impairment tests will be recorded, for each DAR system and each type of impairment, at the following levels of impairment: CA, TOA1, TOA2, TOA3, S1, S2, ..., POF where:

CA = coded audio in a clear channel TOA1, TOA2, TOA3 = three stimulii in the close neighborhood of the approximate TOA (Threshold of Audibility) S1, S2, ... = other intermediate levels of impairments POF (Point Of Failure)

For each of the above impairment levels, the C/N (or D/U) ratios should be noted. Except for CA, each of the above impairment levels should be separated by 0.5 dB in the C/N (or D/U) ratios.

4.3 Subjective test procedures

As proposed in [6], two separate experiments would be performed for each type of impairment considered:

Experiment 1: Threshold of Audibility

The purpose of this experiment is to provide a sensitive and reliable measurement of the TOA for various type of channel impairments. This experiment would be performed using CA (the coded audio in a clear channel) as the reference and the three stimulii recorded in the close neighborhood of the approximate TOA (i.e. TOA1, TOA2 and TOA3). The goal is to determine which of TOA1, TOA2 and TOA3 is the true TOA. Double-stimulus (A-B) presentation would be used along with a categorical rating scale (e.g. "B identical to A" or "B different from A"). Some training would also be provided.

Experiment 2: Failure Characteristic

The purpose of the second experiment is to determine how the subjective quality of each DAR system degrades with increasing levels of impairment. This would be established by using the uncoded CD original as the reference against which the following stimulii would be compared and rated: CA, TOA (as determined in experiment 1 above), S1, S2, ..., POF.

Because some of the degradations to be rated will be small (e.g. for CA and TOA), the triple-stimulus A-B-C presentation would be used, where "A" is the known reference (unprocessed CD signals). One of "B" or "C" will be the stimulus to be rated and the other one will be the hidden reference (i.e. a perfect replica of "A"). The assignment of the stimulus and hidden reference to "B" and "C" will not be known to the judges and will be arranged to be unpredictable to the listeners from trial to trial. For each trial, judges would be asked to rate the *difference* between the known reference "A" and version "B" as well as the difference between "A" and "C" using the same CCIR continuous 5-grade impairment scale as in the quality experiment discussed in section 3 (see Table 1). The numerical results of the Failure Characteristic

experiment could then be plotted for each DAR system and for each type of impairment in a manner similar to that shown in Figure 2.

Since the two first stimulii suggested above (i.e. CA and TOA) will include mostly small degradations, group training would be provided. Unlike the quality test, training would be performed on a *subset only of the items to be rated*. This subset would include all the items with small impairments, namely all CA and TOA versions, as well as a cross-section of materials from the other levels of impairments (i.e. S1 to POF versions). The training procedure would be identical to that of the quality test (see section 3.2).

The listening panel would include a minimum of 6 expert-listeners. Blind rating would be performed individually by each listener for both experiment 1 and 2. Listeners will use the disk-based playback system described in section 2 which allows seamless switching between the stimulii to be compared. Listeners will be able to take as much time as they need on each trial, switching as often as they like, until satisfied with the numerical ratings they are asked to assign. They would also be free to use either the loudspeakers or headphones to make a judgement, whichever they felt was the most critical transducer on any trial. Finally, they would be free to record any additional observations or commentaries they feel appropriate for any trial.

The items to be rated would be ordered in an quasi-random, unpredictable way over the course of the complete test procedure to ensure that time-correlated factors (such as fatigue) would not differentially affect any level of any of the factors under test.

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- [5] Beaton R.J. and Wong P., *A Disk-based System for the Subjective Assessment of High Quality Audio*, Preprint 3497, AES 94th Convention, Berlin, Germany, 16-19 March 1993.
- [6] Jayant N.S., Johnston J.D. and Sundberg C.E.W., *Channel Inpairment Tests*, AT&T document submitted to EIA-DAR Working Group B, 4 May 1993.

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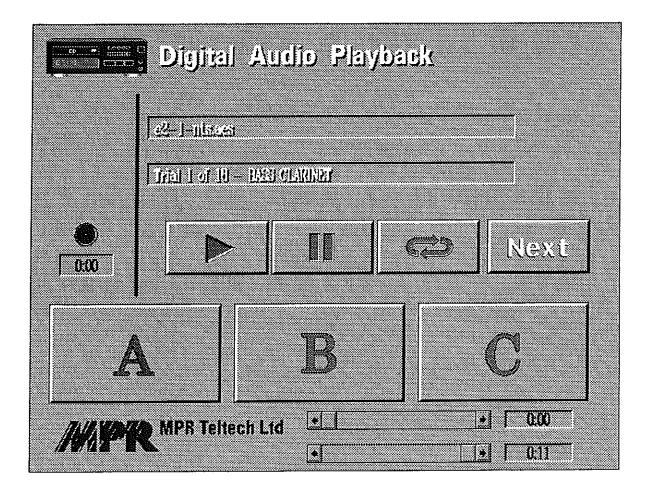


Figure 1 Screen used by listeners during training and blind rating sessions

APPENDIX D

Delay Spread / Doppler Procedures

TEST C-5 DELAY SPREAD/DOPPLER

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Enclosed is the committee approved CRC test procedure.

Each point was monitored for at least 30 seconds.

Classical audio was used for detecting impairments.

Three levels of impairment were recorded by the laboratory experts:

- 0 = Unimpaired
- 1 = Small Impairment
 2 ≥ POF Level of Impairment

Multipath "Stress Testing" of DAR Systems

B. McLarnon (CRC) Revised July 1994

1. Introduction

This document suggests an approach to testing DAR system performance over a wide range of simulated multipath conditions so that the performance limits of a given system can be approximately established. The channel simulation system proposed for use in these tests is the HP 11759C. The idea is to evaluate system performance while varying the multipath parameters until the Point of Failure (POF) of the system under test is reached, and to repeat these tests until the domain of operation of that system is defined. The tests themselves would include informal listening tests to define POF. In addition, a test would be performed at each setting of the simulator to measure the point of reacquisition of the system under test after loss of signal.

2. Simulation Hardware

The HP 11759C simulator can generate 6 independent paths, and two units can be operated in landem to generate 12 paths. The simulator has three modes:

- (1) "Simulation" mode: the delay, amplitude and phase of each path can be set independently. The amplitude of a given path can either be fixed, with a constant Doppler shift or phase shift, or it can be made to fade with a Rayleigh distribution and specified maximum Doppler shift. The latter mode produces a classical "U-shaped" Doppler spectrum for the corresponding path. In the Simulation mode, the simulation runs continuously; however, the actual fading pattern on a given path repeats after about 27 seconds.
- (2) "Travel" mode: in this case, a simulation of a 15 km trip at constant speed through an area with up to six multipath reflectors (or 12 with two simulators) is performed. The user specifies the location and loss factor of each reflector, along with the location of the transmitter. The delay and path loss for each path is updated at a rate of 10 times per second, and a log-normal amplitude distribution is applied to simulate wide-area fading (shadowing).
- (3) "Dynamic" mode: the settings for each path are completely specified by the user. The length of the simulation runs depends on the number of settings in the data file and the update rate at which they are fed to the simulator.

In all three cases, a data file containing the simulator path settings is prepared before the simulation run. During the run, the settings are transmitted from the PC controller to the simulator at a fixed update rate. This rate can be varied from 1 Hz to 2.3 kHz. The "Simulation" mode uses the maximum update rate of 2.3 kHz, whereas the Travel mode uses an update rate of 10 Hz. In the Dynamic mode, the update rate can be specified by the user.

Within the constraints imposed by the simulator, we can vary the following parameters:

- (1) Multipath delay spread
- (2) Maximum Doppler spread
- (3) Shape of the power delay profile
- (4) Temporal behavior of each path

The first two parameters are those which will be varied in order to plot the system performance limits on a delay vs. Doppler plot. Since different multipath profiles having the same delay spread may have

different effects on system performance, it is also useful to try several different ones. The temporal variations possible with a given path vary with the mode, as explained above.

3. Channel Models

For these tests, the Simulation mode is most appropriate. In order to determine the settings for the individual paths, it is necessary to adopt a model. It is suggested that, in developing the settings, the COST 207 recommendations developed in Europe for GSM digital cellular system tests be followed. In fact, these recommendations are the basis for the predefined multipath profiles which come with the HP simulator. Although the COST 207 model was developed for testing 900 MHz systems, there is no reason to believe that the real-world power delay profiles at VHF, L-band or S-band are radically different from those seen at 900 MHz (some significant differences between terrestrial and satellite channels may be expected, however).

The COST 207 report develops multipath profiles for four different environments. Each profile has a different shape and a different delay spread, as follows:

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RA (rural area, non-hilly): sawtooth shape, delay spread = 0.1 ms

TU (typical urban, non-hilly): sawtooth shape, delay spread = 1 ms

BU (bad urban, hilly): double sawtooth shape, delay spread = 2.5 ms

HT (hilly terrain): two separated sawtooth shapes, delay spread = 5 ms

The four profiles are shown in the accompanying figure.

The report also provides recommended settings for 6 and 12-path simulators to achieve these profiles. Each path is characterized by a time delay, attenuation, and type of Doppler spectrum. Rayleigh fading is applied to each path. Four types of Doppler spectra are defined:

CLASS: classical "U-shaped" spectrum extending between the maximum Doppler shift limits

GAUS1: two narrow Gaussian spectra, with the larger of the two on the negative side of the Doppler axis

GAUS2: two narrow Gaussian spectra, with the larger of the two on the positive side of the Doppler axis

RICE: combination of an unscattered component (direct path) and the classical spectrum

In the recommendation, the CLASS type spectrum is used for paths with small delays (< 0.5 ms), and the two Gaussian spectra are used for the longer-delayed paths. The RICE spectrum is used only in the RA profile. This type of spectrum is also appropriate for satellite channels. One limitation of the HP simulator is that it cannot generate the Gaussian Doppler spectra. Rayleigh fading paths can only have classical U-shaped Doppler spectra (or no Doppler shift at all). Thus it uses the CLASS type spectrum for all paths in its predefined GSM test profiles, except for the single instance of RICE in the RA profile. The COST 207 recommendation allows the substitution of classical spectrum for Gaussian, stating that "this approach is a worst case of the time variance of the mobile radio channel".

The predefined multipath profiles for each environment were generated for one delay spread, presumably reflecting typical conditions for that environment. For the purposes of exercising the DAR systems, it will be necessary to produce a family of profiles, with progressively increasing delay spread, for each environment. This can be done with a simple scaling procedure applied to the individual path delays.

4. Test Procedure: Outline

The basic test procedure for a given simulated multipath environment is to load the profile of simulator settings for the smallest delay spread, and then adjust the Doppler spread in increments until the POF is found (note that in this context, "profile" refers to a file stored on the PC which contains all of the settings for a simulation run). This is easily done from the menu screen of the Simulation Mode of the simulator. Changing the Doppler setting on any one of the paths automatically changes it for all of the others. Since the most stressful situations tend to be at the extremes of the Doppler settings (i.e., very slow and very rapid fading) it is possible that more than one POF will be found for a given delay profile setting. For each Doppler setting before POF, a signal reacquisition test should be performed. Once the performance of the system is fully characterized for that delay spread, a new profile is loaded for the next increment in delay spread, and Doppler is again varied in order to find the next POF points. This process is continued for increasing delay spreads until a locus of points mapping out the limits of system performance on a delay vs. Doppler plot is obtained (see Figure 2). The procedure then can be repeated for the other environment types, to see if system performance is dependent upon the nature of the power delay profile.

Preparation for the tests involves generation of a set of scaled multipath profiles for each of the basic profile types, and generation of a new set of Rayleigh Fading Data files for the appropriate Doppler shifts. The profiles have been generated at CRC, and are contained on floppy disk. Generation of the Rayleigh files is done with the HP IQMAKE utility, and should be done on the target PC. Each of these files is about 3.1 MB, so there must be sufficient disk space available for them. The Doppler values in the following table are suggested as a starting point.

Additional values, if needed, can easily be generated by the IQMAKE facility. Note that the maximum Doppler shift which can be generated by the simulator is 425 Hz. It is suggested that the tests for a given delay profile begin with the Doppler setting corresponding to a vehicle speed of 30 km/h at the frequency of the system under test. From that point, the simulated speed can be varied downwards towards zero and upwards towards the maximum speed.

5. Preliminary Setup

Begin by copying the contents of the floppy disk to the directory where the channel simulation software resides. It is assumed that this directory is C:\CHANSIM. If it is not, it may be necessary to edit the multipath profile files to change the IQDATA_DIR entry. To avoid confusion with the new data files to be created, it would be a good idea to temporarily move any existing fading data files (with filenames RAY'.IQ) to another directory. Do the same for the CHANSIM.PRO file, if one exists.

Maximum Doppler	Vehicle Speed (km/h)				
Shift (Hz)	100 MHz	1.5 GHz	2.36 GHz		
0.093	1	0.067	0.043		
0.28	3	0.2	0.13		
0.46	5	0.33	0.21		
0.93	10	0.67	0.43		
1.39	15	1	0.64		
2.78	30	2	1.3		
4.63	50	3.3	2.1		
6.94	75	5	3.2		
9.26	100	6.7	4.2		
13.89	150	10	6,4		
20.83	225	15	9.5		
27.78	300	20	12.7		
41.67	450	30	19.1		
69.44	750	50	31.8		
104.17	1125	75	47.7		
138.89	1500		63.6		
208.33	2250	150	95.3		
312.5	3375	225	143		
416.67	4500		191		

Building the Rayleigh Fading Files:

It is recommended that the fading files be created on the PC which controls the simulator, to avoid the need to transfer the files (the individual files are too large to fit on a floppy disk, and they do not compress significantly). The steps required to create the fading data files are as follows:

(a) Ensure that there is sufficient hard disk space available on the PC. Building the data files for all of the Doppler shifts given in the table will require about 60 megabytes of disk space. It is also important to maximize the amount of RAM available, by loading DOS high and removing

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any unnecessary resident programs and drivers (or loading them into high memory). The DOS *mem* command should show at least 560K of available memory before you start.

(b) Enter the CHANSIM directory containing the simulation software and make sure that the IQMAKE.EXE program is present. All of the data files in the table (a total of nineteen files) can now be created by simply entering:

MAKE

This is simply a batch file which contains the command:

IQMAKE -r 100E6 1 3 5 10 15 30 50 75 100 150 225 300 450 750 1125 1500 2250 3375 4500

Depending upon the speed of the PC, this may take 6 hours or more to complete.

RAY1.IQ, RAY3.IQ, etc., and each should be 3,147,241 bytes in length. If it becomes necessary to build additional fading files later, use the IQMAKE program directly. For example, to build a file which provides the Doppler shift encountered at 100 MHz by a vehicle travelling at 200 kph, the command would be:

IQMAKE -r 100E6 200

6. Running the Simulation

Rather than using the CHANSIM program directly, the simulations are run by means of a batch program which provides a menu for selecting different families of multipath profiles. This allows the user to avoid dealing with a limitation in the simulator software: it allows access to only 10 user-defined profiles at a time. To start the simulation, go to the CHANSIM directory and enter SIMUL. The following menu should be displayed:

Select one of the following environments:

- 1. Rural Area, 12 taps, Delay Spread = 0.02 ~ 0.2 us
- 2. Typical Urban, 12 taps, Delay Spread = 0.2 ~ 2 us
- 3. Hilly Terrain, 12 taps, Delay Spread = 5 ~ 7.5 us
- 4. Bad Urban, 12 taps, Delay Spread = 1 ~ 10 us
- 5. Bad Urban, 12 taps, Delay Spread = 11 ~ 20 us

Select 1-5, or q to quit:

Note that the terms 'tap' and 'path' are used interchangably here. After a selection is made, the CHANSIM program is called, and its main menu will be displayed. Proceed as follows:

- 1. Select S for the Simulation mode. This brings up the Simulation mode screen, which will initially show all paths turned off and the parameters set to zero.
- 2. Set the RF and LO frequencies which are appropriate for the system under test, by moving the cursor to the corresponding locations on the screen and hitting Enter, and then entering the frequency in MHz. This has to be done for each group of three paths (and, unfortunately, has to be re-entered each time you leave the CHANSIM program to select a new environment type, since when the program restarts, the RF frequency reverts to the default of 900 MHz).
- 3. Now it is time to select a multipath profile from the environment type previously selected. Hit ALT-P and then R to recall the stored profile. This will bring up a selection of 10 different profiles from that environment, with a range of delay spreads as indicated. Select a profile (typically from the middle of the range if you are using that profile for the first time) and hit Enter.
- 4. When the profile loads, the parameter values in the Simulation screen will be filled in. The simulation will become active as soon as the load is completed, and will run continuously until a new selection is made or the Simulation mode is exited.
- 5. The Doppler shift can now be varied while the delay profile remains fixed. When the profile is initially loaded, the Doppler will be set to a default value of 2.78 Hz, corresponding to a vehicle speed of 30 kph at 100 MHz. This is a reasonable starting point for VHF tests, but for the UHF systems, a new starting Doppler value should be selected which corresponds to roughly 30 kph vehicle speed at the test frequency. For example, for 1.5 GHz, it would be appropriate to select the file which provides a Doppler shift of 41.67 Hz. To select a new Doppler shift, move to any one of the RAY entries in the Spectrum selection part of the screen and hit Enter. The various Rayleigh fading files for different Doppler shifts that were created previously will be displayed. Use the cursor keys (and PgUp/PgDn) to select the file desired and hit Enter. The simulation will resume with the new Doppler setting as soon as the file loads, and the Simulation screen will show the new Doppler/vehicle speed settings.
- 6. Tests do not necessarily have to be done for every Doppler value. The basic idea is to explore the domain of operation of a system under test and show the results on a graph which plots delay spread vs. Doppler spread (or vehicle speed). For a given environment type, we hold the delay spread constant and perform tests for different Doppler values. In order to reduce the total number of tests, the increments in Doppler can be coarse at first, becoming finer as necessary to zero in on the parts of the domain where problems become evident.
- 7. When tests with different Doppler shifts are completed for a given delay spread, return to step 3 and select a new profile with a different delay spread (again, the increments in delay spread chosen may initially be fairly large, until the limits of the system under test become more evident). When tests are completed for the environment and delay spread range available from the stored profile, hit Esc to exit from the Simulation mode. Then at the main menu, select Q to quit the CHANSIM program. This will return you to the environment selection menu. Make a new selection and return to step 1 above.

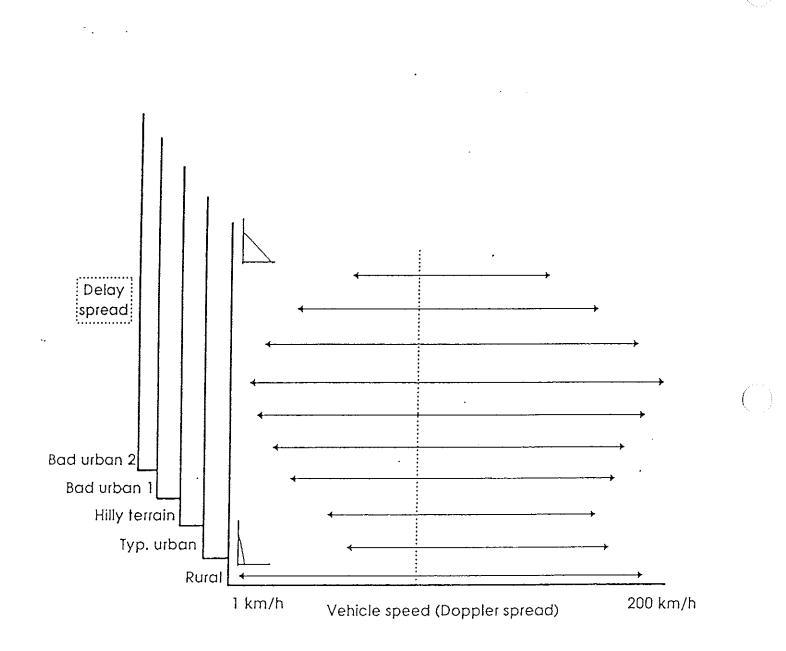


Figure 2: Typical results for one DAR system

APPENDIX E

HP 11759C Multipath Simulator

MULTIPATH SCENARIOS DOPPLER AND RAYLEIGH VHF, L, & S BANDS

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FILE NUMBER	TES	I DESCRIPTION
DAR90100.PRO	B-3	VHF RAYLEIGH 9 PATH SIMULATIONS
DAR90110.PRO	E	VHF RAYLEIGH 9 PATH SIMULATIONS WITH CO-CHANNEL
DAR90120.PRO	C-6	VHF DOPPLER 9 PATH SIMULATIONS
DAR90130.PRO	E	VHF DOPPLER 9 PATH SIMULATIONS WITH CO-CHANNEL
DAR90140.PRO	B-3	L-BAND RAYLEIGH 9 PATH SIMULATIONS
DAR90150.PRO	E	L-BAND RAYLEIGH 9 PATH SIMULATIONS WITH CO-CHANNEL
DAR90160.PRO	C-6	L-BAND DOPPLER 9 PATH SIMULATIONS
DAR90170.PRO	E	L-BAND DOPPLER 9 PATH SIMULATIONS WITH CO-CHANNEL
DAR90180.PRO	B-3	S-BAND RAYLEIGH 9 PATH SIMULATIONS
DAR90220.PRO	SPEC	VHF DOPPLER 3 PATH SIMULATIONS
DAR90240.PRO	C-3	VHF Airplane Flutter Simulations
DAR90250.PRO	C-3	L-BAND AIRPLANE FLUTTER SIMULATIONS
DAR90260.PRO	C-5	BAD URBAN 1 VHF CRC DELAY / SPREAD DOPPLER
DAR90270.PRO	C-5	BAD URBAN 2 VHF CRC DELAY / SPREAD DOPPLER
DAR90280.PRO	C-5	HILLY TERRAIN VHF CRC DELAY / SPREAD DOPPLER
DAR90290.PRO	C-5	RURAL VHF CRC DELAY / SPREAD DOPPLER
DAR90300.PRO	C-5	TYPICAL URBAN VHF CRC DELAY / SPREAD DOPPLER
DAR90310.PRO	C-5	BAD URBAN 1 L-BAND CRC DELAY / SPREAD DOPPLER
DAR90320.PRO	C-5	BAD URBAN 2 L-BAND CRC DELAY / SPREAD DOPPLER
DAR90330.PRO	C-5	HILLY TERRAIN L-BAND CRC DELAY / SPREAD DOPPLER
DAR90340.PRO	C-5	RURAL L-BAND CRC DELAY / SPREAD DOPPLER
DAR90350.PRO	C-5	TYPICAL URBAN L-BAND CRC DELAY / SPREAD DOPPLER

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DAR90100.PRO

----- Profile #3 2:TITLE "RURAL FAST RAYLEIGH" 2:IQDATA_DIR "C:\chan_new" 2:CORR_MODE 3X3 2:DELAY_RES LOW 2:CHAN ATTEN 0.000000e+00 0.000000e+00 0.000000e+00 0.000000e+00 2:GROUP_LOSS 0.000000e+00 2:IQFILE "RAY131.IO" "" "" 2:DELAY_US 0.0000 0.3000 0.5000 0.9000 1.2000 1.9000 2.1000 2.5000 3.0000 0.0000 1.0000 2.0000 2:DOPPLER_HZ_13.0785 13.0785 13.0785 13.0785 13.0785 13.0785 13.0785 13.0785 13.0785 13.0785 0.0000 2.0000 1.5000 2:PHASE DEG 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 2:ATTEN DB 4,0000 8,0000 0.0000 5,0000 16,0000 18,0000 14,0000 20,0000 25,0000 0.0000 2,0000 2,0000 2:CORRELATION 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 ----- Profile #4 #_____ _____

DAR90110.PRO

#------Profile #1
0:TITLE "URBAN SLOW WITH CO"
0:IQDATA_DIR "C:\chan_new"
0:CORR_MODE 3X3
0:DELAY_RES_LOW
0:CHAN_ATTEN 0.000000e+00 0.000000e+00 0.000000e+00 0.000000e+00
0:GROUP_LOSS 0.000000e+00
0:SPECTRUM_RAY_RAY_RAY_RAY_RAY_RAY_RAY_RAY_RAY_DOP_DOP_DOP
0:GPLAY_US 0.0000 0.2000 0.9000 1.2000 1.4000 2.0000 2.4000 3.0000 0.0000 1.0000 2.0000
0:DOPPLER_HZ 0.1744 0.1744 0.1744 0.1744 0.1744 0.1744 0.1744 0.1744 0.1744 0.1744 0.1744 0.1744 0.1744 0.1740 0.0000 2.0000 1.0000
0:PHASE_DEG 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000
0:ATTEN_DB 2.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000

3:TITLE "TERRAIN OBSTRUCTED WITH CO" 3:IQDATA_DIR "C:\chan_new" 3:CORR_MODE 3X3 3:DELAY_RES_LOW 3:CHAN_ATTEN_0.000000e+00 0.000000e+00 0.000000e+00 0.000000e+00 3:GROUP_LOSS 0.000000e+00 3:SPECTRUM_RAY_RAY_RAY_RAY_RAY_RAY_RAY_RAY_RAY_DOP_DOP_DOP 3:IQFILE "RAY52.IQ" "RAY52.IQUDD

DAR90120.PRO

----- Profile #4 "TERRAIN OBSTRUCTED DOPPLER VHF" 3:TITLE 3:IQDATA DIR "C:\chan_new" 3:CORR_MODE NONE 3:DELAY_RES_HIGH 3:CHAN ATTEN 0.000000e+00 0.000000e+00 0.000000e+00 0.000000e+00 3:GROUP LOSS 0.000000c+00 3:SPECTRUM DOP DOP DOP DOP DOP DOP DOP DOP DOP OFF OFF PE IN DERT ALER ANPE MELE HER TH 3:IQFILE 3:DELAY US 0.0000 1.0000 2.5000 3.5000 5.0000 8.0000 12.0000 14.0000 16.0000 0.0000 1.0000 2.0000 3:DOPPLER_HZ 0.8719 1.7438 2.1798 2.6157 3.0517 3.4876 3.9236 4.3595 5.2314 0.0000 2.0000 1.5000 3:PHASE_DEG 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 3:ATTEN DB 10.0000 4.0000 2.0000 3.0000 4.0000 5.0000 2.0000 8.0000 5.0000 0.0000 2.0000 2.0000 3:CORRELATION 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000

DAR90130.PRO

----- Profile #2 1:TITLE "URBAN FAST DOP WITH CO VHF #2" 1:IQDATA_DIR "C:\chan_new" 1:CORR_MODE NONE 1:DELAY_RES_HIGH 1:CHAN ATTEN 0.000000e+00 0.000000e+00 0.000000e+00 0.000000e+00 1:GROUP LOSS 0.000000e+00 1:IQFILE 1:DELAY_US 0.0000 0.2000 0.5000 0.9000 1.2000 1.4000 2.0000 2.4000 3.0000 0.0000 1.0000 2.0000 1:DOPPLER HZ 0.8719 1.7438 2.1798 2.6157 3.0517 3.4876 3.9236 4.3595 5.2314 0.0000 2.0000 1.5000 1:PHASE_DEG 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 1:ATTEN DB 2.0000 0.0000 3.0000 4.0000 2.0000 0.0000 3.0000 5.0000 10.0000 0.0000 2.0000 2.0000 I:CORRELATION 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000

DAR90140.PRO

----- Profile #6 **#**--5:TITLE "L-BAND URBAN FAST" 5:IODATA DIR "C:\chan new" 5:CORR_MODE 3X3 5:DELAY_RES_LOW 5:CHAN_ATTEN 0.000000e+00 0.000000e+00 0.000000e+00 0.000000e+00 5:GROUP LOSS 0.000000e+00 5:SPECTRUM RAY RAY RAY RAY RAY RAY RAY RAY OFF OFF "RAY817.IQ" "RAY817.IQ" "* ** ** ** 11 11 11 11 11 11 5.IOFILE 5;DELAY US 0,0000 0,2000 0,5000 0,9000 1,2000 1,4000 2,0000 2,4000 3,0000 0,0000 0,0000 0,0000 5:DOPPLER_HZ 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 0.0000 0.0000 0.0000 5:ATTEN DB 2.0000 0.0000 3.0000 4.0000 2.0000 0.0000 3.0000 5.0000 10.0000 0.0000 0.0000 5:CORRELATION 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000

----- Profile #8

DAR90150.PRO

#------Profile #6
5:TITLE "L-BAND URBAN FAST WITH CO"
5:IQDATA_DIR "C:\CHAN_NEW"
5:CORR_MODE 3X3
5:DELAY_RES_LOW
5:CHAN_ATTEN 0.000000e+00 0.000000e+00 0.000000e+00 0.000000e+00
5:GROUP_LOSS 0.000000e+00
5:SPECTRUM_RAY_RAY_RAY_RAY_RAY_RAY_RAY_RAY_RAY_DOP_DOP_DOP
5:OFILE "RAY817.IQ" "R

#------Profile #7 6:TITLE "L-BAND RURAL FAST WITH CO " 6:QDATA_DIR "C:\CHAN_NEW" 6:CORR_MODE 3X3 6:DELAY_RES LOW 6:CHAN_ATTEN 0.0000000e+00 0.0000000e+00 0.000000e+00 6:GROUP_LOSS 0.000000e+00 6:GROUP_LOSS 0.000000e+00 6:SPECTRUM RAY RAY RAY RAY RAY RAY RAY RAY RAY DOP DOP DOP 6:IQFILE "RAY2043.IQ" "RAY200 "RAY200 "RAY2043.IQ" "RAY2043.IQ" "RAY2043.IQ" "

6:DELAY_US 0.0000 0.3000 0.5000 0.9000 1.2000 1.9000 2.1000 2.5000 3.0000 0.0000 1.0000 2.0000 6:DOPPLER_HZ 204.3083 204.3083 204.3083 204.3083 204.3083 204.3083 204.3083 204.3083 204.3083 0.0000 2.0000 1.5000 6:PHASE_DEG 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 6:ATTEN_DB 4.0000 8.0000 0.0000 5.0000 16.0000 18.0000 14.0000 20.0000 25.0000 0.0000 2.0000 2.0000 6:CORRELATION 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000

----- Profile #8 7:TITLE "L-BAND TERRAIN OBSTRUCTED WITH CO" 7:IQDATA_DIR "C:\CHAN_NEW" 7:CORR_MODE 3X3 7:DELAY_RES LOW 7:CHAN_ATTEN 0.000000e+00 0.000000e+00 0.000000e+00 0.000000e+00 7:GROUP LOSS 0.000000e+00 7:SPECTRUM RAY RAY RAY RAY RAY RAY RAY RAY RAY DOP DOP DOP "RAY817.IQ" "RAY817.IQ" "RAY817.IQ" "RAY817.IQ" "RAY817.IQ" "RAY817.IQ" "RAY817.IQ" "RAY817.IQ" 7.IOFILE "RAY817.IO" "" 7:DELAY_US 0.0000 1.0000 2.5000 3.5000 5.0000 8.0000 12.0000 14.0000 16.0000 0.0000 1.0000 2.0000 7:DOPPLER HZ 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 0.0000 2.0000 1.5000 7:PHASE_DEG 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 7:ATTEN DB 10.0000 4.0000 2.0000 3.0000 4.0000 5.0000 2.0000 8.0000 5.0000 0.0000 2.0000 2.0000 7:CORRELATION 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000

DAR90160.PRO

#_

----- Profile #1 0:TITLE "L-BAND URBAN SLOW DOPPLER " 0:IQDATA_DIR "C:\chan_new" 0:CORR_MODE NONE 0:DELAY RES HIGH 0:CHAN_ATTEN 0.000000e+00 0.000000e+00 0.000000e+00 0.000000e+00 0:GROUP LOSS 0.000000e+00 0:SPECTRUM DOP DOP DOP DOP DOP DOP DOP DOP DOP OFF OFF 0:IOFILE 0:DELAY US 0.0000 0.2000 0.5000 0.9000 1.2000 1.4000 2.0000 2.4000 3.0000 0.0000 1.0000 2.0000 0:DOPPLER HZ 0.6810 0.9534 1.2258 1.6345 1.7707 1.9069 2.1793 2.4517 2.7241 0.0000 31.1911 23.4274 0;PHASE DEG 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0:ATTEN_DB 2.0000 0.0000 3.0000 4.0000 2.0000 0.0000 3.0000 5.0000 10.0000 0.0000 2.0000 2.0000 0:CORRELATION 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 ----- Profile #2

1:TITLE "L-BAND URBAN FAST DOPPLER " 1:IQDATA_DIR "C:\chan_new" 1:CORR_MODE NONE 1:DELAY RES HIGH 1:CHAN_ATTEN 0.000000e+00 0.000000e+00 0.000000e+00 0.000000e+00 1:GROUP LOSS 0.000000e+00 1:SPECTRUM DOP DOP DOP DOP DOP DOP DOP DOP OFF OFF . 1:IQFILE 1:DELAY US 0.0000 0.2000 0.5000 0.9000 1.2000 1.4000 2.0000 2.4000 3.0000 0.0000 1.0000 2.0000 1:DOPPLER_HZ 13.6206 27.2411 34.0514 40.8617 47.6719 54.4822 61.2925 68.1028 81.7233 0.0000 31.1911 23.4274 1:PHASE DEG 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 1:ATTEN DB 2.0000 0.0000 3.0000 4.0000 2.0000 0.0000 3.0000 5.0000 10.0000 0.0000 2.0000 2.0000 1:CORRELATION 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000

----- Profile #3 2:TITLE "L-BAND RURAL FAST DOPPLER" 2:IQDATA_DIR "C:\chan_new" 2:CORR MODE NONE 2:DELAY_RES HIGH 2:CHAN ATTEN 0.000000e+00 0.000000e+00 0.000000e+00 0.000000e+00 2:GROUP_LOSS 0.000000e+00 2:SPECTRUM DOP DOP DOP DOP DOP DOP DOP DOP DOP OFF OFF 2:IOFILE 2:DELAY US 0.0000 0.3000 0.5000 0.9000 1.2000 1.9000 2.1000 2.5000 3.0000 0.0000 1.0000 2.0000 2:DOPPLER HZ 40.8617 122.5850 204.3083 68.1028 136.2056 149.8261 95.3439 170.2569 177.0672 0.0000 31.1911 23.4274 2:PHASE DEG 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 2:ATTEN_DB 4.0000 8.0000 0.0000 5.0000 16.0000 18.0000 14.0000 20.0000 25.0000 0.0000 2.0000 2.0000 2:CORRELATION 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000

----- Profile #4 #-----"L-BAND TERRAIN OBSTRUCTED DOPPLER" 3:TITLE 3:IQDATA_DIR "C:\chan_new" 3:CORR_MODE NONE 3:DELAY_RES_HIGH 3:CHAN ATTEN 0.000000e+00 0.000000e+00 0.000000e+00 0.000000e+00 3:GROUP LOSS 0.000000e+00 3:SPECTRUM DOP DOP DOP DOP DOP DOP DOP DOP DOP OFF OFF AND AND AND AND THE THE THE THE AND AND 3:IQFILE 3:DELAY US 0.0000 1.0000 2.5000 3.5000 5.0000 8.0000 12.0000 14.0000 16.0000 0.0000 1.0000 2.0000 3:DOPPLER_HZ 13.6206 27.2411 34.0514 40.8617 47.6719 54.4822 61.2925 68.1028 81.7233 0.0000 31.1911 23.4274 3:PHASE_DEG_0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 3:ATTEN DB 10.0000 4.0000 2.0000 3.0000 4.0000 5.0000 2.0000 8.0000 5.0000 0.0000 2.0000 2.0000 3:CORRELATION 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000

DAR90170.PRO

DAR90180.PRO

#..... ----- Profile #I 0:TITLE "S-BAND URBAN SLOW 0:IQDATA_DIR "C:\CHAN_NEW" 0:CORR_MODE 3X3 0:DELAY RES LOW 0;CHAN ATTEN 0.000000e+00 0.000000e+00 0.000000e+00 0.000000e+00 0:GROUP_LOSS 0.000000e+00 0:SPECTRUM RAY RAY RAY RAY RAY RAY RAY RAY RAY OFF OFF "RAY38.IQ" "RAY38.IQ" "RAY38.IQ" "RAY38.IQ" "RAY38.IQ" "RAY38.IQ" "RAY38.IQ" "RAY38.IQ" "RAY38.IQ" " 0:10FILE 0:DELAY_US 0.0000 0.2000 0.5000 0.9000 1.2000 1.4000 2.0000 2.4000 3.0000 0.0000 0.0000 0.0000 0:DOPPLER HZ 3.7619 3.7619 3.7619 3.7619 3.7619 3.7619 3.7619 3.7619 3.7619 3.7619 0.0000 0.0000 0:PHASE DEG 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0:ATTEN_DB 2.0000 0.0000 3.0000 4.0000 2.0000 0.0000 3.0000 5.0000 10.0000 0.0000 0.0000 0.0000 0:CORRELATION 0,0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 #----- Profile #2 "S-BAND URBAN FAST 1:TITLE I:IQDATA DIR "C:\CHAN NEW" 1:CORR_MODE 3X3 1:DELAY_RES LOW 1:CHAN ATTEN 0.000000e+00 0.000000e+00 0.000000e+00 0.000000e+00 1:GROUP LOSS 0.000000e+00 I:SPECTRUM RAY RAY RAY RAY RAY RAY RAY RAY RAY OFF OFF "RAY1129.IQ" 1:IOFILE "RAY1129.IQ" "RAY1129.IQ" "" "" 1:DELAY_US 0.0000 0.2000 0.5000 0.9000 1.2000 1.4000 2.0000 2.4000 3.0000 0.0000 0.0000 0.0000 1:DOPPLER HZ 112.8560 1 1:PHASE DEG 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 1:ATTEN DB 2.0000 0.0000 3.0000 4.0000 2.0000 0.0000 3.0000 5.0000 10.0000 0.0000 0.0000 0.0000 1:CORRELATION 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 #----- Profile #3 2:TITLE "S-BAND RURAL FAST " 2:IQDATA DIR "C:\CHAN NEW" 2:CORR MODE 3X3 2:DELAY_RES LOW 2;CHAN_ATTEN 0.000000e+00 0.000000e+00 0.000000e+00 0.000000e+00 2:GROUP LOSS 0.000000e+00 2:SPECTRUM DOP RAY RAY RAY RAY RAY RAY RAY RAY RAY OFF OFF OFF 2:IQFILE "RAY2821.IQ" "RAY2821" "RAY2821" "RAY284" "RAY "RAY2821.IQ" "RAY2821.IQ" "RAY2821.IQ" "" 2:DELAY_US 0.0000 0.3000 0.5000 0.9000 1.2000 1.9000 2.1000 2.5000 3.0000 0.0000 0.0000 0.0000 2:DOPPLER_HZ 282.1401 282.1401 282.1401 282.1401 282.1401 282.1401 282.1401 282.1401 282.1401 282.1401 0.0000 0.0000 2:PHASE DEG 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 2:ATTEN DB 0.0000 18.0000 10.0000 15.0000 26.0000 28.0000 24.0000 30.0000 35.0000 0.0000 0.0000 0.0000 2:CORRELATION 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 --- Profile #4 #---"S-BAND TERRAIN OBSTRUCTED " 3 TITLE 3:IQDATA_DIR "C:\CHAN_NEW" 3:CORR MODE 3X3 3:DELAY RES LOW 3:CHAN_ATTEN_0.000000e+00 0.000000e+00 0.000000e+00 0.000000e+00 3:GROUP LOSS 0.000000e+00 3:SPECTRUM RAY RAY RAY RAY RAY RAY RAY RAY RAY OFF OFF "RAY1129.IQ" "RAY1129.IQ" "RAY1129.IQ" "RAY1129.IQ" "RAY1129.IQ" "RAY1129.IQ" "RAY1129.IQ" 3:IOFILE "RAY1129.IQ" "RAY1129.IQ" "RAY1129.IQ" "RAY1129.IQ" 3:DELAY_US 0.0000 1.0000 2.5000 3.5000 5.0000 8.0000 12.0000 14.0000 16.0000 0.0000 1.0000 2.0000 3:DOPPLER_HZ_112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 0.0000 2.0000 1.5000 3:PHASE DEG 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 3:ATTEN DB 10.0000 4.0000 2.0000 3.0000 4.0000 5.0000 2.0000 8.0000 5.0000 0.0000 2.0000 2.0000 3:CORRELATION 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000

DAR90220.PRO

DAR90240.DAT

#---------- Profile #2 "AIRPLANE FLUTTER SCENARIO #2 VHF 1:TITLE 1:IQDATA_DIR "C:\CHAN_NEW" I:CORR_MODE 3X3 1:DELAY RES LOW 1:CHAN ATTEN 0.000000e+00 0.000000e+00 0.000000e+00 0.000000e+00 1:GROUP_LOSS 0.000000e+00 1:SPECTRUM PHA DOP OFF OFF OFF OFF OFF OFF OFF OFF OFF LIOFILE. 1:DELAY_US 0.0000 13.6761 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 1:DOPPLER HZ 333.5646 17.4380 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 1:PHASE DEG 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 1:ATTEN DB 0.0000 6.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 1:CORRELATION 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 ----- Profile #3 #---"AIRPLANE FLUTTER SCENARIO #3 VHF

DAR90250.PRO

#------- Profile #1 0:TITLE "AIRPLANE FLUTTER #1 L-BAND" 0:IQDATA_DIR "C:\CHAN_NEW" 0:CORR_MODE 3X3 0:DELAY RES LOW 0:CHAN_ATTEN 0.000000e+00 0.000000e+00 0.000000e+00 0.000000e+00 0:GROUP LOSS 0.000000e+00 0:SPECTRUM PHA DOP OFF OFF OFF OFF OFF OFF OFF OFF OFF 0:IOFILE 0:DELAY_US 0.0000 27.5000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0:DOPPLER HZ 333,5646 424,9613 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0:PHASE DEG 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0:ATTEN_DB 0.0000 8.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0:CORRELATION 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000

DAR90260.PRO

DAR90270.PRO

DAR90280.PRO

DAR90290.PRO

DAR90300.PRO

DAR90310.PRO

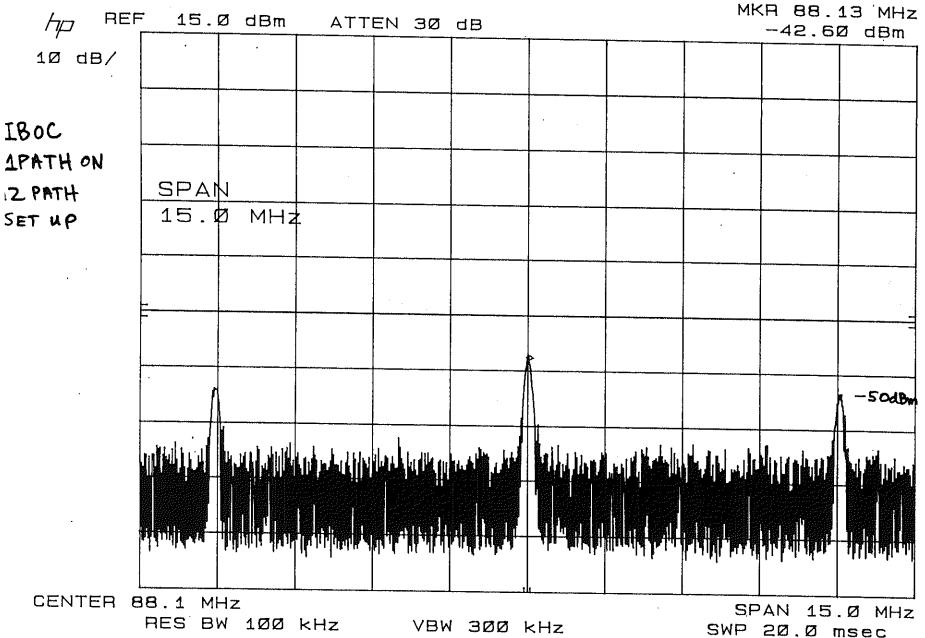
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DAR90330.PRO

DAR90340.PRO

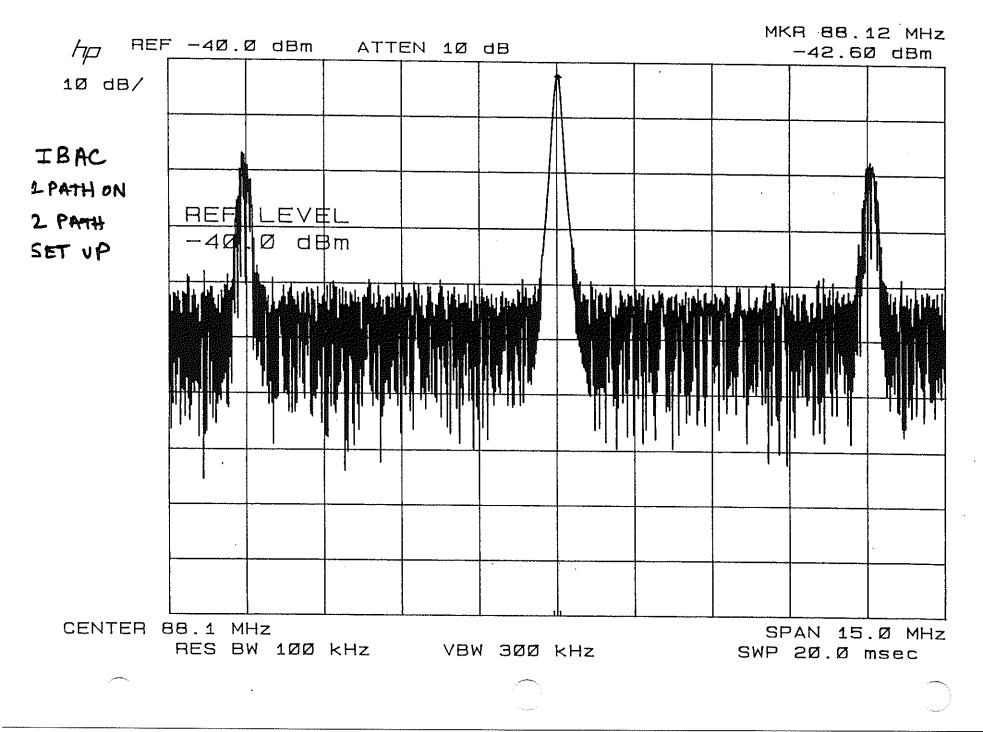
DAR90350.PRO





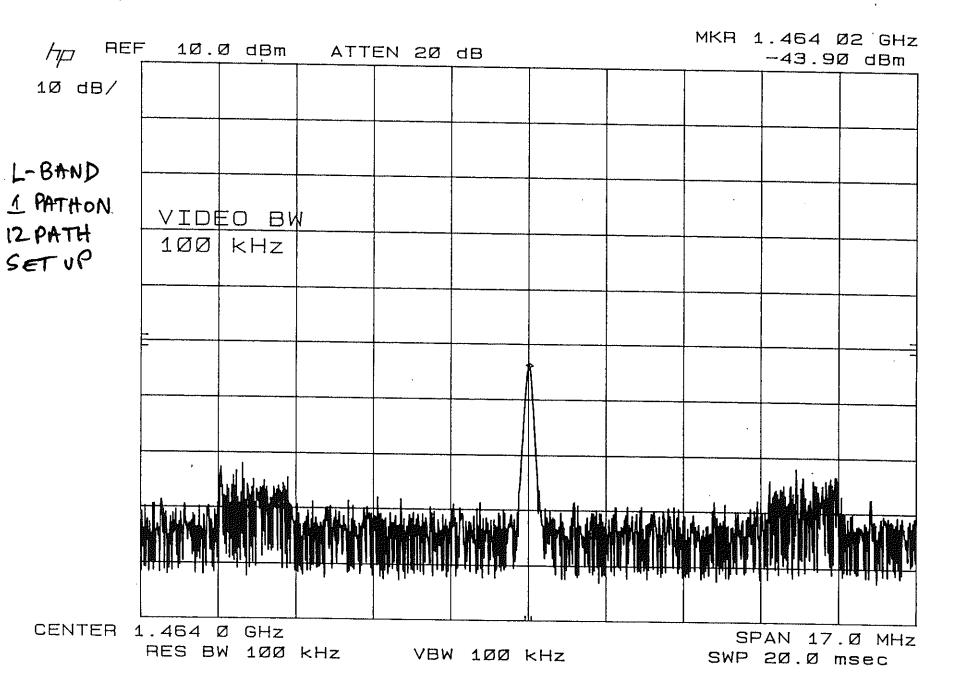
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NRSC Document Improvement Proposal

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