NRSC REPORT

NATIONAL RADIO SYSTEMS COMMITTEE

NRSC-R50
Digital Audio Radio
IBOC Laboratory Tests

Transmission Quality Failure Characterization and Analog Compatibility of IBOC Systems

August 11, 1995

Part II - Appendices A through L



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NRSC-R50

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NRSC-R50

FOREWORD

NRSC-R50, Digital Audio Radio – IBOC Laboratory Tests – Transmission Quality Failure Characterization and Analog Compatibility of IBOC Systems, presents the results of digital radio system tests conducted jointly by the Electronics Industries Association (EIA, precursor to CEA) Subcommittee on Digital Audio Radio (DAR) and the NRSC Digital Audio Broadcasting (DAB) Subcommittee (now the DRB Subcommittee).

Seven different digital radio systems were involved in the joint EIA/NRSC test program—three FM in-band/on-channel (IBOC) systems, one FM in-band/adjacent channel (IBAC) system, one AM IBOC system, the Eureka-147 DAB system (operating at L-band), and a satellite system (operating at S-band). The FM and AM band systems were the only ones considered by the NRSC and consequently the L-band and S-band test results are not included in NRSC-R50. The NRSC chairman at the time of the submission of NRSC-R-50 was Charles Morgan.

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 over-the-air radio broadcasting systems in the United States.

Appendix A – System Descriptions

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 commonly-encountered 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 any 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.

change interesce, and (b) the less common first-adjacent change interesce.

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 frequency-dependent, 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 average 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 flow-controled; 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 µs, and the cyclic prefix is 14.5 µs: 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 raised-cosine 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 single-sidelobe 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

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

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,

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 µs. 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.

REFERENCES

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- [3] J.A.C. Bingham "In-Band Digital Audio Radio: an Update on the AT&T/Amat1 PAC/DMT Solution", 2nd Intl. Symp. for DAB, Toronto, March, 1994.
- [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

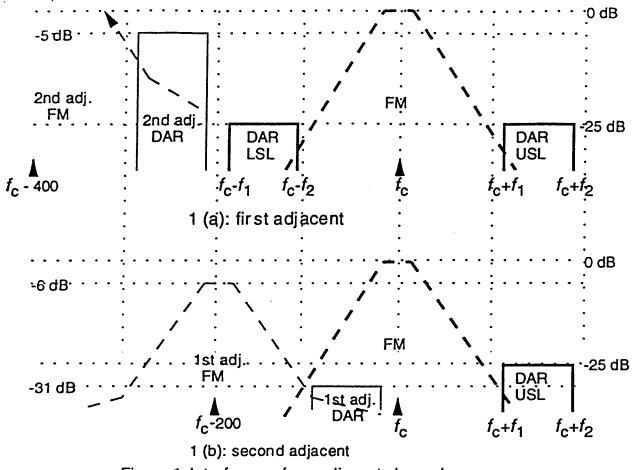


Figure 1. Interference from adjacent channels

AT&T-DAR Systems

Ed Chen and Nikil Jayant

NAB Conference and Exposition, April 1995

Perceptual Audio Coding (PAC)

Refined psychoacoustic models for:

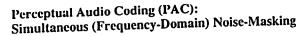
Frequency-domain and time-domain noise-masking Dynamic switching between short and long analysis windows Joint coding of multiple channels (in 2-and 5-channel audio)

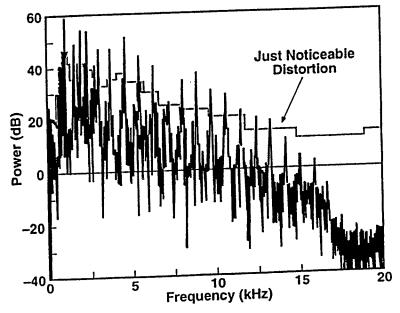
Transmission error concealment

Inexpensive single-chip decoder

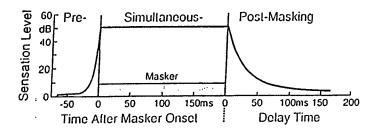
Leading candidate in NBC-contest for MPEG-2 standard

Ideal match to the needs of DAR technology





Perceptual Audio Coding (PAC): Non-Simultaneous (Time-Domain) Noise-Masking



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

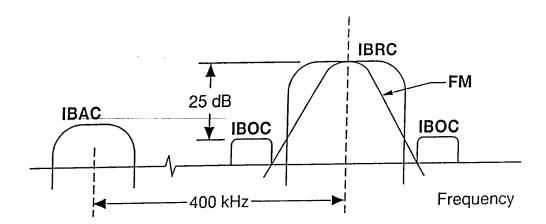
NSJ 3.28.94

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.

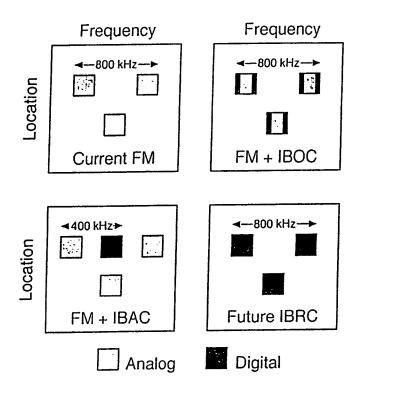
- AT&T (In-Band Adjacent-Channel System)
- AT&T-AMATI (In-Band On-Channel System)
- VOA/NASA/JPL (Satellite System)
- CD-Radio (Satellite System)

Audio Broadcasting in the FM-Band



MH4P866Y03

Audio Broadcasting in the FM-Band



MH4P865Y02

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 $6.6~\mathrm{W}$ and $6.7~\mathrm{KW}$

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

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

KABL Channel 251 (98.1 MHz) at Mt. Beacon, San Francisco
Simultaneous operation of FM and DAB channels
Composite 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

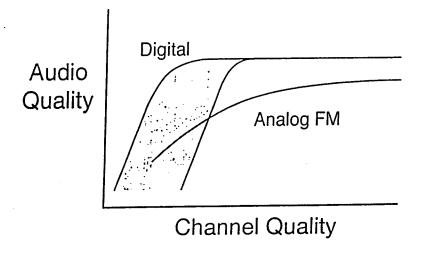
The proposed IBAC transmitter power is 5.0 KW ERP

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 air
IBRC 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.

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 planned 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 very 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 wream, and a proprietary error-concealment procedure at the receiver. The concealment according 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.

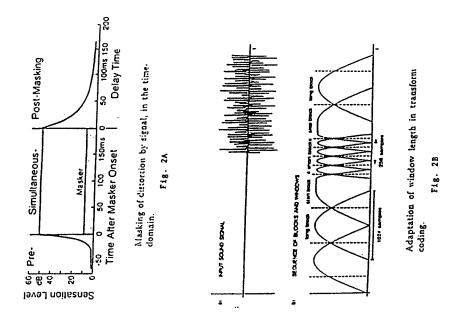
The additional data capacity in the DAR system is on the order of 10 to 20 kbps. This includes synchronous data that are multiplexed to the PAC bit stream as well as asynchronous data that can be added when the (constant-quality, variable-rate) PAC algorithm does not need the allocated constant coding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

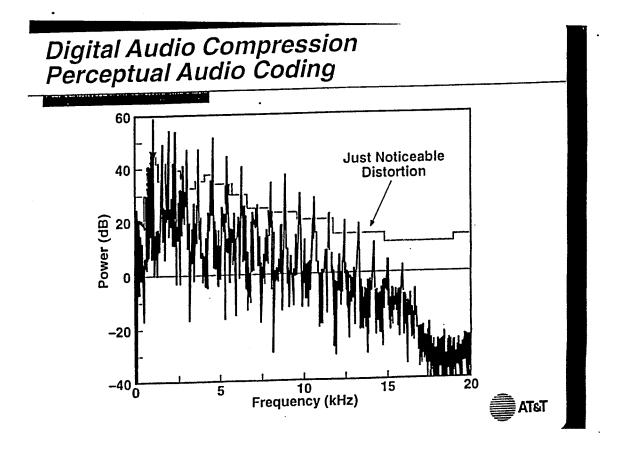
3 The Multichannel Perceptual Audio Coder (MPAC)

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.





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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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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.

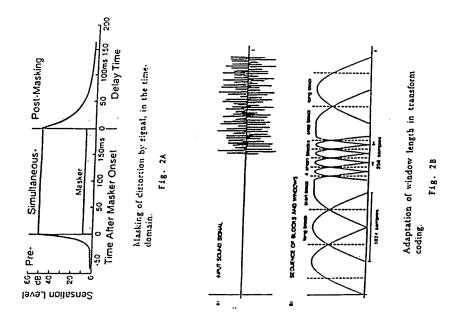
A block diagram of the stereo PAC coder appears in Figure 3. Although the stereo encoder is fairly sophisticated, its design is guided by the need for robust implementation in current signal processing technology. The stereo decoder is quite simple, and it is currently implemented on a single general-purpose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

2 The Application of PAC to DAB Technology

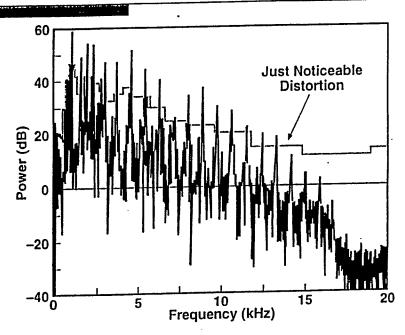
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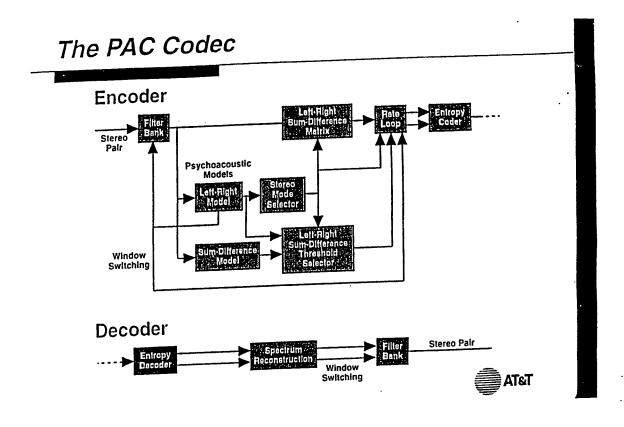
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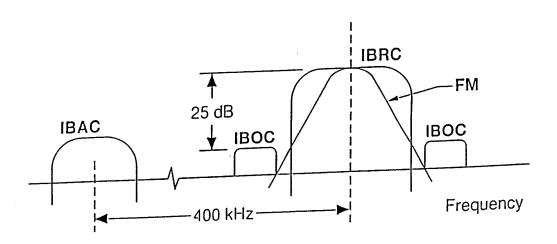
Digital Audio Compression Perceptual Audio Coding

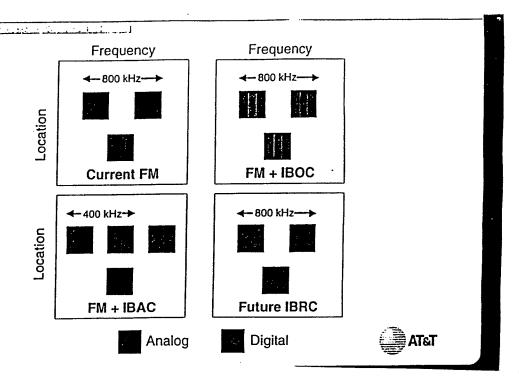






Audio Broadcasting in the FM-Band





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 Dolby AT&T	(MPEG-Layer 2) (AC3) (MPAC)	1	0
		2	1
		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



USA Digital Radio

AM Band In-Band, On-Channel System

332 South Michigan Avenue Suite 605 Chicago, Illinois 60604 Phone, 312, 987, 4454 Fax, 312, 427, 9851 1-800-33-USADR





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.

A.J. Vigil

Sincerely

Engineering Manager

enclosures

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? 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.

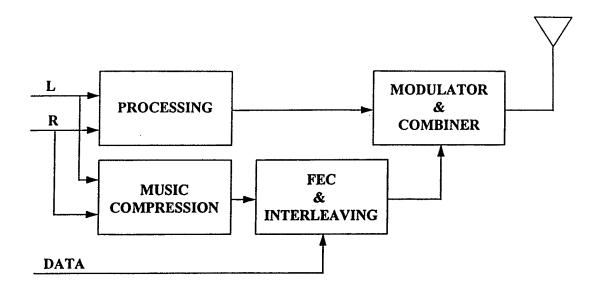


Fig. 1. USADR AM IBOC DAB System Block Diagram.

USA DIGITAL RADIO

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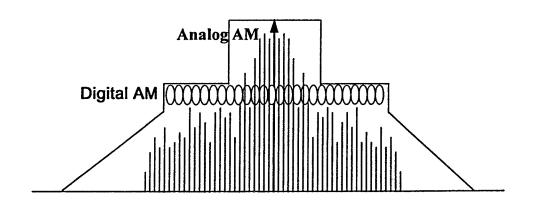


Fig. 2. USADR AM IBOC DAB Modulation Spectrum.



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.

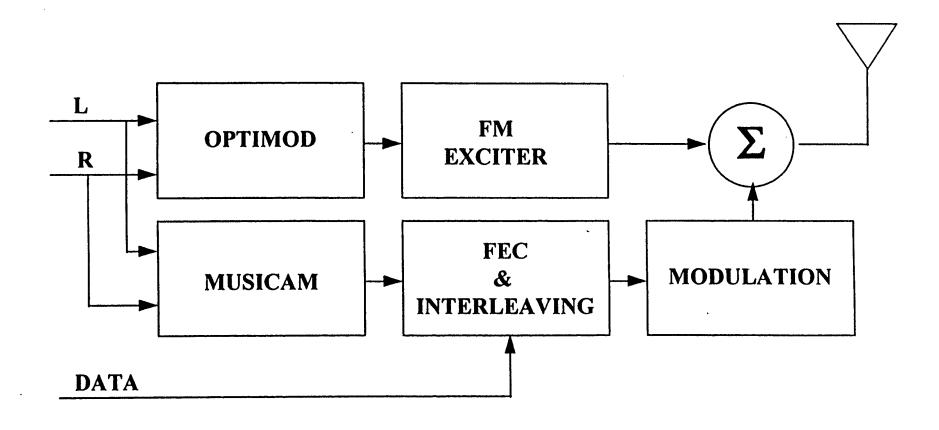
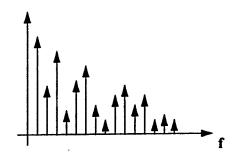


Fig. 1. USADR FM-1 IBOC DAB System Block Diagram.



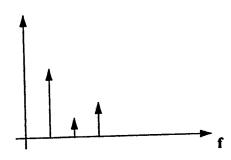


a) High Data Rate Audio Coding

Fig. 2. Variable Data Rate Music Coding.

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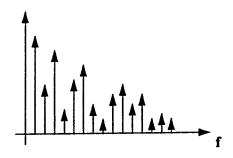
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b) Low Data Rate Audio Coding

Fig. 2. Variable Data Rate Music Coding.



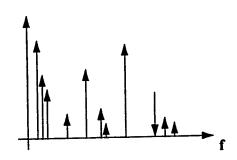


a) Accurate Reproduction

Fig. 3. Comparison of Audio Failure Modes.

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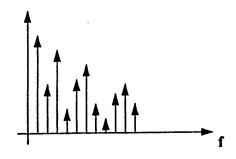
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b) Catastrophic Failure

Fig. 3. Comparison of Audio Failure Modes.



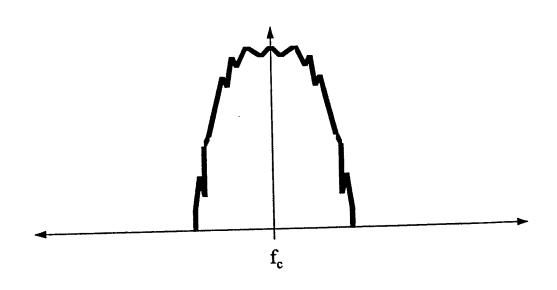


c) Graceful Degradation

Fig. 3. Comparison of Audio Failure Modes.

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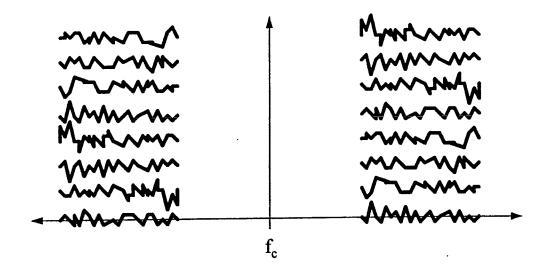
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a) FM Spectrum

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



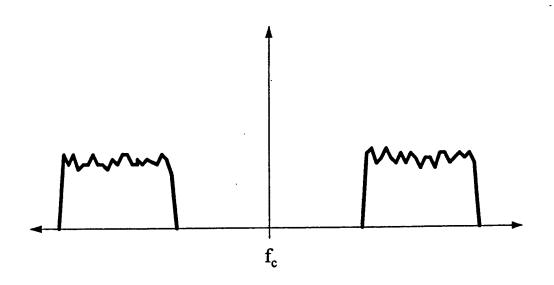


b) DAB Subchannel Spectra

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

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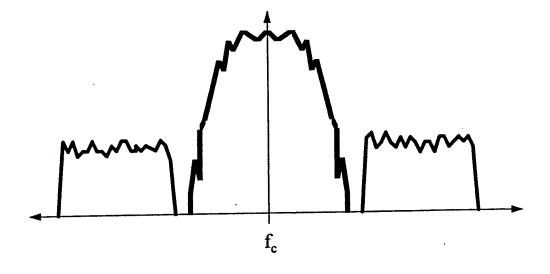
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c) DAB Composite Spectrum

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



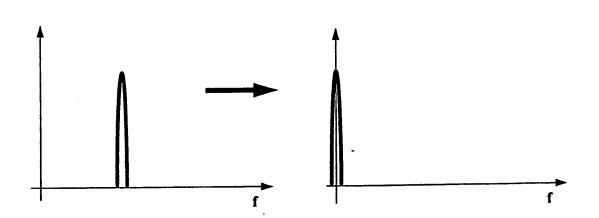


d) Combined FM + DAB Spectrum

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



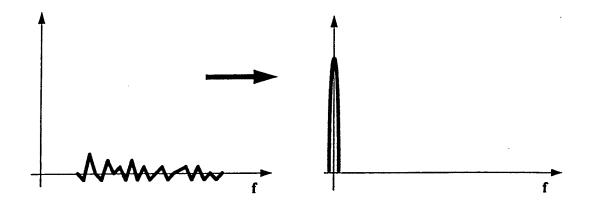
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a) Clear Propagation Channel, Narrowband Data Subchannel

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



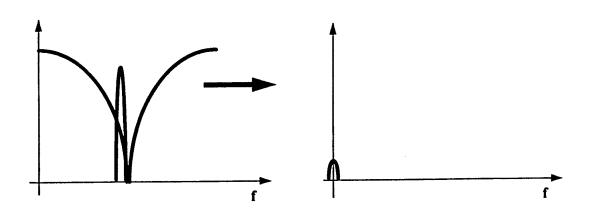


b) Clear Propagation Channel, Wideband Data Subchannel

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

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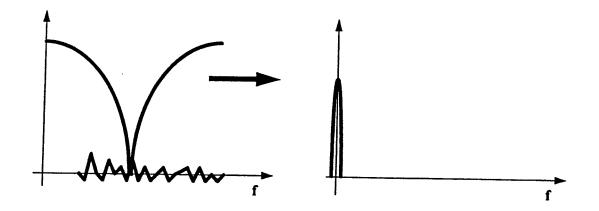
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c) Multipath Propagation Channel, Narrowband Data Subchannel

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





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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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 representation of the USADR FM-2 IBOC DAB modulation composite, subject to 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 fact that the orthogonality 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.

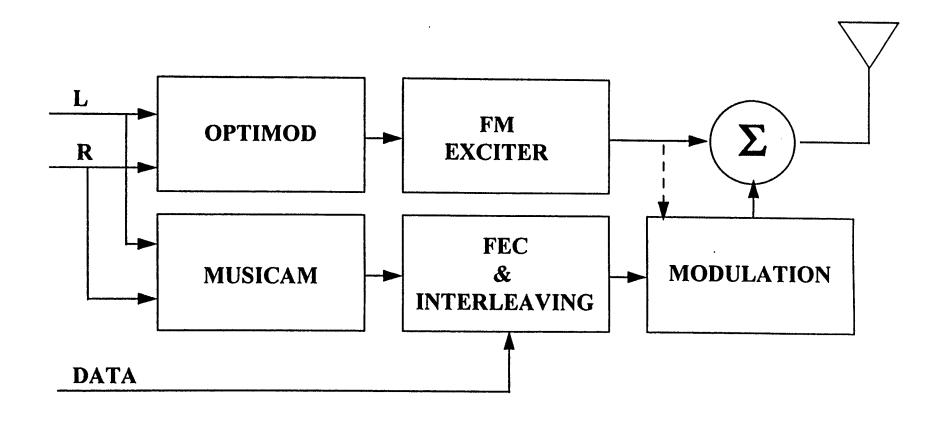
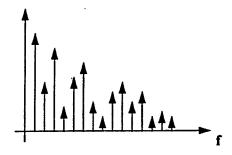


Fig. 1. USADR FM-2 IBOC DAB System Block Diagram.



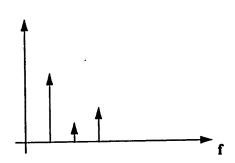


a) High Data Rate Audio Coding

Fig. 2. Variable Data Rate Music Coding.

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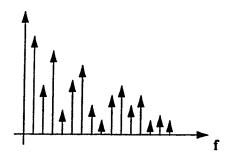
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b) Low Data Rate Audio Coding

Fig. 2. Variable Data Rate Music Coding.



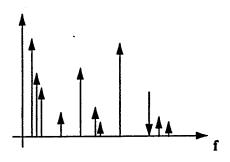


a) Accurate Reproduction

Fig. 3. Comparison of Audio Failure Modes.

USA DIGITAL RADIO

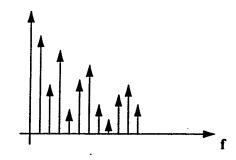
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b) Catastrophic Failure

Fig. 3. Comparison of Audio Failure Modes.



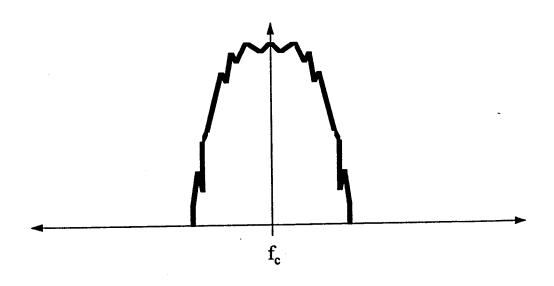


c) Graceful Degradation

Fig. 3. Comparison of Audio Failure Modes.

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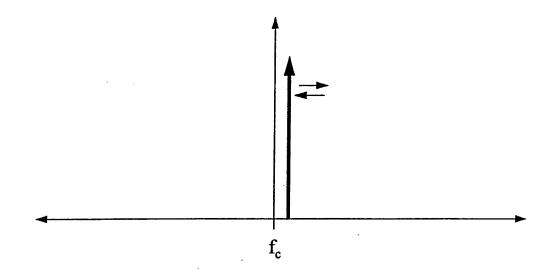
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a) FM, Spectral Representation

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



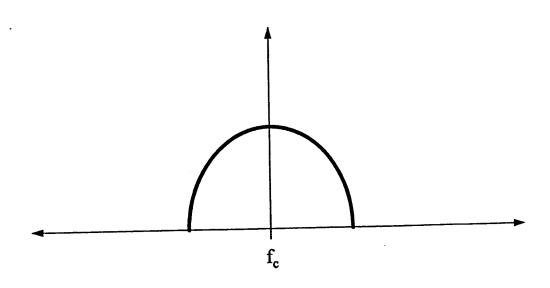


b) FM, Instantaneous Frequency Representation

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

USA DIGITAL RADIO

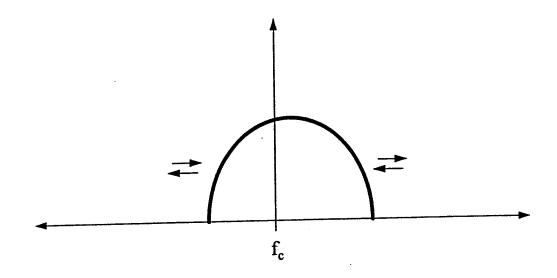
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c) Stationary DAB, Spectral Representation

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



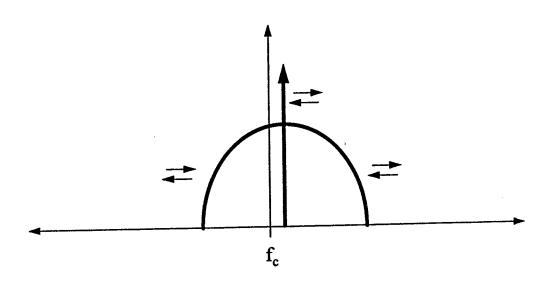


d) DAB, Instantaneous Frequency Representation

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

USA DIGITAL RADIO

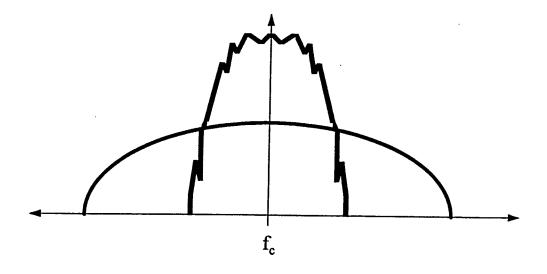
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e) FM + DAB, Instantaneous Frequency Representation

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





f) FM + DAB, Spectral Representation

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



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Appendix B – Unified DAR Laboratory Test Procedures

DAR Laboratory Test Procedures

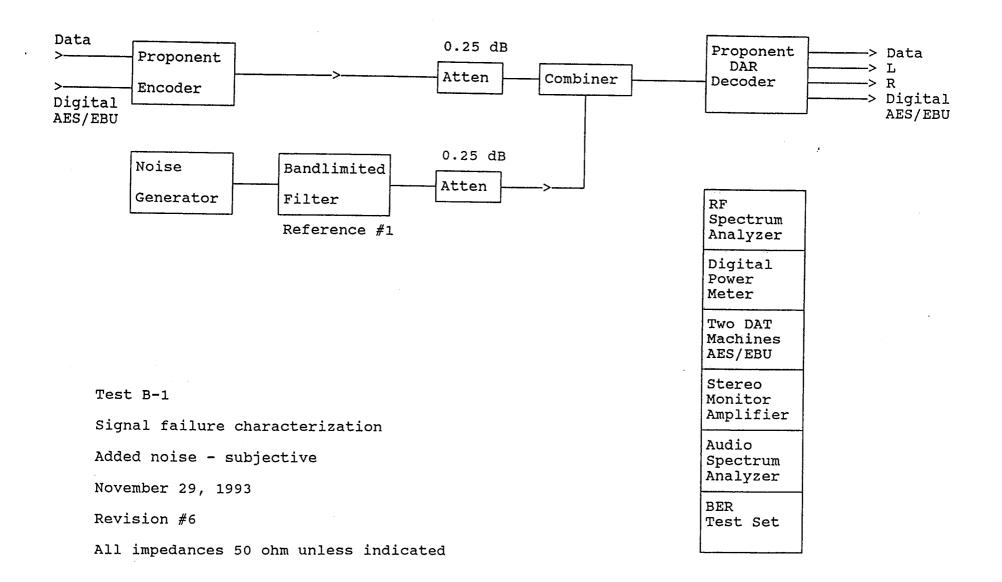
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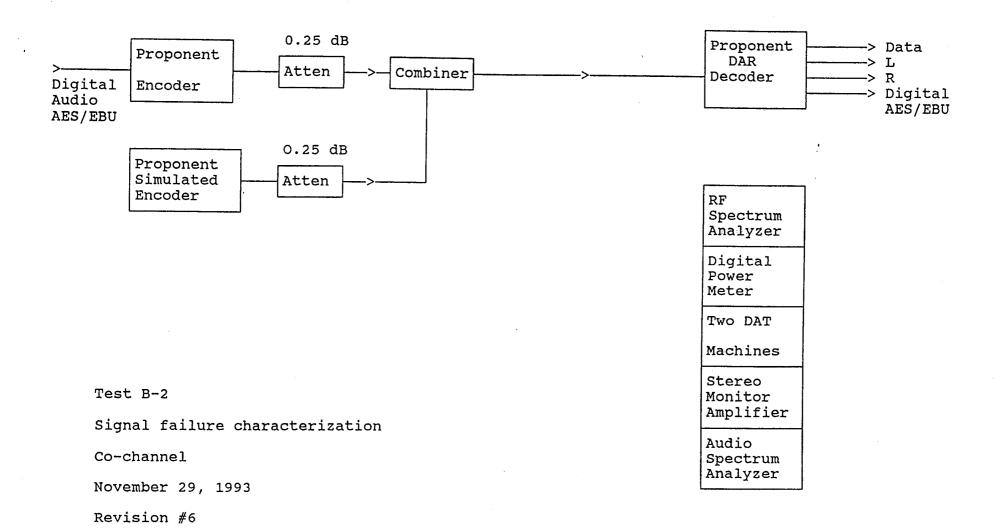
	DESCRIPTION	EVALUATION
Α.	Calibration 1. Daily Check System Power 2. Daily System Spectrum Plot 3. Daily POF 4. Daily audio recording 5. Weekly analog channel proof 6. Reference analog TX proof 7. Weekly calibrate modulation monitors 8. Proponent self check 9. Monthly calibrate test bed	
В.	Signal Failure Characterization 1. Noise 2. Co-Channel 3. Multipath and Noise 3a. AM -> DAR first	- Subjective - Lab EO&C
c.	DAR Performance with Impairment 1. Impulse Noise 2. CW 3. Airplane Flutter 4. Weak Signal Failure 5. Delay Spread/doppler 6. Additional Multipath with Noise 7. Environmental Noise (AM band)	- Lab EO&C
D.	DAR -> DAR with no other Impairments: Co-Channel, First, and Second Adjacent	- Lab EO&C
E.	DAR -> DAR with Multipath: Co-Channel, First, and Second Adjacent	- Lab EO&C
F.	DAR -> Analog no other Impairment: Co-Channel, First, and Second Adjacent	ObjectiveSubjective (analog) EO&C
G.	DAR -> Analog with Multipath on FM: Co-Channel, First, and Second Adjacent	ObjectiveSubjective (analog) EO&C
н.	Analog -> DAR no other Impairment: Co-Channel, First, and Second Adjacent	- EO&C in Lab
I.	Analog -> DAR with Multipath: Co-Channel, First, and Second Adjacent	- EO&C in Lab
J.	Reacquisition (Hysteresis) 1. Failure due to simulated weak signal 2. Failure due to multipath	- EO&C in Lab
к.	Transmission Quality 1. Test materials selection 2. Transmission quality	- Subjective - Subjective - Lab EO&C
L.	IBOC -> Host Analog 1. Proof of Analog channel 2. Interference to host Analog 3. Interference to host Analog 4. Interference with multipath	- Objective - " - Subjective - Lab EO&C
м.	Host Analog -> IBOC 1. Host Analog to DAR 2. Host Analog to DAR with multipath	- Lab EO&C
N.	Multiple Spurious 1. DAR + FM -> FM	- Lab EO&C
0.	Outline of DAR/Subcarrier Compatibility Tests	

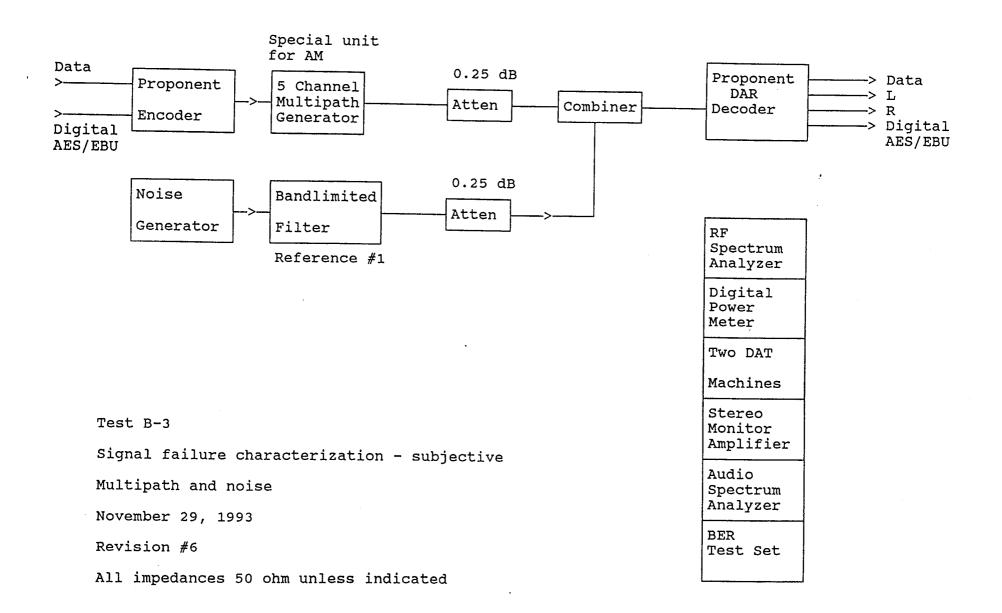
Test Group	Test &	A September 2, 1994 UNIFIED DAR LABORATORY TESTS										
·	Impairment	TEST PROCEDURE Note: The audio impairment test material will be used for the POF test.	Type of Eval	Sig Lev dBf	NB	IBAC	AM IBOC	FM IBOC	DBS	Test Results Data to		
A Calibration	(Daily)	 The average and peak power 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	х	Х	х	x ,	х	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	М	х	х	х	х	X /	Spectrum record in lab log		
	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	х	х	х	х	х	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	х	х	х	х	х	Digital audio recording for the record in lab log		
	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	х	х	NA	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	х	х	х	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	ΝA	NA	NA .	ΝA	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 need	х	х	х	х	×	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	NA	NA	NA	NA	NA	NA	Calibration record in lab log		

REVISION	N #9A Septer	mber 2, 1994 UNIFIED DAR LABORATORY TES	TS			"			···········	
Test Group	Test & Impairment	Note: 1. Impairment audio test material will be used for the digital audio signal. 2. 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. 3. At least three audio segments will be used for each impairment test. 4. 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 characterization of DAR signal failure	1 Noise	1. Gaussian noise will be increased to POF (0.25 dB steps) and the level logged. 2. 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. 3. Steps #1 & #2 will be repeated for each of the three impairment audio segments. 4. The data channel BER will be measured.	Subj & EO&C in Lab	М	х	х	X	×	х	Noise Level at TOA & POF
	2 Co- channel	1. The co-channel interference will be increased to TOA and POF (0.25 dB steps) and the level logged. 2 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. 3. Steps #1 & #2 will be repeated for each of the three impairment audio segments.	Subj & EO&C in Lab	М	х	х	х	х	х	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 TOA and POF are found. (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 two equal points between TOA and POF, and at POF. These digital recordings are for expensiblective assessment.) The noise level and the multipath parameters will be recorded in the laboratory log. The data channel BER will be measured. 	Subj & EO&C in Lab	М	х	х	NA	×	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	х	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.

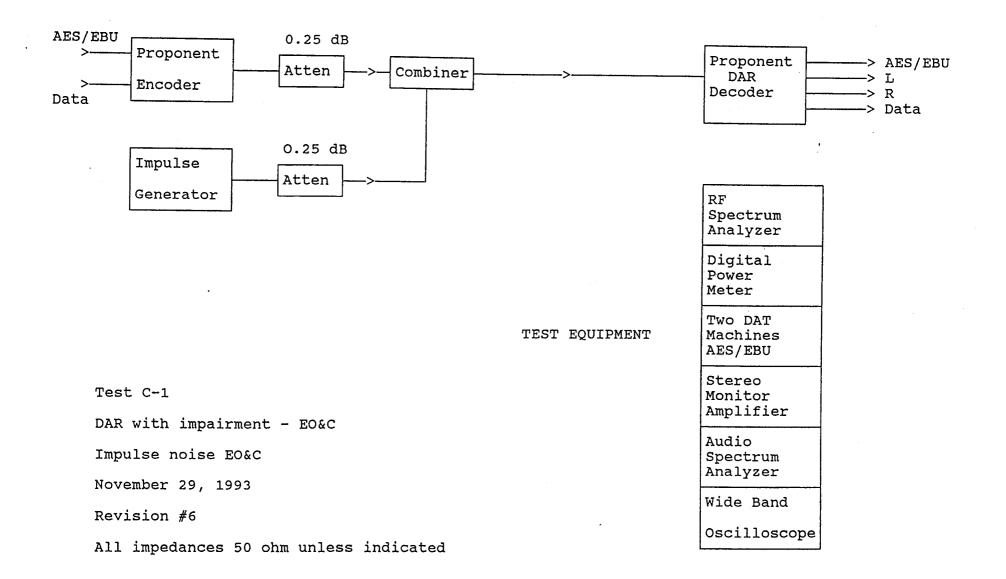


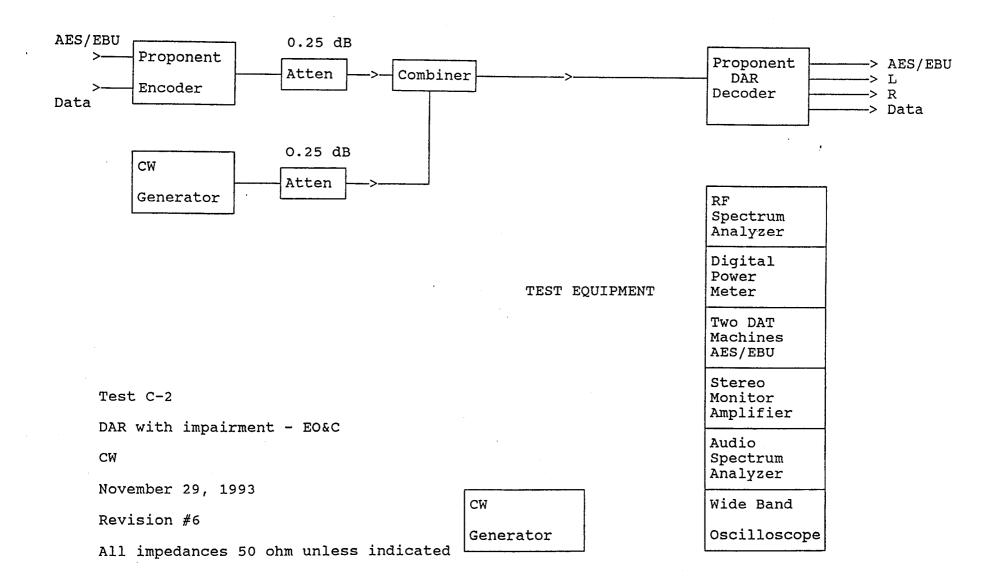


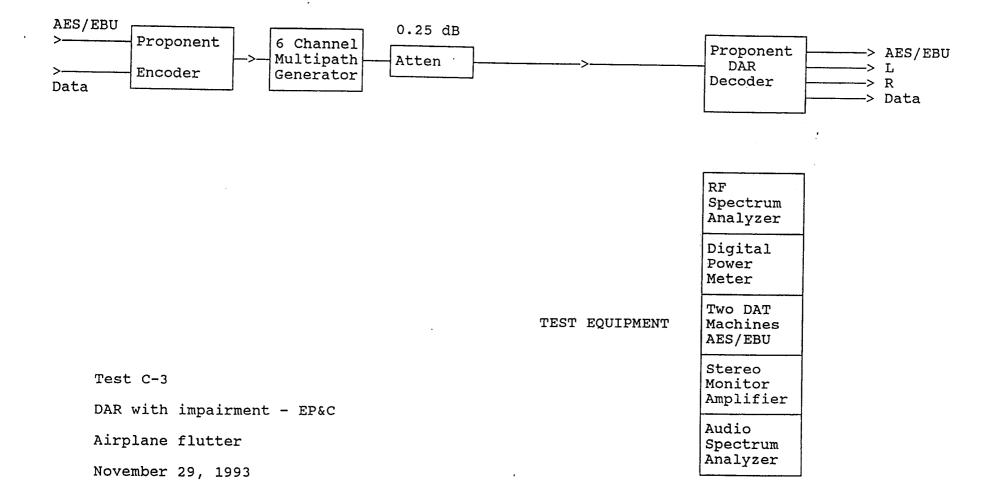


Test Group	V #9A Septen	TEST PROCEDURE	Type of	Sig Lev	NB	IBAC	AM	FM	DBS	T . D . L D
	Impairment	Note: 1. The DAR audio impairment test material will be used for these tests.	Eval	dBf		IBAC	IBOC	IBOC	DB2	Test Results Data to be Recorded
DAR with special impairment	1 Impulse noise	1. A generator capable 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.) 2. The pulse generator output will be mixed with the DAR signal. 3. The amplitude of the pulses will be increased until the TOA and POF is heard by the laboratory specialist.	EO&C	М	х	х	х	х	, x	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 deleted)	х	х	х	х	х	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	W&M	х	х	NA	х	NA	Multipath parameters at TOA & POF
	4 Weak signal	1. Starting with a medium signal level, the signal will be reduced to TOA & POF (0.25 dB steps) (Steps were changed to 1 dB.) 2. A single audio impairment recording will be used for this test.	EO&C	Varying	х	х	х	X	х	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	х	х	NA	х	х	Signal level at TOA & POF
	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	х	x	NA .	х	NA	Signal level at TOA & POF
n	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	х	NA	NA	Interference amplitude at TOA & POF

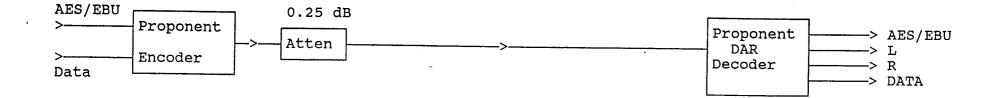
Note: Test procedure changes of September 2, 1994 for tests, C.1.1, C.2 Signal Level, and C.4.1 are shown in Italics







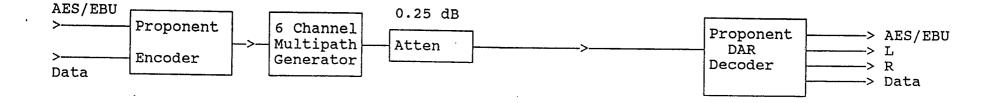
Revision #6



Test C-4
Signal failure characterization - subjective
Weak signal
November 29, 1993
Revision #6

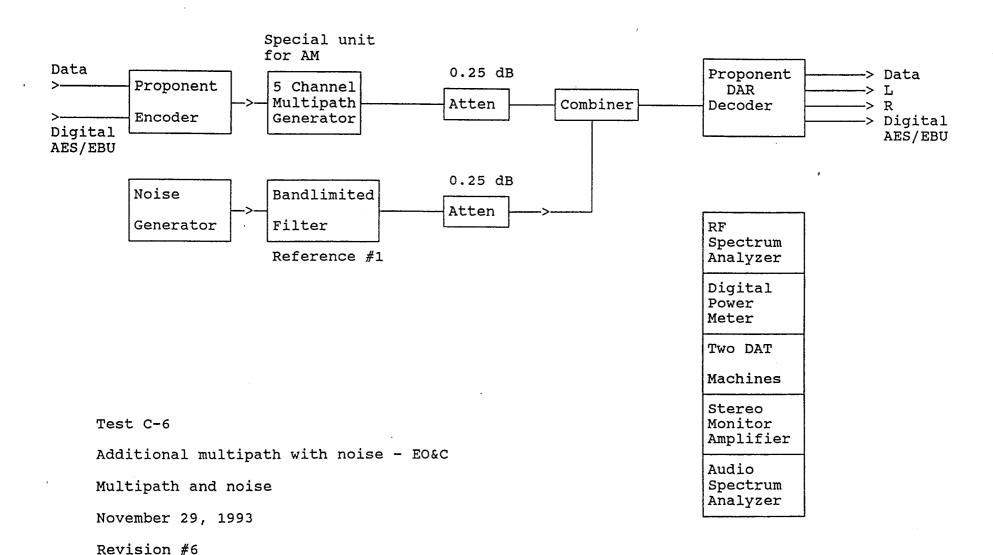
All impedances 50 ohm unless indicated

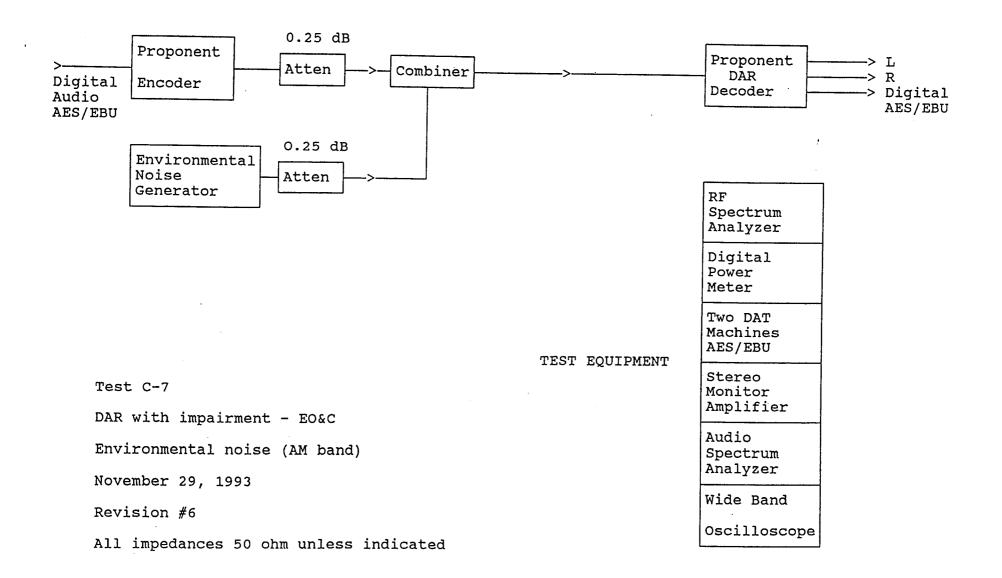
RF
Spectrum
Analyzer
Digital
Power
Meter
Two DAT
Machines
AES/EBU
Stereo
Monitor
Amplifier
Audio
Spectrum
Analyzer



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

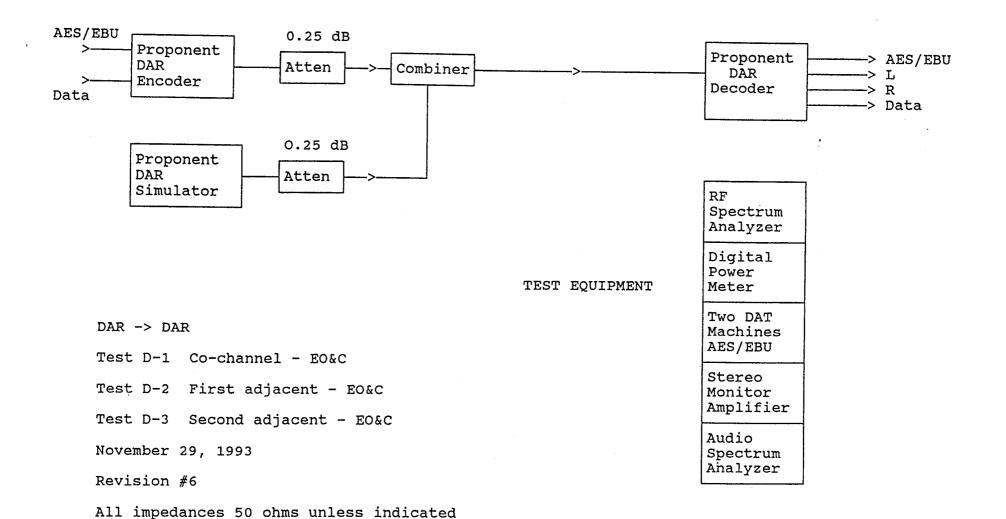
Revision #6





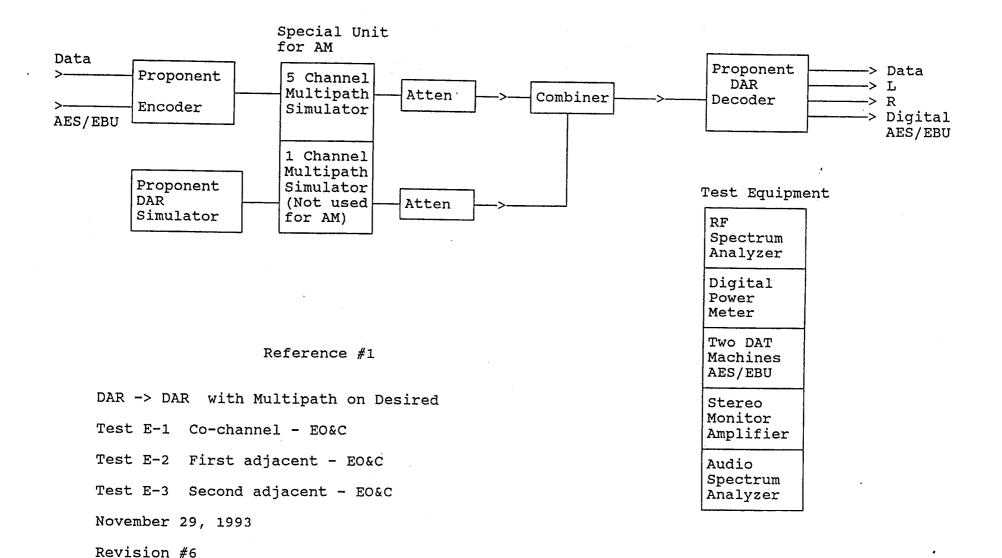
REVISION #	9 November	29, 1993 UNIFIED DAR LABORATOI	RY TESTS	·····						
Test Group	Test Number and Impairment	TEST PROCEDURE Note: 1. The undesired DAR signal will be generated with a simulator supplied by the each proponent. 2. The desired DAB signal will be modulated with unprocessed impairment test audio sequences.	Type of Eval	Sig Lev dBf	NB	IBAC	AM IBOC	FM IBOC	DBS	Test Results & Data to be Recorded
D DAR -> DAR	1 Co-channel	1. The undesired co-channel DAR signal will be increased until the TOA and POF are heard by the lab specialist (0.25 dB resolution). 2. 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	х	х	х	х	Х	D/U & Levels at TOA & POF Data BER at POF -1 dB
	2 First adj	1. The undesired first adjacent DAR signal will be increased until the TOA and POF are heard by the lab specialist (0.25 dB resolution). 2. 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	х	х	×	х	х	D/U & Levels at TOA & POF Data BER at POF -1 dB
	3 Second adj	1. The undesired second adjacent DAR signal will be increased until the TOA and POF are heard by the lab specialist (0.25 dB resolution). 2. 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	х	X	х	NA	D/U & Levels at TOA & POF Data BER at POF -1 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.

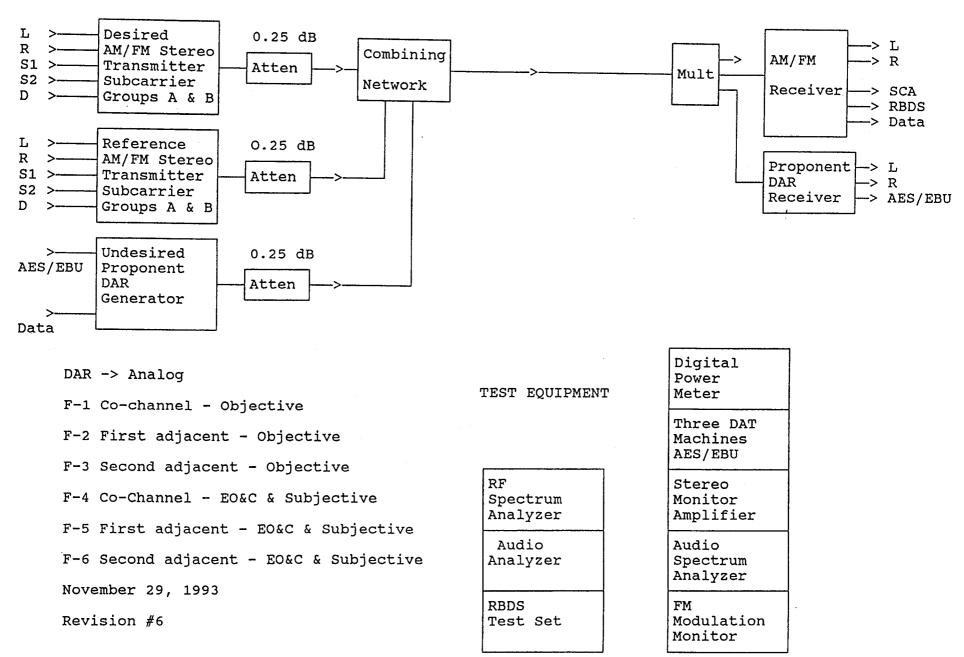


	9 November	29, 1993 UNIFIED DAR LABORATOI	RY TESTS							
Test Group	Test Number and Impairment	Note: 1. The undesired DAR signal will be generated with a simulator supplied by the each proponent. 2. The desired DAB signal will be modulated with unprocessed impairment test audio sequences.	Type of Eval	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 and un-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	М	x	X	X	х	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	х	х	х	х	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	М	NA	х	х	х	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.



REVISION #	9 November	29, 1993 UNIFIED DAR LABORATORY	TESTS							
Test Group	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 Recorded
F DAR -> analog Interference to an analog receiver with no other impairments	1 Co-channel objective	1. 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. 2. 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. 3. 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. 4. 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		х	х	NA	D/U with audi S/N specified RBDS errors for all conditions SCA noise for all conditions Test digital sub
	2 First adjacent	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	Objective	М	NA	х	х	х	NA	Same as F-1
	3 Second adj	below the desired signal frequency, and the results averaged.	Objective	М	NA	х	х	х	NA	Same as F-1
	4 Co-channel subjective	1. 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. 2. The analog transmitters will be setup using the procedure in test F-1 step #2. 3. The DAR transmitter will be gated on and off at 3 second intervals for this test. 4. The desired audio signal will be a moderately processed FM stereo signal. 5. Classical music, rock music, silence, and spoken voice will be used for the audio. 6. 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 tests, the signal/noise will be set at 25 and 40 dB. The noise and analog interference is the reference. 7. 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	х	х	х	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	х	х	х	NA	NA

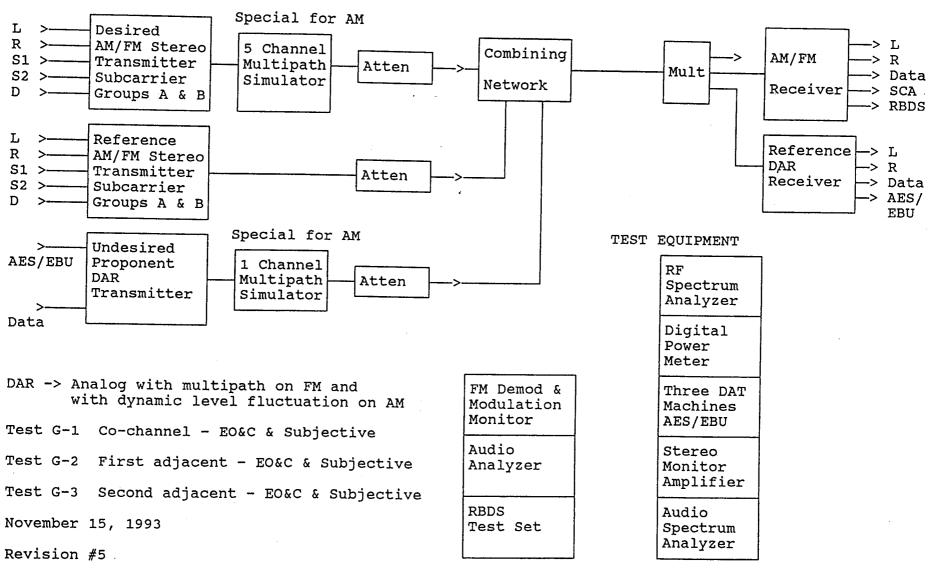


Note: No subcarriers on AM

REVISION #	9 November	29, 1993 UNIFIED DAR LABORATOF	RY TESTS							
Test Group	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
DAR -> analog with multipath Interference to an analog receiver with multipath on the desired and undesired signals	1 Co-channel subjective	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. The AM test will be conducted with amplitude signal level fluctuations. 2. 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. 3. At least one of the FM stereo receivers should have poor AM rejection. 4. The desired audio signal will be a moderately processed AM or FM stereo signal. 5. Classical music, rock music, silence, and spoken voice will be used for the audio. 6. 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. 7. 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. 8. 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 above	Subjective & EO&C in Lab	М	NA	х	х	х	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	х	NA	NA

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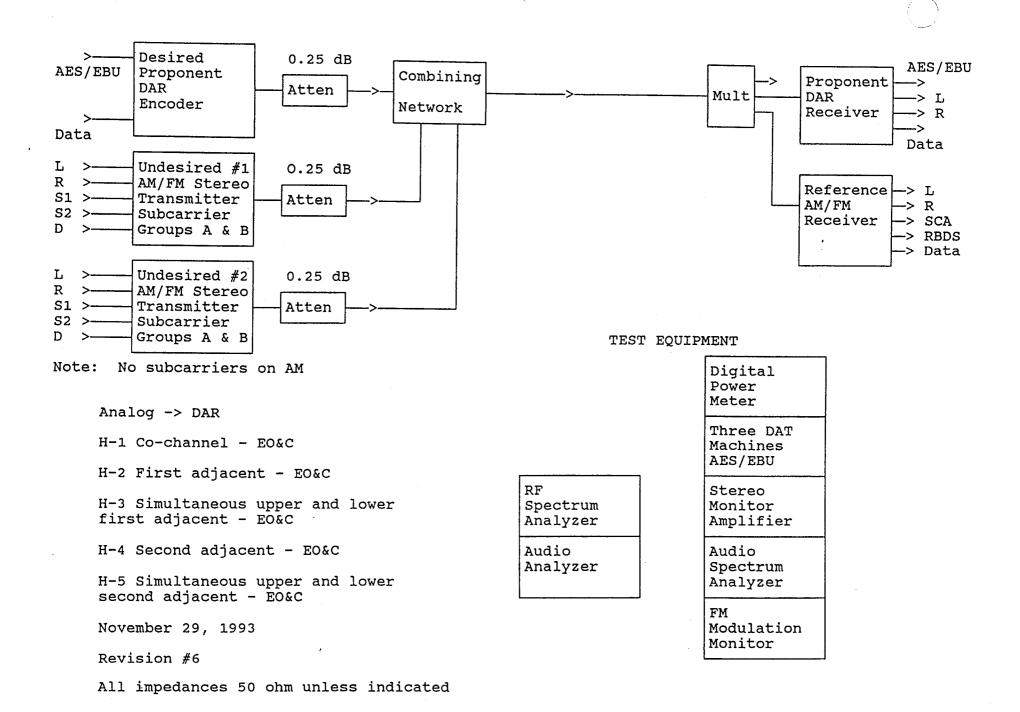
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Note: No subcarriers on AM

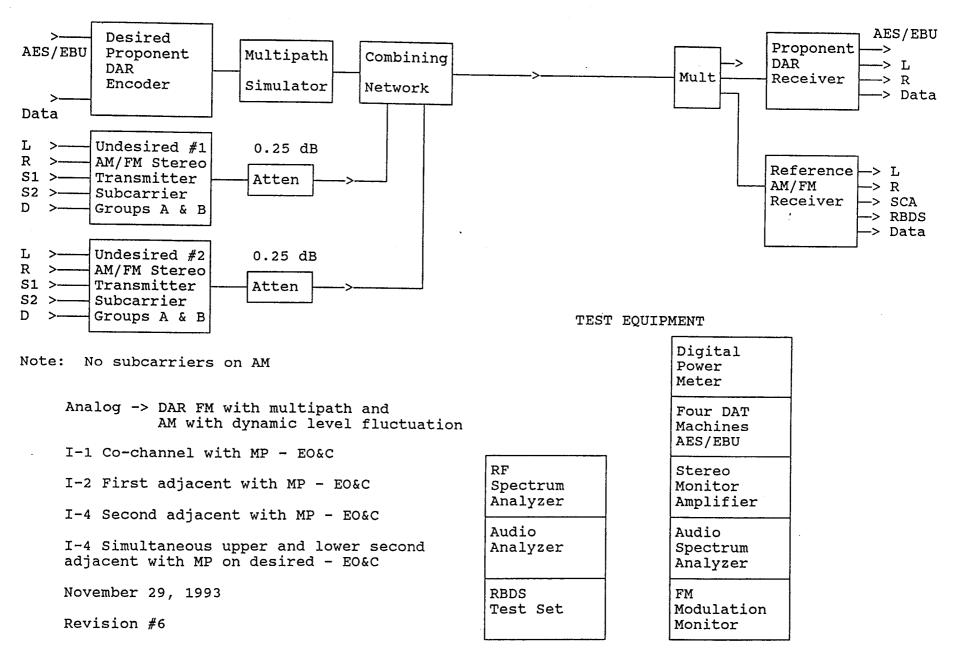
REVISION #9	November	29, 1993 UNIFIED DAR LABORATOR	Y TESTS							
Test Group	Test Number and Impairment	Note: 1. The analog signal will be modulated with processed rock stereo. The FM transmitters will use subcarrier group A and subcarrier group B. 2. The DAB signal will be modulated with the impairment 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-channei	1. 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). 2. For the FM test step #1 will be repeated with subcarrier group B. 3. The TOA will be recorded on digital tape for the record.	EO&C in Lab	М	NA	х	х	х	NA	D/U at POA & POF
impairments	2 First adj	1. 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). 2. For the FM test step #1 will be repeated with subcarrier group B. 3. 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	х	х	х	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	х	х	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	х	х	х	NA	D/U at POA & POF
	5 Simultaneous upper and lower second adjacent	Same as simultaneous upper and lower first adjacent test, H-3.	EO&C in Lab	М	NA	X .	х	х	NA	D/U at POA & POF

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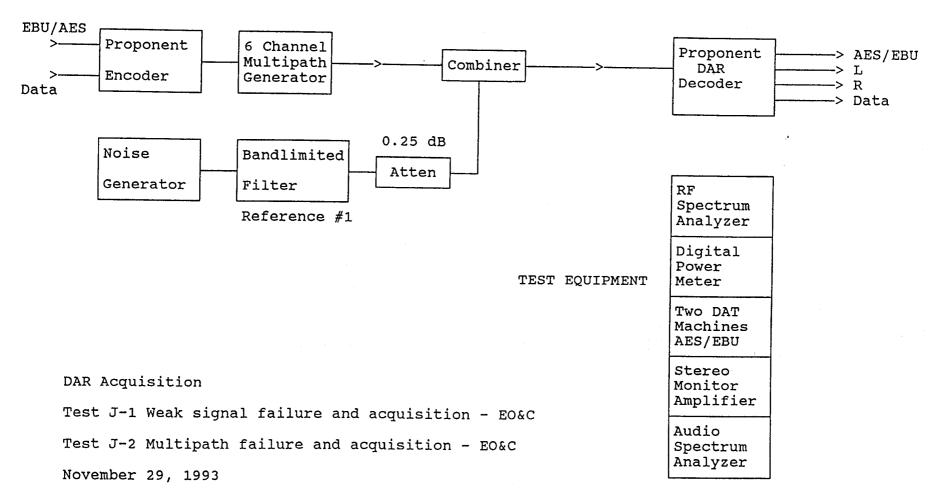
REVISION #9	November 29	, 1993 UNIFIED DAR LABORATOR	y tests							***
Test Group	Test Number and Impairment	Note: 1. The FM signal will be modulated with processed rock stereo, subcarrier group A, and subcarrier group B. The AM audio signal will be processed rock music. 2. The DAB signal will be modulated 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	1. This test will be conducted four times, each with different multipath scenario. The multipath scenarios will be specified by the channel characterization subgroup of Working Group B. 2. 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	х	x	х	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	х	х	х	NA	Same as 1-1
	3 Simultaneous upper and lower first adjacent with multipath	1. This test will be conducted four times, each with different multipath scenario specified by the channel characterization sub-group of Working Group B. 2. 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	х	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	х	х	х	NA	Same as 1-1
	5 Simultaneous upper and lower second adjacent with multipath	Same as simultaneous first adjacent channel test, 1-3. 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	х	х	х	NA	Same as I-1

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REVISION #	9 November	29, 1993 UNIFIED DAR LABORATORY TES	TS					<u></u>		
Test Group	Test Number and Impairment	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	х	х	х	х	х	1. Level at POF 2. Acquisition time at each noise level 3. Hysteresis
	2 Simulated acquisition with multipath and noise	1. 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. 2. Noise will be added until the signal fails. 3. The DAR transmitter will be disconnected from the receiver to assure loss of lock. 4. A different scenario will be selected. 5. 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. 6. 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. 7. For IBOC only, EO&C comments will be made on the quality of the analog signal.	EO&C in Lab	М	х	x	NA	х	х	Acquisition time for each multipath and noise scenario.

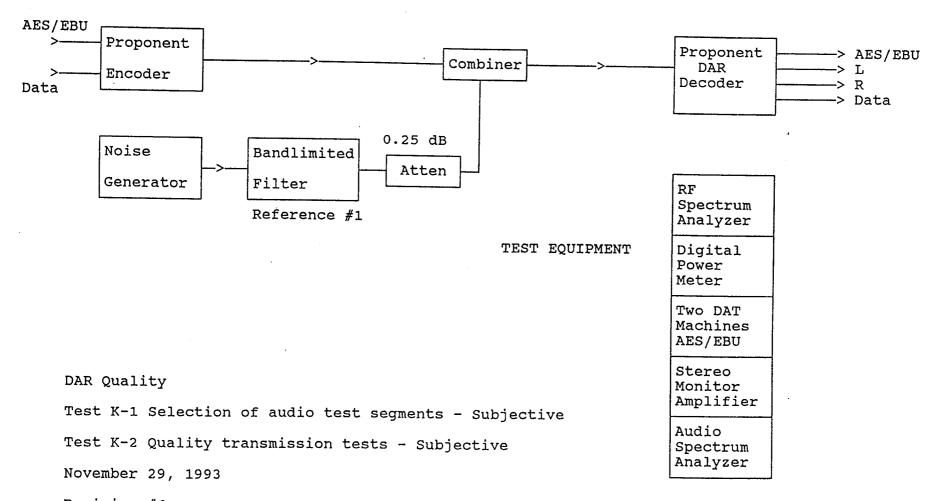
Note: The detailed procedure for noise measurements is supplied in a separate document.



Revision #6

REVISION	#9 November	29, 1993 UNIFIED DAR LABORATOR	RY TESTS							
Test Group	Test Number and Impairment	Test Description	Type of Eval	Sig Lev dBf	NB	IBAC	AM IBOC	FM IBOC	DBS	Test Results & Data to be Recorded
K DAR quality	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 8 segments that are capable of evaluating system quality. Select at least 3 segments for testing systems with transmission impairments. 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	М	x	х	х	х	х	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	х	х	x	х	х	Level at POF Noise at POF

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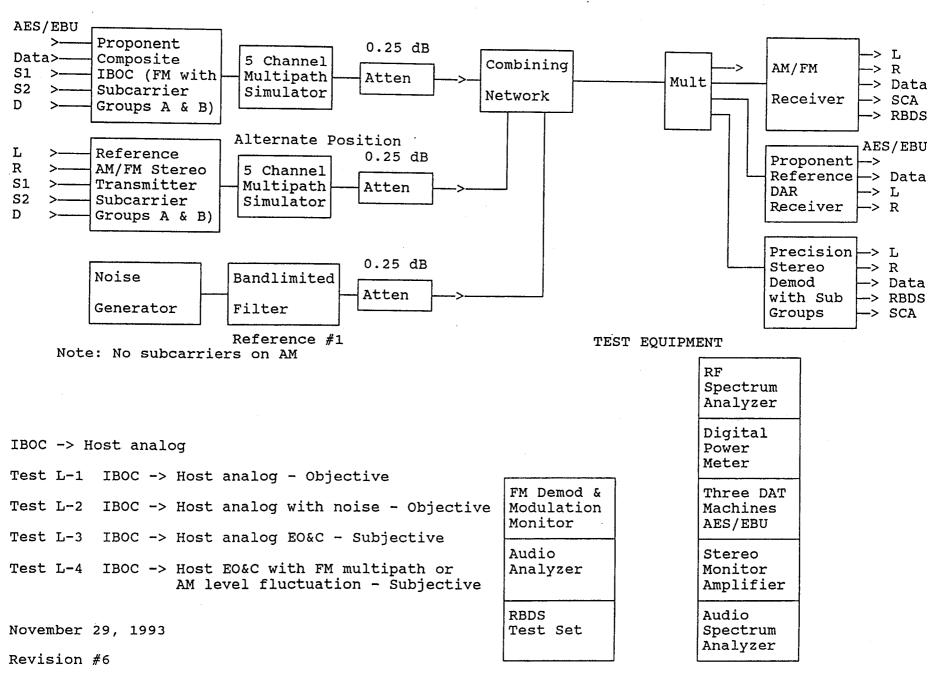
Revision #6

REVISION #9	November 29,	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	NВ	IBAC	AM IBOC	FM IBOC	DBS	Test Results & Data to be Recorded
L DAR -> analog	1 Host analog quality	1. 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	х	x	NA	Performance with and without subs.
IBOC to host analog	2 IBOC to host analog	1. 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. 2. 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. 3. A separate analog transmitter will be used for the reference. 4. For each test receiver, a plot of the input/output characteristics of the IBOC analog signal and the analog reference will be made. 5. The performance of both subcarrier groups will be objectively measured.	Objective	Varying W to S	NA	NA	х	х	NA	Plot input/output audio (distortion & freq. response) Subcarrier performance both digital & analog
	3 IBOC to host analog	1. The same receivers used for test L-2 will be used for this test. 2. The desired audio signal will be moderately processed. 3. Classical music, rock music, silence, and spoken voice will be used for the audio. 4. 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. 5. For each test receiver, a digital audio recording will be made of the IBOC analog audio signal and each receiver with the analog reference. 6. Both subcarrier groups will be monitored for quality.	Subjective EO&C	w&s	NA	NA	х	х	NA	Subjective
	4 IBOC to host analog with multipath	1. The same receivers used for test L-2 will be used for this test. 2. The desired audio signal will be a moderately processed. 3. Classical music, rock music, silence, and spoken voice will be used for the audio. 4. The procedure outlined in L-3 step #4 will be used to setup the transmitters. 5. Both subcarrier groups will be used for this test. 6. 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. 7. 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	х	NA	RBDS and digital errors

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g week to the

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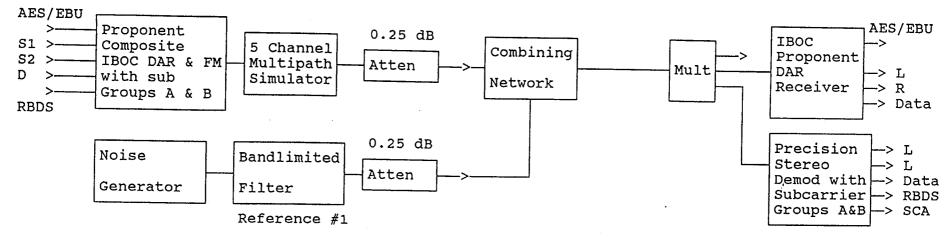


REVISION #9	November 2), 1993 UNIFIED DAR LABORATORY	TESTS							
Test Group	Test	TEST PROCEDURE Note: 1. The signal level for the composite IBOC is for both digital and host FM in dBf. 2. IBOC digital always includes the host analog signal. 3. The analog signal will be heavily modulated with processed 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 dBf	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	1. Four multipath scenarios selected by the RF channel characterization subgroup of WG B will be used for this test. 2. Amplitude varying signals will be used for the AM test. 3. The IBOC analog modulation will be alternately switched on and off while listening to the DAR audio for impairments. 4. Both FM subcarrier modes (A & B) will be switched on and off while listening to the DAR audio for changes in impairments 5. The test results will be recorded on digital audio tape.	EO&C in Lab	W&M	NA	NA	х	х	NA	EO&C

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Note: No subcarriers on AM

TEST EQUIPMENT

		RF Spectrum Analyzer
IBOC Host FM -> DAR		Digital Power Meter
Test M-1 Host FM -> DAR - EO&C Test M-2 Host FM -> DAR with multipath - EO&C	FM Demod & Modulation Monitor	Two DAT Machines AES/EBU
November 29, 1993	Audio Analyzer	Stereo Monitor Amplifier
Revision #6 All impedances 50 ohm unless indicated	RBDS Test Set	Audio Spectrum Analyzer

	·	UNIFIED DAR LABORATORY TESTS							
Test Group	Test Number and Impairment	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)	1. 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.		NA	NA	х	NA	NA	30 dB Audio 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.		NA	х	х	NA	NA	Changes in Subjective Interference

July 17, 1995

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

Subcarrier Test Gro	up A	Subcarrier Test	Group B	Subcarrier Test Group C (Undesired)	Subcarrier Test Group D			
RBOS	3X	RBOS	10%	67 kHz 10%				
66.5 kHz Digital	8.5%	67 kHz	10%	92 Khz 10%				
92 kHz	8.5%				92 kHz HS Digital Data 10%			

	Subcarri Dar -> Analog er		DAR								
Description	Test	67 kHz SCA Analog Quality	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					
DAR -> Anwilog Interference to analog receiver with no	F-2 First adjacent objective	x	x	x	x	x					
	F-3 Second adjacent objective	х	x	x	X	x					
	F-4 Co-channel subjective	. х	x	NA.	. NA	NA					
	F-5 First adjacent subjective	х	x	NA	NA	NA					
other impairments	F-6 Second adjacent subjective	х	×	NA	NA	NA					
G	G-1 Co-channel subjective	х	x	X	x	X					
DAR -> enelog with multipath	G-2 first adjacent subjective	x	x	х	Х	х					
	G-3 Second adjacent subjective	х	x	x	x	х					
H	H-1 Co-channel	FM SIGNAL SUBCARRIER TO DAR COMPATIBILITY TEST									
Analog ->	H-2 First adjacent	и									
DAR no other impairments	H-3 Simultaneous upper & lower first	neous upper & lower first "									
EO&C	H-4 Second adjacent	0									
TOA & POF	H-5 Simultaneous upper & lower second	п									
I	I-1 Co-channel with multipath · FM SIGNAL SUBCARRIER TO DAR COMPATIBILITY TEST										
Analog -> DAR with multipath	1-2 First ed] with multipath	11									
	1-3 Simultaneous up & low with multipath	0									
EO&C TOA & POF	I-4 Second adj with multipath										
	1-5 Simultaneous second up & low with MP	nultaneous second up & low with MP "									
L DAR ->	L-1 IBOC host analog quality objective with end without subcarriers	FM subcarrier -> stereo program audio objective test Subcarrier groups A and B will be independently used									
enalog IBOC to host	L-2 IBOC -> host analog objective	X	х	x	x	χ					
enalog	L-3 180C -> host analog subjective	x	X	NA	NA	NA					
	L-4 IBOC -> host analog with MP subjective	X	х	x	х	х					
M Host Analog to IBOC DAR EO&C in Lab	M-1 Host analog -> IBOC digital	FM SIGNAL SUBCARRIER TO DAR COMPATIBILITY TEST									
	M-2 Host analog -> IBOC digital multipath	FM 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



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 $2\frac{1}{2}$ 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.

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5.0 -- Imperceptible
4.0 -- Perceptible but not annoying
3.0 -- Slightly annoying
2.0 -- Annoying
1.0 -- Very annoying
```

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.

References

- [1] CCIR Draft New Recommendation, Subjective Assessment of Audio Systems with Small Impairments Including Multichannel Sound Systems, Task Group 10/3, Geneva, Switzerland, 1 November 1993.
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- [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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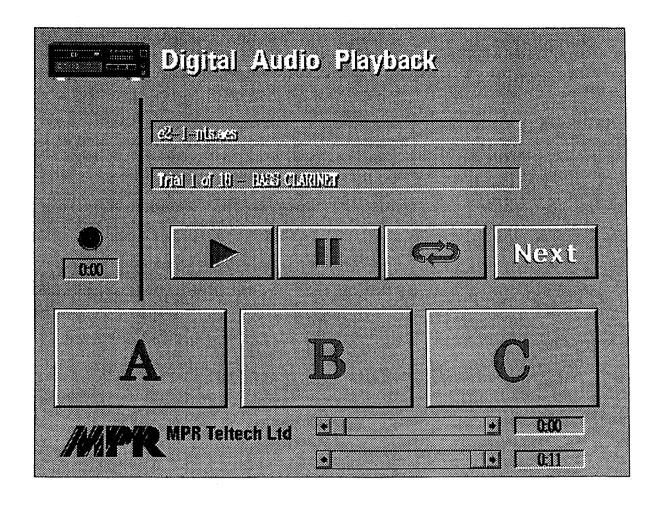


Figure 1 Screen used by listeners during training and blind rating sessions

Appendix D – Delay Spread / Doppler Procedures

TEST C-5 DELAY SPREAD/DOPPLER

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 tandem to generate 12 paths. The simulator has three modes:

- "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.
- "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).
- "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:

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	100	63.6
208.33	2250	150	95.3
312.5	3375	225	143
416.67	4500	300	191

Building the Rayleigh Fading Files:

It is recommended that the fading files be created on the PC which confrols 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

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.

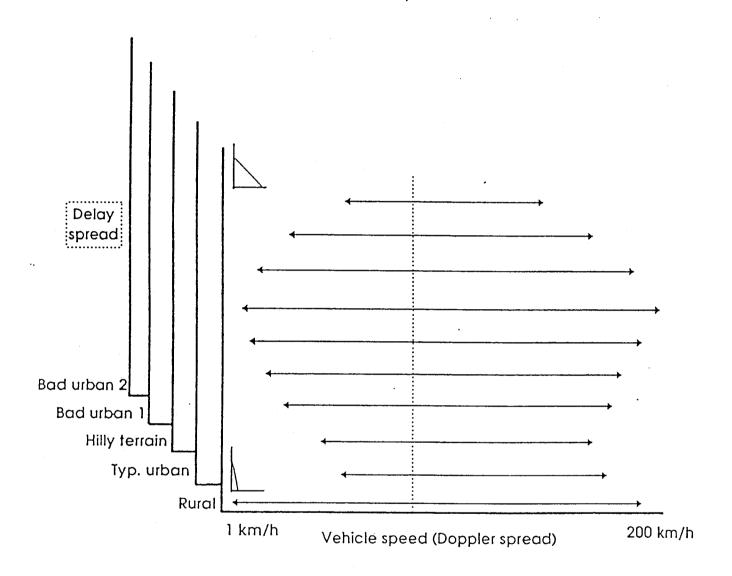


Figure 2: Typical results for one DAR system

Appendix E – HP 11759C Multipath Simulator

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

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

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:SPECTRUM RAY RAY RAY RAY RAY RAY RAY RAY RAY OFF OFF "RAY131.IQ" "RAY131.IQ" "RAY131.IQ" "RAY131.IQ" "RAY131.IQ" "RAY131.IQ" "RAY131.IQ" "RAY131.IQ" 2:IQFILE "RAY131.IQ" "" "" " 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 13.0785 13.0785 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 0.0000

DAR90110.PRO

- Profile #3 2:TITLE "RURAL FAST WITH CO" 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 RAY RAY RAY RAY RAY RAY RAY RAY DOP DOP "RAY131.IQ" "RAY131.IQ" "RAY131.IQ" "RAY131.IQ" "RAY131.IQ" "RAY131.IQ" "RAY131.IQ" "RAY131.IQ" "RAY131.IQ" 2:IOFILE "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 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

DAR90120.PRO

DAR90130.PRO

- Profile #1 0:TITLE "URBAN SLOW DOP WITH CO VHF #1" 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:IQFILE 0:DOPPLER HZ 0.0436 0.0610 0.0785 0.1046 0.1133 0.1221 0.1395 0.1569 0.1744 0.0000 2.0000 1.5000 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 0.0000 - Profile #2 "URBAN FAST DOP WITH CO VHF #2" 1:TITLE 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: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 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 0.0000 - Profile #3 2:TITLE "RURAL FAST DOP WITH CO VHF #3" 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 0.000000c+00 2:SPECTRUM DOP DOP DOP DOP DOP DOP DOP DOP DOP 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 2.6157 7.8471 13.0785 4.3595 8.7190 9.5909 6.1033 10.8988 11.3347 0.0000 2.0000 1.5000 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 3:TITLE "TERRAIN OBSTRUCTED FAST DOP WITH CO VHF #4" 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: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: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

DAR90140.PRO

```
Profile #5
4:TITLE
                          "L-BAND URBAN SLOW '
4:IODATA DIR "C:\chan new"
4:CORR_MODE 3X3
4:DELAY RES LOW
4:CHAN ATTEN 0.000000e+00 0.000000e+00 0.000000e+00 0.000000e+00
4:GROUP LOSS 0.000000e+00
4:SPECTRUM RAY RAY RAY RAY RAY RAY RAY RAY RAY OFF OFF
                        "RAY27.IQ" "" "" ""
4: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
4:DOPPLER_HZ 2.7241 2.7241 2.7241 2.7241 2.7241 2.7241 2.7241 2.7241 2.7241 0.0000 0.0000 0.0000
4: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
4: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
4: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 #6
                          "L-BAND URBAN FAST"
5:TITLE
5:IQDATA_DIR "C:\chan_new"
5:CORR_MODE 3X3
5:DELAŸ 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" "" "" "" "" "" "" "" ""
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 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 8
5: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
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 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 0.0000 0.0000
                                                                     -- Profile #7
6:TITLE
                          "L-BAND RURAL FAST "
6:IQDATA DIR "C:\chan new
6:CORR_MODE 3X3
6:DELAY_RES LOW
6:CHAN ATTEN 0.000000e+00 0.000000e+00 0.000000e+00 0.000000e+00
6:GROUP LOSS 0.000000e+00
6:SPECTRUM RAY RAY RAY RAY RAY RAY RAY RAY RAY OFF OFF 6:IQFILE "RAY2043.IQ" "RAY2045.IQ" "RAY2045.IQ" "RAY2045.IQ" "RAY20
6: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
6:DOPPLER HZ 204.3083 204.3083 204.3083 204.3083 204.3083 204.3083 204.3083 204.3083 204.3083 204.3083 0.0000 0.0000
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 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 0.0000 0.0000
6:CORRELATION 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000
                                                                      - Profile #8
                          " L-BAND TERRAIN OBSTRUCTED "
7:TITLE
7:IQDATA DIR "C:\chan_new"
7:CORR MODE 3X3
7:DELAY_RES LOW
7:CHAN ATTEN 0.000000c+00 0.000000c+00 0.000000c+00 0.000000c+00
7:GROUP LOSS 0.000000e+00
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 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
```

DAR90150.PRO

```
- Profile #5
   4:TITLE
                                 "L-BAND URBAN SLOW WITH CO"
   4:IQDATA DIR "C:\CHAN NEW"
   4:CORR MODE 3X3
   4:DELAY RES LOW
   4:CHAN ATTEN 0.000000e+00 0.000000e+00 0.000000e+00 0.000000e+00
   4:GROUP LOSS 0.000000e+00
   4:SPECTRUM RAY RAY RAY RAY RAY RAY RAY RAY DOP DOP
                               "RAY27.IQ" "RAY27.IQ" "RAY27.IQ" "RAY27.IQ" "RAY27.IQ" "RAY27.IQ" "RAY27.IQ" "RAY27.IQ" "RAY27.IQ" "RAY27.IQ"
   4: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
  4:DOPPLER_HZ 2.7241 2.7241 2.7241 2.7241 2.7241 2.7241 2.7241 2.7241 2.7241 2.7241 0.0000 2.0000 1.5000
  4: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
  4:CORRELATION 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000
                                                                                    - Profile #6
                                "L-BAND URBAN FAST WITH CO"
  5:TITLE
  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 DOP DOP
                               "RAY817.IQ" "RAY81
  "RAY817.IO" "" "
 5: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
 5:DOPPLER_HZ 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 8
 5: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
 5: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
 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 #7
                              "L-BAND RURAL FAST WITH CO"
 6:TITLE
 6:IQDATA_DIR "C:\CHAN_NEW"
 6:CORR MODE 3X3
 6:DELAY_RES_LOW
6:CHAN_ATTEN 0.000000c+00 0.000000c+00 0.000000c+00 0.000000e+00
 6:GROUP LOSS 0.000000e+00
 6:SPECTRUM RAY RAY RAY RAY RAY RAY RAY RAY DOP DOP
 6:IQFILE "RAY2043.IQ" "RAY2043.IQ" "RAY2043.IQ" "RAY2043.IQ" "RAY2043.IQ" "RAY2043.IQ" "RAY2043.IQ" "RAY2043.IQ"
 "RAY2043.IO" "" "" ""
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 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 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 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
                           "RAY817.IQ" "RAY817.IQ" "RAY817.IQ" "RAY817.IQ" "RAY817.IQ" "RAY817.IQ" "RAY817.IQ" "RAY817.IQ" "RAY817.IQ"
"RAY817.IQ" "" "" ""
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 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 81.7233 8
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 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 0.00

DAR90160.PRO

DAR90170 PRO

```
--- Profile #1
0:TITLE
         "L-BAND URBAN SLOW DOPPLER WITH CO #1"
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: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: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 0.0000
                         -- Profile #2
          "L-BAND URBAN FAST DOPPLER WITH CO #2"
1:TITLE
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: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: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 #3
2:TITLE
         "L-BAND RURAL FAST DOPPLER WITH CO #3"
2:IQDATA DIR "C:\chan_new"
2:CORR_MODE NONE
2:DELAY RES HIGH
2:CHAN ATTEN 0.000000c+00 0.000000e+00 0.000000c+00 0.000000c+00
2:GROUP LOSS 0.000000e+00
           DOP DOP DOP DOP DOP DOP DOP DOP DOP DOP
2:SPECTRUM
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: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 #4
3:TITLE
         "L-BAND TERRAIN OBSTRUCTED DOPPLER WITH CO #4"
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: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
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DAR90180.PRO

```
--- Profile #1
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" "
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
1.TITLE
                        "S-BAND URBAN FAST"
1: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
1:SPECTRUM RAY RAY RAY RAY RAY RAY RAY RAY OFF OFF
                     "RAY1129.IQ" "RAY1
"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
1:DOPPLER_HZ 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560
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 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 0.0000
                                                             --- Profile #3
                       "S-BAND RURAL FAST
2:TITLE
2:IQDATA DIR "C:\CHAN NEW"
2:CORR_MODE 3X3
2:DELAY_RES LOW
2:CHAN ATTEN 0.000000e+00 0.000000c+00 0.000000e+00 0.000000c+00
2:GROUP LOSS 0.000000e+00
2:SPECTRUM DOP RAY RAY RAY RAY RAY RAY RAY RAY OFF OFF
                     "RAY2821.IQ" "RAY2821.IQ" "RAY2821.IQ" "RAY2821.IQ" "RAY2821.IQ" "RAY2821.IQ" "RAY2821.IQ" "RAY2821.IQ" "RAY2821.IQ"
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: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 #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 OFF OFF
                     "RAY1129.IQ" "RAY1129.IQ" "RAY1129.IQ" "RAY1129.IQ" "RAY1129.IQ" "RAY1129.IQ" "RAY1129.IQ" "RAY1129.IQ"
"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 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560 112.8560
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
```

DAR90220.PRO

```
-- Profile #1
 0:TITLE
          "Obstructed Path (San Fran 4)"
 0:IQDATA DIR "C:\chan new
 0:CORR MODE NONE
 0:DELAY_RES HIGH
 0:GROUP LOSS 0.000000e+00
 0:SPECTRUM DOP DOP DOP OFF OFF OFF OFF OFF OFF OFF
 0:DOPPLER_HZ 4.4700 3.8700 3.1600 2.6157 3.0517 3.4876 3.9236 4.3595 5.2314 0.0000 2.0000 1.5000
 0:ATTEN DB 8.0000 8.0000 1.0000 3.0000 4.0000 5.0000 2.0000 8.0000 5.0000 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 0.0000
                         -- Profile #2
1:TITLE
          "Rural Highway (Salt Lake City Utah)"
 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 OFF OFF OFF OFF OFF OFF OFF
1:DELAY_US 0.0000 2.0000 12.0000 3.5000 5.0000 8.0000 12.0000 14.0000 16.0000 0.0000 1.0000 2.0000
1:DOPPLER_HZ 4.4700 3.8700 3.1600 2.6157 3.0517 3.4876 3.9236 4.3595 5.2314 0.0000 2.0000 1.5000
1:CORRELATION 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000 0.0000
          "Suburban (Westside Highway 9)"
2:TITLE
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 OFF OFF OFF OFF OFF OFF OFF
2:IOFILE
2:DELAY_US 0.0000 9.0000 2.0000 3.5000 5.0000 8.0000 12.0000 14.0000 16.0000 0.0000 1.0000 2.0000
2:DOPPLER_HZ 4.4000 3.8000 3.1000 2.6157 3.0517 3.4876 3.9236 4.3595 5.2314 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 0.0000
2:ATTEN_DB 10.0000 0.0000 5.0000 3.0000 4.0000 5.0000 2.0000 8.0000 5.0000 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 0.0000
                         - Profile #4
         "Dense Urban (Nova 4)"
3:TITLE
3:IQDATA_DIR "C:\chan_new'
3:CORR_MODE NONE
3:DELAY_RES HIGH
3:CHAN ATTEN 0.000000c+00 0.000000e+00 0.000000e+00 0.000000c+00
3:GROUP LOSS 0.000000e+00
3:SPECTRUM DOP DOP DOP OFF OFF OFF OFF OFF OFF
3:IOFILE
3:DELAY_US 0.0000 20.0000 15.0000 3.5000 5.0000 8.0000 12.0000 14.0000 16.0000 0.0000 1.0000 2.0000
3:DOPPLER_HZ 4.4000 3.8000 3.1000 2.6157 3.0517 3.4876 3.9236 4.3595 5.2314 0.0000 2.0000 1.5000
3:ATTEN DB 14.0000 3.5000 0.0000 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 0.0000
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HP 11759C MULTIPATH PROFILES

DAR90240.DAT

DAR90250.PRO

DAR90260.PRO

DAR90270.PRO

DAR90280.PRO

DAR90290.PRO

DAR90300.PRO

DAR90310.PRO

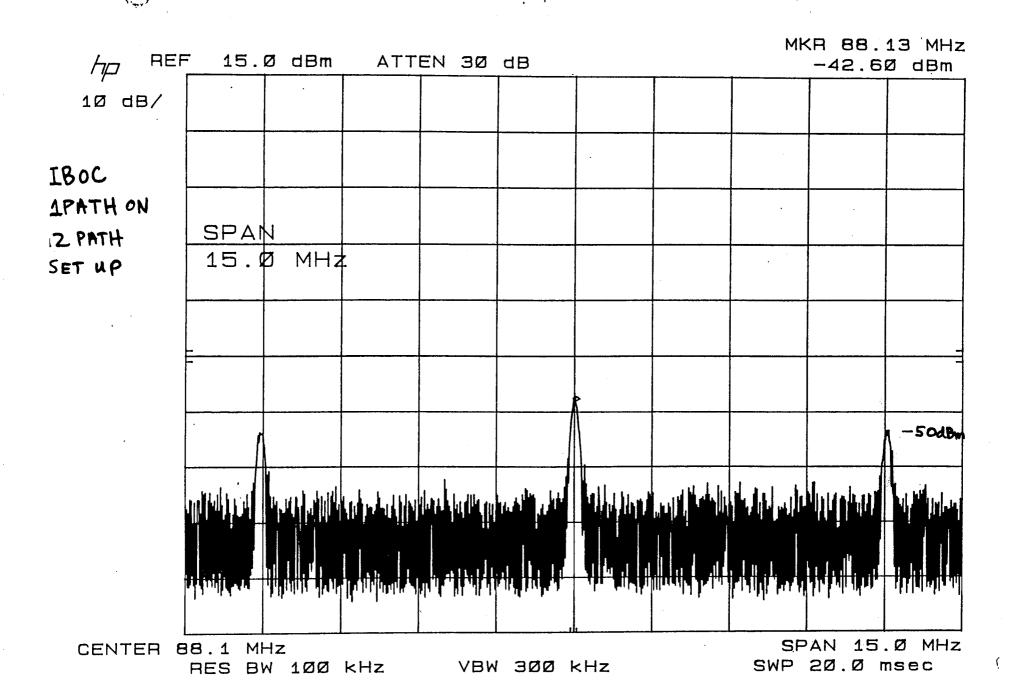
DAR90320.PRO

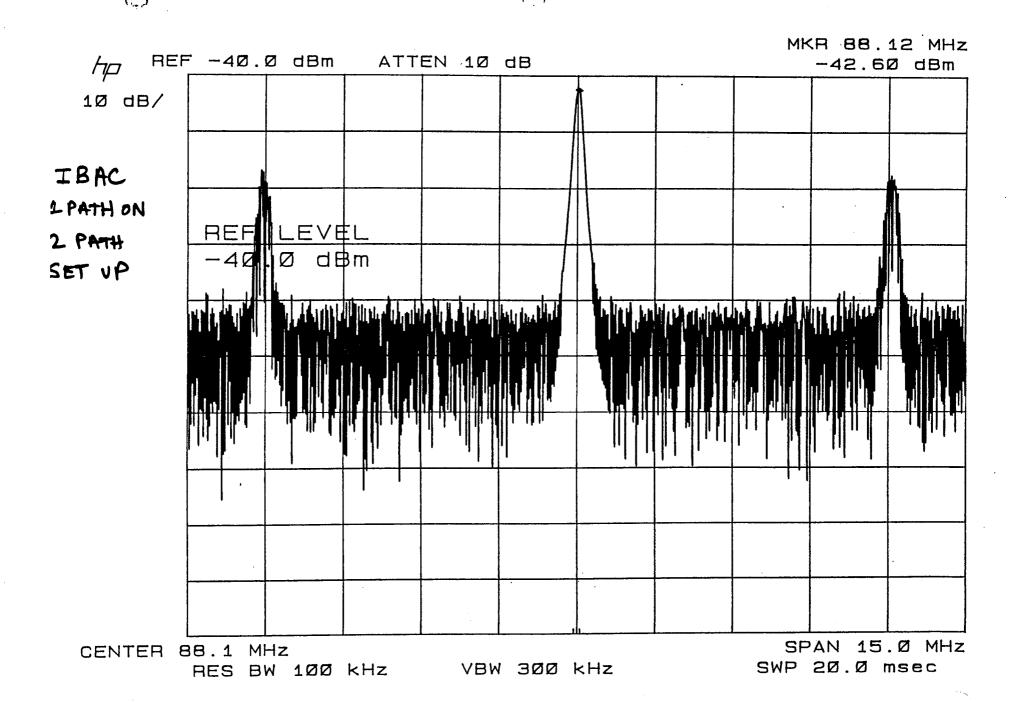
DAR90330.PRO

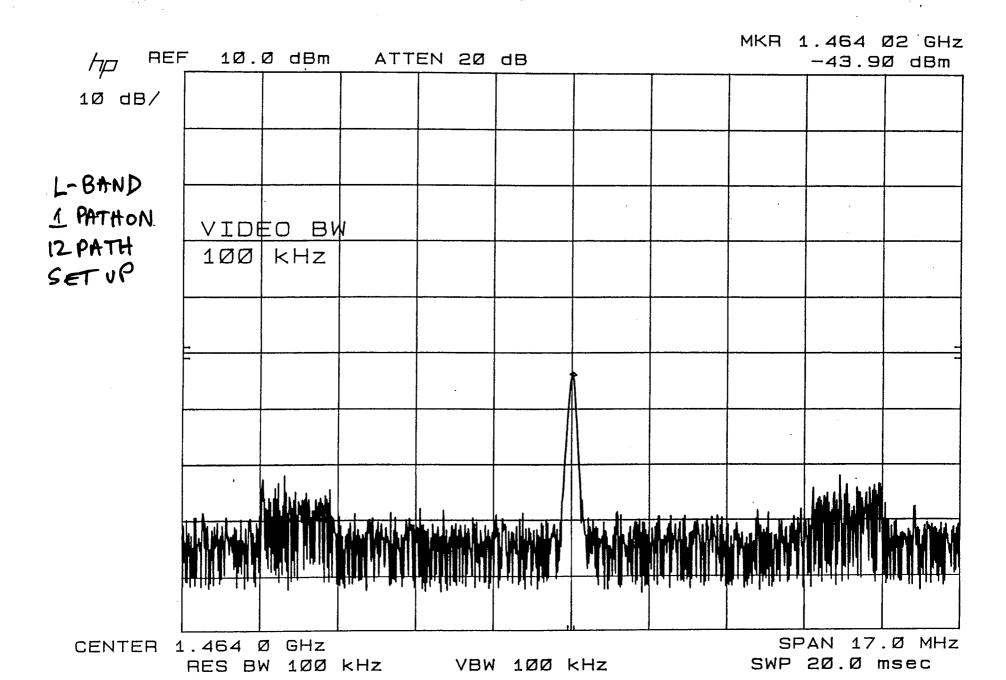
HP 11759C MULTIPATH PROFILES

DAR90340.PRO

DAR90350.PRO







Appendix F – Description of Multipath Profiles

SUMMARY OF:

THE FINAL REPORT OF THE CHANNEL CHARACTERIZATION TASK GROUP; THE DERIVATION AND RATIONAL FOR MULTIPATH SIMULATION PARAMETERS FOR THE EIA-DAR LABORATORY TESTING

NOTE:

This is a condensed version of the above titled report. It follows different section headings but with the same appendix reference as the full report. Only Appendix J is attached to this summary. The full report is available as a separate document.

MULTIPATH CHARACTERIZATION AND CHANNEL SIMULATION BACKGROUND

At the January 22, 1992 meeting of the EIA-DAR Committee, the "Digital Audio Radio Technical Performance and Service Objectives" were discussed and adopted. The requirement for multipath performance testing was set.

The candidate laboratory channel simulator could be directly programmed using time domain values; the relative attenuations, doppler frequencies (or relative phases) and time delays. Searching the literature for channel characteristics in the time domain for direct application to the simulator revealed very little information. A source of direct information on time domain parameters, the characterization test, was required.

In early 1993 the Delco channel test plan system was disclosed. The Delco system description evolves over a number of months as detailed in the series of memoranda and reports in Appendix A. The Channel Characterization data would be collected and then processed to extract the time domain parameters that would then be applied to the laboratory channel simulation. The processing plans are described in Appendix B.

The Hewlett Packard simulator, model No. 11759C was chosen for the laboratory testing. It can be programmed in the Direct mode with the individual channel parameters to simulate a multipath condition. Use of this technique to achieve a dynamic simulation at fixed steps along a path (sequential snapshots) was discussed and the direct control of the simulator based on the actual measured channel characteristics was pursued.

CHANNEL TEST NEEDS; EQUIPMENT, VENUE, ETC.

The channel test program is summarized in Appendix C. Only one city could reasonably be used for channel characteristic testing because of cost and time limits, therefore the test venue should contain many areas that represent as many "difficult" environments as possible.

By May 1993, plans were underway for conducting a channel characterization test

in Charlotte, N.C. Those early tests revealed system and operational limitations in conducting such tests, detailed in the July Subcommittee meeting and in a report in Appendix D. Plans were made to revise the equipment and test at another venue.

Bonneville Broadcasting, a long time participant in the EIA-DAR test program, offered its transmitter site in Salt Lake City on Farnsworth Mountain. The site was investigated and the equipment was delivered and set up at the site with testing beginning in late September and continuing to early October of 1993.

CHANNEL TEST DATA COLLECTED; FINDINGS; ENVIRONMENTS, SPEED, DATA, VOLUME COLLECTED, PROCESSING, ETC.

In early October 1993 the actual Salt Lake City channel characterization test data was collected over approximately one week. Appendix E is a description of the measurements and the data collected. As data was collected along each path, the environment around the area was described. Four major "Environments" quickly emerged: Urban, Suburban, Rural and Terrain Obstructed. Appendix F is a March 7, 1994 memorandum discussing the data collected, its analysis and the certification of the test method.

The data extraction strategy was studied and modified from its initial frequency domain dependent version to one which selected reflections based on time domain values in order of the strongest reflections with their accompanying delay and relative phases.

By April 1994 the VHF Channel Characterization data had been analyzed, providing the overall range of reflection magnitudes verses time delay for the four significant environments in Salt Lake City as shown in Appendix G. Further analysis then extracted the individual channel reflection vs. time information on a file by file basis as explained in the memorandum report dated April 17, 1994 in Appendix H.

The measured VHF reflection time vs. magnitude information was studied to arrive at the range of data appropriate for challenging multipath Environments. Information from other sources was also compared to the measured VHF channel data so that the simulation could include the 1.4 GHz UHF channel DAR system as well. The Canadian CRC investigation relating to L-Band characterization lead to the exchange of several documents, a selection of which are included in Appendix I.

The listings of time delays and magnitudes with appropriate doppler velocities and Rayleigh file parameters for each of the three environments (four tests), as adopted by the EIA-DAR test laboratory and is indicated in the attached Appendix J.

SIMULATOR LIMITATIONS; ATTENUATOR RATE OF CHANGE & "ARTIFACTS"

The direct control method, described in the Simulator operating manual, was tested

in early 1994 and limitations quickly appeared. The simulator attenuator control circuits have a significantly slow time constant which will allow only slow changes in the simulation channels, far slower than were measured. This was implemented and found to function properly.

Tests were run on the simulator using sample direct control data and observing the simulator effect in the frequency domain. Upon close observation it could be seen that artifacts were being generated. Appendix K is a memorandum and report of July 12, 1994 describing the findings. The report and attachments indicate that the frequency domain artifacts are generated by the step changes in the simulator channels.

In an attempt to resolve the Frequency Domain artifacts the data was "smoothed", as indicated in Appendix K, by limiting the rate of change of some of the parameters and approximating missing data between data files. The artifacts decreased but still remained. It was decided, not knowing the impact of the artifacts on the systems under test, that the direct control of the simulator was not possible.

IMPLEMENTING SIMULATION MODE; MAINTAINING VARIABILITY

The Simulation or "sim" mode of operation allows for two variations. The first is the "Doppler" mode, a fixed parameter mode with a fixed cyclical simulation, and the second applies a Rayleigh variation characteristic on the "Doppler" control parameters. This Rayleigh characteristic will restore some of the channel variability lost in going from the Direct to the Sim mode of channel simulation.

The Rayleigh fading characteristic is imparted on the simulator action by a control file that is generated by the HP program IQMAKE. The Rayleigh fading values are oriented about the basic channel parameters, those which were measured in Salt Lake City, not any other parameters associated with any standard Cellular or Land Mobile system. The resulting simulation has the overall characteristic of the measured control values but with the Rayleigh variation characteristics specific for the frequency and velocity of interest for the test, impressed on each of the control channels. This effect on each of the individual channels then generates a combined effect on the overall variation of the combined R.F. channel output.

DISCUSSION OF APPROPRIATENESS OF DOPPLER VS. RAYLEIGH CHARACTERISTICS

Much discussion centered about the proper use of the Doppler or Rayleigh simulation modes. Objections were raised citing the HP instruction manual with various references to the Rayleigh model defined for mobile cellular radio. Concerns were expressed relative to whether or not a Rayleigh faded channel was appropriate for the mobile environment. The use of particular sections of the Salt Lake City measured channel characteristics and then only that one venue was questioned.

Many individual experts in mobile communications reviewed the questions and concerns and have supported this simulation concept as appropriate for the laboratory testing. Appendix L contains the observations, comments and responses regarding the Rayleigh and Doppler simulations.

As a result of the questions regarding the use of Rayleigh simulation a decision was made to incorporate <u>both</u> Doppler and Rayleigh simulations in an expanded laboratory test.

CHANNEL TEST AND SIMULATION; LESSONS LEARNED

This channel characterization project and the channel simulation in the laboratory has again confirmed the immense variability that exists in an R.F. propagation path which can not be carried to and totally duplicated in laboratory simulation. For laboratory purposes, however, capturing all that variability would be counter-productive. For example, in an average environment, much of the time the R.F. channel may be quite benign with few if any interesting and stressful multipath conditions. The laboratory testing is meant to be a critical test of the systems. It is the relatively rare but stressful conditions that need to be reliably and rapidly repeated in the laboratory. This goal guided the extraction of "significant" multipath segments from the four environments to concentrate on those areas that generally would yield harsh tests.

Ideally, the original laboratory test would have used the actual channel parameters measured in the field, complete with their variability along the measurement path, to control the channel simulator as if driving along that same path. Hardware limitations prohibited this. The same general channel characteristics for the difficult path segments were used but with the parameter variability now supplied by the Rayleigh fading profile applied to those characteristics.

The channel simulation testing has been applied uniformly to all of the proponent systems, even to the extent of testing in both Rayleigh and Doppler modes. The systems individual relative performances will be determined by the systems themselves, not by the design of the testing. If the testing were designed so that all systems were to fail the test, or where all were to easily pass the simulation test, the results would be useless. The only valid test is one that spans the range of performance from perfect to failed for all systems under test and hence determines a threshold of actual performance. The laboratory simulation provides such a test.

APPENDIX J - DATA PROCESSING AND INTEGRATION TO SIMULATOR(S); ACTUAL SIMULATION PARAMETERS

- J-1 Memo Report (L&C), "Suggested Nine Path Multipath Simulation Settings" -- 7/26/94
- J-2 Letter from D. Londa, EIA, "Channel Simulator Parameters" -- 8/16/94

SUGGESTED NINE PATH MULTIPATH SIMULATION SETTINGS 7/26/94

SIMULATOR SETTINGS - DELAY IN MICROSECONDS, ATTENUATION IN dB **URBAN SLOW-FAST** RURAL **OBSTRUCTED** PATH **DELAY** ATTN. **DELAY** ATTN. DELAY ATTN. 1 0.0 2 0.0 4 0.0 10 2 0.2 0 0.3 8 1.0 4 3 0.5 3 0.5 0 2.5 2 4 0.9 4 0.9 5 3.5 3 5 1.2 2 1.2 16 5.0 4 6 14 0 1.9 18 8.0 5 7 2.0 3 2.1 14 12 2 8 2.4 5 2.5 20 14 8 9 3.0 10 3.0 25 16

URBAN - SLOW; Use Rayleigh doppler path at 1 KPH at 94.1MHz RF test frequency. NOTE; this slow speed may not allow full development of all possible Rayleigh states. Be prepared to try 2 and 4 KPH to see if there is a difference.

URBAN - FAST: Use Rayleigh doppler at 60 KPH.

RURAL (FAST): Use Rayleigh doppler at 150 KPH.

SUBURBAN/TERRAIN (FAST); Use Rayleigh doppler at 60 KPH.

The suggested settings above are based on a comparison of the original EIA SIM A - D files and the Canadian UHF suggested 12 path urban and 8 path rural settings. A thorough review of the Salt Lake City direct control files will be made to determine the average delays and magnitudes for the four environments to extract 9 path settings to be applied to this test. Those revised settings will be coordinated with Canada. The goal is a uniform Simulation Mode test for all bands.

NOTE: 7/26/94 The EIA VHF test data has been reviewed and coordinated with Canada. The above table represents consolidated Urban and Rural environment parameters for both VHF and UHF tests. The Obstructed file is based only on the VHF measurement further analysis.

August 16, 1994

Mr. Robert D. Culver Lohnes & Culver 8309 Cherry Lane Laurel, MD 20707-4830 Phone:(301) 776-4488

Fax:(301) 776-4499

Dear Bob,

Here is the multipath characterization video tape.

The channel simulator parameters used for the Urban environment are as follow:

PATH	DELAY (us)	DOPPLER (kmh)	ATTEN (dB)		
1	0.0	2 or 60	2.0		
2	0.2	2 or 60	0.0		
3	0.5	2 or 60	3.0		
4	0.9	2 or 60	4.0		
5	1.2	2 or 60	2.0		
6	1.4	2 or 60	0.0		
7	2.0	2 or 60	3.0		
8	2.4	2 or 60	5.0		
9	3.0	2 or 60	10.0		

The channel simulator parameters used for the Rural environment are as follow:

PATH	DELAY (us)	DOPPLER (kmh)	ATTEN (dB)
1	0.0	150	4.0
2	0.3	150	8.0
3	0.5	150	0.0
4	0.9	150	5.0
5	1.2	150	16.0
6	1.9	150	18.0
7	2.1	150	14.0
8	2.5	150	20.0
9	3.0	150	25.0

The channel simulator parameters used for the Suburban / Terrain Obstructed environment are as follow:

PATH	DELAY (us)	DOPPLER (kmh)	ATTEN (dB)		
1	0.0	60	10.0		
2	1.0	60	4.0		
3	2.5	60	2.0		
4	3.5	60	3.0		
5	5.0	60	4.0		
6	8.0	60	5.0		
7	12.0	60	2.0		
8	14.0	60	8.0		
9	16.0	60	5.0		

The IQMAKE.EXE utility was used twice to create the Rayleigh fading data. The command parameters used are as follow:

IQMAKE -R 94.1E6 2 60 150 and similarly

IQMAKE -R 1.47E9 2 60 150.

These command lines then created the Rayleigh fading data files and the appropriate filename was indicated in the simulation mode menu under the Spectrum filename heading. Simulation profiles were stored with the appropriate Raleigh fading data filenames, delay and attenuation and the profiles recalled and video taped.

The first four segments on the tape (00:00-05:42) are the VHF simulations and the last five segments (05:42-12:22) include a reference of the signal unimpaired followed by the simulations in L-Band.

Call me if you have any questions.

Best regards,

David M. Londa RF Test Manager

Appendix G – System Specific Tests and Procedures

Proposed System Specific Third Mode Test for Amati/AT&T IBOC
System

May 10, 1994

The DAR test plan calls for the Amati/AT&T IBOC System to be tested in two modes. The first and primary mode uses the upper and lower extremity of the FM channel mask, and the second mode uses the upper or lower extremity of mask for the transmission of the DAR signal. Because of possible variations in the test receiver, limited testing of both combinations for the second mode should be conducted.

Proposal:

- 1. The proponent will recommend the sideband to be used for the second mode testing.
- 2. For the side band not used for general testing, the following tests will be conducted:

B-1 Noise for EO&C only .

B-3 Multipath and noise for EO&C only

System specific tests for the Eureka-147 DAB system

System operation in a active echo environment

A- Purpose of the tests:

These tests are to confirm the capability of the Eureka-147 DAB system, as claimed by the European system developers, to work in presence of large echoes produced by on-channel re-transmitters. These re-transmitters, operating on the same frequency, are expected to be used to fill gaps within coverage areas (gap-filler); extend the coverage of the conventional single transmitter (coverage extenders); or stretch the coverage of broadcast stations over a larger area using a network of synchronized on-channel transmitters (single frequency network - SFN). In all these three cases, the on-channel repeaters will generate, depending on where the receiver is located relative to these repeaters, active echoes that may fall either before of after the reference signal coming from the main transmitter, and the level of these active echoes could be either smaller or larger than the reference signal. The goal of this test will be to confirm the operation of the Eureka-147 system in presence of these echoes.

Further, the system developers claim that the system has improved performance in presence of echoes. More precisely, it is claimed that it makes constructive use of these whoes radical diam being negatively impacted by mem. In fact, the performance of the system would be related to the power addition of these echoes. The goal of these tests is also to confirm this claim and clarify the circumstances and conditions under which this happens.

B-Proposed test procedure:

Test 1: Echo weighting template

It is proposed to use the HP 11759C channel simulator to simulate active echoes that would be injected in the transmission channel before reception by the Eureka-147 receiver. Figure 1 depicts the setup required to conduct the tests

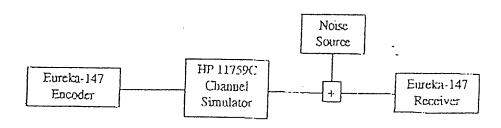


Figure 1: Block diagram of the test semp

One path of the channel simulator would be used to provide the main signal with a constant amplitude and phase. This would be the reference signal for the test. A second simulator path would carry the active echo. The phase of this path would be constantly changing (implemented in the simulator as a fixed Doppler shift). The rate of phase shift could be a minimal and a maximal amount, within the limits of the Doppler spread of the channel (i.e., simulating a vehicle displacement orthogonal to the main signal path, and parallel and orthogonal to the on-channel repeater's signal path).

It is proposed to sweep the delay of the active echo from 0 µsec to +186 µsec (limit of the channel simulator) referenced to the main path in steps of 10 µsec. At each step, the amplitude of the echo will be brought from a low level and increased until the "point of failure" (POF) is reached by listening to the audio material (measuring the BER on the data channel and identifying the echo amplitude for a BER of 10⁴ would make this test more repeatable). The echo amplitude at the POF will be noted at each echo delay. If the POF is not reached for an echo higher that 10 dB above the reference signal, there is no need to go beyond this point. This sweep should be repeated for the case where there is no noise in the channel and also for the cases where noise levels of 15 dB and 12 dB below the level of the main path are injected in the channel.

The results of this test should be similar to the predicted template shown in Figure 2. This template can, in fact, be used to weight the effect of echoes as a function of their delay. This template should theoretically be symmetrical relative to the main path since it is claimed that pre- and post-echoes are handled the same way by the system. Although the system is to work equally well with pre-echoes, the left side of the template is not expected to be reproduceable by laboratory measurements since the synchronization algorithm is programmed to latch on to the first echo of a certain power, whether it is a pre-echo or the reference signal.

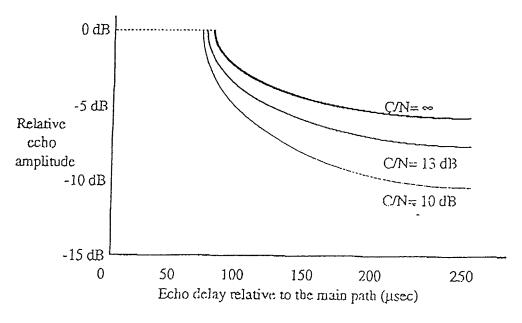


Figure 2: Typical echo weighting template

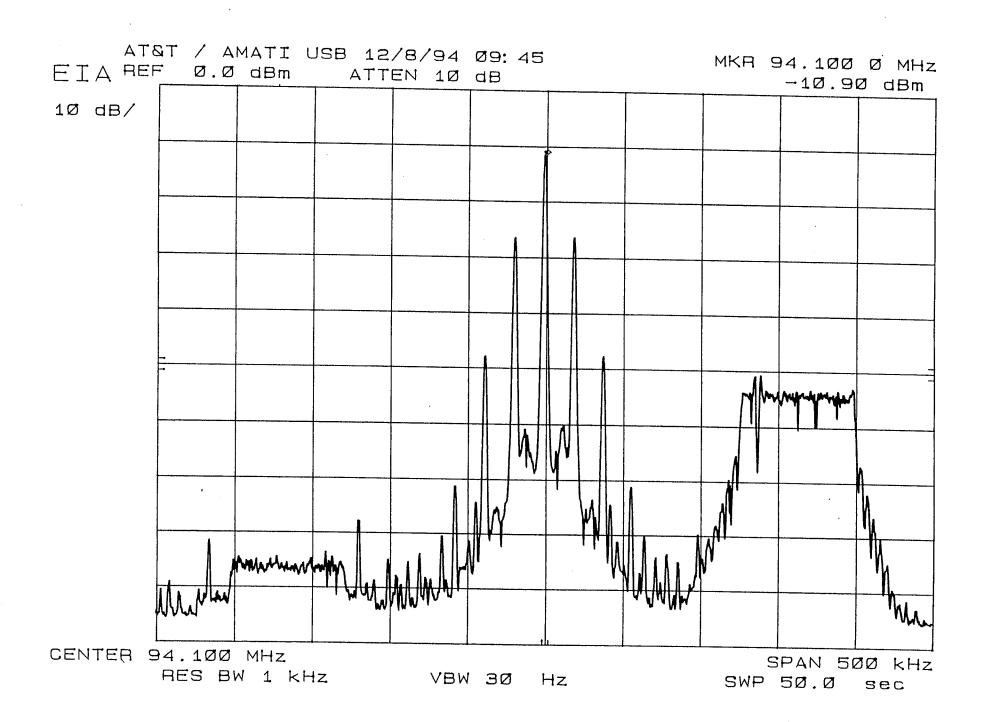
Test 2: Effect of echoes on the system's C/N performance.

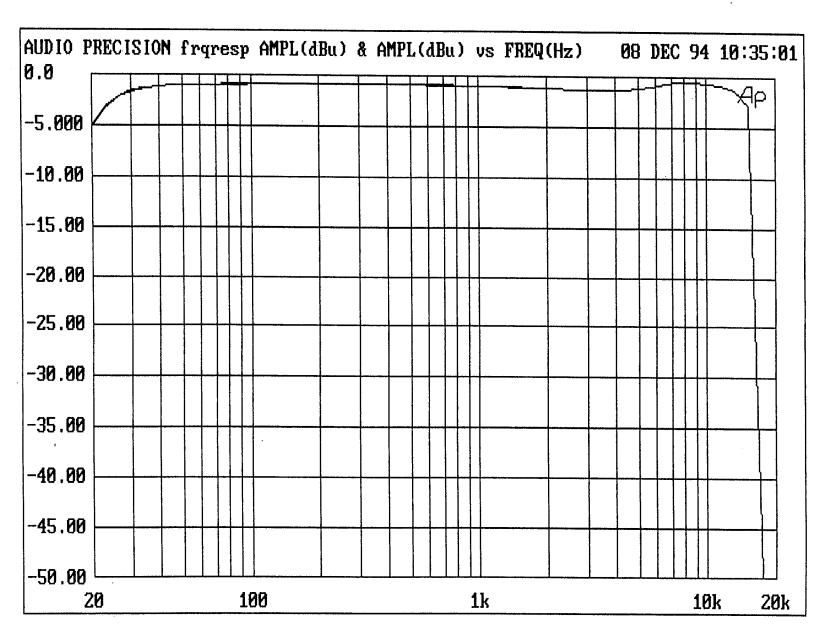
- Test 2.1: It is proposed to inject a calibrated level of noise in the channel path and raise this level of noise in presence of the main signal only, until the POF is reached (or BER= 10⁻⁴). This value is noted.
- Test 2.2: Then an echo with a varying phase as described in Test 1 above is injected at the same power as the main signal. The delay between the main signal and this echo is such that the echo falls within the range where the POF could not be reached with an increase of the echo power in Test 1 (i.e., within the system guard interval). The noise power is then increased until the POF is reached (or BER=10⁻¹). This C/N value is noted. In this case, "C" represents the added power of the two signals, thus 3 dB higher than in the preceding case. The increase in C/N value, compared to that of the previous Test 2.1. represents the impact of moving the operation of the system from a Gaussian channel to a Rayleigh channel.
- Test 2.3: A second echo of the same power and varying phase should then be injected. This new echo should be within the range where the POF could not be reached as performed under Test 2.1 (i.e., within the guard interval). The noise power is then increased until the POF is reached (or BER= 10⁻¹). This new value of C/N at POF should be recorded. In this case, "C" represents the total power of the three echoes (i.e., 4.8 dB higher than in the case of the single signal). The C/N value is then compared to the case of Test 2.2 to verify that the principle of power addition of echoes applies as claimed once the system operates in a Rayleigh environment (i.e., for one echo and more). This principle can be verified further with the injection of more echoes at the same level and the identification of the C/N at POF (or BER= 10⁻¹).

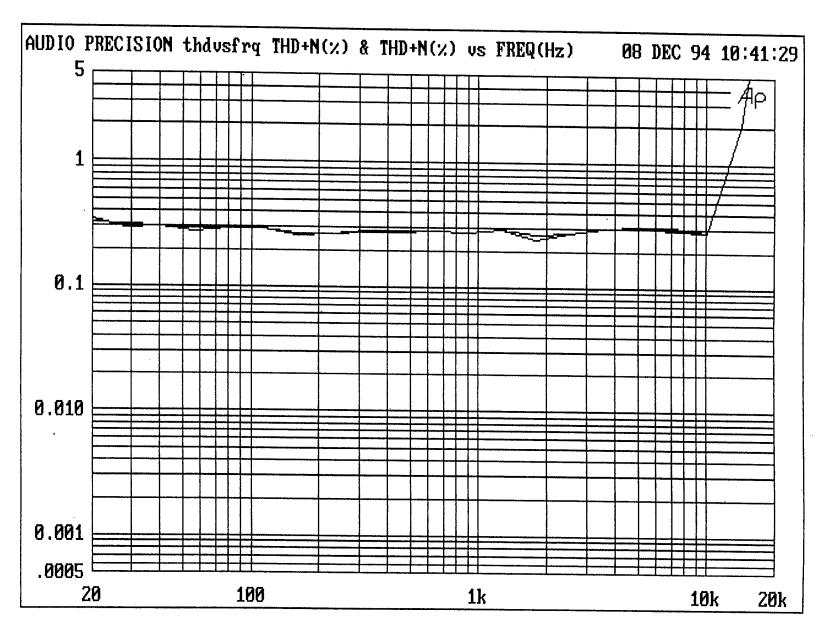
Gérald Chouinard CRC 94.05.04

Test	B-1			
AT&T Amati		System Spec	si6a	
Upper Side Ba	ınd	Gaussian N		
		Gaussian	TUISE	
				Units
Glockenspiel		TOA	POF	
	Attenuator	32.50	29.00	dB
-	Co/No	20.61	17.11	dB
	TOA	Small break in recovered audio.		
EO&C				
	POF	Excessive muting and a loud squeal.		
	B-2			
AT&T Amati		System Spec	ific	
Upper Side Ba	nd	Co-Chan	nel	
Glockenspiel	110/ - Tue-rus.	TOA	PAR .	Units
	Attenuator	20.50	POF 18.00	
	d/u	21.36	18.86	dB
-	TOA	Small warble or flutter and some static pops		dB
EO&C				
	POF	Excessive muting and fluttering.		
Addition	al Comments:	In unimpaired conditions at medium signal s the recovered audio has artifacts with glocke	trength	
		program material. Artifacts sound like a mod on the first note of some glockenspiel arpegg	proco type rattle	
Testers: I Date:	DML,RMc 8-Dec-94			

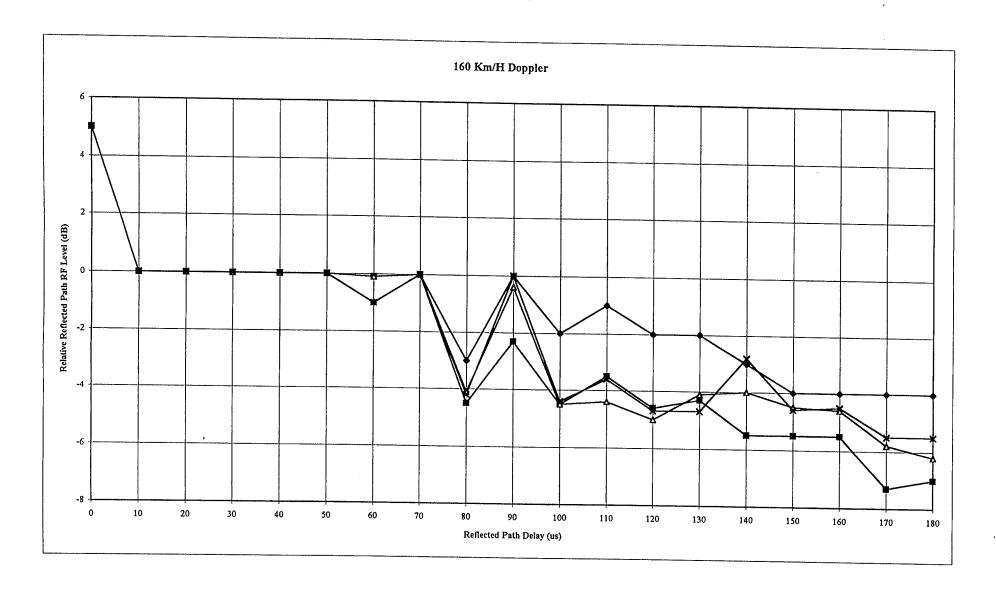
Test AT&T Amati USB	B-3	System Spe	cific Multi	path (Raylei	igh)				
Program Material:	Glockensp	iel	i							·
Scenario										
	Level	Attn	Co/No	Units		EO&C				
#1 Urban Slow	TOA	63.75	52.34	dB		Mutes, p Worse th	ops and flutt an POF.	ers.		
	POF	63.75	52.34	dΒ						
#2 Urban Fast	TOA	63.75	52.34	dΒ			pops and mu l of impairm			
	POF	63.75	52.34	dΒ						
#3 Rural Fast	TOA	63.75	52.34	dB		Mutes, fl Worse th	utters and la	ge pops		
	POF	63.75	52.34	dΒ						
#4 Terrain Obstructed	TOA	63.75	52.34	dB			ation mutes orse than PO		nd	
	POF	63.75	52.34	dΒ						
Test Date:	8-Dcc-94	'		De	sired			<u> </u>	Voise	
Testers:	DML, RM	c	Signal IL 3WIN	4	31.46 10.79 72.25	dB	BV 0dB Re	/ 6.45I	E+06 Hz	. 1
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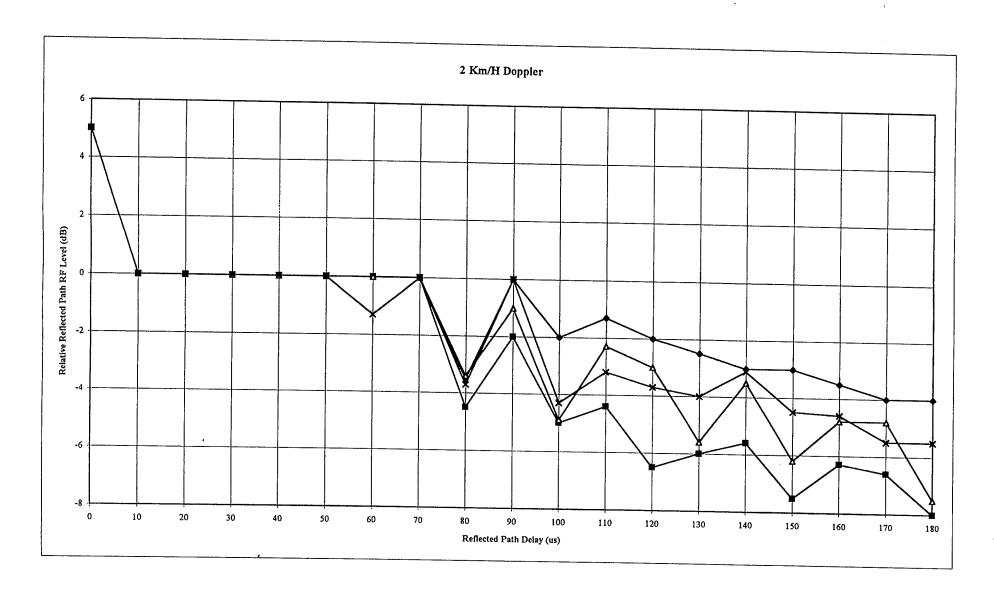




											 								
30-Nov-9																			
E-147 Sys																			
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PHASE: 2																			
CLEAR (CHANN	EL TE	ST			DELA	Y (us)												
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POF	5	0.00	0.00	0.00	0.00	0.00	-1.00	0.00	-3.00	0.00	-2.00	-1.00	-2.00	-2.00	-3.00	-4.00	-4.00	-4.00	-4.00
																	1.00	17.00	-4,00
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													<u> </u>			L.,		1	
C/N = 10							Y (us)												
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POF	- 5	0.00	0.00	0.00	0.00	0.00	-0.10	0.00	-4.50	-2.30	-4.50	-3.50	-4.60	-4.30	-5.50	-5.50	-5.50	-7.30	-7.00
W V/N 13.	AR THEST	r				PART A	¥2 ()												
C/N = 13	dB TEST		20	20	40		Y (us)									T			
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	0	10				50	60	70											
	0	10				50	60	70											
	0	10				50	60	70											
POF	5	0.00				50 0.00	60	70											
	0 5 22dB TE	10 0.00 ST	0.00	0.00	0.00	50 0.00 DELA	60 -0.10 -Y (us)	70	-4.10	-0.40	-4.50	-4.40	-5.00	-4.10	-4.00	-4.50	-4.60	-5.80	-6.20
POF Co/No = 2	0 5 22dB TE	10 0.00 ST 10	20	30	0.00	50 0.00 DELA 50	60 -0.10 Y (us)	70 0.00	-4.10	-0.40 90	-4.50 100	-4.40 110	-5.00 120	-4.10 130	-4.00 140	-4.50 150	-4.60 160	-5.80 170	-6.20
POF	0 5 22dB TE	10 0.00 ST	0.00	0.00	0.00	50 0.00 DELA	60 -0.10 -Y (us)	70	-4.10	-0.40	-4.50	-4.40 110	-5.00	-4.10	-4.00	-4.50	-4.60	-5.80	180 -6.20
POF Co/No = 2	0 5 22dB TE	10 0.00 ST 10	20	30	0.00	50 0.00 DELA 50	60 -0.10 Y (us)	70 0.00	-4.10	-0.40 90	-4.50 100	-4.40 110	-5.00 120	-4.10 130	-4.00 140	-4.50 150	-4.60 160	-5.80 170	-6.20 180
POF Co/No = 2	0 5 22dB TE	10 0.00 ST 10	20	30	0.00	50 0.00 DELA 50	60 -0.10 Y (us)	70 0.00	-4.10	-0.40 90	-4.50 100	-4.40 110	-5.00 120	-4.10 130	-4.00 140	-4.50 150	-4.60 160	-5.80 170	-6.20 180
POF Co/No = 2	0 5 22dB TE	10 0.00 ST 10	20	30	0.00	50 0.00 DELA 50	60 -0.10 Y (us)	70 0.00	-4.10	-0.40 90	-4.50 100	-4.40 110	-5.00 120	-4.10 130	-4.00 140	-4.50 150	-4.60 160	-5.80 170	-6.20 180
POF Co/No = 2	22dB TE	10 0,00 ST 10 0.00	20 0.00	30 0.00	0.00 40 0.00	50 0.00 DELA 50 0.00	60 -0.10 Y (us) 60 -1.00	70 0.00 70 0.00	-4.10 80 -4.20	90 0.00	-4.50 100 -4.40	-4.40 110 -3.60	-5.00 120 -4.70	-4.10 130 -4.70	-4.00 140 -2.80	-4.50 150	-4.60 160	-5.80 170	-6.20
POF Co/No = 2	22dB TE 0 5 5 8: The va	10 0.00 ST 10 0.00	20 0.00	30 0.00	0.00 40 0.00	50 0.00 DELA 50 0.00	60 -0.10 Y (us) 60 -1.00	70 0.00 70 0.00	-4.10 80 -4.20	90 0.00	-4.50 100 -4.40	-4.40 110 -3.60	-5.00 120 -4.70	-4.10 130 -4.70	-4.00 140 -2.80	-4.50 150	-4.60 160	-5.80 170	-6.20 180



I	EIA Digital Audio Radio Test Laboratory																		
30-Nov-9																	· · · · · · · · · · · · · · · · · · ·		
	stem Spec																		
l .	R: 2 Km/																		
1	2.724 Hz																		
CLEAR	CLEAR CHANNEL TEST DELAY (us)																		
	0	10	20	30	40	50	60	70	80	90	100	110	120	130	140	150	160	170	180
POF	5	0.00	0.00	0.00	0.00	0.00	0.00	0.00	-3.50	0.00	-2.00	-1.30	-2.00	-2.50	-3.00	-3.00		-4.00	-4.00
																3.00	3.50	-7.00	-4.00
				<u> </u>															
					- 						1				<u> </u>	<u> </u>	·		
C- (N-	40.10 mm	· ~ m																	
Co/No =	10dB TE		T 20	1 20	T		Y (us)												
707		10	20	30	40	50	60	70	80	90	100	110	120	130	140	150	160	170	180
POF	- 5	0.00	0.00	0.00	0.00	0.00	0.00	0.00	-4.50	-2.00	-5.00	-4.40	-6.50	-6.00	-5.60	-7.50	-6.30	-6.60	-8.00
			L		<u> </u>				<u> </u>										
Co/No =	13dB TE	err				PART A	** / \												
C0/110 -	0	10	20	30	40	DELA		70	I 00										
POF	- 5				40	50	60	70	80	90	100	110	120	130	140	150	160	170	180
PUF		0.00	0.00	0.00	0.00	0.00	0.00	0.00	-3.40	-1.00	-4.90	-2.30	-3.00	-5.60	-3.50	-6.20	-4.80	-4.80	-7.50
	-						<u>.</u>												
Co/No =	22dB TE	ST				DELA	V (no)												
0077.0	000	10	20	30	40	50	60	70	80	90	100	110	:20	120					
POF	5	0.00	0.00	0.00	0.00	0.00					100	110	120	130	140	150	160	170	180
101	-	0.00	0.00	0.00	0.00	0.00	-1.30	0.00	-3.70	0.00	- 4.30	-3.20	-3.70	-4.00	-3.10	-4.50	-4.60	-5.50	-5.50
	-																		
			L																
27-4-	The	1 19	4. 1			~ .													
Note	s: The va	lues lis	ted are	the dela	iyed or	reflecte	d path l	RF leve	ls in dE	relativ	e to the	non-de	elayed p	oath RF	level				
	at a PC	r ievei	of imp	airmen	ι.														i



Engineers:

RMc/DL

DATE:

5/24/95

TEST N

MULTIPLE SPURIOUS

This test is IBAC specific. In addition to co-channel and adjacent channel separations, Part 73.207 of the FCC rules also specifies the minimum distance separation requirement for FM stations operating at 10.6MHz and 10.8MHz (10.7MHz IF) above and below the operating channel. Using the interference caused by two FM stations operating at the 10.7MHz separation as the reference, Test N will compare the two FM station's reference with the interference caused by an IBAC and FM station at the same power level.

1) Reference

The following frequencies and procedures will be used to characterize the reference receiver.

RF GEN. 1 = 94.1 MHz

Rec. Freq. = 99.95MHz

RF GEN. 2 =

104.8MHz

RF1 = The RF level required from a single generator to give 30dB S/N ratio at the receiver tuning frequency RF2 = The RF level required from both interfering generators to give 30dB S/N ratio. (RF Level GEN 1 = RF Level GEN 2) Result:

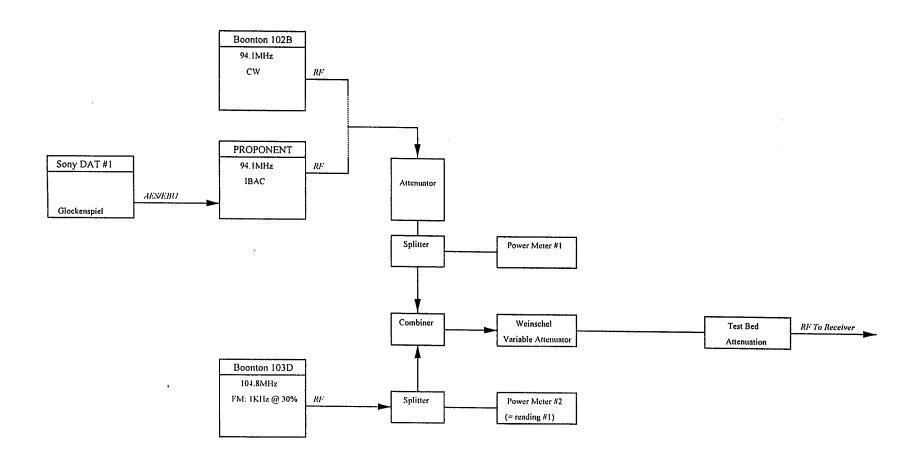
 $IF_REJ. = 20 \log RF2/RF1(uV) \text{ or } RF2 - RF1(dBm)$

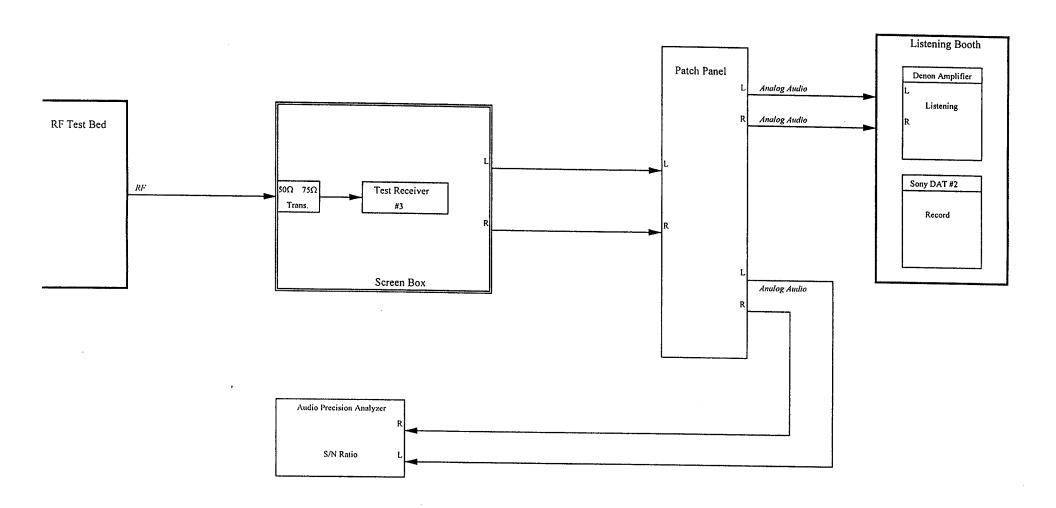
Subjective EO&C of the test receiver audio quality

Tes

Using the test receiver, the proponent IBAC system will replace RF GEN. 1 (94.1MHz). The average power of the IBAC transmitter will be set to the same level as RF GEN. 2. Any difference in subjective interference will be noted in the EO&C.

Test N Multiple Sp Receiver					EO&C	
PANASON						
			RF (dBm)	Proponent Digital Audio Input = Glockenspiel	Proponent Digital Audio Input = 0	
Analog / A					Topolicit Digital Addition input = 0	Without Digital Signal
30dB S/N	94.1MHz 89.25+99.95MHz	RFI		NA	NA	The pudio is relativity of
IF Reject.	89.25+99.95MHz	RF2	-38.95			The audio is relativly clean sounding with some background hiss typical of a 30dB S/N ratio
ir Kejeci.		D/U	-59.25			suckground hiss typical of a 30dB S/N ratio
30dB S/N	99.95MHz	RF1	-96.95			
	94.1+104.8MHz	RF2	-37,2			
IF Reject.		D/U	-59.75	<u></u>		
AT&T - IB.	AC				Sounds worse than Analog reference and	
304B 6VVI	94.1+104.8MHz	חרם	22.0	Sounds worse than Analog reference W/	worse than when Proponent W/Glockensniel	NA
IF Reject.	74.1 F 104.6 WITZ	RF2 D/U	-37.2 -59.7	occational low level buzzing sound	Audio is torn up by a low frequency (approx 100Hz)	
		2,0	-32,7		sound	
				•		
i						
		1				
		- [
		ĺ				
		1	1			ļ
			ļ			
Notes:	The Boonton 103D F	RF gene	erator used f	or S/N ratio and IF reject, tests		
	The Boonton 102B R	RF gene	rator used a	s second generator in IF reject, tests (analog only	1	
	The "Lower" generat	or or P	roponent at	94.1 MHz is not modulated (CW only)	,	
	The "Upper" generate	or is m	odulated wit	th 1KHz at 30% (22.5KHz)		
······································	inis test is an IBAC	test on	ly. Other da	a included as additional information only		





Appendix H – Receiver Characterizations

APPENDIX H

Receiver Characterizations (Consumer Analog)

Enclosure #1

Five Analog FM Receiver Characterization Reports

Enclosure #2

Additional Information of the Five FM Receivers

Enclosure #3

Three Analog AM Receiver Characterization Reports

DAR FM TEST RECEIVER DATA

Receiver Lab #1

Type Auto

Index

Page	Description
1	Laboratory FM -> FM D/U Ratios
2	Radio Characterization/Confirmation
3	Signal, Noise, & Separation VS RF Level
4	Graph of Signal & Filtered Noise VS RF Level
5	Graph of Separation VS RF Level
6	Graph of Signal, Noise, Filtered Noise, & Separation VS RF Level
7	Woodstock Engineering Receiver Test Report
8	Audio VS RF Frequency Test
9	Receiver Upper 1st Adjacent Interference/Noise
LO	Receiver Lower 2nd Adjacent Interference/Noise
1	Receiver Upper 2nd Adjacent Interference/Noise

FM -> FM Laboratory Measurements for the Delco Model 16192463

Laboratory Receiver #1

Type: Auto

Measurements were made at a moderate signal level of -62 dBm.

The signal to noise ratio was set at 45 dB and this measurement was made using a 15kHz low pass and a CCIR filter with quasi-peak detection. For the second adjacent tests, 45 dB S/N was not attainable on the test bed with this receiver so 47 dB was used.

Test Results:

Co-Channel	D/U	36.17	dB	
Lower First	Adjacent	D/U	4.09	dВ
Upper First	Adjacent	D/U	5.41	đВ
Lower Second	d Adjacent	D/U	-24.17	đΒ
Upper Second	d Adjacent	D/U	-24.17	đВ

ELECTRONIC INDUSTRIES ASSOCIATION

Digital Audio Radio Laboratory

Engineers:

RMc/DL

DATE:

2/21/95

PROJ.:

RADIO CHARACTERIZATION/CONFIRMATION

- * Key point measurements for comparison to Grossjean data
- * Additional data with regard to audio performance VS RF level

TEST SET-UP

- * Delco Radio Gra
- Graphic EQ Flat, Loudness Off, Fader & Bal.- Centered
- * Test Bed, W/Orban Stereo Gen & Harris Exciter as Signal Source
- * Boonton RF Gen used for crosscheck verification
- * Delco Dummy Antenna
- * Audio Reference Level: OdB = 1 Watt (2Vrms)

Load Imp. = 4Ω

* Audio measurements made with Audio Precision as rms unweighted

FM TESTS

(TEST FQ. 94.1MHZ)

S/N RATIO - 1KHZ, 30% MOD

20dB S/N	-105 dBm
30dB S/N	-102 dBm
50dB S/N	-92 dBm

S/N RATIO - 1KHZ, 100% MOD

USABLE	50dB S/N	-95dBm	(Boonton Gen.)
USABLE	50dB S/N	-96dBm	(Test Bed)
MAX	59.7dB	-60dBm	(Boonton Gen.)
MAX	59.4dB	-62dBm	(Test Bed)

THD - 1KHZ, 100% MOD (-50dBm)

MONO	0.80 %	(Boonton G
MONO	0.65 %	(Test Bed)
STEREO	2.04 %	(Test Bed)

LIMITING THRESHOLD

(Tracability; Grossjean/RF Generator/Test Bed)

	Boonton RF Ge	Through Test Bed	
98.1MHZ (Gross		94.1MHZ (lab freq.)	94.1MHZ
Audio -1dB	-101 dBm	-100.4 dBm	-101 < LThresh. < -100dBm

HIGH CUT THRESHOLD

Audio: 10KHZ, L+R, 100% Mod, Pilot off

-3dB = -85dBm

Note:

Same result with Pilot On

SEPARATION @ -62dBm

Freq.	L->R	R->L	
1KHZ	36dB	32dB	(W/O Pre-Emph)
10KH7	17dB	17dB	(W/O Pre-Emph)

SIGNAL, NOISE & SEPARATION VS RF LEVEL

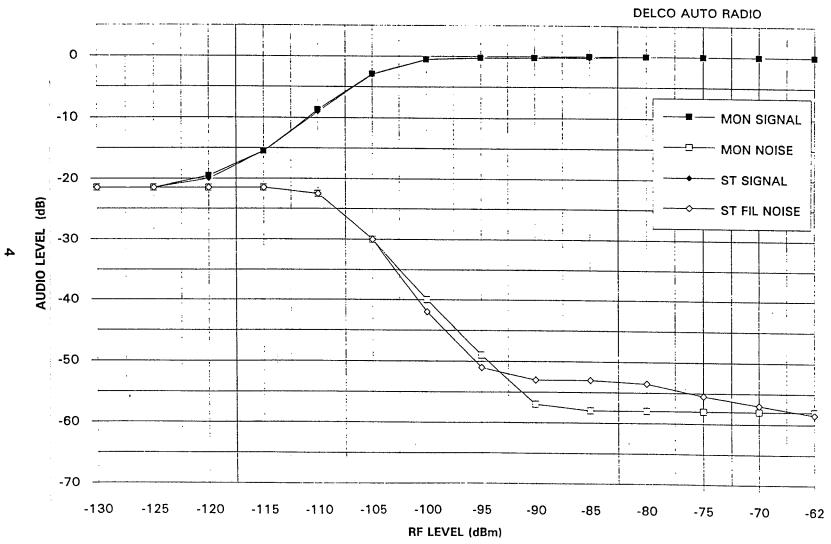
- Left channel used as the measurement channel for Signal and Noise data
- Left channel driven (L only) for separation data
- Audio test frequency = 1KHZ
- Note: There were no significant improvements in performance at RF levels above 62dBm

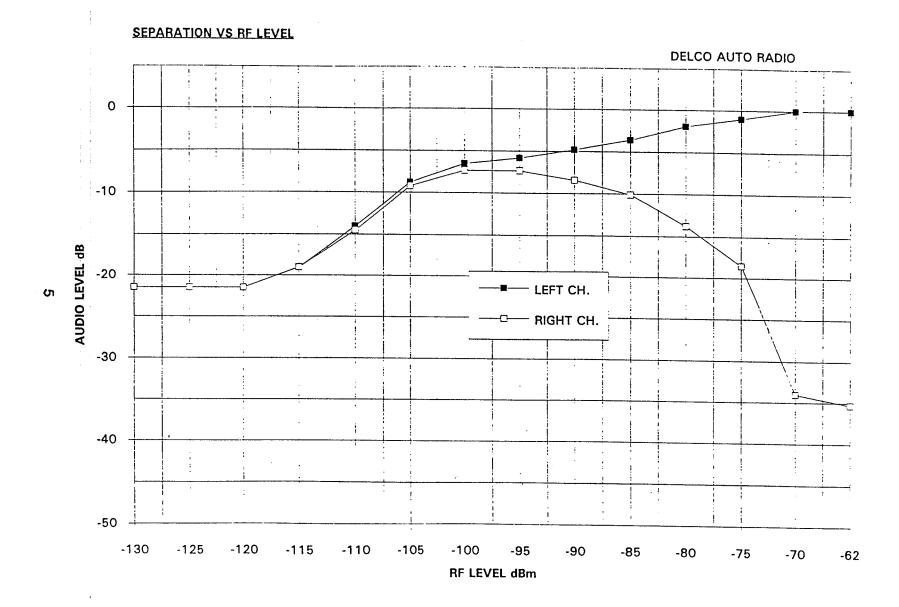
CURVE DATA

SIGNAL, NOISE & SEPARATION VS RF LEVEL

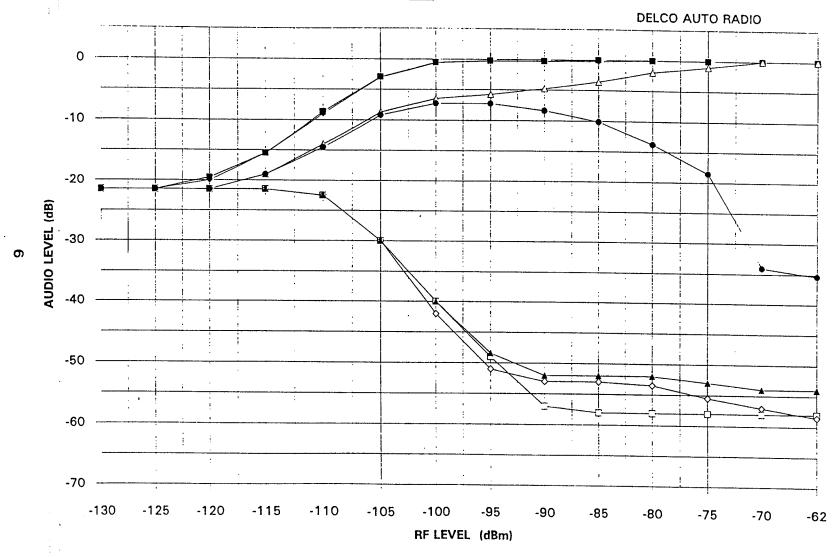
	mon	o (L)		Stereo (L)			Separation	L->R
RF Level	Signal	Noise	Signal	Filt. Noise	Noise	RF Level	Left	Right
dBm	dB	dB	dB	dB	dB	dBm	dB	dB
-130	-21.5	-21.5	-21.5	-21.5	-21.5	-130	-21.5	-21.5
-125		-21.5	-21.5	-21.5	-21.5	-125	-21.5	-21.5
-120		-21.5	-20	-21.5	-21.5	-120	-21.5	-21.5
-115		-21.5	-15.5	-21.5	-21.5	-115	-19	-19
-110		-22.5	-9	-22.5	-22.5	-110	-14	-14.5
-105	-2.9	-30	-3	-30	-30	-105	-8.75	-9.2
-100		-40	-0.57	-42	-40	-100	-6.5	-7.3
-95	-0.23	-49	-0.29	-51	-48.4	-95	-5.8	-7.3
-90	-0.21	-57	-0.26	-53	-52	-90	-4.8	-8.4
-85	0	-58	-0.21	-53	-52	-85	-3.6	-10.1
-80	0	-58	0	-53.5	-52	-80	-1.92	-13.75
-75	0	-58	0	-55.5	-53	-75	-1	-18.5
-70	0	-58	0	-57	-54	-70	0	-34
-62	O	-58	0	-58.5	-54	-62	0	-35.3
-57						-57		
								<u> </u>







SIG., NOISE, FILT. NOISE & SEPARATION VS RF LEVEL



	GEN	KCAK		••••
TUNER TEST DATA Manufacturer: Model Number: Serial Number:	Delco 16192463 1000499 car			
Type: FM 30% modulation(98.1MHz	٠.١			
.ra 504 modulation(50111111	Úsing IEE	E/EIA 1	LOΩ,	2,10Ω,45Ω resistive pad
20 dB S/N	1.4	0.7	μV	7 8.1 dBr -110.1 dbm
30 dB S/N	1.9		μV	7 10.8 dBf -107.4 dBm
50 dB S/N	4.8	2.4	μV	7 18.8 dBf -99.4 dBm
Interstation Noise	-10.0		dB	3
Mute start Level	very soft		дp	3.0 μ V receiver input
High cut at 10KHz	3.0	16.0		
FottIF rejection	178.0	89.0	υV	
Image rejection	170.0	07.0	۳.	
FM 100% MODULATION MONO				7 11.2 dBf -107.0 dBm
Usable Sensitivity	2.0	1.0		100 O dBm
50dB S/N	3.2	1.6	μv dB	2010 412
Maximum S/N	55.0 0.6		uв	
THD % AM Rejection at 1mV	44.8		dВ	3
FM 100% MODULATIONSTEREO	4400			
				dBf dBm
OBUDIC Delit-	BLEND		μV μV	inc dom
	BLEND 65.0		dB	must be measured with volume
Maximum S/N THD %	1.3			set just below clipping
1KHz separation	. 31.0		đВ	
10KHz separation	24.8		dΒ	3
Stero Blend action:				39.2 dBf -79.0 dBm
Separation at 25µVreceive	14.9		dB	,
67KHz SCA Rejection	54.0		dВ	
δF=5KHz	-40.0		dВ	3
19 and 38KHzproducts	-40.0		~	•
FM TWO SIGNAL TESTS (98.1 708 µV (-50 dBm)	MHz)			
Capture Ratio	7.5		dΒ	`
Selectivity@ 200KHz	10.0		٦5	3
for 30dB S/N	10.0		dB	
for 50dB S/N	-6.0		dB	•
Selectivity@ 400KHz	>63		dВ	3
for 30dB S/N for 50dB S/N	48.0		dB	3
IM Rejection	20.0	10.0	mV	91.2 dBf -27.0 dBm
(98.9 and 99.7)				
2MHz IM rejection	100.0	50.0	mV	/ 105.2 dBf -13.0 dBm
(99.1 and 100.1)		25.0	17	99.2 dBf -19.0 dBm
IF mix rejection	50.0	25.0	mv	/ 99.2 UBL 1510 UE
(96.4 and 107.2)				
AM 30% MODULATION MONO				
DUMMY ANTENNA:				- 00 0 dp-
20dB S/N	16.0	16.0		
Max S/N	49.0		dB %	3
THD at max S/N	0.2 0.5		8	
THD at 80% mod	0.5		•	
-3dB Audio Response 600KHz	1680.0		Hz	z 2140 in AM stereo position
1400KHz	1680.0		Ηz	z.
±10KHz Selectivity	30.0		dΒ	limited by local AGC
±20KHz Selectivity	nm		dB	
Iocal AGC action:			العسد	
level for -3dB 600KHz des	sired sign	al redu	icti	/ -10.0 dBm
1400KHz	100.0	70.7 70.7	mν	/ -10.0 abiii
10MHz	100.0 100.0	70.7		
27MHz	100.0	70.7	Tr: A	
IF mix rejection (1400 & 945 or 950)	>100		mV	J
(1400 & 945 Or 950) AM stereo:			•	The second of th
50% modulation			_	
Separation	30.0		dB	
max S/N	>45.0		dB	5

DELCO Channel Characteristics

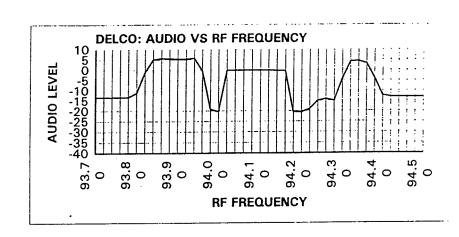
94.1MHZ

Audio VS RF Frequency

Note:

- * The results here represent a chacteristic receiver input signature based on sweeping the RF signal through the desired channel
- * The test signal is modulated with 1khz @ 100%
- * The measurements are made using 15khz low pass and CCIR filters with quasi-peak detection
- * RF level is -62dBm

RF	AUDIO
FREQ.	LEVEL
93.70	-13.2
93.72	-13.2
93.74	-13.2
93.76	-13.2
93.78	-13.2
93.80	-11.1
93.82	-1
93.84	5.14
93.86	5.8
93.88	5.64
93.90	5.5
93.92	5.52
93.94	5.87
93.96	-0.3
93.98	-19.1
94.00	-20.3
94.02	-0.15
94.04	0
94.06	0
94.08	0
94.10	0
94.12	0
94.14	-0.22
94.16	-0.42
94.18	-20.37
04.20	20.57



Tuning Frequency

94.12	0
94.14	-0.22
94.16	-0.42
94.18	-20.37
94.20	-20.57
94.22	-18.92
94.24	-14.9
94.26	-14.06
94.28	-14.97
94.30	-4.3
94.32	4.2
94.34	4.35
94.36	3.16
94.38	-3.9
94.40	-12.3
94.42	-13.1
94.44	-13.1
94.46	-13.1
94.48	-13.1
94.50	-13.1

RMc

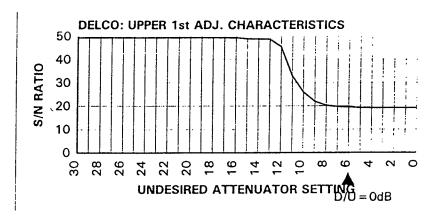
Delco Adjacent Channel Characteristics

Upper first adj. channel 94.3mhz

Note

- * The results here represent a chacteristic receiver input signature based on ramping the undesired signal up in 1dB increments and recording the signal to noise ratio.
- * The measurements are made using a 15khz low pass and CCIR filters with quasi-peak detection
- * The interfering signal is modulated with clipped pink noise
- * SCA's (group B) are employed on both the desired and the undesired signals.

UNDES.	RADIO
ATTEN.	S/N (dB)
40	
39	
38	
37	
36	
35	
34	
33	
32	
31	
30	49.5
29	49.5
28	49.5
27	49.5
26	49.5
25	49.5
24	49.5
23	49.5
22	49.5
21	49.5
20	49.5
19	49.5
18	49.5
17	49.5
16	49.5
15.	49
14	48.9
13	48.8
12	45.5
11	33
10	26
9	22
8	20.2
7	19.7
6	19.5
5	19.3



D/U = 0dB

19.2 19.2

19.2

19.2

19.2

3

1

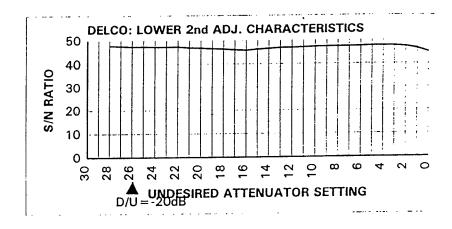
DELCO Adjacent Channel Characteristics

Lower second adj. channel 93.7mhz

Note:

- * The results here represent a chacteristic receiver input signature based on ramping the undesired signal up in 1dB increments and recording the signal to noise ratio.
- * The measurements are made using a 15khz low pass and CCIR filters with quasi-peak detection
- * The interfering signal is modulated with clipped pink noise
- * SCA's (group B) are employed on both the desired and the undesired signals.

UNDES.	RADIO
ATTEN.	S/N (dB)
40	
39	
38	
37	
36	
35	
34	
33	
32	
31	
30	
29	
28	47.75
27	47.5
26	47.35
25	47.3
24	47.2
23	47.2
22	47.24
21	46.9
20	46.75



D/U = -20dB

46.5 19 46.3 18 46 17 16 45.8 46.25 15 14 46.5 13 46.8 46.9 12 47 11 47.2 10 9 47.25 47.3 8 47.3 7 6 47.4

47.5

47.6

47.5

47.2

46.4

44.9

5

4

3

1

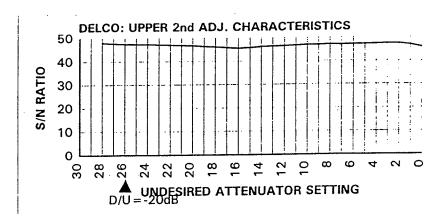
Delco Auto Radio Adjacent Channel Characteristics

Upper second adj. channel 94.5mhz

Note:

- * The results here represent a chacteristic receiver input signature based on ramping the undesired signal up in 1dB increments and recording the signal to noise ratio.
- * The measurements are made using a 15khz low pass and CCIR filters with quasi-peak detection
- * The interfering signal is modulated with clipped pink noise
- * SCA's (group B) are employed on both the desired and the undesired signals.

UNDES.	RADIO	
ATTEN.	S/N (dB)	
40		
39		
38		
37		
36		
35		
34		
33		
32		
31		
30		
29		•
28	47.9	
27	47.6	
26	47.4	D/U = -20dB
25	47.3	
24	47.3	
23	47.1	
22	47	
21	46.8	
20	46.7	
19	46.3	
18	46.1	
17	45.8	
16	45.5	
15	45.7	
14	46.1	
13	46.3	
12	46.4	
11	46.6	
10	46.8	
9	46.9	
8	47	
7	46.9	
6	47	
5	47.1	
4	47.2	
3	47.3	
2	47.3	
1	46.8	
0	45.8	



DAR FM TEST RECEIVER DATA Receiver Lab #2 Type High End Home Hi-Fi

Index

Page	Description
1	Laboratory FM -> FM D/U Ratios
2	Radio Characterization/Confirmation
3	Signal, Noise, & Separation VS RF Level
4	Graph of Signal & Filtered Noise VS RF Level
5	Graph of Separation VS RF Level
6	Graph of Signal, Noise, Filtered Noise, & Separation VS RF Level
7	Woodstock Engineering Receiver Test Report
8	Audio VS RF Frequency Test (no measurement made)
9	Receiver Upper 1st Adjacent Interference/Noise
10	Receiver Lower 2nd Adjacent Interference/Noise
17	Receiver Upper 2nd Adjacent Interference/Noise

FM -> FM Laboratory Measurements for the Denon Model TU-380 RD Laboratory Receiver #2

Type: High end home Hi-Fi

Measurements were made at a moderate signal level of -62 dBm.

The signal to noise ratio was set at 45 dB and this measurement was made using a 15kHz low pass and a CCIR filter with quasi-peak detection.

Test Results:

Co-Channel	D/U	43.39	đВ
Lower First Adjacent	D/U	23.61	đВ
Upper First Adjacent	D/U	12.46	đВ
Lower Second Adjacent	D/U	-24.67	đВ
Upper Second Adjacent	D/U	-33.18	đВ

ELECTRONIC INDUSTRIES ASSOCIATION

Digital Audio Radio Laboratory

Engineers:

RMc/DL

DATE:

2/21/95

PROJ.:

RADIO CHARACTERIZATION/CONFIRMATION

- * Key point measurements for comparison to Grossjean data
- * Additional data with regard to audio performance VS RF level

TEST SET-UP

- Denon TU-380RD * Receiver:
- * Ant. Net:

50/75 ohm resistive pad (-7.8dB insertion loss)

- * Audio Ref: 724mVrms
- * Receiver in "Auto" Mode for stereo tests
- * Receiver in manual mode for mono tests
- * Test Bed, W/Orban Stereo Gen & Harris Exciter as Signal Source
- * Audio measurements made with Audio Precision as rms unweighted

(TEST FQ. 94.1MHZ) **FM TESTS**

S/N RATIO - 1KHZ, 100% MOD

70dB MAX

(mono)

THD - 1KHZ, 100% MOD (-50dBm)

MONO

0.17 %

STEREO

0.24 %

LIMITING THRESHOLD (Audio -1dB)

-106dBm

HIGH CUT THRESHOLD

Audio: 10KHZ, L+R, 100% Mod, Pilot off

NA due to mute

SEPARATION @ -62dBm

SELVINY	011 6 0200		
Freq.	L->R	R->L	
1KHZ	-38dB	-37dB	(W/O Pre-Emph)
10KHZ	-35dB	-34dB	(W/O Pre-Emph)

SIGNAL, NOISE & SEPARATION VS RF LEVEL

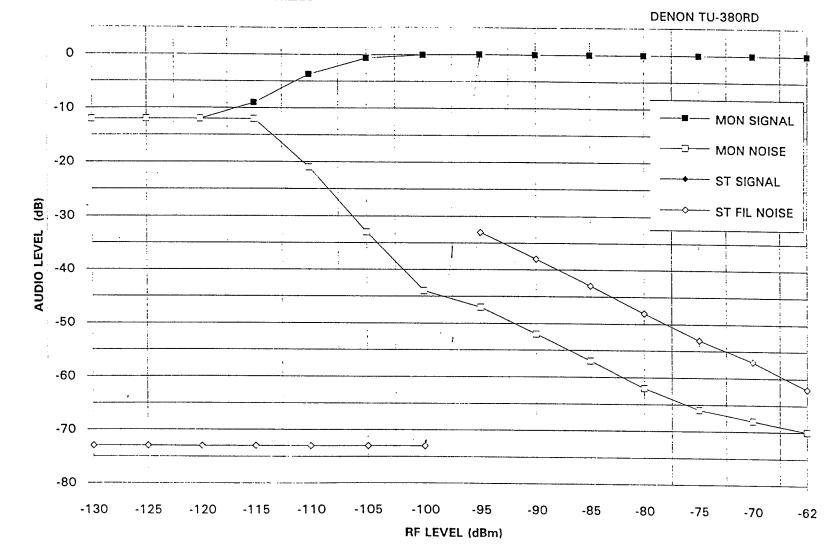
- Left channel used as the measurement channel for Signal and Noise data
- Left channel driven (L only) for separation data
- Audio test frequency = 1KHZ
- * Receiver in "Manual" mode for Mono measurements, "Auto" mode for stereo measurements
- * RF levels represent power into the receiver after 50/75 ohm conversion

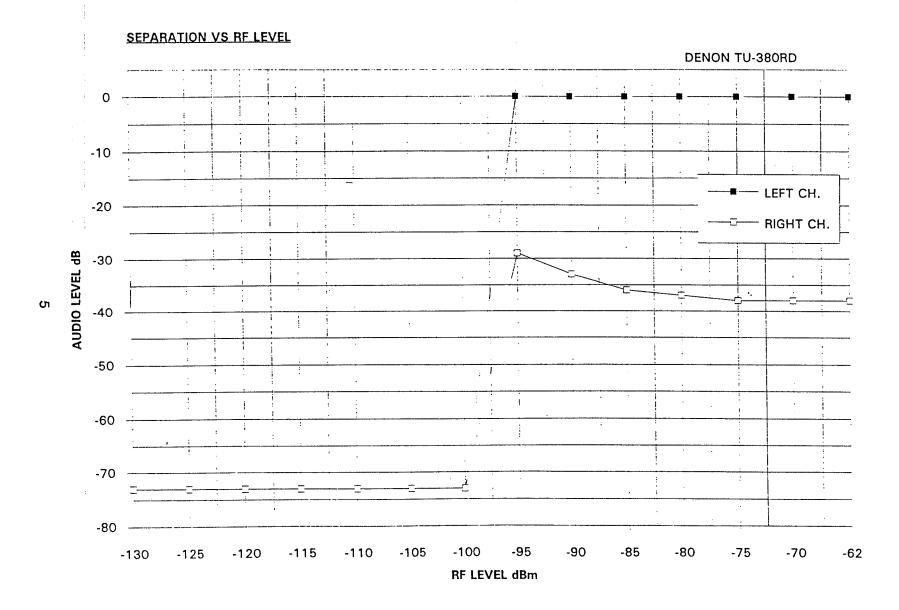
CURVE DATA

SIGNAL, NOISE & SEPARATION VS RF LEVEL

	mon	o (L)		Stereo (L)	, ,,,,,,		Separation	L->R
RF Level	Signal	Noise	Signal	Filt. Noise	Noise	RF Level	Left	Right
dBm	₫B	dB	dB	dB	dB	dBm	dΒ	dB
-130	-12	-12	-73	-73	-73	-130	-73	-73
-125	-12	-12	-73	-73	-73	-125	-73	-73
-120	-12	-12	-73	-73	-73	-120	-73	-73
-115	-9	-12	-73	-73	-73	-115	-73	-73
-110	-3.7	-21	-73	-73	-73	-110	-73	-73
-105	-0.6	-33	-73	-73	-73	-105	-73	-73
-100	-0.05	-44	-73	-73	-73	-100	-73	-73
-95	0	-47	0	-33	-33	-95	0	-29
-90	0	-52	0	-38	-38	-90	0	-33
-85	0	-57	0	-43	-43	-85	0	-36
-80	0	-62	0	-48	-48	-80	0	-37
-75	0	-66	0	-53	-53	-75	0	-38
-70	0	-68	0	-57	-57	-70	0	-38
-62	0	-70	0	-62	-62	-62	0	-38
-57	0	-70	0	-64	-64	-57	0	-38
						ļ		
						<u>. </u>		

SIGNAL & FILTERED NOISE VS RF LEVEL





Engineers: RMc/DL

Date: 6/12/95

Print Date: 8/7/95

PROJECT: RECEIVER CHARACTERIZATION

Radio Mfg.: PANASONIC Model No.: RX-FS430 Serial No.: GR3J01184

Index

Page	Description
1	Cover
2 & 3	AM data regarding test set up and measurements including; Distortion, selectivity and signal & noise levels at various RF levels ranging from -45dBm to -130dBm
4	Plot of signal and noise VS RF level

Notes:

- * All measurements made as RMS, unweighted.
- * RF levels represent power at the receiver
- * Tone control set full clockwise for minimum high cut
- * Output set to standard output level of 1V rms

Engineers: RMc/DL

Date: 6/12/95

Print Date: 8/7/95

PROJECT: RECEIVER CHARACTERIZATION

Radio Mfg.: Panasonic Model No.: 16192463 Serial No.: 1000499

AM TESTS (TEST FREQ. 1660KHz)

TEST SET-UP

Ant. Network: None

(radio modified for 50 ohm input)

Audio Ref.: 1,0Vrms

(0dB)

Rec. set up: Tone control maximum clockwise

Test Bed: Test Bed: Boonton RF generator used as signal source

2 & 3 Audio measurements made with Audio Precision as rms unweighted

THD - 400HZ, 80% MOD (-47dBm) 4 1.20 %

STEREO

%

5

Selectivity (RF level = -95dBm)

+ 10KHz = 17dB

+20KHz = 30dB

-10KHz = 14dB

-20KHz = 27dB

Average = 15.5dB

Average = 28.5dB

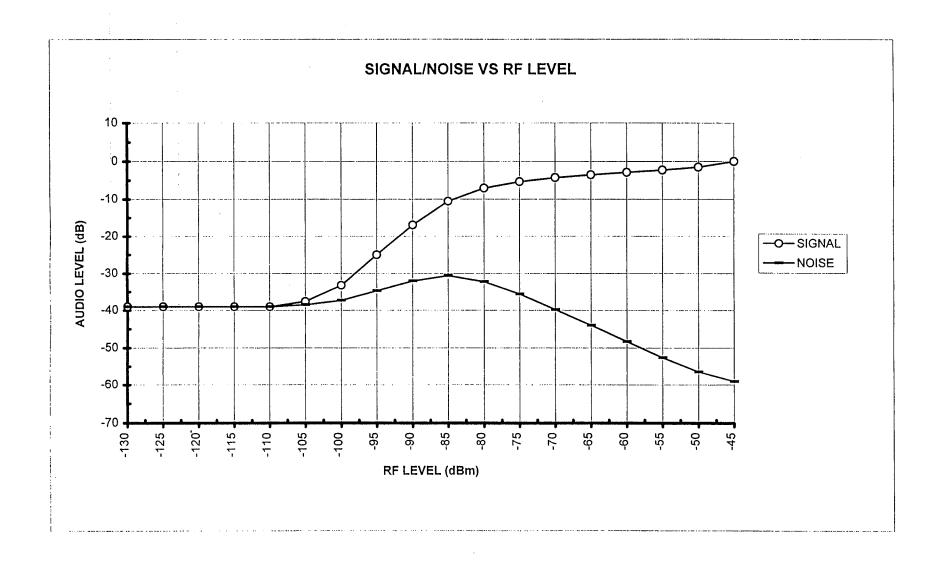
AUDIO VS RF LEVEL MEASUREMENTS

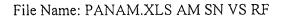
- * Left channel used as the measurement channel for Signal and Noise data
- * Audio test frequency = 400HZ
- * "Signal" modulation level = 80%
- * Audio line transformer (1:1 ratio) used to eliminate ground loop introducing RF noise into the screen box This receiver only was affected in this manner

CURVE DATA

Audio VS RF Level

RF Level							ر او د د در د داو	
	Signal	Noise	Signal	Noise	Signal	Noise	Left	Right
dBm	dB	dB	dB	dB	dB	dB	dB	dB
-130	-39	-39				71.7		
-125	-39	-39						
-120	-39	-39						
-115	-39	-39	•					
-110	-39	-39						
-105	-37.6	-38.5						
-100	-33.2	-37.3						
-95	-25	-34.7						
-90	-16.9	-32						
-85	-10.5	-30.6						
-80	-7	-32.2						
-75	-5.29	-35.6						
-70	-4.26	⁷ -39.8						
-65	-3.5	-44						
-60	-2.85	-48.3						
-55	-2.24	-52.6				•		
-50	-1.5	-56.5						
-45	0	-59						





Engineers: RMc/DL

Date: 6/12/95 **Print Date:** 8/7/95

PROJECT: RECEIVER CHARACTERIZATION

Radio Mfg.: DELCO Model No.: 16192463 Serial No.: 1000499

Index

Page		Description
1		Cover
2 &3		AM data regarding test set up and measurements including; Distortion, selectivity and signal & noise levels at various RF levels ranging from -45dBm to -130dBm
4		Plot of signal and noise VS RF level with the receiver set for "Narrow Band" operation
5		Plot of signal and noise VS RF level with the receiver set for "Wide Band" operation
6	i	Plot of receivers audio frequency response in both wide and narrow band modes
Notes:		
	* * *	All measurements made as RMS, unweighted. RF levels represent power at the Dummy Antenna input Automobile receivers output connected to four ohm loads and set to standard reference output level of 1 Watt. Automobile receivers balance and fade controls set to "detent" positions. Tone controls set for "flat" operation.

Engineers: RMc/DL

Date: 6/12/95

Print Date: 8/7/95

PROJECT: RECEIVER CHARACTERIZATION

Mfg.: Delco Model No.: 16192463 Serial No.: 1000499

AM TESTS

(TEST FREQ. 1660KHz)

TEST SET-UP

Ant. Network: JFW composite antenna dummy

Audio Ref.: 2.0Vrms Load Imp = 4 ohms

Rec. set up: Loudness off, graphic EQ flat, balance & fade centered

Test Bed: Test Bed: Boonton RF generator used as signal source

Meas.: Audio measurements made with Audio Precision as rms unweighted

THD - 400HZ, 80% MOD (-47dBm)

MONO 1.30 % STEREO %

Selectivity (RF level

(RF level = -105dBm)

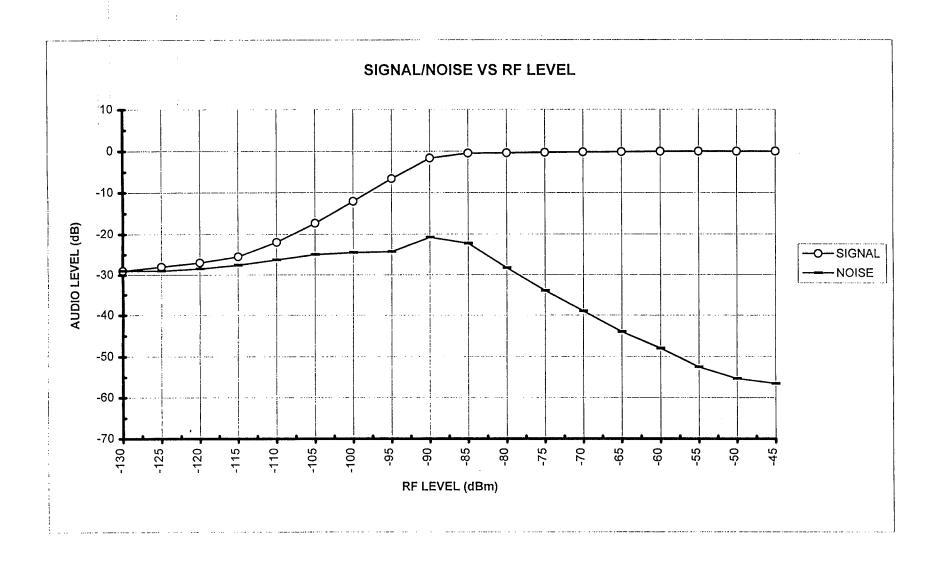
Narrow	· Wide	Narrow	Wide	
+10KHz = 33dB	+10KHz = 22dB	+20KHz = NA	+20KHz = NA	
-10KHz = 35dB	-10KHz = 11.5dB	-20KHz =	-20KHz =	
Average = 34dB	Average = 16.75dB	Average =	Average =	

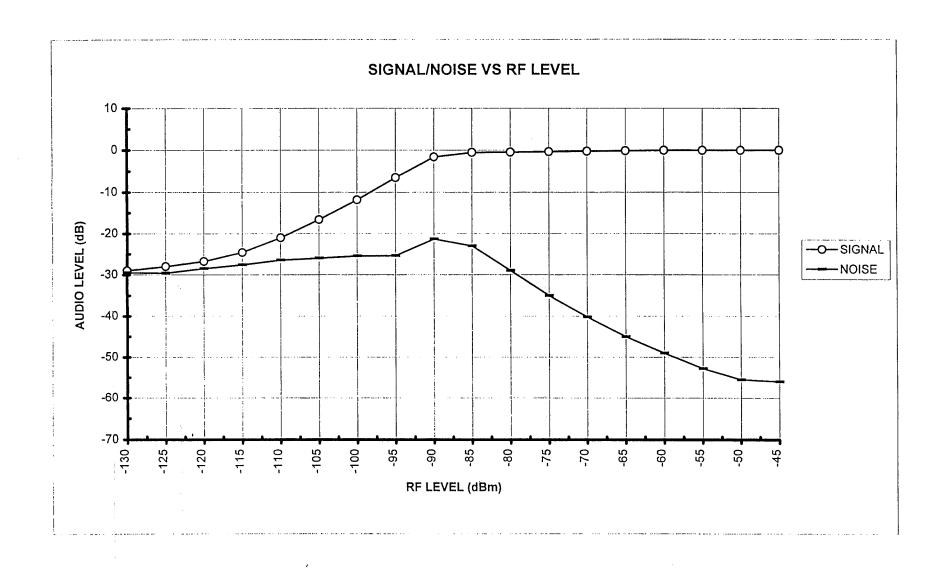
- * Left channel used as the measurement channel for Signal and Noise data
- * Audio test frequency = 400HZ * "Signal" modul 35.3dB
- * "Wide Band" refers to "Am-St" selected
- * "Narrow Band" refers to "Am-St" off

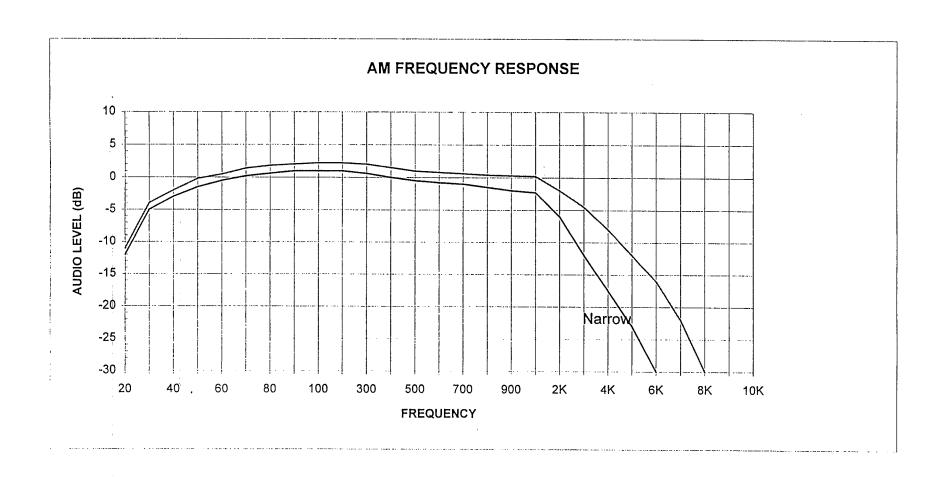
CURVE DATA

Audio VS RF Level

RF Level	Wide Band		Narrow Band				1 1 1 1 1 1 1 1 1	are the end of
	Signal	Noise	Signal	Noise	Signal	Noise	Left	Right
ent power into th	dB	dB	dB	dB	dB	dB	dB	dB
-130	-29	-29	-29	-29.5			QD .	ub .
-125	-28	-29	-28	-29.6				
-120	-27	-28.5	-26.8	-28.5				
-115	-25.5	-27.6	-24.6	-27.6				
-110	-22	-26.3	-21	-26.5			 	
-105	-17.3	-25	-16.6	-26			-	
-100	-12.1	-24.5	-11.8	-25.5			 	
-95	-6.6	-24.3	-6.5	-25.4	-		 	
-90	-1.65	-20.8	-1.6	-21.3			 	
-85	-0.44	-22.3	-0.5	-23				
-80	-0.38	-28.3	-0.45	-29		·		
-75	-0.27	-34	-0.32	-35			 	
-70	-0.15	-39	-0.2	-40.2	-			
-65	-0.1	· -44	-0.1	-45				
-60	0	-48	0	-49				
-55	0	-52.5	0	-52.8			 	
-50	0	-55.3	0	-55.5			-{ 	
-45	0	-56.5	0	-56			 	







Engineers: RMc/DL

PROJECT: RECEIVER CHARACTERIZATION

Radio Mfg.: Denon

Model No.: TU-680NAB Serial No.: 2092400103

Index

Page	Description
1	Cover
2 & 3	AM data regarding test set up and measurements including; Distortion, selectivity and signal & noise levels at various RF levels ranging from -45dBm to -130dBm
4	Plot of AM signal and noise VS RF level with the receiver set for "Narrow Band" operation
5	Plot of AM signal and noise VS RF level with the receiver set for "Wide Band" operation
6	Plot of AM audio frequency response in both wide and narrow band modes
Notes:	·
*	All measurements made as RMS, unweighted. RF levels represent power at the receiver after 50/75 ohm conversion

File Name: D680AM.XLS Cover

Print Date:8/7/95

Engineers: RMc/DL

PROJECT: RECEIVER CHARACTERIZATION

Radio Mfg.: Denon

Model No.: TU-680NAB Serial No.: 2092400103

AM TESTS

(TEST FREQ. 1660KHz)

TEST SET-UP

Ant. input: 75 ohms

Audio Ref.: 672mVrms (0dB)

Rec. set up: "Bandwidth" wide or narrow selected for specific tests

Test Bed: Test Bed: Boonton RF generator used as signal source

Meas.: Audio measurements made with Audio Precision as rms unweighted

THD - 400HZ, 80% MOD (-47dBm)

1111	TOURALI	0070	11202	.,,
MONO			3.50) %
STERE	0			%

Selectivity	(RF level = -95dBm)
	AND THE PROPERTY OF THE PROPER

File Name: D680AM.XLS AM DATA

Narrow	, Wide	Narrow	Wide
+ 10KHz = 10.5dB	+ 10KHz = 10.5dB	+20KHz = 65dB	+20KHz = 64dB
-10KHz = 7dB	-10KHz = 7dB	-20KHz = 51dB	-20KHz = 51dB
Average = 8.75dB	Average = $8.75dB$	Average = 58dB	Average = 57.5dB

Page 2 of 6

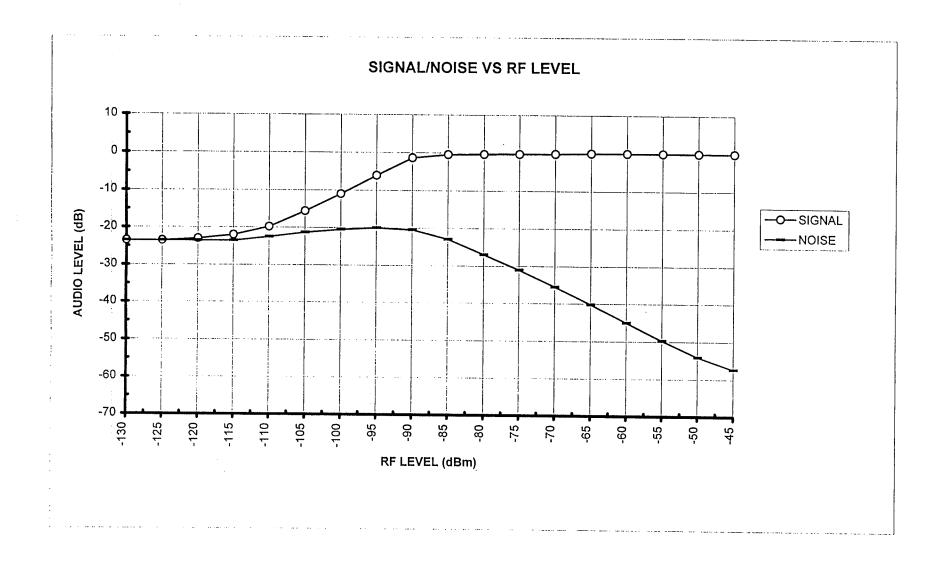
AUDIO VS RF LEVEL MEASUREMENTS

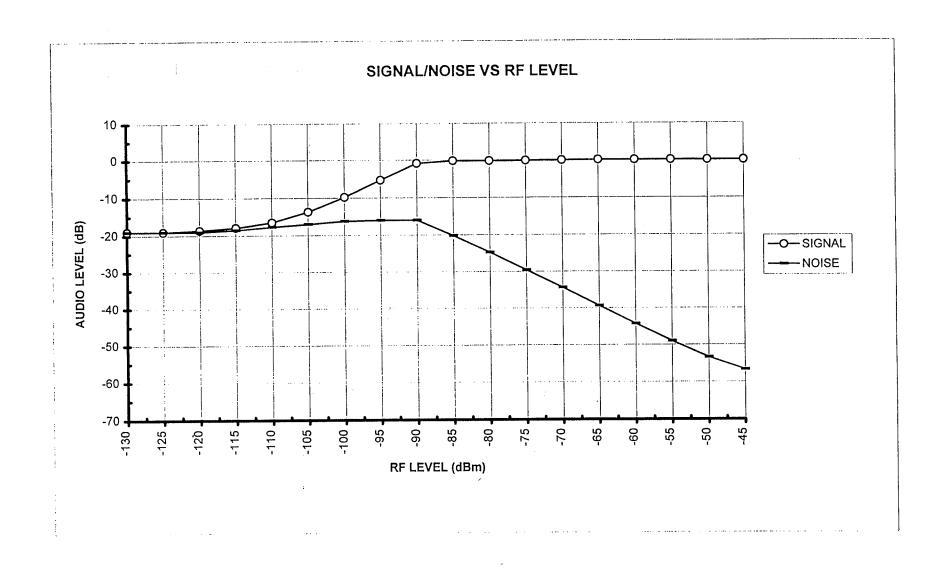
- * Left channel used as the measurement channel for Signal and Noise data
- * Audio test frequency = 400HZ
- * Signal modulation level = 80%

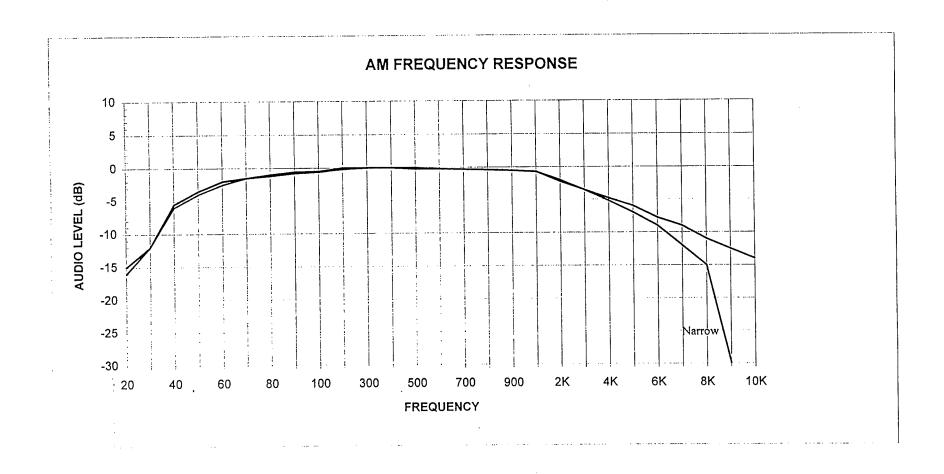
CURVE DATA

Audio VS RF Level

RF Level	Wide Ba	ind	Narrow B	and				
	Signal	Noise	Signal	Noise	Signal	Noise	Left	Right
dBm	dB	dΒ	dB	₫B	dB	dB	dB	dB
-130	-19	-19	-23.5	-23.5				
-125	-19	-19	-23.5	-23.5				
-120	-18.6	-19	-23	-23.5				
-115	-17.9	-18.5	-22	-23.5				
-110	-16.5	-17.7	-19.8	-22.5				
-105	-13.7	-17	-15.6	-21.3				
-100	-9.8	-16.2	-11	-20.5		**		
-95	-5.3	-16	-6	-20			****	
-90	-0.85	-16	-1.35	-20.5			······································	
-85	-0.2	-20.2	-0.42	-22.9				
-80	-0.18	-24.8	-0.32	-27				
-75	-0.13	-29.6	-0.25	-31				
-70	-0.08	-34.4	-0.14	-35.5				
-65	0	-39.4	0	-40.2				
-60	0	-44.3	0	-45			4	1
-55	0	· -49	0	-49.7			****	
-50	0	-53.4	0	-54.1		·····		
-45	0	-56.7	0	-57.3				







Appendix I – Subcarrier Calibration

SCAcal

ELECTRONIC INDUSTRIES ASSOCIATION

Digital Audio Radio Laboratory

Cutting Edge stereo gennerator used

Engineers:

RMc/DL

DATE:

5/12/95

SCA CALIBRATION

Measurements made with Belar Wizard Modulation Monitor
Typical measurements reflect numbers that one should see on this monitor
The Monitor indicates 1% at CW and 101% at 100% (with reference to Besel null method)
All individual SCA's were cross checked on Seiko RPA Spectrum Analyzer
Seiko RPA Spectrum Analyzer calibration checked with HP8566B Spectrum Analyzer
Group output from SCA Mixer box used for both group and individual SCA's
Individual SCA levels set in SCA Mixer box

GROUP A

GROUP A					
FREQ.	INJ. LEV.	MEAS.	SOURCE	TYPE	COMMENTS
57khz	3.00%	4.00%	RE533	RDS	Phase locked to pilot sync port
66.5khz	8.50%	11.00%	SEIKO	HS Data	Phase locked to pilot sync port
				•	Injection level accurately set with RPA utility
92khz	8.50%	9.00%	CRL	Analog	
TOTAL	20.00%	21.00%			
GROUP B					
FREQ.	INJ. LEV.	MEAS.	SOURCE	TYPE	COMMENTS
57khz	10.00%	11.00%	RE533	RDS	Phase locked to pilot sync port
67khz	10.00%	11.00%	CRL	Analog	
TOTAL	20.00%	21.00%			
GROUP C					
FREQ.	INJ. LEV.	MEAS.	SOURCE	TYPE	COMMENTS

FREQ.	INJ. LEV.	MEAS.	SOURCE	TYPE	COMMENTS	
67khz	10.00%	11.00%	CRL	Analog		
92khz	10.00%	11.00%	CRL	Analog		
TOTAL	20.00%	21.00%				

GROUP D

FREQ.	INJ. LEV.	MEAS.	SOURCE	TYPE	COMMENTS	
92khz	10.00%	11.00%	Mainstream	Data		
TOTAL	10.00%	11.00%				

ELECTRONIC INDUSTRIES ASSOCIATION

Digital Audio Radio Laboratory

Engineers:

RMc/DL

DATE:

5/12/95

SCA GROUP / ANALOG CALIBRATION

EQUIPMENT:

CUTTING EDGE STEREO GENERATOR BELAR WIZARD MODULATION MONITOR

SET-UP GUIDE

CUTTING EDGE:

PROCESSORS SET FOR CLIPPED PINK NOISE SETTINGS

(C.E.)

STEREO MODE

: ON

PILOT

: ON

PILOT LEVEL ATTENUATOR:

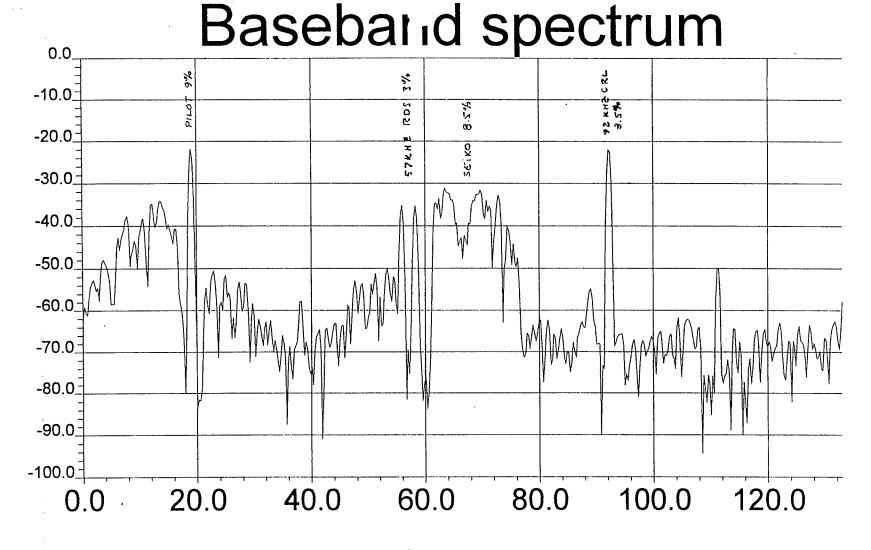
82

OUTPUT LEV ATTENUATOR:

SEE CHART

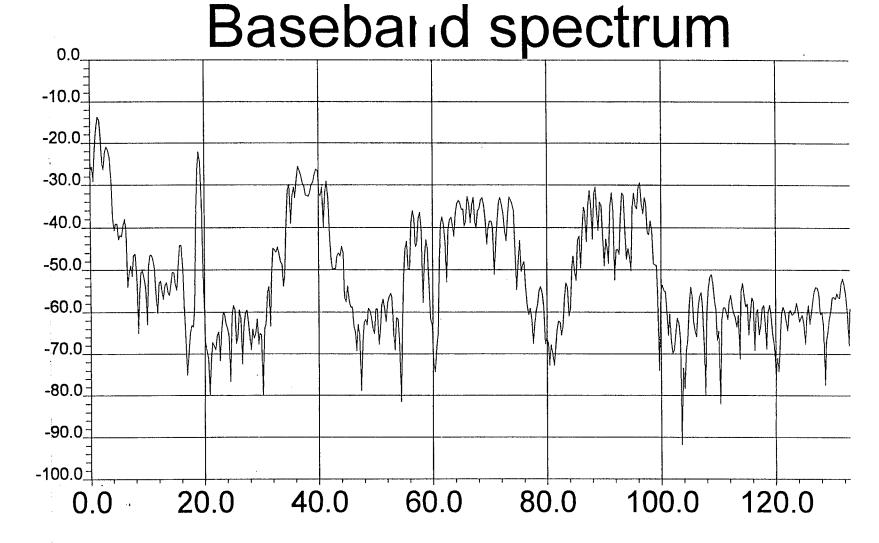
CALIBRATION

	C.E. Output	BELAR %
SIGNAL	Level Setting	Measurement
CW (PILOT OFF)	47	4
PILOT ONLY	47	13
PILOT+SCA GRP A	42	30
PILOT+SCA GRP B	42	30
PILOT+SCA GRP D	47	23
PILOT +PINK NOISE	47	100
PINK NOISE+GRP A	42	110
PINK NOISE+GRP B	42	110
PINK NOISE+GRP D	47	110



Frequency (kHz) Bandwidth 494 Hz

RF input -32.6 dBm 5/11/95 RMS/DL CUTTING COGG PILOT + GROUP A SCA'S Frequency 94.1 WEIA

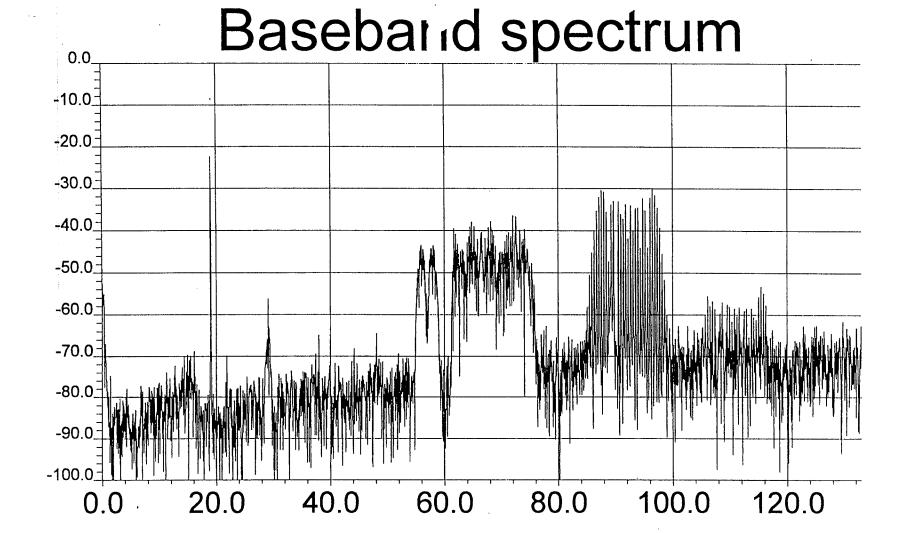


Frequency (kHz) Bandwidth 494 Hz

LZ FM REF SCA GRP A

RF input -47.0 dBm

Frequency 94.1 WEIA



Frequency (kHz) Bandwidth 62 Hz

6/07/95

SCA GROUP A 57KHZ RDS, 66.5KHZ SEIKO, 92KHZ (MODULATED)

RF input -47.7 dBm Frequency 94.1 WEIA

Pilot phase: 62.1

(SEIKO)

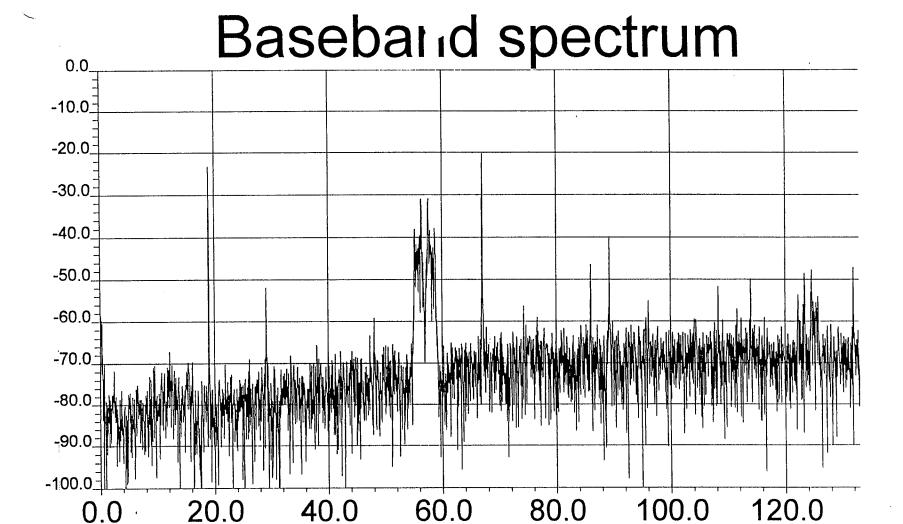
Pilot Injection: 8.9%

Pilot Frequency: 18999.1

Subcarrier Injection: 8.5% (SEIRO)

RF input -32.6 dBm

Frequency 94.1 WEIA

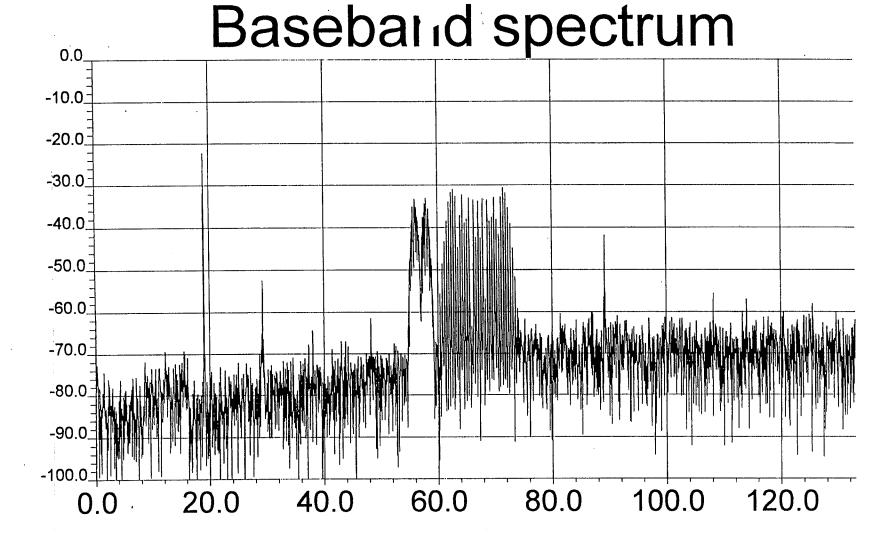


Frequency (kHz) Bandwidth 62 Hz

6/07/25

SCA GROUP B 57 KHE RDS & 67KHZ (UNMODULATED)

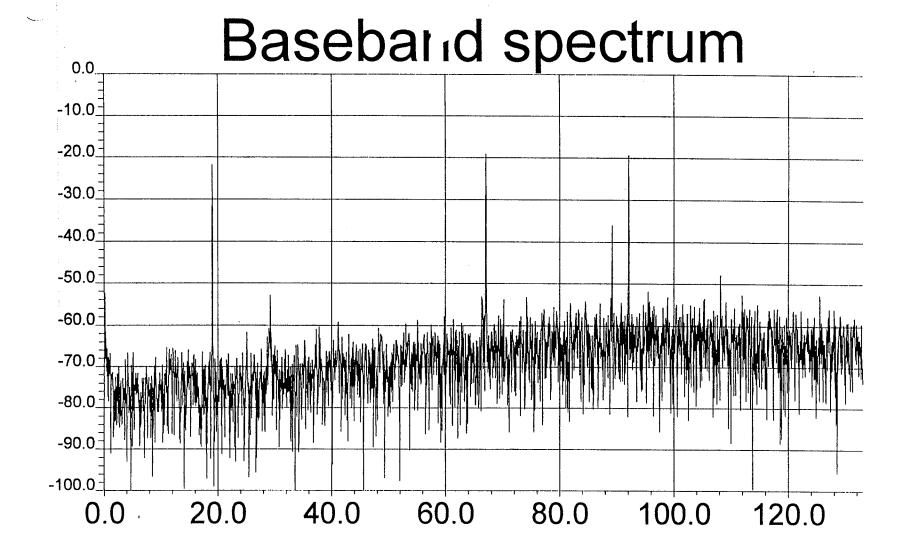
RF input -53.0 dBm Frequency 94.1 WEIA



Frequency (kHz) Bandwidth 62 Hz

6/07/95 SCA GROUP TO STRHZ ROS & GTKHZ (MODULATED)

RF input -50_4 dBm Frequency 94.1 WEIA



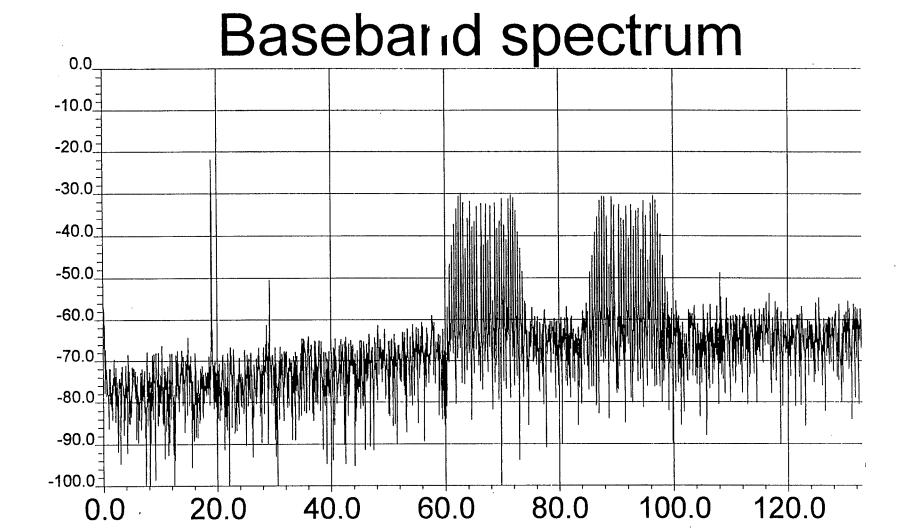
Frequency (kHz) Bandwidth 62 Hz

6/07/95

SCA GROUP C (67KHZ & 92KHZ) UNMODULATED

RF input -56.2 dBm

Frequency 94.1 WEIA

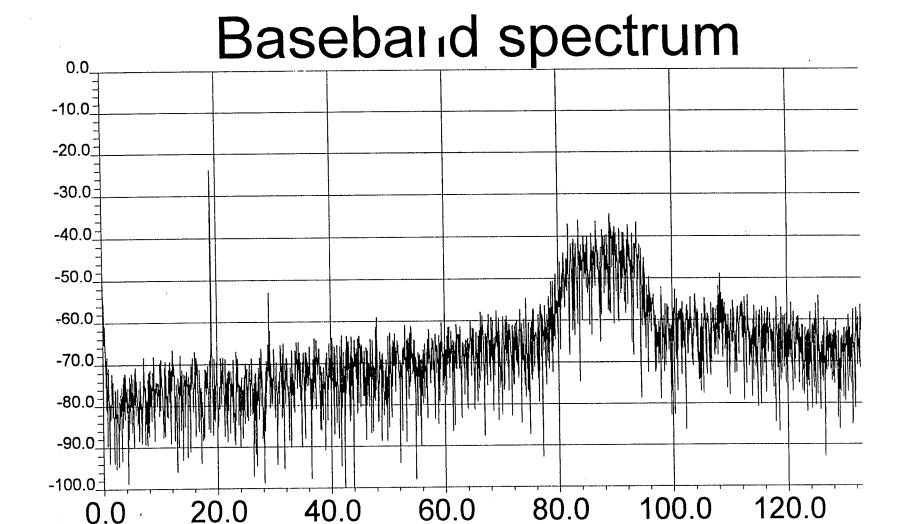


Frequency (kHz) Bandwidth 62 Hz

6/07/95

SCA GROUP C (67KHZ & 92KHZ) MODULATED

RF input -56.3 dBm Frequency 94.1 WEIA



Frequency (kHz) Bandwidth 62 Hz

G/07/95 SCA GROUP D 92KHZ MAINSTREAM DATA

RF input -56.5 dBm Frequency 94.1 WEIA

June 14, 1995

Mr. Robert McCutcheon, EIA DAR Testing Lab NASA Lewis Research Center

Dear Mr. McCutcheon,

Thank you for hosting me at the DAR testing facility on June 7th. During the visit I was able to determine that the Mainstream Data FM subcarrier equipment had been installed correctly, and was operating at 10% injection. Upon addition of RF noise the signal degradation initially produced "first-level" error correction activity as expected, and with additional signal degradation the "second-level" error correction also became active, as expected.

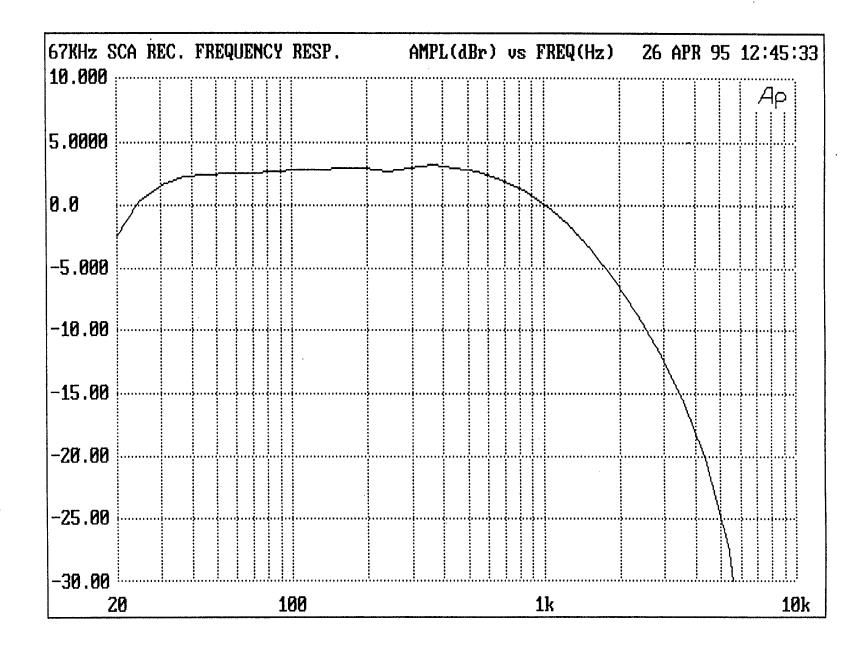
As we discussed during the visit there is a threshold of errors above which we deem the performance unacceptable. Admittedly, this is not a sharp line, given that some customers are more sensitive to errors than others. Our installers attempt to orient the customer antenna to eliminate ANY error correction activity during the several minutes they would be allotted to stay and watch the LCD display. This is achievable in most cases. If the antenna cannot be adjusted to bring the error rate below five to ten first-level error corrections per minute then the site is deemed a non-FM site and other arrangements are made to deliver the data to the customer by other more costly means. Rapid first-level error events, while they may all be corrected over any short observation period, indicate that there is not enough performance margin left to trust the site performance over a variey of weather and other anomalous conditions.

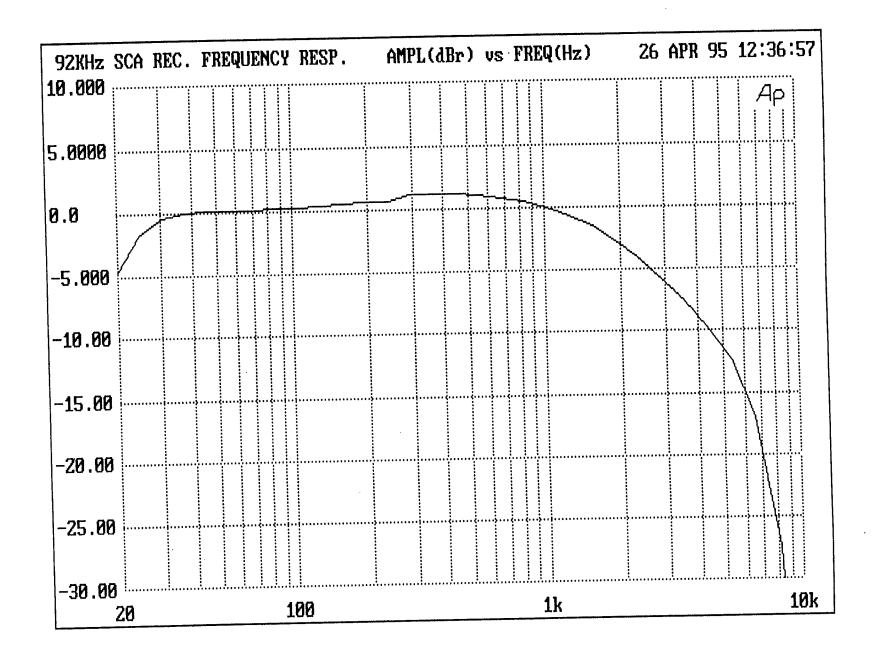
Sincerely,

Bruce Rothaar V.P. Engineering

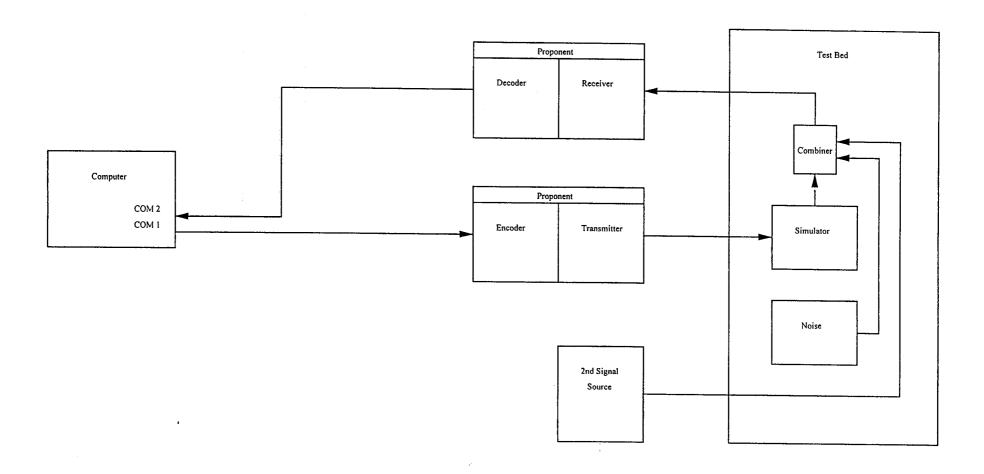
Bruce Rothan

Mainstream Data (801) 584-2800





Appendix J – Ancillary Data Channel



Testing DAR ancillary data channels

Proponent ancillary and auxillary data channels must conform to PC-compatible COM1, COM2 port electrical and mechanical specifications. Electrical specifications follow RS-232 conventions. Mechanical specification are that it is a male DB-9 connector with signals as follows:

pin	signal	sense	function
2	RxD	in	input data
3	TxD	out	output data
5	GND		
7	RTS-	out	input flow control
8	CTS-	in	output flow control

The test platform ("EIA Test PC") is an IBM-compatible 486DX PC running DOS 5.0 or higher with COM1 and COM2 ports and VGA display. In addition, two "Null Modem" cables are required to connect the test PC to the proponent transmitter and receiver. Test PC COM1 connects to the proponent transmitter, and Test PC COM2 connects to proponent receiver. Figure 1 shows the connections for each of the proponents that uses the PAC codec. A "Null Modem" cable has connections as follows:

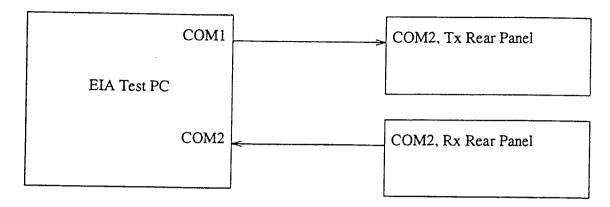
from	to
pin	pin
2	3
3	2
5	5
7	8
8	7

The Test PC runs the supplied C-language program $ck_data.c$ that transmits a known sequence of 8-bit bytes out the Test PC's COM1 port and checks that the Test PC receives them on its COM2 port. Currently the sequence is the 8-bit values 0, 1, 2, ... 255, 0, 1, ... although this can be easily changed in the test program source code. This C-language test program requires the following commercial software:

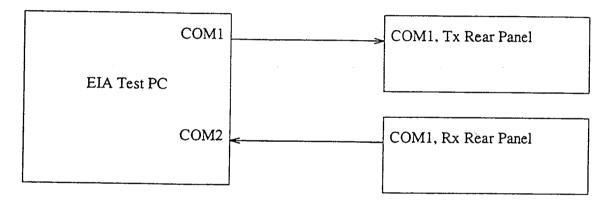
Microsoft C/C++ compiler version 7.0 (also called "Visual C++").
Greenleaf Commlib level 2, version 4.0 or higher
Greenleaf Software, Inc.
(214) 248-2561

On executing the data check program $ck_data.exe$ the Test PC will display several statistics, all referenced to when the test began or to when the test counts were reset. They are: the current elapsed time for the test, the data rate over the channel under test, expressed in bits per second (bps), the total number of bytes received, and the number of errored bytes received.

TESTING AT&T ANCILLARY DATA CHANNEL



TESTING AT&T AUXILLARY DATA CHANNEL



TESTING AT&T/AMATI OR JPL ANCILLARY DATA CHANNEL

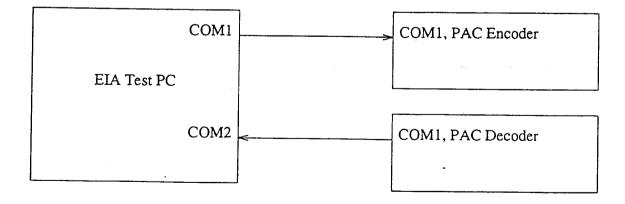


Figure 1. Block Diagram for Testing DAR Data Channels

```
inogram to test ancillary data channels
 * COM1 transmits data test pattern
   COM2 receives data test pattern
    COM1 and COM2 run with hardware flow control (CTS-/RTS-)
         at 19200 baud.
   the output test pattern is byte sequence of i%255 for i=0.1,2... or
         0,1.2,...255,0.1...
    any desired output sequence can be generated by modifying
         function "next_byte()'
   the input data is flagged as "illegal" until SYNCLIM correct bytes
         are received, after which the consecutive correct bytes
         are counted and the received bits per second (bps) are computed.
 *
        with each received byte counted as 8 bits. Elapsed time is displayed.
   from keyboard:
         "ESC" key quits program
         "R" key resets counts
 × /
#define DEBUG
#include (stdio.h)
#include (stdlib.h)
#include (graph.h)
#include (time.h)
#include (string.h)
#include (math.h)
#include 'commlib.h'
#include 'asclidef.h'
#include "ibmkeys.h"
#include (io.h)
#define TITLE
                "EIA DAR Ancillary Data Channel Test"
#define BAUD
                19200L
#define SYNCLIM 32
#define T_ROW
                                 /* title */
                1
#define C_ROW
                5
                                 /* com port status */
#define L_COL
                0
                                 /× left margin */
#define M_COL
                                 /* middle tab stop */
                60
PORT
        *port[2]:
int
        com_port[2], com_char. com_ref:
long
        com_bytes. com_sync. com_err:
        buffer[80];
char
double
       interval;
extern void main(int argo.char **argv );
extern void open_com(int i);
extern
        void put_port_char(int i);
        void get_port_char(int i):
extern
extern
        int next_byte(int *i);
extern
       void show_port_stat(void):
extern
        void clr_com_stats(void):
        int time_int(int mode.double *interval.int step):
extern
        void show_time(double interval);
extern
void
main(argo, argv)
int argc:
```

char **arav:

```
int o. i. j:
         /* set defaults */
         com_port[0] = COM1:
         com_port[1] = COM2:
         com_char = 0;
         if (DEBUG) (
                 printf("sending on COM1 at %ld baud\n". BAUD);
                 printf("receiving on COM2 at %ld baud\n", BAUD);
                 printf("%s. %s\n\n".
                         "ESC to quit",
                         "'R' to reset counts");
                 printf("CR to proceed\n");
                 getchar():
         ł
         /* open ports */
        open_com(B);
        open_com(1);
        clr_com_stats():
        time_int(0. &interval. 1):
        while (1) {
                 if ( gfkbhit() ) {
                         c = getkey():
                         if ( c == ESC ) {
                                 PortClose( port[0] ):
                                 PortClose( port[1] ):
                                 _settextposition(23, D):
                                 exit( 0 ):
                         else if ( c == 'R' ) {
                                 clr_com_stats():
                                 time_int(0. &interval. 1):
                         }
                put_port_char(0):
                get_port_char(1):
                if (time_int(1. &interval. 1)) {
                         show_time(interval):
                         show_port_stat():
                ł
ł
void
open_com(i)
int i;
        port[i] = PortOpenGreenleaf( com_port[i], BAUD. 'N'. 8, 1 );
        if ( port[i]->status ( ASSUCCESS ) {
                printf( "Failed to open COM1 port. Status = %d\n".
                          port[i]->status ):
                exit( 1 ):
        UseRtsCts( port[i], 1 ):
void
put_port_char(i)
```

int 1:

```
if (WhiteChar( port(x), com_char ) == ASSUCCESS) { .
                 next_byte(&com_char):
         }
 1
void
get_port_char(i)
int 1:
         int c. match:
         c = ReadChar( port[i] ):
         if ( c ( ASSUCCESS ) return:
        match = (next_byte(&com_ref) == c) ? 1 : 0;
        com_bytes++:
        if (com_sync++ < SYNCLIM) {
                 if (!match) {
                         com_sync = 0:
                         com_ref = c:
                 ŀ
        else {
                 if (!match)
                         com err++:
                         com_sync = 0:
                         com_ref = c:
                         return:
                 eise {
                         com_sync = SYNCLIM:
                 ł
        ł
ł
int
next_byte(int *p)
        int i = ((*p)+1) & Oxff:
        return(*p = i):
void
show_port_stat()
        long rate:
        rate = 8.0*(double)com_bytes/interval:
        sprintf(buffer,
                "Received: %61d bps at %61d baud". rate. BAUD);
        _settextposition(C_ROW, L_COL):
        _outtext(buffer);
        if (com_sync == SYNCLIM) {
                sprintf(buffer, "%12ld %12s %12ld bytes",
                        com_err. "errors in". com_bytes):
        }
        else {
                sprintf(buffer, "%12ld %12s %12ld bytes",
                        com_sync. "invalid data". com_bytes):
         settextposition(C ROW+1. L COL):
```

```
void
clr_com_stats()
         _clearscreent _GCLEARSCREEN1;
        sprintf(buffer, "%s", TITLE):
         _settextposition(T_ROW, L_COL):
         _outtext(buffer);
        com_bytes = 0:
        com_sync = 0:
        com_err = 0;
}
int
time_int(mode, interval, step)
int mode:
double *interval;
int step:
        long now:
        static long start, prev:
        if (mode == 0) {
                start = ElapsedTime()/1000:
                 prev = start:
                return 0:
        1
        else {
                 now = ElapsedTime()/1000:
                 if (now < (prev + step)) {
                         return 0:
                 }
                 else {
                         *interval = (double)(now - start):
                         prev = now:
                         return 1:
        ł
show_time(interval)
double interval:
        long s = (long)interval:
       -sprintf(buffer, "TIME: %3dh %2dm %2ds".
                (int)(s/(60*60)),
                (int)(s*(60*60))/60.
                (int)(s%60));
        _settextposition(T_ROW, M_COL);
        _outtext(buffer);
```

Appendix K – IBOC System Modifications





600 Mountain Avenue P.O. Box 636 Murray Hill, NJ 07974-0636 908-582-3000

TO:

TOM KELLER, EIA WG B

cc: Dave Londa, DAR Test Lab Manager John Bingham, Amati Communications Corp.

FROM:

Edward Y. Chen, AT&T

DATE:

Friday March 17, 1995

SUBJECT:

AT&T/AMATI EQUIPMENT MODIFICATIONS

- 1. On January 27, the following changes were made in the AT&T/Amati IBOC transmitters,
 - a. In the primary transmitter, resistor R25 (1 K) was bypassed with a wire.
 - b. In the secondary transmitter, an external 2K resistor was replaced by a 750 ohm resistor.
 - c. The changes resulted in an increase in digital power level for both transmitters. Prior to the changes, the digital power was measured to be -28.43 dBm with the composite power (analog + digital) at -7.10 dBm. After the changes, the digital power was increased by 8 dB to -20.44 dBm with the composite power at -7.31 dBm.
- 2. On February 3, 1995, permission was given to allow the Amati analog FM signal to be disabled and substituted by an FM signal generated by a Harris THE -1 FM exciter. The Harris FM signal was combined with the Amati digital signal through an external 10 dB directional coupler. The final power levels were adjusted to be identical to what they were previously with Amati's original FM. The final composite power was -7.16 dBm with analog FM at -7.56 dBm and digital power at -20.46 dBm.
- 3. On March 15, 1995, a front end box was installed and became part of the AT&T/Amati IBOC receiver. This additional box functioned as a broadband attenuator which protected the receiver from being overdriven. Its insertion loss was 2.24 dB.
- 4. The 3 dB bandwidth of the Amati DAB signal is 73.3 Khz per sidelobe, or 146.6 Khz total.

Sincerely,

Edward Y. Chen



March 6, 1995

Tom Keller Consultant and Chairman Working Group B Digital Audio Radio Subcommittee Electronic Industries Association 6721 Clelia Ct Springfield, Virginia 22152-3033

Dear Mr. Keller:

The following changes were made by USADR to its FM-1 IBOC-DAB receiver system, returned today to Cleveland.

- Receiver Equalizer: One field programmable gate array (FPGA) was replaced and demodulator software was upgraded to increase equalizer speed. Software upgrades have been supplied on a microdiskette in a set of DOS files in a directory named "RXCODE5." Software which drives the Transmitter, found in a subdirectory named "TXCODE2," has not been changed.
- 2. Source Decoding: The software upgrade (PROM change) to correct the design error causing unintentional time delay in music decoding was not made. However, a PROM change was made to the error correction decoder. This change is addressed in the attached letter.
- 3. **RF Front End.** Two changes were made. The first change was the removal and replacement of a rotary encoder switch used for tuning the FM frequency. Before this repair, the FM-1 receiver front end would either tune in one direction only or not tune at all. This repair allows the FM-1 receiver front end to tune, in frequency, up and down as well as wrap around.

The second change to the receiver RF front end was the removal of a notch filter assembly, which was never used (system was always operated with this filter assembly bypassed), from an internal compartment labeled "Notch Filters." The notch filter assembly was replaced, in this compartment, with an FMIF bandpass filter, TTE # KC6 10.7 MHz BPF, 500 kHz BW. This bandpass filter improves IF selectivity to a limited degree.

A 6 dB attenuator was added in line between the RF front end and the demodulator to make up for the reduced insertion loss of the bandpass filter installed compared to the notch filter assembly removed.

Tom Keller, EIA WG B March 6, 1995 Page 2 of 2

I certify that no other changes were made to the FM-1 RF front end, demodulator or error correction decoder. I certify that no changes at all were made to the CDQ 2001 MUSICAM decoder. Please refer to the attached letter regarding the changes to the error correction decoder.

Sincerely,

A.J. Vigil

Engineering Manager

cc: Ralph Justus

Electronic Industries Association Consumer Electronics Group 2500 Wilson Blvd

Arlington, VA 22201-3834

cc: Dave Londa

EIA/DAR Test Laboratory NASA-Lewis Research Center 21000 Brookpark Road MS 54-2

Cleveland, OH 44135



March 6, 1995

Tom Keller Consultant and Chairman Working Group B Digital Audio Radio Subcommittee Electronic Industries Association 6721 Clelia Ct Springfield, Virginia 22152-3033

Dear Mr. Keller:

USA Digital Radio thanks the EIA for this system modification and lab retest opportunity.

In the process of executing various system changes which are described in the attached letter, USADR also found it necessary to modify its FM-1 interleaver and deinterleaver.

USADR has executed changes in Chicago to its FM-1 error correction decoder as well as changes in Cleveland, today, to its error correction encoder.

In detail, the following changes were made:

- 1. Error Correction Decoder: Removal and replacement of 3 28-pin PROMS, those PROMS labelled U6, U26 and U46.
- 2. Error Correction Encoder: Removal and replacement of 3 28-pin PROMS, those PROMS labelled U6, U26 and U46.

Although these changes were not previously allowed, USADR respectfully requests the EIA to accept these changes. USADR believes the following reasons justify allowing these changes.

Tom Keller, EIA WG B March 6, 1995 Page 2 of 2

- The modulation waveform remains unchanged, significantly simplifying retesting
 Unimpaired and and and are the significantly simplifying retesting
- 2. Unimpaired audio codec performance is unaltered, significantly simplifying retesting.
- A previous proponent system change was requested which did not impact unimpaired audio codec performance. This proponent change was accepted for similar reasons.
- Retesting, for which USADR has already agreed to pay, has already been scheduled and does not need to be altered.

I certify that no changes were made to the USADR FM-1 system other than those described here and those described in the attached letter documenting previously allowed changes.

Sincerely,

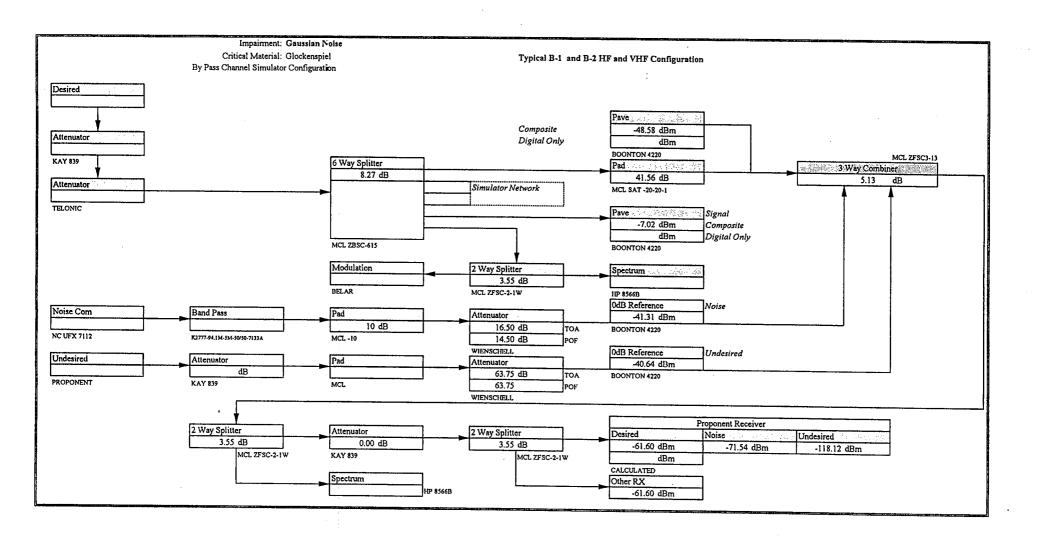
A.J. Vigil

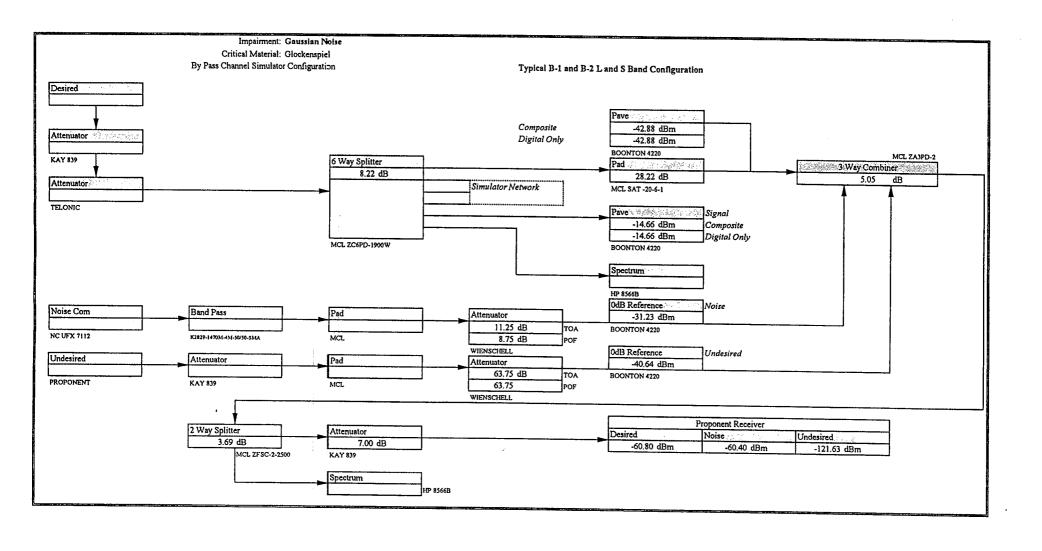
Engineering Manager

cc: Ralph Justus
Electronic Industries Association
Consumer Electronics Group
2500 Wilson Blvd
Arlington, VA 22201-3834

cc: Dave Londa
EIA/DAR Test Laboratory
NASA-Lewis Research Center
21000 Brookpark Road
MS 54-2
Cleveland, OH 44135

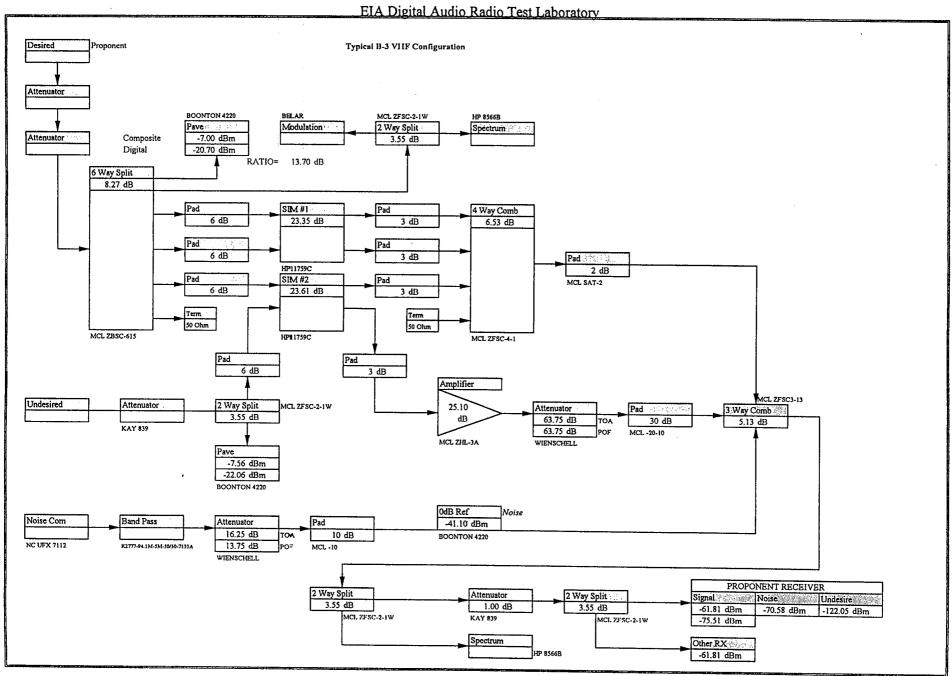
Appendix L – Laboratory RF, Audio and Composite Stereo Distributions

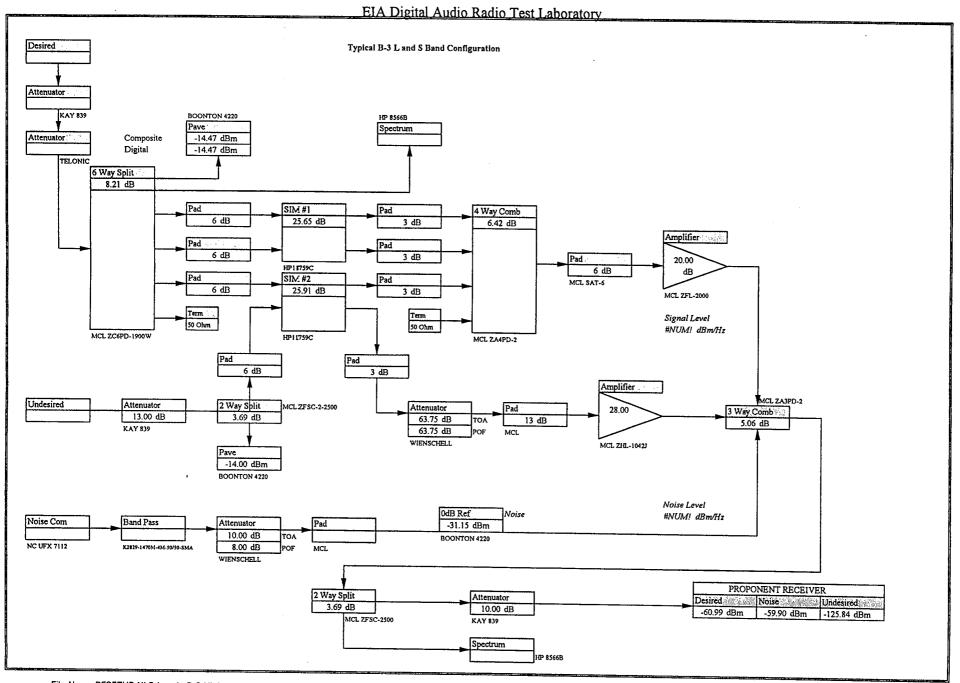




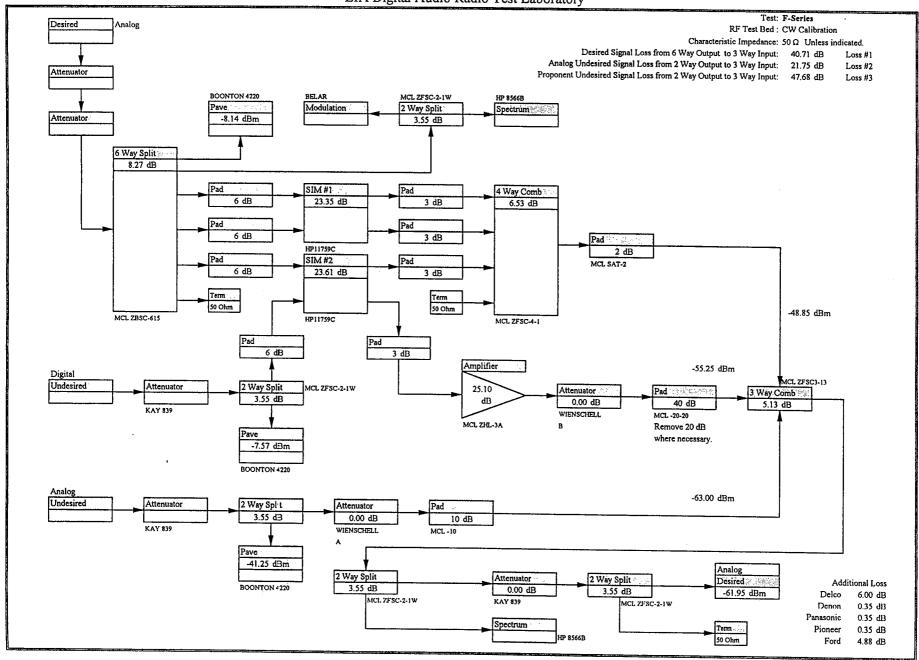


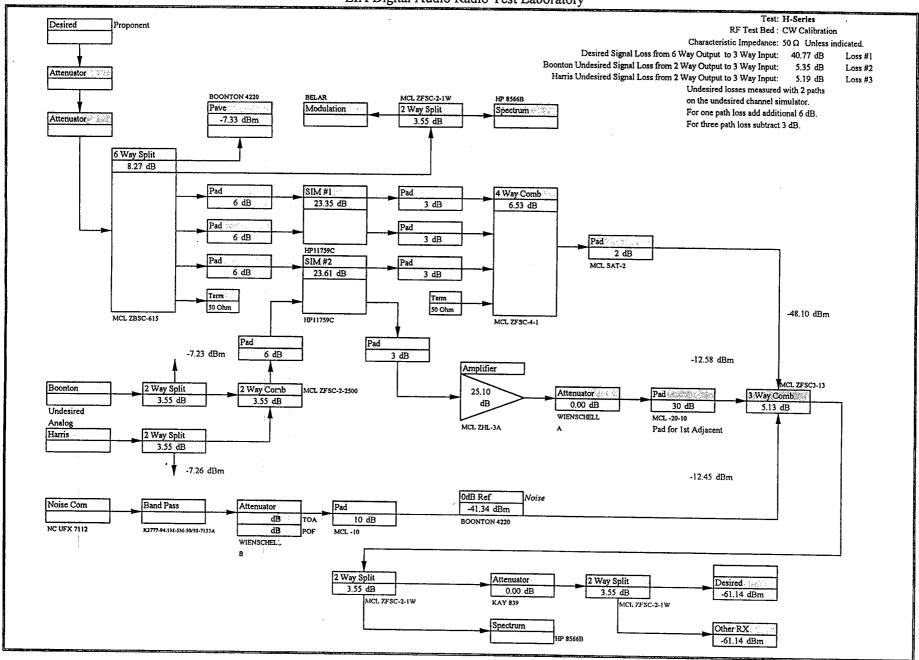
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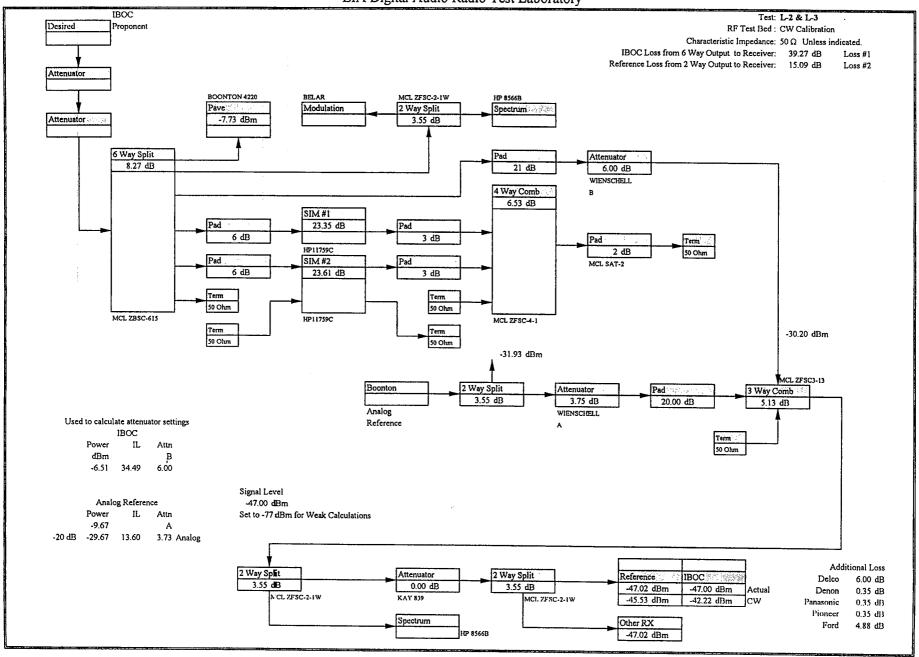


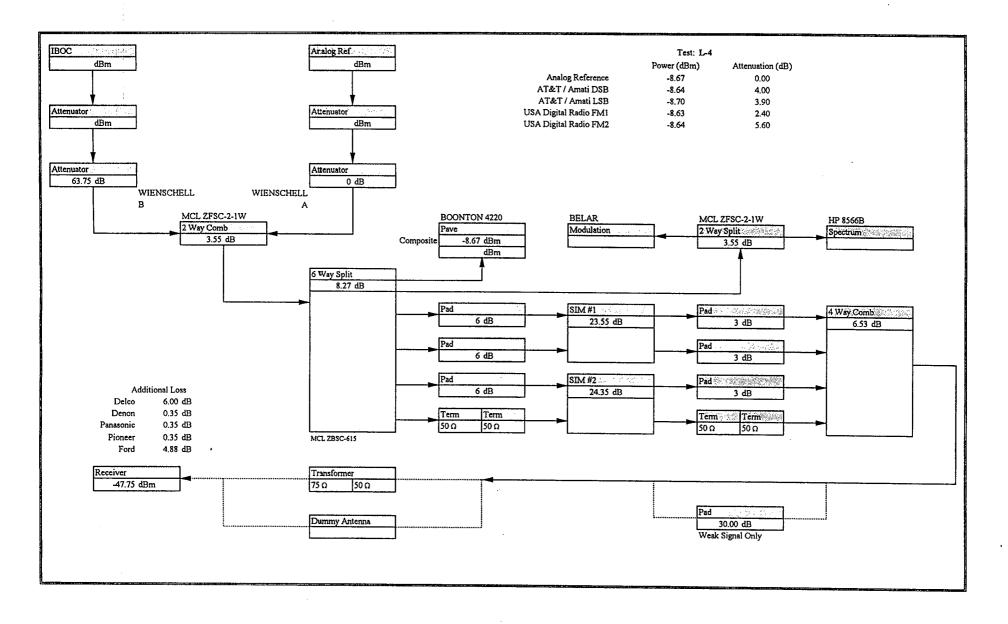


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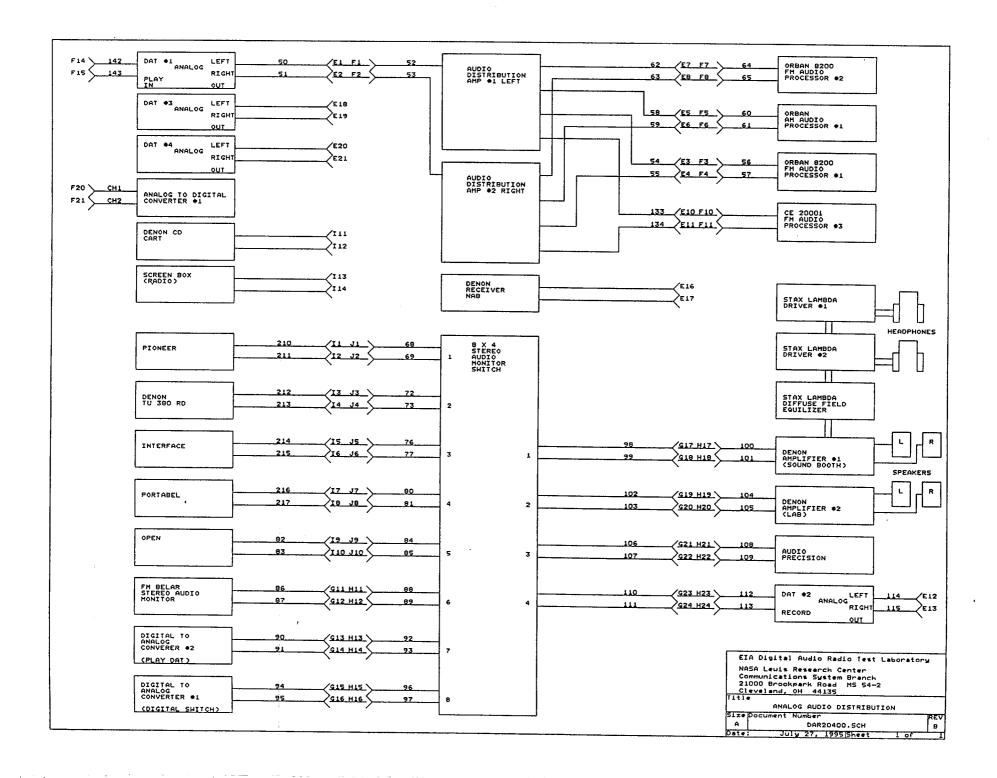


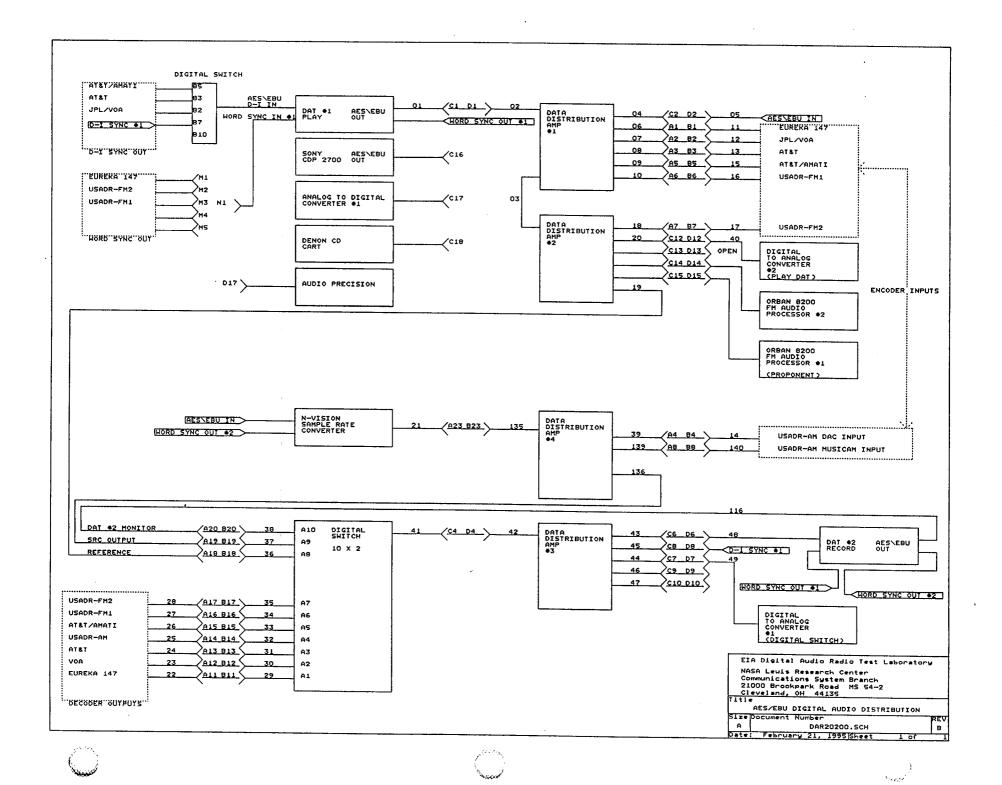


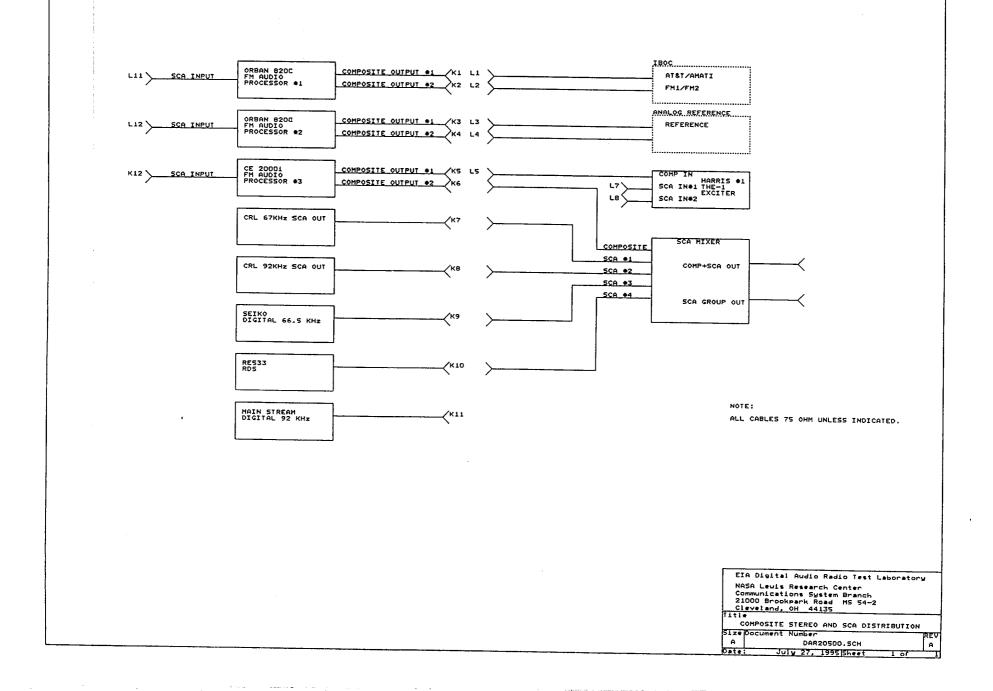


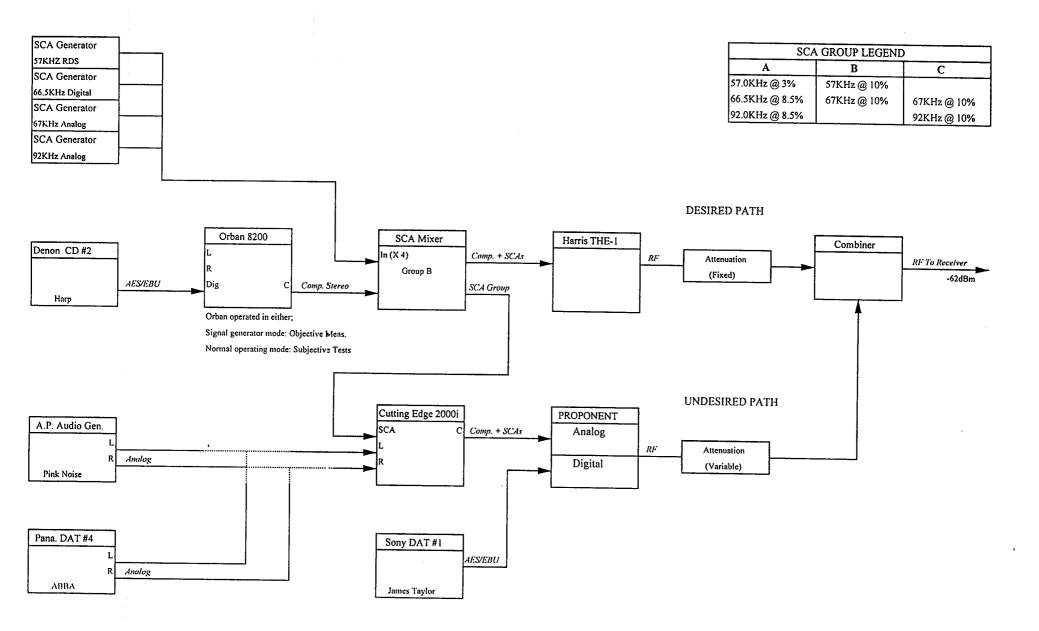


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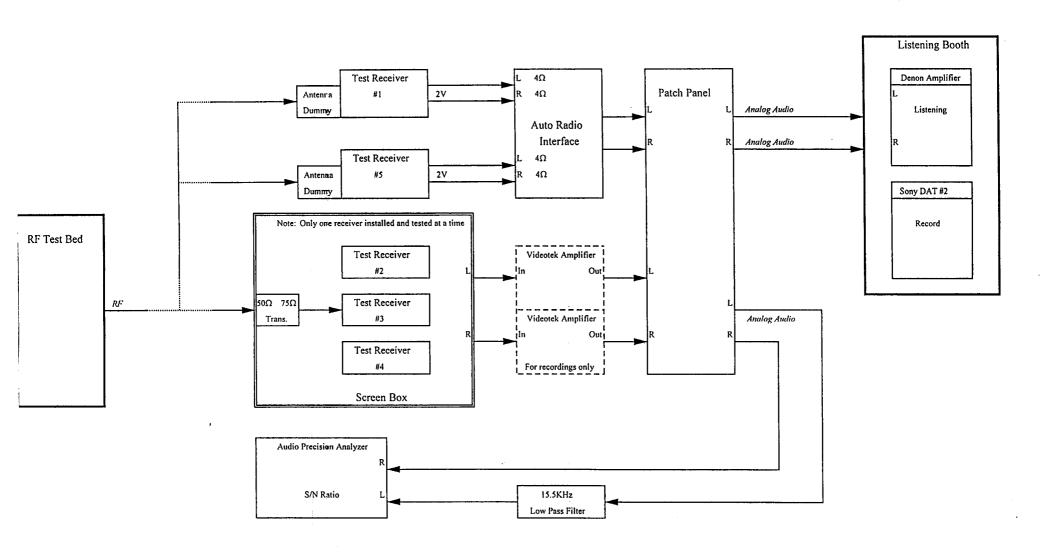


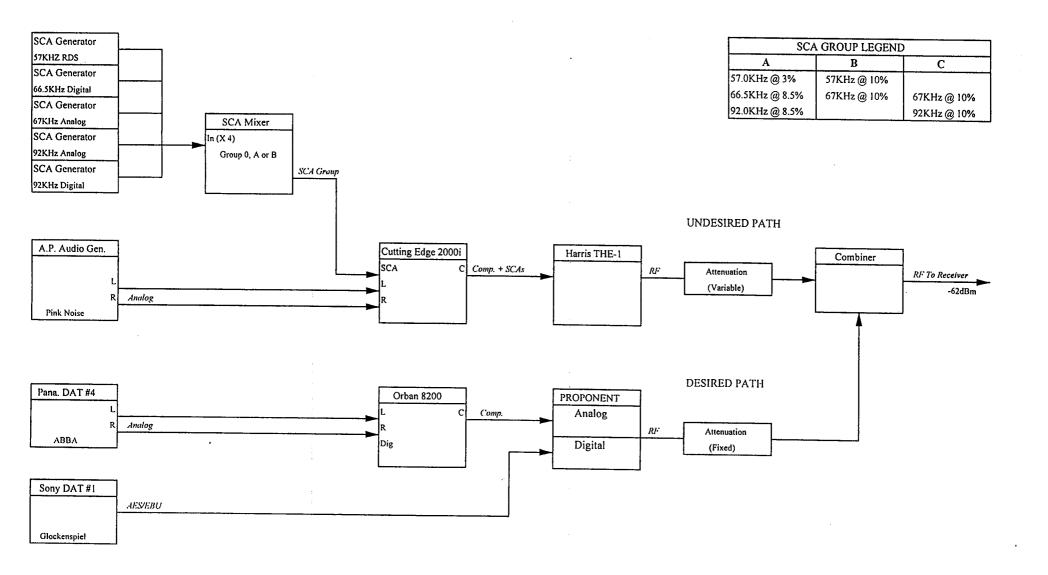




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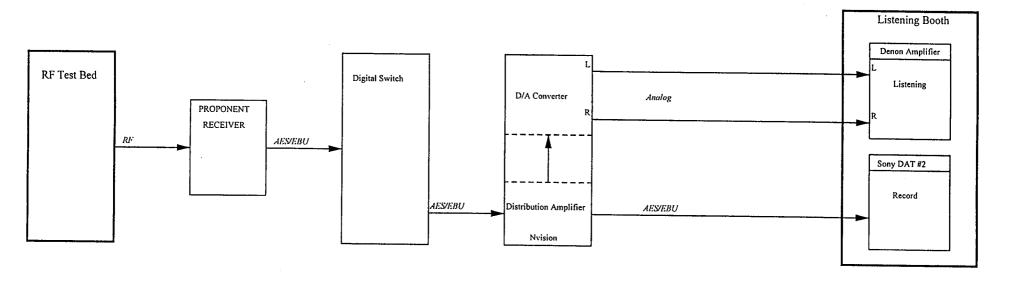
F&G Trans Block

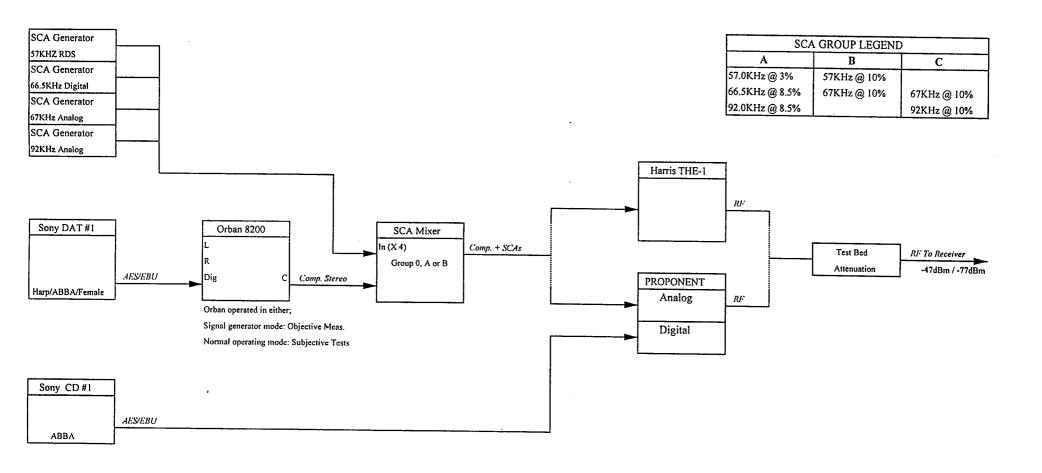




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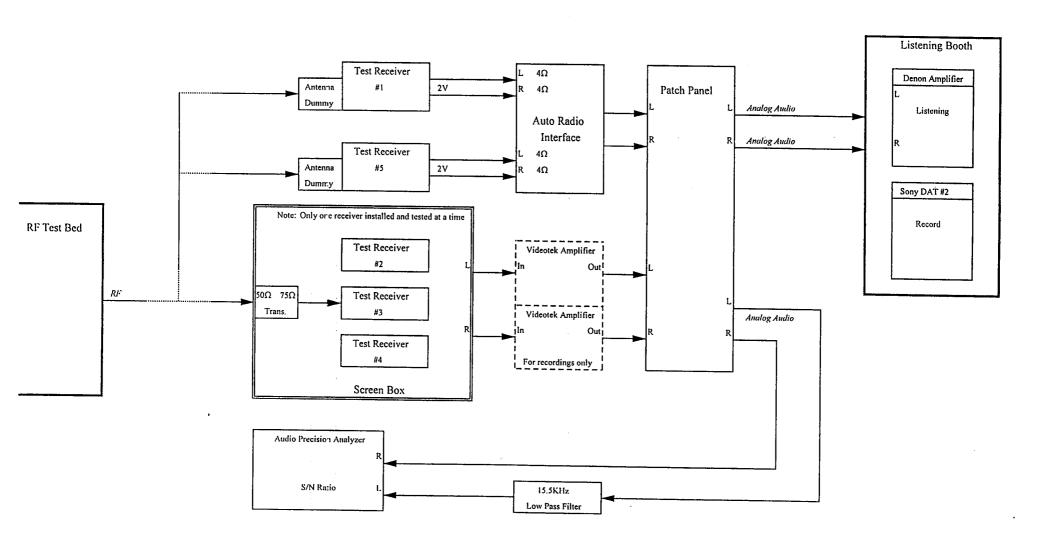
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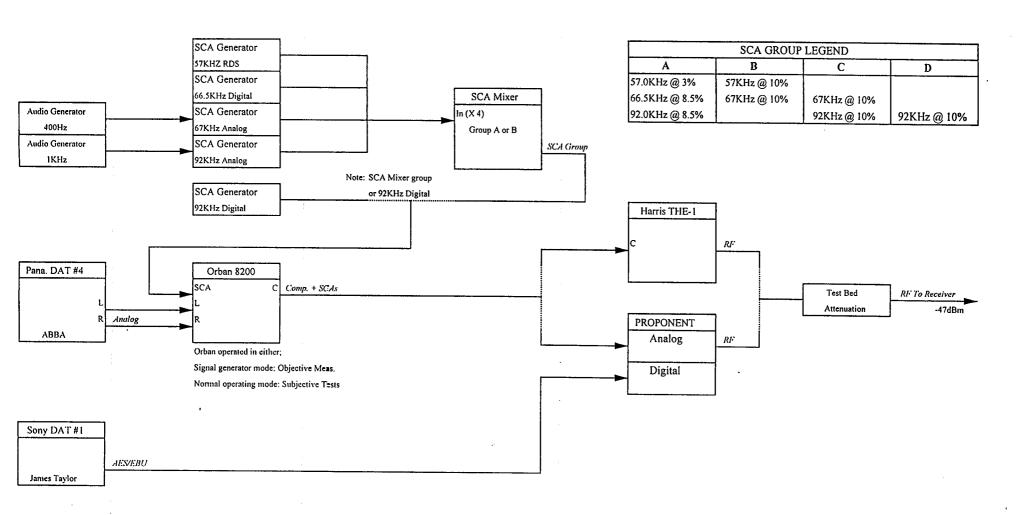




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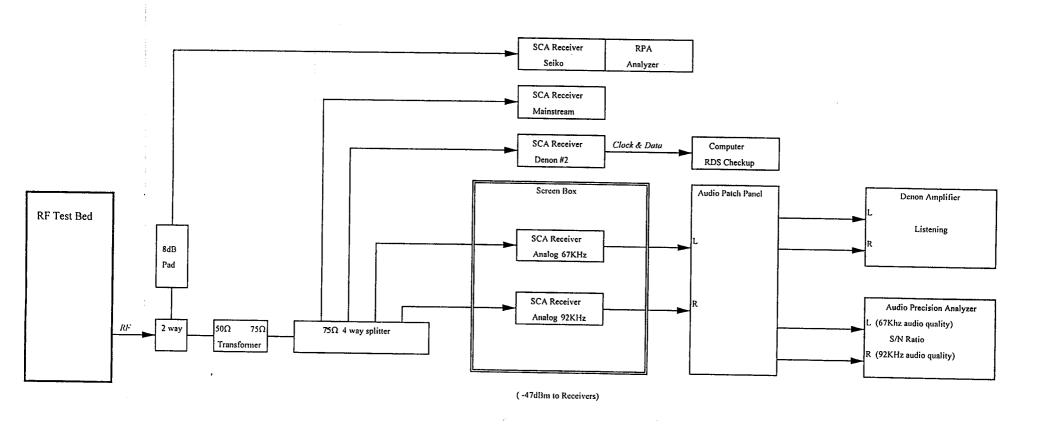
L Trans Block

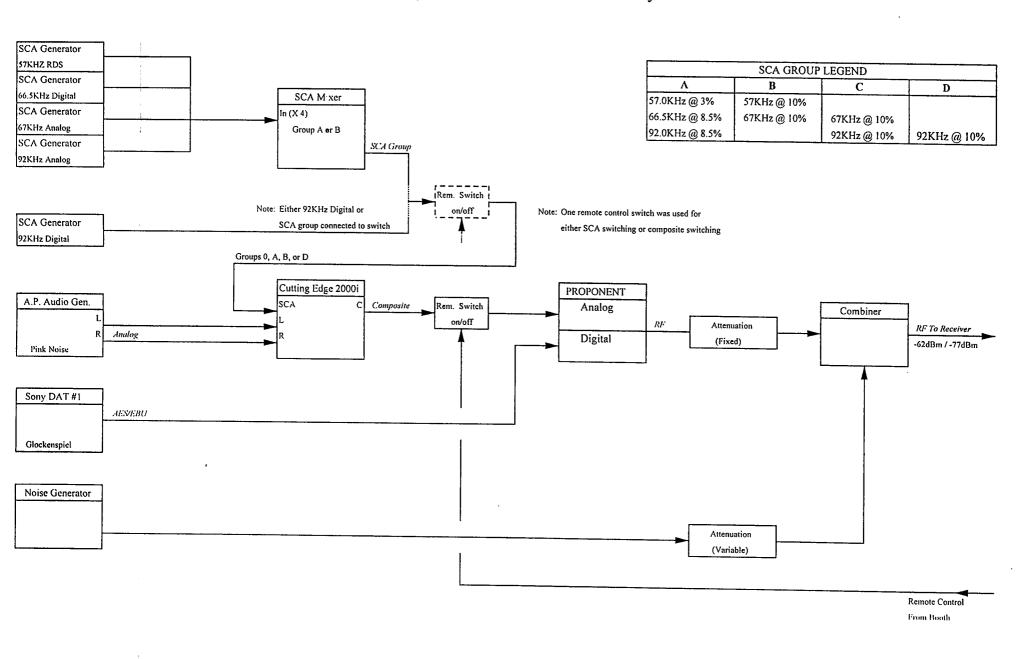




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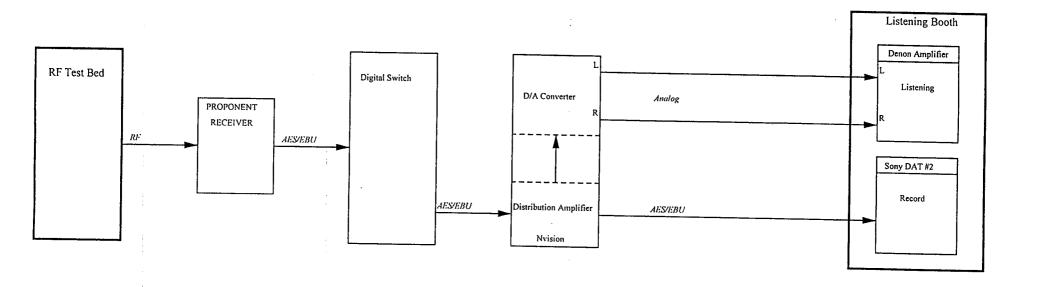
L (SCAs) Trans Block

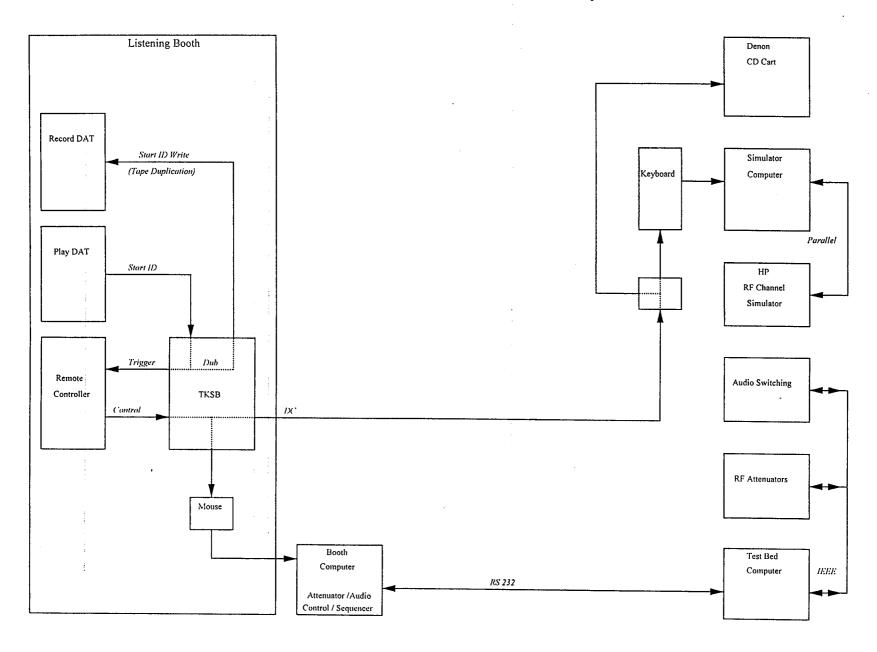




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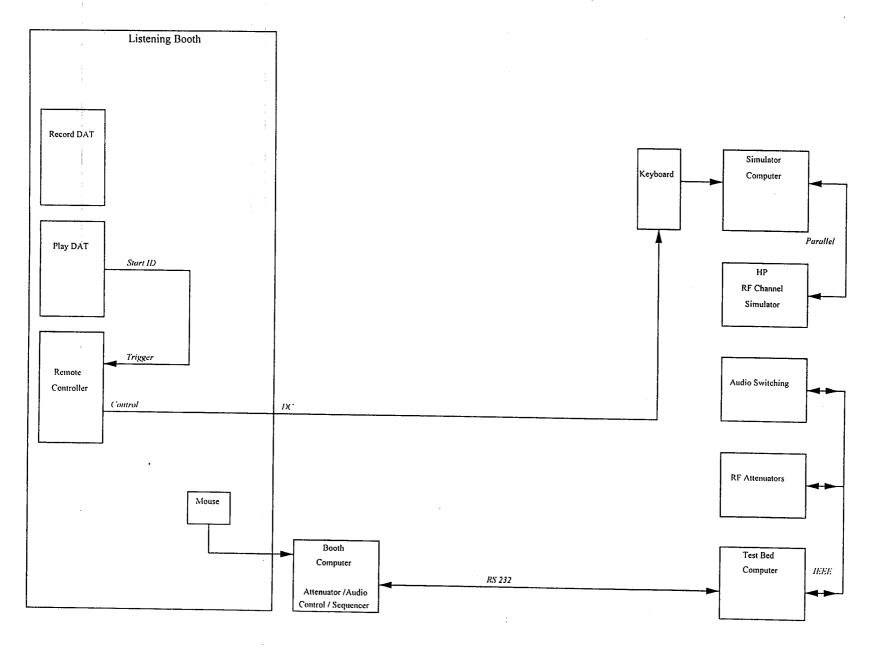
M Trans Block





File Name:BLOCKDIA.XLS

Rem. Cont. G & L



NRSC-R50

NRSC Document Improvement Proposal

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Email: standards@ce.org

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