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1996

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Annual Broadcast Engineering Conference

J BALLARD

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1996

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Annual Broadcast Engineering Conference

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FOREWORD

This year marks the 50th anniversary of the NAB Broadcast Engineering Conference. In this *Proceedings* you will find papers presented by many of the conference speakers. We hope you enjoy the publication and use it as a technical reference in your profession.

During the past 50 years, engineers and technicians have been instrumental in leading broadcast technology from monophonic to stereophonic and from black-and-white to color. Our industry is now in the process of evolving into the age of digital. The theme of this year's conference is *50 Years of Creating the Future*. As broadcast professionals, we are now being called upon to chart the future of broadcasting for the next 50 years. Our engineering talent is needed more than ever before to educate non-technical colleagues and to assist our management teams in the decision making processes that will define the future of our industry.

Exciting times lie ahead for broadcasters and we salute all who have devoted their time and energy to make this special conference a success. We especially thank the Society of Broadcast Engineers (SBE) for its efforts in co-producing the program. Again, this year the IEEE Broadcast Technology Society and our colleagues at the Society of Motion Picture and Television Engineers have developed exceptional tutorials for broadcast engineers facing the transition from analog to digital. Also, the European Broadcasting Union organized a session on planning for digital video broadcasting in Europe. Our thanks go out to these fine organizations as well.

As always, the NAB/SBE Conference Planning Committee welcomes your comments not only on this year's conference, but also for future conferences. We invite you to call or write anytime.

Jerry Whitaker
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Lynn Claudy, National Association of Broadcasters, Washington, DC

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William Hassinger
Former Assistant Bureau Chief for Engineering
FCC Mass Media Bureau
Washington, DC

*Paper not available at the time of publication.

DAB: U.S. AND WORLDWIDE PLANNING: PART I

Sunday, April 14, 1996
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Session Chairperson:
Milford Smith, Greater Media, Inc., East Brunswick, NJ

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Washington, DC

*Paper not available at the time of publication.

SUMMARY OF FM BAND IBOC LABORATORY TESTS RESULTS

Thomas B. Keller
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Consultant/EIA
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ABSTRACT

The Electronic Industries Association Subcommittee on Digital Audio Radio and the National Radio Systems Committee DAB Subcommittee have completed the laboratory tests for digital radio systems. Working Group B (Testing) of the EIA DAR Subcommittee was responsible for conducting the laboratory tests for the seven proposed DAR systems. Of these, four systems operate in the VHF 88 MHz to 108 MHz FM band, one in the medium wave band (AM), one in the satellite band, and one in a new terrestrial DAR band. Of the four systems intended to operate in the FM band, one is designed to operate on adjacent channels, and the remaining three are intended to share existing channels. The in-band/on-channel (IBOC) DAR system laboratory tests were conducted in collaboration with the National Radio Systems Committee.

This paper is intended to focus only on the tests of three FM band IBOC systems and only on those tests that effect the performance of the digital signal and the IBOC systems in-band compatibility. Due to presentation limitations, this paper will not focus on the system audio quality, multipath performance, or subcarrier performance. The complete laboratory test results for all seven systems is available from EIA (1).

INTRODUCTION

The DAR tests were conducted in two laboratories, the transmission laboratory at NASA Lewis Research Center, Cleveland, Ohio, and the expert subjective tests at the CRC Subjective Quality Assessment Laboratory, Ottawa, Ontario. The tests at NASA, Cleveland were in two phases, digital and in-band compatibility. The digital phase evaluated the signal quality and failure characteristics. Additionally, the digital test included multipath, co-channel, and adjacent channel impairments. The in-band compatibility phase of the transmission tests, conducted at the transmission test laboratory, included a series of tests to measure possible interference to the existing analog program service caused by the introduction of the in-band DAR signal. Comprehensive tests were also conducted to measure possible interference to the ancillary subcarrier service channels by the in-band DAR signal. For the in-band compatibility tests, the committee approved the selection of a group of receivers that is representative of the existing analog consumer receiver population.

The Cleveland laboratory digital transmission tests were conducted using subjective detection of the Threshold Of Audibility (TOA) and Point Of Failure (POF) by the laboratory specialist. The results of signal failure characterization transmission tests that were assessed by the Cleveland specialists were digitally recorded at the transmission laboratory and sent to the CRC for additional assessment by a larger group of expert listeners.

In-band compatibility objective tests were conducted at the transmission laboratory. Digital audio recordings were made at the output of the analog compatibility receivers for subjective evaluation by a group of industry experts.

WORKING GROUP B (TESTING)

The EIA DAR Subcommittee Working Group B on testing started meeting in the summer of 1992. The working group completed the laboratory test plan, established the transmission test laboratory, selected the subjective testing laboratory, characterized the transmission multipath for the FM band (88MHz to 108MHz), and completed the laboratory tests. Three FM band IBOC systems were presented by the system proponents for testing. Table 1 lists the IBOC proponents with the location of the digital signal.

Table 1. Systems	
Proponent	Description
AT&T/Amati Mode 1	Dual Side Band
AT&T/Amati Mode 2	Lower Side Band
USADR-FM 1	Dual Side Band
USADR-FM 2	Under FM

SYSTEM WAVEFORM

The IBOC systems differ in the location of the digital signal in the FM channel.

The AT&T/Amati Double Side Band (DSB) mode places the digital signals on both sides of the FM signal using the first 100 kHz of the first upper and lower adjacent channel, see Figure 1A. Each digital sideband is 73.5 kHz wide for a total digital bandwidth of 147 kHz. The total composite channel bandwidth is 400 kHz.

The USADR FM-1 system digital signal is also located in upper and lower first adjacent channel. Figure 1C shows that the digital signal extends 120 kHz into the first adjacent channel. The half power bandwidth of each digital signal is 100 kHz for a total digital bandwidth of 200 kHz. The FM-1 system total composite channel bandwidth is 440 kHz.

The proponents maintain that the digital sideband signals are within the guidelines established by FCC 73.317 (2) of the FCC code. This rule states that "Any emission appering on a frequency removed from the carrier by between 120 kHz and 240 kHz inclusive must be attenuated at least 25 dB below the level of the unmodulated carrier".

The AT&T/Amati system is capable of operating in three modes; Double SideBand (DSB) Figure 1A, Lower SideBand (LSB) Figure 1B, and Upper SideBand (USB). The LSB and USB modes are designed to be used to alleviate known adjacent channel interference. The receiver will automatically select the transmitted mode. The DSB and LSB modes were tested at the EIA DAR Laboratory.

The digital signal for the USADR FM-2 system is designed to be completely orthogonal to the host analog FM. A spectrum analyzer plot of the composite signal is shown in Figure 1D. The digital energy is transmitted under the analog and spreads into the adjacent channels at a decreasing level.

Because of different interpretations of FCC 73.317 (2) by the IBOC proponents, the subcommittees allowed the AT&T/Amati to increase the digital power (with respect to the analog FM) to a power level comparable to that used by the USADR FM-1 system. The AT&T/Amati power change was interpreted as a major change by the subcommittees which warranted other proponents to also make system changes. Besides the power change, AT&T/Amati made a single receiver modification. USADR made several changes to their FM-1 system. The proponents completed the system changes by spring of 1995, and the digital systems were retested. The performance data in this paper is from the two retested systems. All of the test data are contained in the EIA DAR Laboratory Report of August 11, 1995. (1)

A second transmitter is used for the IBOC digital signal. The configuration is shown in Figure 2. The digital signal is not an FM subcarrier, but is broadcast from a second transmitter and passively combined with the FM analog signal.

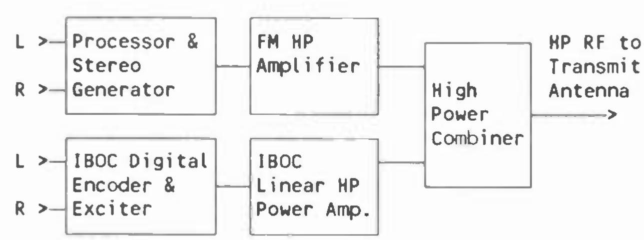

Figure 2. IBOC Transmitter Block Diagram

IBOC to IBOC INTERFERENCE TESTS

The IBOC -> IBOC digital tests were designed to determine the coverage for each IBOC system operating in that system's environment. The composite IBOC to IBOC (digital to digital) tests were conducted on co-channel, first adjacent channel, and the second adjacent channels. The Desired/Undesired (D/U) ratios for the composite IBOC signal can be used to calculate the digital signal coverage. Because the FM band D/U ratios have been set by FCC rules, the D/U ratios for the FM IBOC system have already been established.

To realistically simulate interference from a second digital station, each proponent was required to furnish a second digital transmitter or a system simulator. For the systems that supplied a simulator, the laboratory staff conducted certification tests. In cases where the RF waveform did not match the waveform of the main IBOC

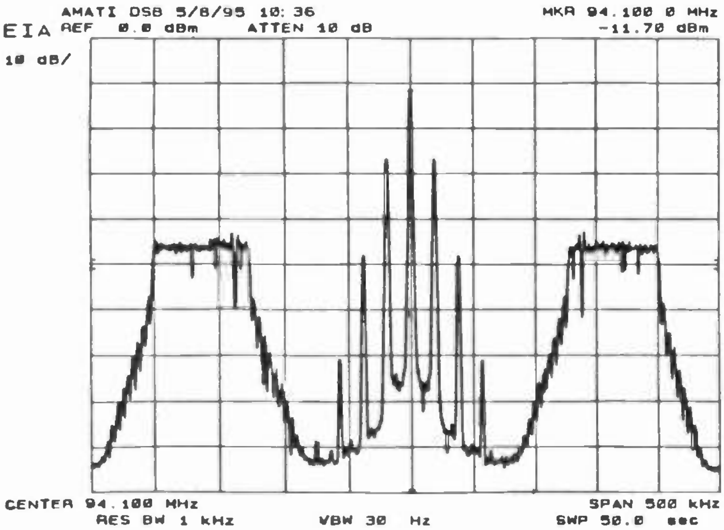

Figure 1A. AT&T/Amati DSB

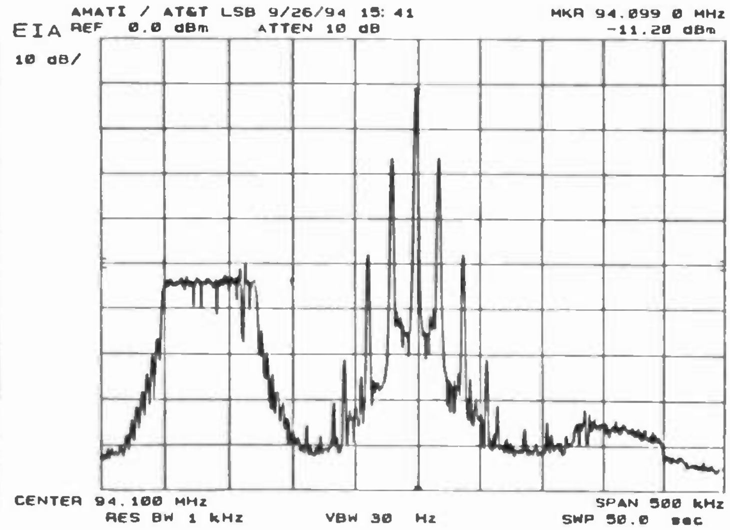

Figure 1B. AT&T/Amati LSB

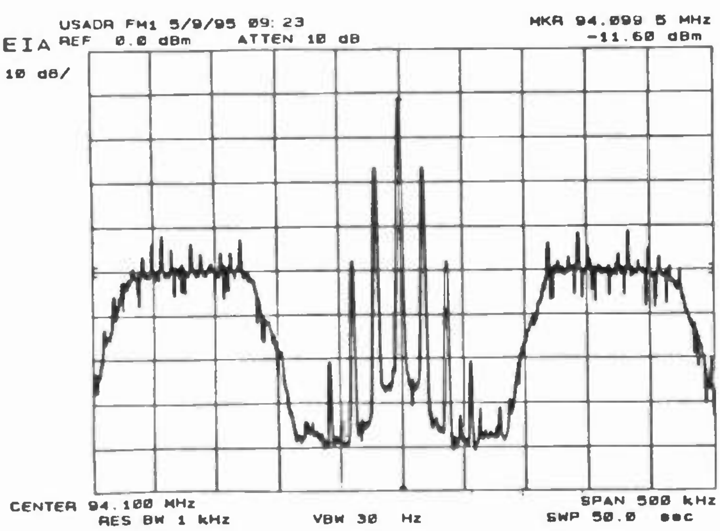

Figure 1C. USADR FM-1

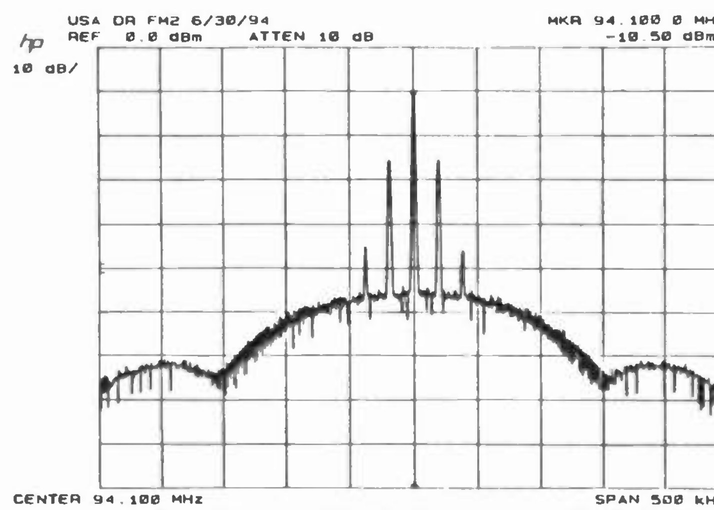

Figure 1D. USADR FM-2

transmitter, the proponents were required to correct the RF waveform.

For the subjective digital transmission tests, the undesired composite IBOC signal was added to the desired composite IBOC signal and increased in amplitude until the TOA and POF was heard listening to digital audio by the laboratory specialists. The D/U was recorded at this point (Figure 3).

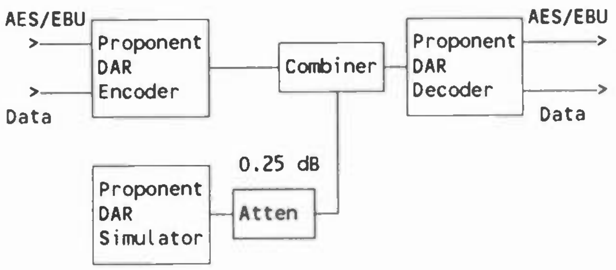

Figure 3. IBOC to IBOC Laboratory Block Diagram

The TOA is the point where the interference is perceptible, but not annoying. After the TOA was found, the interference was increased to the Point of Failure (POF) and again the D/U recorded. The POF is the undesired signal level where the digital audio completely fails, or the audio is very annoying. A signal attenuator with 0.25 dB steps was used to find the TOA and POF. For this series of tests, glockenspiel was used for the critical audio program material. The co-channel TOA and POF found at the transmission laboratory were re-confirmed by the experts at the subjective assessment laboratory at the CRC, Ottawa. The NASA Laboratory also digitally tape recorded several intermediate points between TOA and POF for further system failure characterization tests conducted by the expert listeners at CRC, Ottawa.

Tables 2, 3, and 4 show the performance of each system for the three interference modes. Each system's D/U for POF is shown and a comparison is made between the laboratory D/U and the D/U found in FCC 73.215. Simulated multipath tests were also conducted that showed an increase in interference with multipath.

Co-channel digital performance at the POF exceeds the FCC 20 dB D/U requirement at the protected contour by 11 db for the Amati/AT&T DSB system and by 14 dB for the USADR FM-1 system. The AT&T/Amati LSB system was 3 dB better, and the FM-2 system was 44.3 dB poorer than the FCC 20 dB D/U (Table 2).

First Adjacent digital performance for all systems did not meet the FCC 6 dB D/U criteria. Both the AT&T/Amati and the FM-1 systems were below the FCC D/U by at least 14 dB at POF. Figure 1 shows that the principal interferer for the sideband IBOC systems is the adjacent channel FM signal. With the side band IBOC systems, the digital signal is 15 dB lower than the interfering FM signal (Figures 1A & 1C). The DAR laboratory tests have shown that the co-channel D/U ratios of 10.5 dB at POF can be expected for non-IBOC and IBOC systems. If we add the 10.5 dB D/U for the co-channel and the 15 dB D/U for the IBOC analog to digital power ratio, we have a predicted 25.5 dB D/U at POF for IBOC -> IBOC first adjacent interference. The D/U performance of AT&T/Amati system of 24 dB exceeds this predicted D/U by 1.5 dB. This difference may be explained by the fact that the interferer is the analog signal of the composite undesired IBOC effecting only one half of the desired digital signal (Table 3).

Second Adjacent D/U ratios are important for the sideband IBOC systems because the digital signal is located in the adjacent channel. With the present FCC rules, the spacing of second adjacent class C1 to C1 stations is 52 miles, and class B to B is 46 miles. FCC 73.215, Contour protection for short-spaced assignments, protects all classes of stations within the protected contour with a -40 dB D/U (2). This means that the undesired station can be 40 dB stronger than the desired. The AT&T/Amati DSB system failed (POF) 19 dB below the FCC 40 dB protection criteria, and the USADR/FM-1 failed (POF) 37 dB below the FCC criteria (Table 4).

Figure 4 is a spectrum analyzer plot of the second lower adjacent channel, taken at the desired signal's TOA for the AT&T/Amati DSB system. Figure 5 illustrates the USADR FM-1 performance for the same test. The overlap of the digital signals may explain the second adjacent performance difference.

FM to IBOC INTERFERENCE TESTS

Interference from the analog to the composite IBOC digital tests were conducted for co-channel, lower first adjacent, upper first adjacent, lower second adjacent, upper second adjacent, and simultaneous lower and upper second adjacent.

The IBOC systems that use the adjacent channel for the transmission of the IBOC digital audio had negative D/U ratios for the co-channel tests. This extraordinary co-channel result can be explained by the fact that the

Table 2.	IBOC Co-Channel D -> D Laboratory Test Results Results are for TOA and POF			
	Amati/AT&T DSB D/U	Amati/AT&T LSB D/U	USADR FM-1 D/U	USADR FM-2 D/U
TOA	10.5 dB	17 dB	10.8 dB	44.3 dB
POF	9.0 dB	15.5 dB	7.3 dB	40.8 dB
POF Compared to FCC 73.215 or 20 dB D/U	11.0 dB Less Sensitive to Interference	4.5 dB Less Sensitive to Interference	13.7 dB Less Sensitive to Interference	20.8 dB More Sensitive to Interference

FCC 73.215 Contour protection for short-spaced assignments.

Table 3.	IBOC First Adjacent D -> D Laboratory Test Results			
	Amati/AT&T DSB D/U	Amati/AT&T LSB D/U	USADR FM-1 D/U	USADR FM-2 D/U
TOA	24.1 dB	42.7 dB	27.0 dB	30.2 dB
POF	20.3 dB	38.7 dB	22.8 dB	29.0 dB
POF Compared with FCC 73.215 or 6 dB D/U	14.3 dB More Sensitive to Interference	32.7 dB More Sensitive to Interference	16.8 dB More Sensitive to Interference	23.0 dB More Sensitive to Interference

The tests were done on the upper and lower first adjacent channels and the results averaged.
FCC 73.215 Contour protection for short-spaced assignments.

Table 4	IBOC Second Adjacent D -> D Laboratory Test Results			
	Amati/AT&T DSB D/U	Amati/AT&T LSB D/U	USADR FM-1 D/U	USADR FM-2 D/U
TOA	-17.4 dB	-16.8 dB Lower 2.2 dB Upper	4.5 dB	30.6 dB
POF	-21.2 dB	-19.5 dB Lower -2.3 dB Upper	-3.1 dB	No Test
POF Compared with FCC 73.215 or -40 dB D/U	18.8 dB More Sensitive to Interference	14.8 dB Lower 37.7 dB Upper More Sensitive to Interference	36.9 dB More Sensitive to Interference	70.6 dB More Sensitive to Interference (this is TOA)

Unless otherwise indicated the tests were done on the upper and lower second adjacent channels and the results averaged.
FCC 73.215 Contour protection for short-spaced assignments.

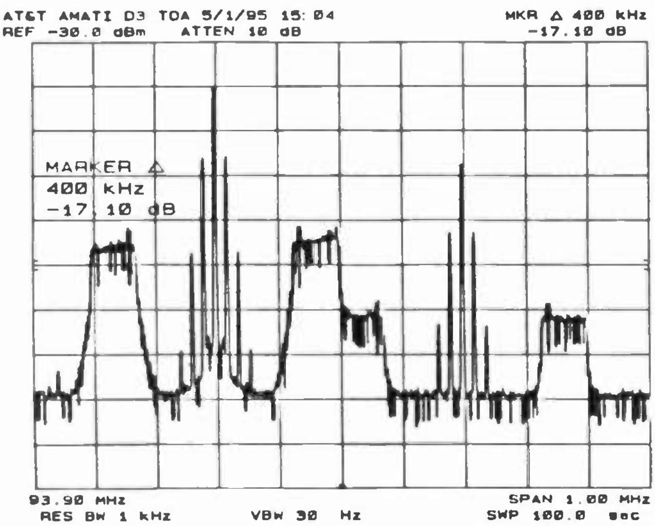

Figure 4. AT&T/Amati DSB Lower 2nd Adjacent

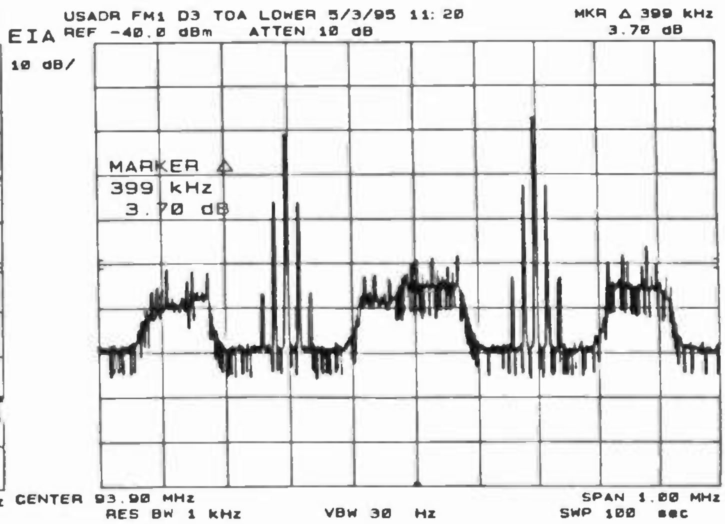

Figure 5. USADR FM-1 Lower 2nd Adjacent

undesired FM station only interferes with the host FM (Figures 1A & 1C).

The FM -> IBOC first adjacent channel tests show slightly less interference from the FM signal than the considerable interference found in the IBOC -> IBOC tests. Eliminating the digital signal from the host FM had a slight effect on the interference experienced in the IBOC -> IBOC tests.

The second adjacent test results show very little interference from the analog to the digital for those systems that used the adjacent channel for digital. The AT&T Amati DSB system came within 2 dB of meeting the FCC -40 dB D/U at POF.

FM -> FM REFERENCE TESTS

To establish a reference for the inband compatibility tests, it was necessary to conduct a series of FM -> FM D/U tests with a representative group of contemporary consumer FM stereo radios. Five FM radios were selected that represent a cross section of receivers in use in the United States. The selection was divided into four categories: auto, portable, high end home Hi-Fi, and competitive Hi-Fi. The two automobile radios were selected because of their large population and their wide difference in the stereo blend. These auto radios also showed high adjacent channel rejection. The portable and personal portable use similar circuitry and have less adjacent channel rejection. The high end home Hi-Fi radios had good 2nd adjacent channel rejection but

Table 5.		REFERENCE FM -> FM RECEIVER TESTS			
Receiver	Type	Co-Channel D/U in dB	1 st Adjacent D/U in dB	2 nd Adjacent D/U in dB	114 kHz Test S/N dB
1. Delco	Auto	36.2	4.7	-24.2	No Change
2. Denon	Hi-Fi	43.4	18.0	-28.9	34.0
3. Panasonic	Portable	40.9	27.3	-10.1	33.6
4. Pioneer	Hi-Fi	44.2	26.6	-15.0	33.1
5. Ford	Auto	35.2	-6.1	-45.3	No Change

For the D/U measurements, the interfering FM signal level was set for a 45 dB audio S/N. This measurement was made using quasi-peak detection, a 15 kHz LP filter, and the CCIR filter.

The upper and lower adjacent D/Us were averaged.

exhibited the similar adjacent channel rejection characteristics found in the portable and home radios.

Table 5 shows the result of the FM -> FM D/U tests that were conducted with the five radios in the DAR laboratory. For the D/U measurements, the undesired signal RF level was set for a 45 dB audio signal to noise ratio. The audio noise measurement was made using quasi-peak detection, a 15 kHz low pass filter, and the CCIR filter. The desired signal level was -62 dBm. Antenna matching networks were used when needed. The portable and home receivers were tested in a shielded box that eliminated interference from other electronic devices in the laboratory. The two auto radios did not need additional shielding.

IBOC TO HOST FM COMPATIBILITY TESTS

The objective of this test was to measure possible interference from the IBOC digital transmitter to a cross section of consumer analog receivers that are tuned to the host FM station. The tests were conducted at strong -47 dBm and weak -77 dBm signal levels. This is the most straightforward test of the entire DAR compatibility series. For reference the test receiver signal to noise was measured with the laboratory THE-1 transmitter. The IBOC digital signal was turned on, and the audio RMS S/N measured. Changes in S/N were then noted (Table 6).

There are many ways of decoding the FM stereo signal. In practice the PLL stereo decoder has become the norm. Because the PLL stereo decoder uses square wave switching, the circuit is able to demodulate baseband signals which are the odd harmonics of 38 kHz, 114 kHz and 190 kHz (3). Without 114 kHz LP filters or special circuitry, the PLL decoders will detect the IBOC digital signal as noise. To further understand this phenomena, a special receiver test was conducted at the DAR laboratory to find out which receivers were sensitive to the 114 kHz signal without using a DAR signal. A CW subcarrier was added to the FM signal at 113 kHz with 10% injection, and the receiver audio output noise measured. Auto receivers #1 and #5 showed little change in S/N with either the 114 kHz tests or the IBOC signals. Receivers #2, #3, and #4 exhibited a large increase in noise with the 113 kHz subcarrier. This noise was the beat tone of 1 kHz between the test signal of 113 kHz and 114 kHz. Table #6 shows the results of the IBOC to host FM and 113 kHz subcarrier test for the five laboratory radios. The test showed that the radios that have a significant increase in noise with the IBOC signal also had an increase in noise with the 113 kHz subcarrier test. The sensitive radios had noise increases of 18 dB to 26 dB with the IBOC signals. The two auto radios that did not have a noise increase with IBOC did not have an audio noise increase with the 113 kHz subcarrier test.

Extended CW subcarrier tests conducted using a 189 kHz subcarrier revealed that the radios were also sensitive to 190 kHz. Injecting a signal at 152 kHz midway between the 114 kHz and 190 kHz did not change the radios output noise level. It appears that the band of frequencies around 152 kHz does not effect the noise performance of the PLL stereo receiver.

Subjective tests were also conducted with industry expert listeners. The S/N degradation results shown in Table 6 were consistent with the observed degradation. The audio output of each of the five FM radios was recorded on digital audio tape, and these tapes were transferred to eight CDs for subjective assessment. The subjective tests rated the digital to FM host interference, as well as the digital -> FM co-channel, first adjacent, and second

Table 6. IBOC DAR -> Host FM
RMS Noise
Signal Level -47 dBm

Receiver	Type Radio	S/N FM Only Reference	S/N 114 kHz Test	S/N AT&T/Amati DSB	S/N USADR FM-1
1. Delco 161924463	Auto	60.0 dB	No Change	60.7 dB	60.3 dB
2. Denon TU-280RD	Hi-Fi High end	68.0 dB	34.0 dB	50.0 dB	44.9 dB
3. Panasonic RX-PS430	Stereo Portable	67.5 dB	33.6 dB	44.2 dB	42.0 dB
4. Pioneer SX-201	HI-Fi	66.0 dB	33.1 dB	40.0 dB	39.2 dB
5. Ford F4XF-19B132-CB	Auto	65.0 dB	No Change	64.0 dB	62.7 dB

adjacent channel tests. These recordings were transferred to eight CDs and sent to the 11 expert listeners. For the IBOC-to-host FM test, the experts compared the FM signal audio to the IBOC FM signal audio. The test segments were assessed by rating changes between the reference and the test segment. The subjective effects of the interference were rated using the seven point CCIR numerical rating scale.

3	Much Better
2	Better
1	Slightly Better
0	Same
-1	Slightly Worse
-2	Worse
-3	Much Worse

The results of these tests showed that the experts rated the 40 to 50 dB S/N ratios worse to much worse than the reference. The S/N degradation shown in Table 6 were consistent with the expert observed degradation.

IBOC TO FM INTERFERENCE TESTS

A comprehensive set of compatibility tests were conducted for co-channel, first adjacent channel, and second adjacent channel IBOC -> FM interference. All five of the selected consumer radios were used for this test series. The first step for the objective tests was to establish an FM -> FM reference by adjusting the undesired FM RF signal level for an audio S/N of 45 dB at the test radio output. The undesired FM signal was then replaced with the composite IBOC signal and the undesired level adjusted for a 45 dB audio S/N. The test results in Table 7 compare the D/U ratios for the reference FM -> FM tests (existing service) to the D/U ratios for the IBOC -> FM tests. A positive increase in D/U indicates an interference increase. Expert Observation and Commentary (EO&C) tests were also conducted by the laboratory specialists, and subjective listening tests were conducted by the 11 industry experts. The results of these tests are reported in either the EIA DAR Laboratory Test Report or the DAR Subcommittees minutes.

The digital to analog co-channel test results show little difference in interference between the FM -> FM tests and the digital -> FM tests.

The first adjacent tests were conducted at a reference 35 dB and 45 dB audio signal to noise ratios. Because the

Table 7. IBOC -> ANALOG INTERFERENCE

Receivers		Delco D/U in dB	Denon D/U in dB	Panasonic D/U in dB	Pioneer D/U in dB	Ford D/U in dB
Co-Channel Audio S/N 45 dB	Reference	36	43	41	44	35
	AT&T Amati	37	43	41	44	35
	USADR FM-1	35	43	41	44	35
First Adjacent Audio S/N 45 dB	Reference	5	18	27	27	-6
	AT&T Amati	20	28	30	31	19
	USADR FM-1	18	26	29	30	17
First Adjacent Audio S/N 35 dB	Reference	4	7	15	15	-17
	AT&T Amati	8	17	18	20	8
	USADR FM-1	7	15	17	18	6
Second Adjacent Audio S/N 45 dB	Reference	-24	-29	-10	-15	-45
	AT&T Amati	-24	-19	-5	-3	-30
	USADR FM-1	-24	-10	2	4	-27

The undesired signal level was set for either 35 dB or 45 dB audio S/N ratio. The audio noise measurements were made using quasi-peak detection, a 15 kHz filter, and the CCIR filter.
The first and second D/U ratios are the averaged upper and lower D/U measurements.

Panasonic and Pioneer radios are not as selective as the auto radios, the first adjacent FM interference masked additional interference from the IBOC signal. During the test that used a 35 dB audio S/N, the Denon and Ford radios displayed an increase in interference. The Ford auto radio is very selective and is able to detect FM or digital signals transmitted in the first adjacent channel well beyond the FCC protected contour. For the 35 dB S/N tests only, the Denon and Ford showed an increase in digital interference.

For the second adjacent channel tests, four of the radios had an increase in interference. With the close spacing of the second adjacent channels, this interference can be significant.

CONCLUSIONS

Digital to Digital

The IBOC systems that use the first adjacent channel for the transmission of the digital signal have a fundamental problem with the interference from the undesired first adjacent FM channel that can result in a significant reduction of digital coverage as compared to the host FM. The second adjacent interference is critical but can be improved by system design. With the exception of the system that transmits the digital signal under the analog, the co-channel performance exceeded the FCC prescribed D/U ratios (less interference).

FM to Digital

Again, the systems that transmit the digital signal in the first adjacent channel have a significant problem with interference from an undesired FM signal in first adjacent channel. These systems experienced little interference from FM stations operating on co-channel or second adjacent channels.

Digital to FM

An increase in interference to other FM stations operating on the first or second adjacent channels was found. This increase is receiver dependent.

IBOC to FM Host

This interference is most pronounced at moderate to strong RF signal levels. The noise is detected in the PLL stereo decoders and can be eliminated with the use of special circuitry. A large population of stereo receivers are subject to this noise increase.

System Design

Both systems were updated by the proponents in the spring of 1995 prior to start of the second round of digital performance tests.

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2. Code of Federal Regulations, Part 73 Radio Broadcast Services

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IBOC DAB: ITS POTENTIAL FOR BROADCASTERS

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ABSTRACT

As the NRSC/EIA process for evaluating digital audio broadcasting (DAB) systems proceeds, premature conclusions regarding the fate of in-band/on channel (IBOC) techniques abound. Often overlooked in these pronouncements are a number of factors including the relative newness of IBOC technology compared to other systems, in particular the Eureka-147 new-band system, the tradeoffs inherent in an existing-band system and in co-locating with an analog AM or FM signal, and the challenges, both political and financial, of obtaining new spectrum for DAB services.

The purpose of this paper is to present these issues from the perspective of existing radio broadcasters, who support IBOC and are committed to transitioning to a digital method of signal delivery that is clearly superior to existing AM and FM analog methods and at the same time is not detrimental to the industry as a whole.

INTRODUCTION

Radio broadcasting in the United States is a big business. Over 91,000 persons were employed by commercial radio stations in 1994[1]; in 1995 annual advertising revenues are projected to be at the US $12 billion level[2]. Mergers and acquisitions are at an all-time high, due in part to the changes in the regulatory climate. It's a competitive business, and a station's success (or failure) will depend upon not only what it broadcasts, but technical factors, too, in particular its coverage area as well as the coverage area of its competitors.

The introduction of digital audio broadcasting in the US brings with it the promise of quality enhancements and new service opportunities. Broadcasters recognize that the age of digital technology has arrived, and that the radio business needs to "go digital" with all due haste, lest it be set aside by its listeners for any number of competing digital services now coming into their own. It is a new technology, though, and careful consideration must be given as to how this new technology will affect the business aspects of radio.

Currently there are three different techniques being actively considered for terrestrial DAB in the US, which can be characterized by, among other things, their spectral relationship to existing terrestrial AM and FM analog services. They are -- new-band DAB, which would use new spectrum, different from what is currently allocated for terrestrial radio service; in-band/adjacent channel (IBAC) which utilizes digital signals that can co-exist with analog FM signals in the existing analog FM band; and in-band/on channel (IBOC), where a digital carrier is broadcast along with a "host" analog AM or FM signal.

Of these three, IBOC digital radio gives broadcasters their best hope to bring digital quality to the public without eviscerating the current structure of US broadcasting. It brings with it solutions to a number of "thorny" issues which accompany a transition from analog to digital, including :

- Spectrum issues - available spectrum in the US is at a premium; it is not clear where the spectrum for a new-band system would come from. And even if it were identified, if the process currently being undertaken for advanced television (ATV) is any indication, there would almost certainly be talk of an auction for its use. With IBOC, these issues never arise;

- Investment issues - IBOC preserves the existing competitive landscape, allowing radio owners to protect their investment in radio facilities and listeners. In addition, the costs associated with implementing IBOC systems will be significantly less than those incurred for an entirely new transmission facility, as would be required in implementing a new-band system;

- Broadcaster independence - IBOC transmission facilities are independent of one another unlike the multiplexed facilities which are used in the Eureka-147 new-band system.

These advantages are not without penalty, the most serious being that an IBOC digital receiver must contend with the interference caused by its analog "host" existing on the same frequency, unlike a new-band system which has no such in-band interference. And conversely, existing analog signals, both "host" and adjacent, have to contend with potential interference from the IBOC signal. The pros and cons of these differing approaches have been presented elsewhere, with greater detail and eloquence than what is given here; the point to be made is that on balance, broadcasters have chosen IBOC. They are pursuing its implementation through their sponsorship of the National Radio Systems Committee (NRSC) DAB Subcommittee.

Five candidate IBOC systems have undergone laboratory testing as part of the DAR laboratory tests[3], conducted by the Electronic Industries Association (EIA) and the NRSC (which is co-sponsored by EIA and NAB). The results for the IBOC systems were mixed, revealing weaknesses and areas for improvement but also demonstrating under what circumstances IBOC performance is good. More importantly, these results make it possible for broadcasters to understand in more precise terms the tradeoffs and compromises which need to be considered in the implementation of IBOC digital radio. To that end, the NRSC's DAB Subcommittee Report Writing Working Group is in the process of preparing a comprehensive evaluation of the laboratory (and when available, field test) data with the goal of determining which IBOC system is suitable for consideration as a digital broadcasting standard in the US.

Additional data on IBOC will be collected in field trials to be conducted in the San Francisco Bay area during the first half of 1996. This field test data is vital to a comprehensive consideration of what IBOC has to offer, especially since it is anticipated that the IBOC proponents will be submitting improved equipment for field testing compared to that which was evaluated in the lab. This last point highlights an important fact which is often overlooked by the IBOC detractors, which is that the IBOC technology is not yet fully mature, and in fact is at a much earlier stage of development than the new-band systems against which it is compared, especially the Eureka-147 system which was being demonstrated as early as 1988.

In the sections that follow, a brief overview of how broadcasters came to support the IBOC strategy will be given, followed by a discussion of some of the EIA/NRSC DAR laboratory test results for IBOC systems that were released in August of 1995. Then, the scenario under which broadcasters see the IBOC development process proceeding will be given, followed by some concluding remarks about broadcasters and their commitment to IBOC.

A PROCESS OF ELIMINATION

Other than IBOC, there are currently two primary contenders for terrestrial digital radio systems - new band systems, such as Eureka-147, and in-band/adjacent channel systems (IBAC, sometimes referred to as in-band/reserved channel or IBRC), of which one example is currently undergoing evaluation in the previously mentioned EIA/NRSC test effort. Neither of these approaches offers broadcasters the potential for as smooth a transition from analog to digital as does IBOC. In each case, spectrum issues are a primary reason that broadcasters do not endorse these approaches.

In 1991 NAB studied the possibility of implementing a Eureka-147 system in the US and concluded that anywhere from 57 MHz to 130 MHz of new spectrum would be required to accommodate all existing radio stations (only 11,000 at that time)[4], that is, to afford each existing station with a corresponding frequency assignment under a Eureka-based scheme. Even if this amount of spectrum were available, which it is not, the possibility of being able to re-assign each station to a new digital channel in an equitable manner is remote.

Furthermore, the cost of implementing a new-band system would greatly exceed the cost of an in-band system. A new-band system would require new transmitters and antennas to be purchased. New frequency allocation studies would have to be performed throughout the entire country. And, in many cases, new towers would have to be erected. Implementation of an IBOC system, on the other hand, would be much less expensive. It would not require the purchase of new towers or antennas, and would require only a minimum amount of new transmitter equipment. When one contrasts the costs of a new-band system with those of an IBOC system, it becomes evident that an IBOC system is much more cost-effective.

With the IBAC approach, the situation is somewhat different. In theory, an IBAC DAB system for the FM band is an appealing one because under the implementation plan for IBAC, each existing FM broadcaster would be provided with a new digital audio signal, separate from its analog signal. This would make it much easier (compared to the IBOC case) for digital broadcast signals to coexist with analog signals in the same band without causing interference to existing analog receivers. In addition, the IBAC DAB system would not (in theory) require the FCC to allocate any new radio broadcasting spectrum, unlike new-band approaches, since all existing FM analog stations would be provided with a digital channel in the existing FM broadcast band.

Despite these theoretical advantages, in reality two major problems exist. The first is that accommodation of AM broadcasters is not addressed in an IBAC scheme. Any DAB system that benefits only FM broadcasters without providing any opportunities for AM stations is unacceptable to the broadcast industry and counter to the public interest.

The second problem is that there is simply not enough spectrum available in the existing FM broadcast band to provide each existing station with a new digital signal. In an IBAC scheme, a new 200 kHz-wide channel (in the 88-108 MHz band) would be assigned to every existing analog station. This would double the number of channel allocations that exist today. As anyone who has made an attempt to find a vacant channel in a major population center knows, there are not nearly enough unused frequencies available in the United States to accommodate an IBAC system. In fact, in many major metropolitan areas, there are *no* frequencies available that could be used to implement this system.

ENCOURAGING LABORATORY RESULTS

A tremendous amount of data has been collected on IBOC system performance by the EIA/NRSC DAR testing laboratory[5]. This data, while illuminating, does not represent the last word on IBOC performance as some commentaries would have one believe. IBOC technology is still under development, and is not as mature as new-band technology such as Eureka-147. Both IBOC proponents have indicated that significant system improvements have been made since the equipment submitted for testing was built, over 2 years ago, and are expected to submit improved systems for testing in the San Francisco field trials.

Furthermore, as already mentioned, field test results are expected this year and will provide a valuable supplement to the lab data. Field demonstrations of IBOC systems (conducted by the proponents) to-date have been excellent, and the San Francisco tests will be very useful in quantifying the service characteristics of IBOC DAB in real-world terms.

Two important positive results from the laboratory test data are apparent - firstly, the audio quality of unimpaired IBOC signals is demonstrably better than the existing AM and FM signals which they are designed to replace, and second, the IBOC DAB systems appeared compatible to a large degree with existing analog signals when received by certain types of analog receivers.

Audio quality

The sound quality of the various IBOC DAB systems was evaluated by a panel of expert listeners at the Communications Research Centre in Ottawa, Ontario. These listeners assessed the audio quality of the various IBOC systems both in the presence and absence of transmission impairments. Each of the systems was compared, under each test condition, with a known (CD-quality) reference and graded on a five point scale. All of the systems were tested with the same audio segments, chosen by an expert panel such that each of the IBOC systems would find it difficult to deal with at least two of the segments. The five grades used for grading the systems are as follows (they refer to any imperfections noted in the IBOC signals compared to the "perfect" reference signal):

Imperceptible
Perceptible, but not annoying
Slightly annoying
Annoying
Very annoying

The performance of the USA Digital Radio FM-1 system was judged to be statistically the same as that of the highest data rate version (224 kb/s) of the Eureka 147 system, and these two systems were the best performers of all that were tested. In addition, the AT&T/Amati Dual Sideband system, operating at 160 kb/s, performed better than a reduced data rate version of the Eureka 147 system which was operating at 192 kb/s. Overall, the FM-band IBOC systems performed very well in the audio quality tests. All four of them were judged to have imperfections, compared to a "perfect" CD-quality reference, that were "perceptible, but not annoying."

The audio quality of the USA Digital Radio AM-band IBOC system was judged overall by the expert listeners to be "annoying." That is, the imperfections that existed in the audio output of the AM-band system were considered annoying compared to the "perfect" CD-quality reference. It is important to note that in a comparison with an analog AM audio signal, the AM IBOC signal is clearly superior -- it is only with respect to the original CD reference that it is described as "annoying".

One reason for the AM-band system's low audio quality rating is that the AM-band system has the lowest data rate of all systems under test. The data rates for the various IBOC systems tested in the lab are given in Table 1.

Table 1. IBOC System Audio Codec Data Rates

SYSTEM	BIT RATE (KB/S)
Compact disc (source)	*1,411*
USA Digital Radio FM-1	256
USA Digital Radio FM-2	256
AT&T/Amati FM Dual Sideband	160
AT&T/Amati FM Lower Sideband	128
USA Digital Radio AM	96

In order to fit all of the audio information into a data stream of only 96 kb/s (kilobits per second), the AM-band IBOC system has to discard a considerably larger amount of the original audio information than do the FM-band IBOC systems. Throwing away this larger amount of data causes the AM-band IBOC receiver to develop a less accurate recreation of the original signal than its FM-band counterparts. Unfortunately, not a lot can be done to improve the AM-band system's data rate because the bandwidth available for transmitting an AM-band IBOC DAB system is *much* less than that available for transmitting an FM-band IBOC signal. The analog AM radio channel has only about one tenth the bandwidth of the analog FM channel and, in general, wider bandwidths mean that higher data rates are possible (and vice versa).

Compatibility of IBOC DAB signals with existing analog signals

Compatibility with existing analog services is the biggest challenge for IBOC DAB, for several reasons. Primary among these is the design of existing analog receivers, which will be in use (and in large numbers) at the onset of IBOC transmissions.

Analog receivers are designed to filter out (i.e. suppress) all incoming radio signals except for the one to which the receiver is tuned. The filtering circuitry used to accomplish this function varies widely from one model of receiver to another. This is because, in general, the better a receiver's filtering circuitry the more expensive it is to make, motivating receiver manufacturers to use the minimum amount of filtering possible for satisfactory performance of any given radio.

In general, the average car radio has more effective filtering circuitry than the average home or portable receiver. Car radios require better filtering circuitry than other radios because they are constantly moving, and the radio operator has no control over the orientation of the antenna. If a car radio is to continue receiving a particular station as it moves throughout a geographical area, then it must be able to handle situations when the desired station's signal is becoming weaker as an adjacent channel station's signal becomes stronger. In order to do this, the car radio must have sufficient filtering circuitry to enable it to suppress the adjacent channel station (whose strength is increasing) while continuing to receive the desired station (whose strength is decreasing).

On the other hand, a home/portable receiver is typically used in a non-mobile environment so it generally does not have to deal with situations where a desired signal is becoming weaker while an adjacent channel undesired signal is becoming stronger. Instead, these receivers remain in relatively fixed locations, where the ratio between the strength of a desired signal and that of any undesired signals (on co- or adjacent-channels) remains more or less constant. In situations where reception of a particular station is difficult, a home/portable receiver's antenna can be oriented to provide maximum reception capability. (Obviously, this is not an option with a car radio because the orientation of a car radio's antenna depends on the orientation of the car itself.)

Not surprisingly, the analog automotive receivers that were tested in the DAB laboratory demonstrated a greater degree of immunity to the digital IBOC signal than did the non-automotive receivers, presumably due to the more effective filtering circuits just discussed. In fact, in many situations, the performance of an automotive receiver *with* the digital IBOC signal present was superior to the performance of a non-automotive receiver *without* the IBOC signal present. This was true for all compatibility cases considered (to host, co-, 1st-, and 2nd-adjacent channels) and is another reason that the IBOC DAB laboratory test results are encouraging because it indicates that receiver manufacturers would likely be able to design marketable receivers capable of receiving analog radio signals in the presence of IBOC DAB signals.

To illustrate the compatibility of analog and IBOC signals when an automotive receiver is used, laboratory data on the interference experienced by an analog FM station, with an IBOC signal present on its 1st adjacent channel, has been applied to some real-world situations. The particular lab data of interest established the desired-to-undesired ratio (D/U) at which the signal-to-noise ratio (S/N) in the automotive receiver's audio output was equal to 35 dB (which is considered poor performance) as a result of an IBOC signal being present in the first adjacent channel to the analog signal.

Utilizing the resources of Dataworld® (an engineering firm specializing in statistical information relating to the broadcast industry) it was possible to represent the D/U data just described on a signal strength contour map for an existing FM station, and project the impact that the adjacent IBOC signal would have on the FM station's coverage area. Two pairs of short-spaced, 1st adjacent FM stations were selected and their 1, 0.5, and 0.1 mV/m signal-strength contours were plotted (these plots are presented in Figures 1 and 2). The characteristics of these stations are given in Tables 2 and 3. In each case, the contours for the interfering station are based on FCC (50,10) data while those for the desired station are based on FCC (50,50) data (in accordance with FCC procedures for analyzing interference).

Lines corresponding to the D/U ratios measured in the lab, for a S/N ratio of 35 dB in the audio output of the automotive receiver (due to an IBOC interferer), were added to these plots, and shading was placed in the region between the 35 dB S/N D/U line and the protected contour of the desired signal - in Figure 1, this is the 1.0 mV/m contour (since WMGQ is a class A station), while for Figure 2 this is the 0.5 mV/m contour (WMXV is class B). This shaded region, then, represents that portion of the primary listening area of the analog station where the interference from the IBOC carrier is predicted to cause a significant reduction in the analog signal's S/N ratio.

These results are not surprising due to the fact that the IBOC signal energy is co-located with the 1st adjacent channel. Compared to the overall listening area, the shaded regions shown in these figures are small, but in the fiercely competitive radio industry it can be expected that no station will want to lose even one listener to this type of interference. A possible solution

Table 2. Station Characteristics[6] for Figure 1

PARAMETER	WOGL-FM	WMGQ-FM
Frequency	98.1 MHz	98.3 MHz
Location	Philadelphia, PA	New Brunswick, NJ
Class	B	A
ERP	12.5 kW	1.2 kW
HAAT	1000 ft.	525 ft.
Role in simulation	IBOC interferer	Analog desired

Table 3. Station Characteristics[7] for Figure 2

PARAMETER	WDAS-FM	WMXV-FM
Frequency	105.3 MHz	105.1 MHz
Location	Philadelphia, PA	New York, NY
Class	B	B
ERP	16.5 kW	6.0 kW
HAAT	870 ft.	1220 ft.
Role in simulation	IBOC interferer	Analog desired

to this problem would be to simply reduce the digital carrier signal strength so as to reduce the shaded area of interference while presumably still supporting a viable coverage area for the digital signal. Then over time, as the IBOC signal gains wider acceptance and the number of analog signal listeners diminishes, the digital carrier power can then be gradually increased.

THE NEXT STEPS

Given that broadcasters are committed to the IBOC approach, and with the current state of the technology in mind, what steps does the broadcasting industry need to take so that IBOC can become a reality ?

The most important step to be taken involves vigorous support of the IBOC technology development by the broadcast community, in all aspects. As demonstrated by the tests already done, this technology has potential but it is not yet fully developed, and unfortunately, it is at a disadvantage when being compared against more mature systems. Given the level of support for these other systems, in particular for Eureka-147, broadcasters need to fully embrace the IBOC concept and provide the resources needed to make it happen.

A further important step involves the identification of a long-term transition strategy, which defines exactly how broadcasters go from the all-analog signal environment which currently exists, to an introductory period where IBOC transmissions are initiated, and ultimately progressing to the point where IBOC is fully integrated into the radio broadcasting environment. Even now, with only the laboratory data to rely on, some possible transition strategies are emerging. The adjacent channel interference data discussed earlier suggests that IBOC-compatible receivers will need to have tighter IF filters than those found in many classes of receivers currently in use.

Gradually increasing the power of the digital carrier over time is another possible facet of a transition strategy, such that initially the digital power will be lower, resulting in enhanced compatibility with existing receivers that are not "IBOC-compatible". Then, as IBOC-compatible receivers become more prevalent (time scale: years), which would be expected as listeners purchase new radios that can receive the IBOC digital signal, the power level would be increased affording the digital service to a larger area, and by then, minimizing the impact on non-IBOC receivers which are reduced in numbers from when the service was initiated.

It is also important to keep the appropriate goals in mind, which are, to develop a digital radio system which represents a significant improvement -- in sound quality, in data transmission capabilities, in signal robustness -- over existing AM and FM systems, but at the same time protects the investments of the existing broadcast community. These are ambitious goals which will take time to achieve. When one looks at the time it took for the ATV or Eureka standards to fully mature, then it is clear that more time is needed for the development of IBOC into a viable broadcasting standard.

SUMMARY

IBOC DAB gives broadcasters the best hope for digital quality while maintaining the existing structure of the industry. Broadcasters need to maintain their support for the development of IBOC, both individually and through participation in the NRSC-supported testing and standard-setting activities. Critics of IBOC, many of whom have already decided that new-band DAB is the only viable option, are not in a position at the present time to categorically reject IBOC because it is still being developed, and their criticism is premature. Furthermore, they ignore the realities of spectrum policy in the US which at the present time is extremely unlikely to result in new spectrum being allocated for terrestrial radio service.

Patience and perseverance are the key words for IBOC right now. Hopefully the on-going field test program will provide additional valuable insight into this technology, and this knowledge will help to bring an IBOC DAB standard for the US another step closer to reality.

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[1] Source - FCC Mass Media Bureau, "Equal Employment Opportunity Trend Report," December, 1994, page 1945.

[2] Source - Communications Daily, Feb. 13, 1996, page 8.

[3] "Report on Digital Audio Radio Laboratory Tests - Transmission Quality, Failure Characterization, and Analog Compatibility," EIA-CEG, August 11, 1995.

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[5] "IBOC Laboratory Tests Report," by Mr. Tom Keller, 1996 NAB Broadcast Engineering Conference, April 13-18, 1996.

[6] Source - FCC FM Database.

[7] Id.

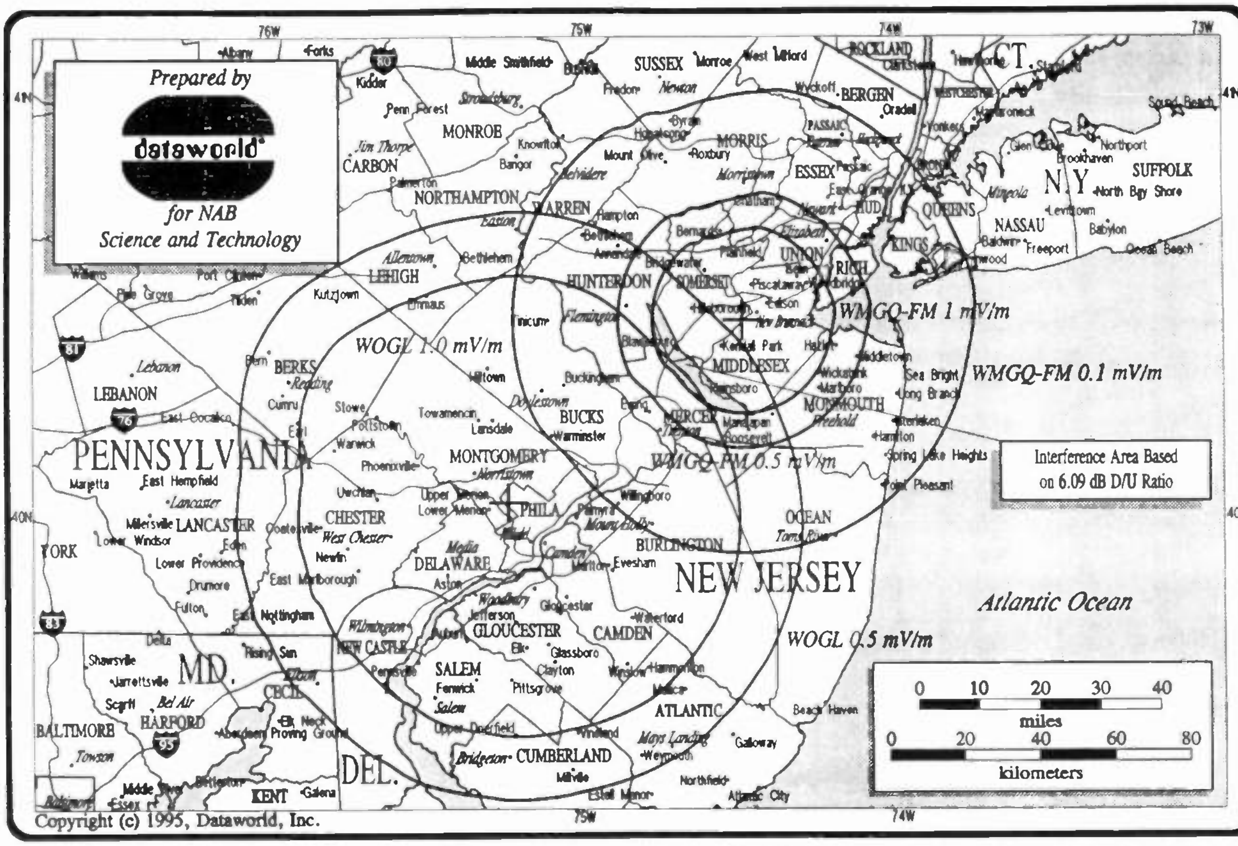

Figure 1. Predicted Interference Caused by a Lower 1st Adjacent IBOC Interferer to an Analog FM Signal Being Received by an Automotive Receiver

Figure 2. Predicted Interference Caused by an Upper 1st Adjacent IBOC Interferer to an Analog FM Signal Being Received by an Automotive Receiver

ON-CARRIER DIGITAL FM TECHNOLOGY: A NEW APPROACH FOR DIGITAL AUDIO BROADCASTING AND EXTRA HIGH SPEED DATA TRANSMISSION

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Abstract. *We examine research and development results pertaining to a new method for on-carrier FM broadcasting of digital data streams up to 200 Kbps, and their applicability for digital audio broadcast (DAB) and data dissemination via high speed subcarrier. Specifically, we look at the potential of the Subsidiary Communications Service (SCS) in terms of bandwidth efficiency (b/s/Hz), reception range, and practical implementation. We conclude that the new on-carrier method offers capabilities similar to those offered by proposed in-band DAB, while maintaining full compliance with current broadcast regulations including CFR Title 47, Part 73* [1].

Introduction

Emerging digital technologies offer FM broadcasters new opportunities previously assumed impractical. One example is the transmission of high speed digital audio, multimedia, and data services in the SCS subcarrier portion of the FM signal. These on-carrier services comply with current FCC regulations, and can provide digital services up to 200 kilobits per second (Kbps) while retaining their analog stereo signal. Up to 400 Kbps can be achieved by stations broadcasting in mono, using additional bandwidth normally taken by the stereo composite (L-R) signal.

Using high speed digital transmission on the FM subcarrier, analog FM transmission can continue while the new digital services are transmitted simultaneously. Bandwidth efficiencies of 5 b/s/Hz and more are feasible using today's technology. Table 1 provides examples of how data transmission rates are tailored to specific applications.

Various digital signal processing technologies such as bandwidth efficient modulation, audio compression, error correction, and fading control techniques are the key to expanding the capacity of the SCS portion of the FM signal. By combining the benefits of these techniques, robust high speed digital signals can be transmitted and received on the FM carrier. Proven multipath mitigation techniques including forward error correction, symbol interleaving, pilot tone tracking and adaptive equalization are employed to ensure quality reception in difficult environments.

TABLE 1. Digital Subcarrier Services as a Function of Data Rate.

Data Rate (Kb/s)	1.2 - 2.4	16-32	128-256	256-512	1,200
Bandwidth Efficiency (b/s/Hz)	<1	<1	>3	>6	>30
Service Examples	• RBDS • Paging • Differential GPS	• ITS (IVHS) • Fax	• Digital audio broadcast	• Digital multimedia (slow video)	• Digital multimedia (full motion)

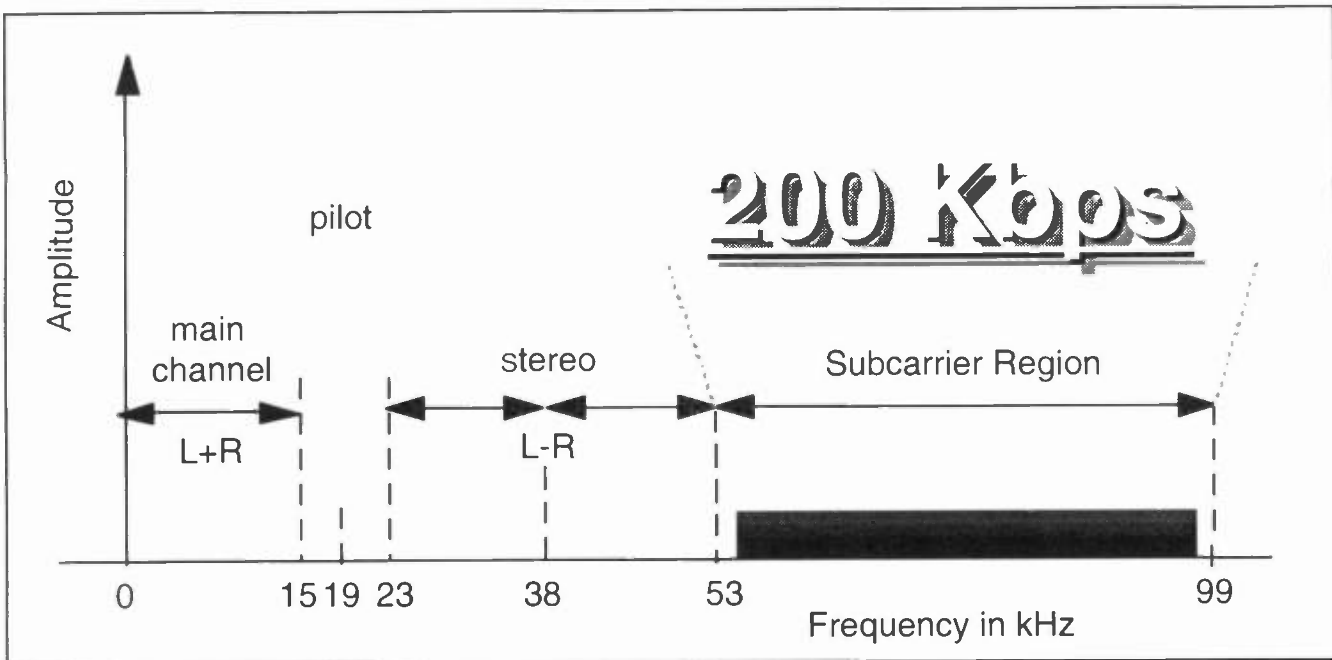

Figure 1. *Data rates of 200 Kbps are possible by transmitting signals with a bandwidth efficiency of 5-6 b/s/Hz over approximately 40 KHz of the subcarrier region Signal injection (10%) is kept within the limits specified by CFR Title 47, Part 73.319.*

Bandwidth Efficient Modulation

In the last quarter century, significant strides have been made in the field of audio frequency digital signal processing, in terms of improved waveforms and cheaper devices. This has enabled telephone modems using leased general switched telephone circuits to advance from 300 bits per second (300 bps) to 14,400 bits per second (14.4 Kbps) and beyond using a variety of different waveforms and equalization schemes (Table 2).

TABLE 2. Bandwidth Efficiency of General Switched Telephone Circuit (GSTN) Audio Baseband Modems

CCITT Recommend	Data Rate (bits/sec)	B/W Efficiency (bits/sec/Hz)
V.21	300	< 1.0
V.22	1200	< 1.0
V.22 *bis*	2400	< 1.0
V.27 *ter*	4800	1.8
V.32	9600	3.6
V.33	14400	5.3

Just as digital signal processing (DSP) technology has increased the bandwidth efficiency of telephone modems, it is also advancing radio broadcast state of the art. Today, audio baseband modulation schemes can support spectral efficiencies of 5 to 6 bits per second per Hertz (b/s/Hz) transmitted over an FM radio subcarrier (Figure 1). Precise implementation of these audio waveforms is made possible by low cost direct digital synthesizer (DDS) chips. The popularity of audio CD players and digitized audio for personal computing and games has driven down the cost of sophisticated digital audio devices. Technology change has been rapid, and approaches considered impossible three to five years ago are possible today.

On-Carrier vs. IBOC

An example of this rapid change in technology can be seen in waveform design for radio broadcast. When the designers of first-generation digital audio broadcast (DAB) systems contemplated transmission of "CD-quality" digital audio over the airwaves, it was not considered feasible then to obtain the requisite bandwidth (128-256 Kbps) necessary for DAB from on-carrier methods such as high speed subcarriers. Thus, the first generation In Band DAB proponents devised techniques such as "In Band, On Channel" (IBOC) which

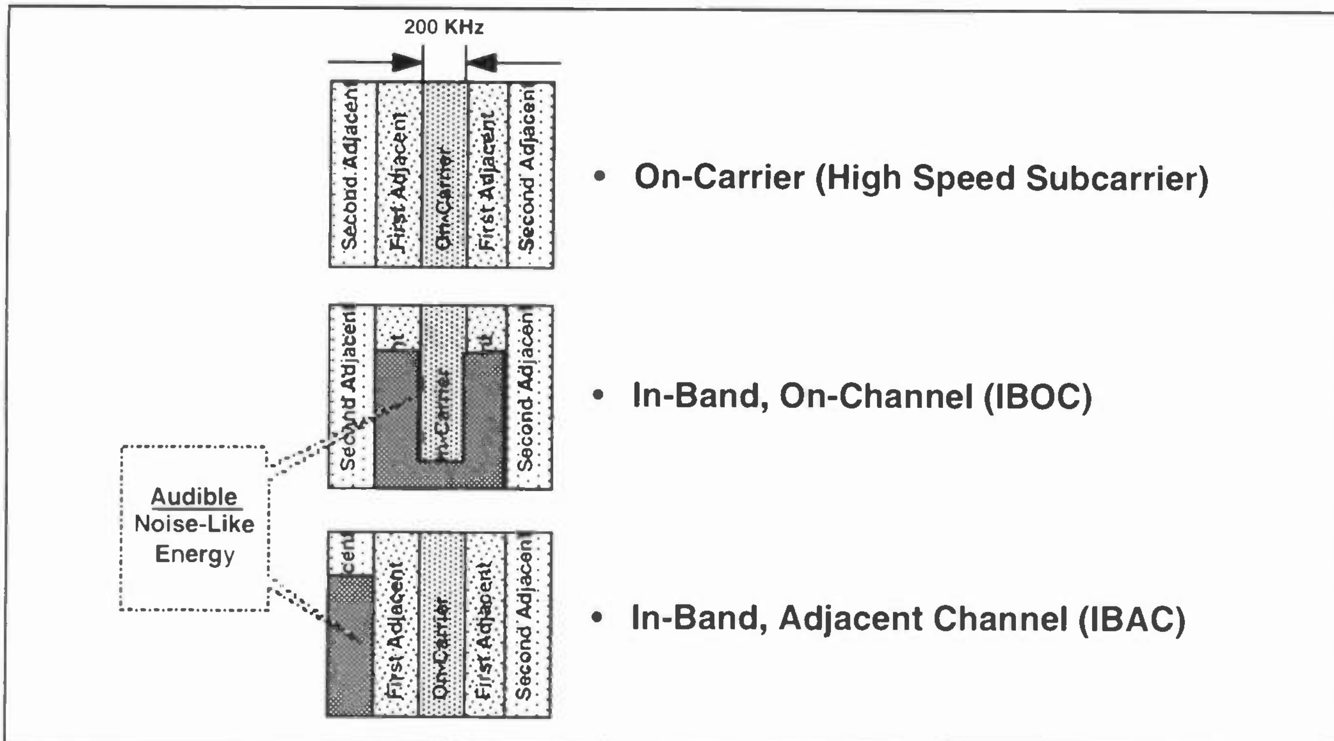

Figure 2. *An on-carrier method, using high speed subcarrier technology, can support DAB while maintaining full compliance with existing FCC regulations. Proposed IBOC systems inject noise-like energy into adjacent channels, and may cause interference with the main analog carrier, subcarriers, and other in-band DAB broadcasts.*

employ signal modulations which are not "true FM", making action by the FCC a possibility. These waveforms may inject digitally modulated, noise-like energy into the licensed channel as well as adjacent channels (Figure 2). During laboratory tests sponsored jointly by the Electronics Industry Association (EIA) and the National Association of Broadcasters (NAB), some proponent IBOC systems were found to cause audible interference to the analog FM signal, and were also found to interfere with subcarrier signals as well as adjacent-channel DAB signals[3]. Some systems also had difficulty when tested under simulated multipath conditions. Over-the-air testing of the proponent DAB systems, originally scheduled for 1995, has been delayed pending settlement of test procedure issues.

Unlike IBOC and IBAC signals, high speed subcarrier technology is an on-carrier FM method, strictly complying with Part 73.319 (subcarrier) and Part 73.317 (RF mask) regulations. Maintaining signal injection within legal limits (10%), high speed subcarriers with efficiencies of 5-6 bits/sec/Hz can be transmitted over 38 KHz of available bandwidth (57 KHz - 95 KHz) on carriers with stereo analog signals. FM stations broadcasting a stereo analog signal can achieve about 200 Kbps of channel data bandwidth. Monaural stations, with 76 KHz of available bandwidth (19 KHz - 95 KHz), can transmit approximately 400 Kbps on their subcarriers. In both cases, the FM subcarrier remains compatible with the analog signal.

No Interference To Main Carrier

Interference to other stations is not an issue because the high speed data subcarrier operates within the occupied bandwidth limitations of FM stations. Interference to the analog stereo signal is prevented with simple digital filtering. In laboratory tests one such filter reduced

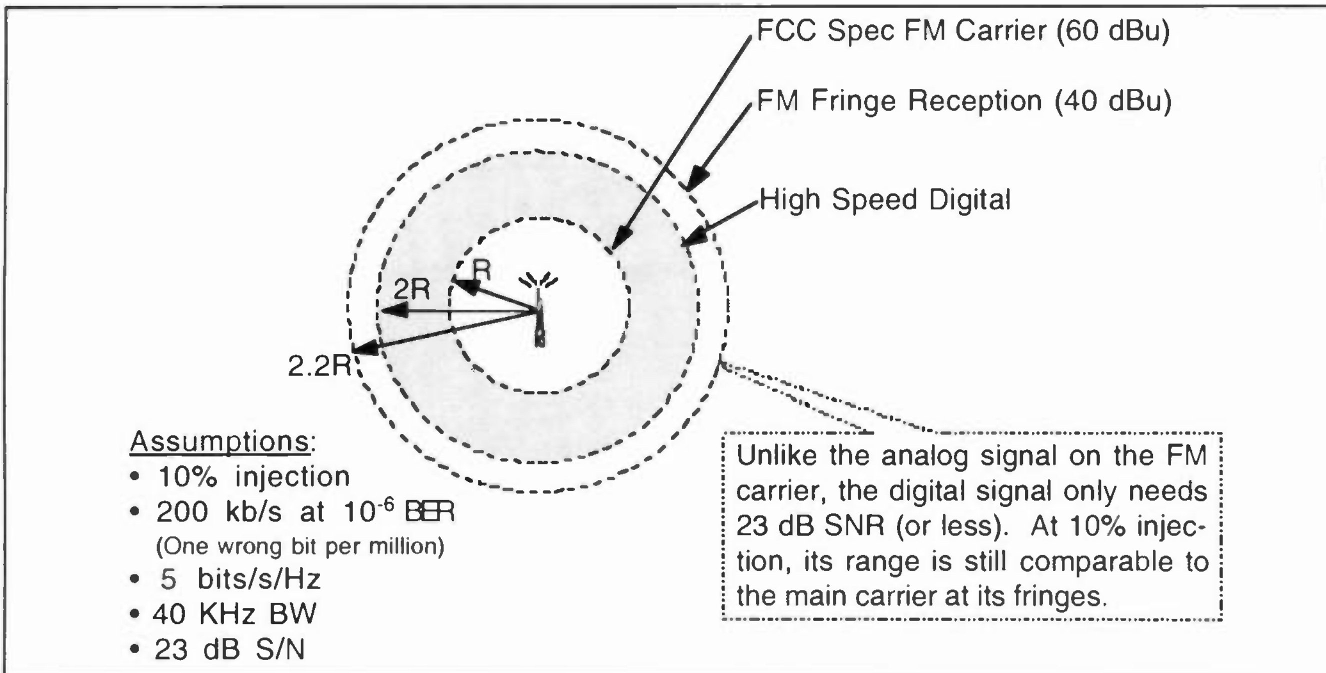

Figure 3. *A 200 Kbps high speed subcarrier signal, at 10% injection, can maintain quality digital service with 10^{-6} BER at transmission ranges comparing favorably with the analog signal at its fringes.*

crosstalk to inaudible levels. Better filtering is readily available if better performance is desired.

On-Carrier Digital FM Broadcast

FM broadcasters can influence how they will use their wideband digital subcarrier channels. Whether the "bit stream" carries full bandwidth stereo, multichannel mono, images, or other forms of data depends entirely on the way FM broadcasters choose to use it.

Stations wishing to maintain their analog stereo format can do so, and still support about 200 Kbps of channel bandwidth. This is sufficient bandwidth to broadcast one stereo, CD-quality DAB signal[4,7]. Stations maintaining a monaural format can double their channel bandwidth to about 400 Kbps. A migration path to fully digital FM also exists. With regulatory approval, analog services can someday be phased out completely and replaced by additional data bandwidth.

The new wideband digital subcarrier is not compatible with other subcarriers signals, such as the RBDS subcarrier at 57 KHz. However, an RBDS-like function can be incorporated by multiplexing the low-bandwidth RBDS data on the new wideband subcarrier .

Digital Subcarrier Reception Range

Modern digital modulation schemes require less signal-to-noise ratio while maintaining high levels of performance (fewer than one wrong bit per million). Even at low levels of injection (10%), digital signals continue to provide good service at distances comparing favorably with the analog main carrier (Figure 3).

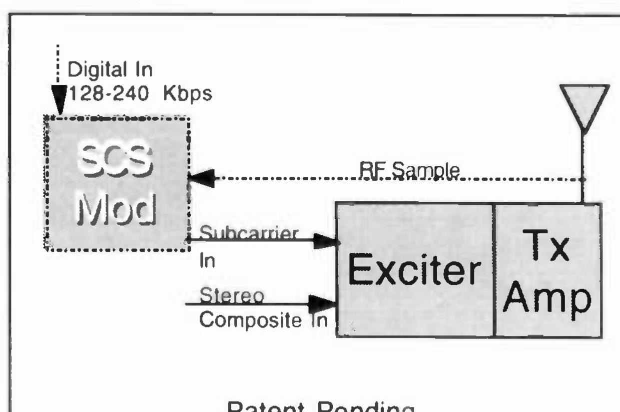

Figure 4. *Upgrading the FM broadcast station for on-carrier FM digital broadcast is straightforward, requiring the addition of a new SCS modulator. An RF sample line provides feedback for adaptive equalization.*

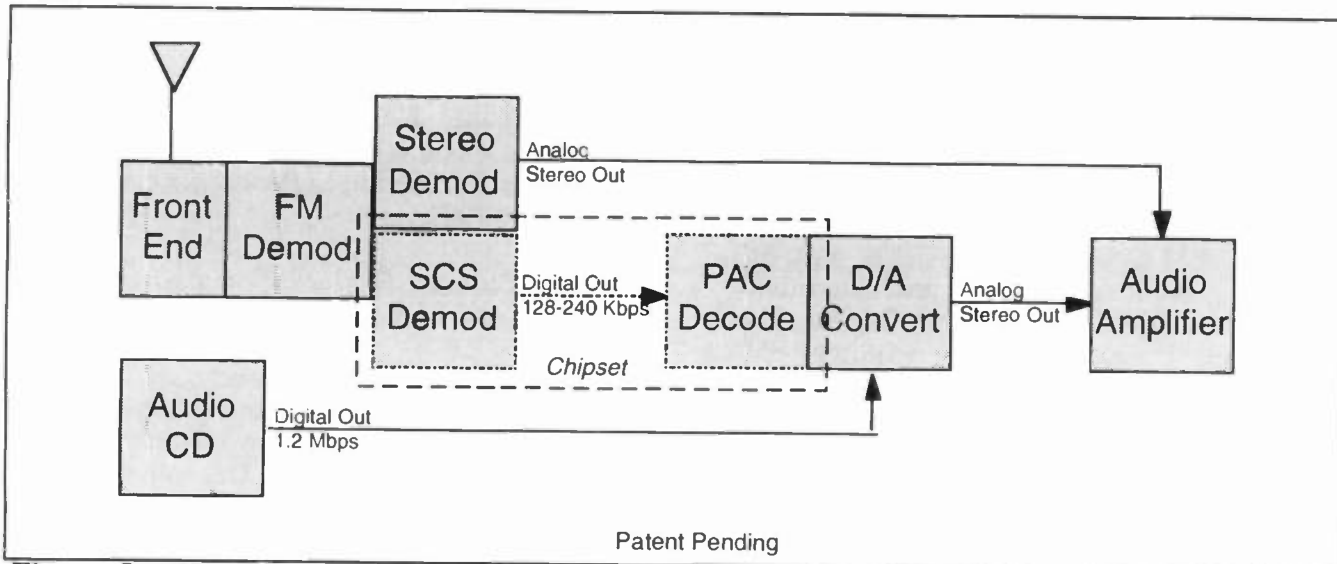

Figure 5. *Block diagram of a typical stereo system with Digital FM and audio CD player. Conversion of the consumer receiver to Digital FM operation is straightforward.*

Upgrading the Broadcast Station

Upgrading the modern FM broadcast station for on-carrier digital FM is a straightforward operation. The key new element is a wideband digital SCS modulator (Figure 4). For stations broadcasting digital audio, a second component called a psychoacoustic decimater (also known as a perceptual audio coder, or PAC) is also necessary to "compress" the bandwidth of an audio CD player (about 1.2 Mbps) to about 160 Kbps - a compression ratio of approximately 7.5:1.

Low Cost Consumer Receivers

Redesign of the consumer receiver for on-carrier digital FM is also straightforward (Figure 5). Two new components are required: the SCS demodulator, and (for digital audio) the PAC decoder. In the near term, these devices will appear initially as "chipsets". These will be installed in new consumer radios, largely derived from existing analog designs.

The SCS demodulator input is connected to the FM discriminator output (as used by the composite stereo demodulator in today's analog receiver). The digital output of the SCS demodulator is input to the PAC decoder and digital to analog (D/A) converter. Since the new SCS and PAC components are audio-frequency devices, comparable to digital audio chipsets on telephone modems or audio CD players, the cost of on-carrier digital FM receivers can be several hundred dollars less than competing DAB implementations, such as IBOC, L-Band or Satellite Broadcast receivers.

Summary

FM broadcasters are closely watching the development of other Digital Audio Broadcasting services, concerned that FM radio will eventually be at an economic and technological disadvantage. On-carrier DAB technology will soon offer the FM broadcaster a rapid and relatively inexpensive way to "go digital." Its high bandwidth efficiency, coupled with emerging PAC coder algorithms, finally makes on-carrier digital audio possible. Decades of FM subcarrier broadcasting experience support new experimental evidence that on-carrier digital audio is fully compatible with existing analog stereo.

On-carrier digital audio as described in this paper requires no new regulations or approvals from the FCC, because it occupies allowed spectrum at allowed signal levels. Preliminary data indicate that coverage area and signal reliability is comparable to or better than the analog signal on the same station. Cost of on-carrier DAB receivers will be comparable to today's analog receivers because it relies on the existing analog FM front end and available, low-cost digital audio technologies. All the building blocks are available for rapid implementation of a new consumer medium delivered by FM broadcasters.

Taking a cue from the television "Grand Alliance," FM broadcasters can define a model for making their signals more productive and more competitive. On-Carrier Digital FM technology presents an opportunity FM broadcasters should not overlook.

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SATELLITE DAB TECHNOLOGY

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ABSTRACT

The paper describes the current technology proposed for use in Digital Audio Broadcasting (DAB) systems in the United States using the frequency band 2310-2360 MHz. The technology will be used to implement a new service, termed Satellite Radio, which provides multiple audio channels of music and voice to low cost radio receivers. The technology discussed includes the capabilities of current geosynchronous satellites, the use of information compression, the transmission encoding of the information and modulation techniques. Methods of mitigating service outages to mobile user vehicles from transmission blockage and multipath are presented.

INTRODUCTION

Digital Audio Broadcasting (DAB) from satellites in the United States currently encompasses a wide variety of possible service requirements. Because of this, various other descriptive titles have been used (e.g., SDARS-Satellite Digital Audio Radio Service; DSR-Digital Satellite Radio; DSB-Digital Satellite Broadcasting, etc.). The major differentiator between satellite DAB systems is whether they serve only fixed locations, such as is done by DirecTV, or whether they serve both mobile platforms and fixed locations. Equivalent service to mobile platforms requires much more powerful satellites and transmission at radio frequencies below 3000 MHz. Major focus for satellite DAB systems has been on the provision of national service. Two reasons for this are the ability to concentrate satellite radiated power on a more limited coverage area and to avoid political problems associated with propaganda. The following discussion will dwell primarily on satellite DAB systems servicing mobile platforms with United States national coverage since implementation plans for such have been published[1].

SYSTEM CONSIDERATIONS

1. Configuration

The most basic satellite DAB configuration is shown in Fig. 1. Essentially a central programming and origination earth station transmits the broadcast radio programs to a geosynchronous satellite using an uplink frequency generally in the low microwave range. The satellite contains a frequency translation transponder (i.e., "bent pipe") which changes the uplink frequency to the desired downlink frequency, provides power amplification and directs through its antenna the resultant energy towards the coverage/service area on earth. The satellite's transmission is received by mobile and fixed users within this coverage area who recover the broadcast programs.

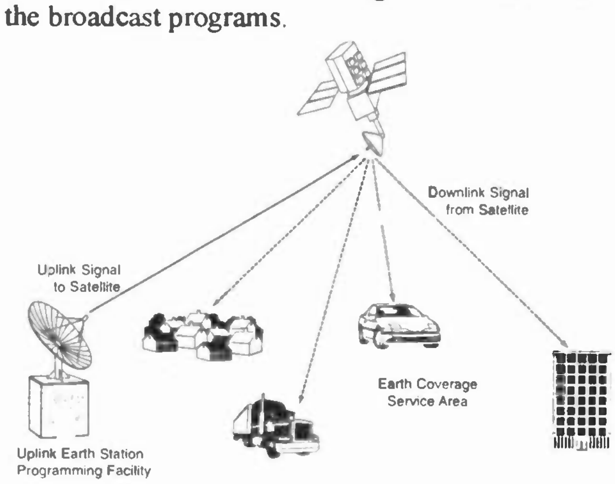

Fig. 1 Basic Satellite DAB System Configuration

a. Orbit The choice of a geosynchronous orbit for a DAB satellite permits coverage of any country situated at moderate latitudes using a single satellite or, if satellite spatial diversity is employed, a pair of satellites. Low or medium earth orbit satellite systems for satellite DAB are economically unattractive since their advantages (i.e., global coverage, lower path loss and small propagation time delay) are generally not applicable to a broadcast service. In this regard, it is assumed that the required coverage area will be fully encompassed by the antennas of either the geosynchronous or the low earth orbit satellites so that the larger required beamwidth of the low earth orbit satellite negates its path loss advantage. Also, the broadcast signal being a one way transmission is unaffected by the length of the propagation time delay. However, countries at high latitudes have low elevation angles to geosynchronous satellites, such as northern Europe. Service outages from blockage and multipath are much more severe at low elevation angles as subsequently discussed. To overcome these deficiencies in high latitude service areas, a satellite DAB system has been proposed[2] which uses several satellites in inclined, highly elliptical orbits with apogees above geosynchronous orbital altitudes at northern latitudes and medium altitude perigees in the southern hemisphere.

b. Uplink Access/Transponder Configuration It is possible to use a DAB satellite transponder which receives and demodulates/demultiplexes the uplink transmission and then remultiplexes/ remodulates the signals (with the same or different modulation) prior to downlink transmission. The complexity of so doing is only employed when special service requirements must be met. For "bent pipe" operation, the uplink signal-to-noise ratio normally can be easily made high enough to be negligible in the overall transmission system so demodulation/remodulation satellite transponders offer no significant benefits in transmission link margin. A DAB service requiring access to the satellite transponder from several uplink earth stations (i.e., multiple broadcast origination centers) or where downlink coverage is provided by multiple beams, as discussed subsequently, can benefit from use of a demodulation/demultiplexer-remultiplexer/remodulation transponder configuration. Similar benefits can occur when the bandwidths of the uplink and downlink must be different due to available International Telecommunications Union (ITU) and national frequency allocations.

c. Operational Radio Frequencies/ Bandwidths There are frequency allocations for DAB downlinks which were made at the World Administrative Radio Conference (WARC) 1992. Many countries chose the 1452-1492 MHz band (i.e., L-band) with certain restrictions on initial usage and implementation timing. The United States has chosen the 2310-2360 MHz band, and the USSR and a few Asian countries chose 2535-2655 MHz (i.e., S-band). From a theoretical system design viewpoint, the lower the radio frequency band and the wider the usable bandwidth, the more effective and economical will be the resulting satellite DAB hardware implementation for equivalent service. It has been shown from a practical implementation viewpoint that effective DAB systems can be built in any frequency band below 3 GHz with a bandwidth of 10-20 MHz apiece[3]. However, it is vital for DAB systems serving mobile platforms that the downlink frequency band chosen be free from significant radio interference everywhere within the coverage areas. This has been and is the key issue in satellite DAB system frequency selection and is under current consideration by the ITU-R[3]. Where more than one satellite DAB system serve a specific coverage area or where they serve adjacent coverage areas, intersystem interference dictates the use of different downlink frequencies by each satellite system (i.e., frequency band segmentation). Four United States DAB systems have applied to the FCC for use of the 2310-2360 MHz radio frequency spectrum using segmentation of the band in fourths. It is possible in some cases to circumvent the use of frequency band segmentation by intersystem cooperative use of Code Division Multiple Access (CDMA), but this severely restricts the use of differing technology, service offerings and innovative systems.

d. Satellite Earth Coverage A single downlink beam from the satellite covering the contiguous United States service area is generally an effective DAB implementation. However, several variations may occur. One is where spot beams are added to cover Alaska, Hawaii and Puerto Rico. For very large national areas, coverage by partially overlapping multiple beams might be used so differences in time zones and regional interests can be accommodated. An extension of this is in satellite DAB systems serving an overall area where individual beams might be used for diverse programming to regions therein. It is also possible to have a satellite DAB system provide full service area coverage of many square degrees and, additionally, narrow beam coverage at an adjacent radio frequency of a few square degrees to those subsections within the service area where additional programming capability is desired. The use of spot or multiple satellite antenna beams can cause spectrum requirement inefficiency, prohibitively high radiated power flux densities, and inter/intra-system interference.

The earth coverage of the satellite DAB uplink beam is generally not a significant technical problem and would be selected to encompass as closely as possible the area(s) containing the origination/uplink center(s). Such directive satellite uplink antenna coverage both improves the uplink signal-to-noise ratio, as previously mentioned, and reduces possibilities for uplink radio frequency interference.

e. Earth Terminals

(1) Fixed. Fixed earth terminals for receiving satellite DAB are technically identical to those used for receiving Direct Broadcast Satellite (DBS) television[4] with appropriate adjustment for difference in frequency of operation and antenna size. Current implementation plans for satellite DAB envision outdoor or through a window indoor reception, since the required satellite radiated power for true urban indoor reception cannot be economically implemented at this time.

(2) Mobile. The emphasis on United States' satellite DAB systems is to provide service for users on mobile platforms including automobiles, trucks, campers, etc. Satellite DAB service is currently being provided to fixed users by DBS television and cable systems, but such service (i.e., many broadcast channels uniformly distributed over a wide geographical area) can only be effectively provided to mobile users by satellite with downlink frequency transmission below 3000 MHz. The mobile satellite DAB platform terminal is relatively straightforward from a technology viewpoint. Its performance is limited by the receiving antenna whose main beamwidth must stay in view of the satellite as the mobile platform turns or otherwise changes its orientation. It is currently not possible to implement a steerable phased array antenna or a mechanically steered antenna that is economical for this service (e.g., cost of under US \$10), ergonomically suitable for mounting on passenger automobiles and maintenance free over several years of operation. Consequently, antennas for this service are low gain with toroidal beamshapes. This allows relatively constant gain in the azimuthal plane (e.g., as the mobile platform turns) and directive gain in the elevation plane, the amount dependent on satellite orbital location(s) and on the geographical extent of the service area. For the contiguous United States and geosynchronous satellites located at 70° and 120° W. Longitude, a toroidal pattern providing 3-5 dBi of gain between 20° and 60° elevation angle can be achieved with a planar array using microstrip construction. An antenna of this type is shown in Fig. 2. A typical G/T for such a mobile terminal is -19 dB/K using HEMP MIC low noise amplifier receivers[5].

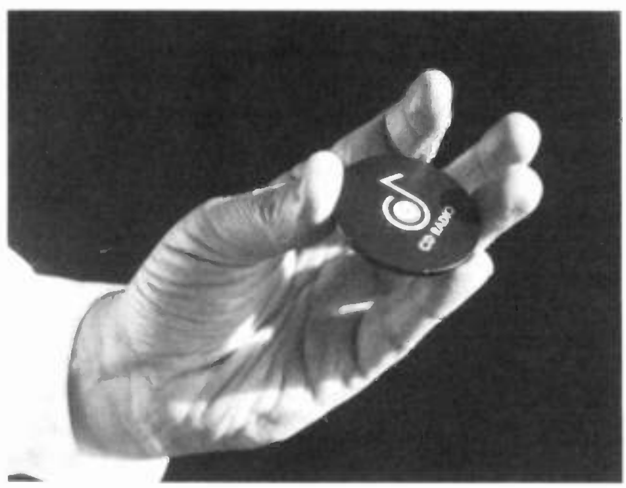

Fig. 2 DAB Automobile Antenna

SERVICE CONSIDERATIONS

1. Service Requirements

The introduction to this paper dealt briefly with the broadness of service requirements. These have been well delineated elsewhere[6]. As a generalization, DAB services in the United States are envisioned to provide premium aural programming which includes stereo CD music in a narrowcast mode (i.e., large numbers of such music channels each with a different genre) and ethnic channels (i.e., due to the broad satellite DAB geographical coverage, ethnic groups too small to be served economically by local terrestrial radio can be accumulated to form a sufficient audience). Ancillary satellite DAB services also would distribute voice for news, entertainment, education, social and political purposes and provide low speed digital services such as paging, facsimile and weather/emergency alert data.

2. Service Quality

Two aspects of service quality are of great importance to system performance. They are digital compression of the audio signals and reception durability (i.e., signal outage).

a) Compression. All satellite DAB systems proposed to date utilize compression of the broadcast programming usually at its transmission point as part of the analog-to-digital conversion process and expansion at the reception point prior to the digital-to-analog conversion. Several compression schemes with associated processing algorithms have been developed and tested such as MUSICAM and PAC (Perceptual Audio Coding)[7]. Currently, stereo CD music quality can be achieved by joint processed compression with a 128 kb/s data transmission rate assuming a transmission error rate of 10^{-5} and monaural FM quality voice with a 32 kb/s data transmission rate.

b) Outage. Mobile platform receivers in DAB systems using a single satellite will experience outage whenever the line of sight between the satellite and receiver is physically obstructed (i.e., blockage) and whenever multipath fading exceeds the mobile platform receiver's transmission service threshold. Blockage type outages can be reduced or eliminated by use of complementary terrestrial repeaters of the satellite signal within the obstructed areas and by satellite spatial diversity where two satellites transmitting the same broadcast signal but located many degrees (e.g., 30°) apart on the geosynchronous orbital arc are both received by each mobile platform. Additionally, very short duration outages can be avoided through extensive error encoding/ interleaving of the transmission stream. The usefulness of this technique is dependent both on the tolerable receiver cost for increased storage and decoding and on the tolerable receiver recovery time after an outage.

Multipath fading in a mobile environment is a more complex matter since, for satellite DAB, there are diffuse components (i.e., quasi-Ricean), specular components (i.e., brief, high intensity interference often associated with reflections from buildings or trucks), and quasi-Rayleigh components (e.g., trees and their leaves). Heavy Rayleigh fades are associated with major transmission path obstructions which satellite DAB systems would address as the previously discussed blockage. Other types of multipath in satellite DAB systems can be mitigated by using higher satellite EIRPs to overcome such fading, higher elevation angles between the mobile platforms and the satellite(s), satellite spatial diversity as previously mentioned either alone or in combination with radio frequency diversity, transmission encoding/interleaving as previously discussed and multipath resistant modulations as will be discussed subsequently. It is possible to combine some or all of the previous techniques in a practical satellite DAB system so that a high service reception durability with regard to outage performance can be attained. It is noted that raising satellite EIRP alone is not an effective solution to Ricean multipath since the satellite would have to be over ten times more powerful. Preliminary measurements at S-band of an emulated satellite DAB combined spatial and frequency diversity system indicate many decibels of multipath mitigation and elimination of frequency selective fading are achievable[8].

c) Modulation/Multiplex. An effective modulation/multiplex for satellite DAB is time division multiplex (TDM) transmitted by offset quadrature phase shift keying (OQPSK). This allows the satellite's output power transmitter to be driven at or near saturation and is spectrum efficient in terms of number of channels per megahertz. Other effective multiplex techniques for satellite DAB are Coded Orthogonal Frequency Division Multiplex (COFDM) and forms of spread spectrum modulation using Code Division Multiplex (CDM). Both these modulations/ multiplexes are advantageous in having properties which mitigate multipath fading (including the rejection of multipath components or the constructive addition of multipath components by use of Rake design receivers with the number of combinable components and their allowable relative delays set by receiver economics), resist interference and allow easy access (e.g., Code Division Multiple Access-CDMA). Both these modulations/ multiplexes require that the satellite's output power transmitter be operated below saturation, the amount dependent on allowable intermodulation levels and linearization techniques. However, this lessened amount of transmitter power can only be regained for equivalent system performance by use of a more powerful satellite.

SYSTEM IMPLEMENTATIONS

Four satellite systems have been proposed for implementation to provide the previously discussed DAB service in the United States for mobile and fixed users at S-band. The systems make various selections of the previously described technology and techniques. Assuming government regulatory approval in the near future, it is possible that one or more of these satellite DAB systems would be operational in 1999.

SATELLITE DAB SYSTEM EXAMPLE

The example satellite system consists of two geosynchronous satellites, one located over the east coast of the United States at 80° West Longitude and the second over the west coast of the United States at 110° West Longitude. The satellites receive in the 6720 MHz band and transmit in two 8 MHz segments of the 2310-2360 MHz band. The satellites each receive the same transmission from the system's up-link/programming center essentially simultaneously and retransmit the signal through an antenna beam covering the contiguous United States. Fig. 3 shows the block diagram of the satellite's transmission payload. The retransmission frequencies of the two satellites are separated by 20 MHz so that both satellite spatial and frequency diversity are achieved, and the beam edge satellite EIRP is 57 dBW. The high EIRP is required due to the low gain of the mobile platform antenna and the large number of broadcast programs. The transmission from the origination center consists of 30 stereo CD music channels. The CD stereo music programs are compressed prior to transmission using a joint processing algorithm based on perceptual audio coding so only a 128 kb/s output data rate is required for each. Several compressed voice programs are also provided. The program channels are digitally multiplexed together (i.e., TDM-time division multiplex) resulting in a 4 Mb/s output signal. The data blocks of the output signal are Reed-Solomon coded, then convolutionally encoded with time interleaving and finally transmitted to the satellites using offset quadrature phase shift keying (i.e., the uplink).

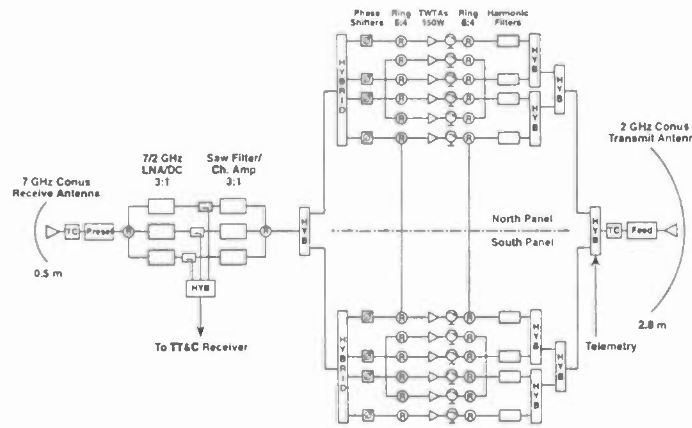

Fig. 3 DAB Satellite Communications Block Diagram

Fig. 4 gives a summary transmission link budget. The satellite system cost is approximately US $400 million.

Satellite EIRP (1)	57	dBW
Path Loss (2)	-192	dB
Vehicle Antenna Gain (3)	3	dBi
Received Power at Vehicle	-132	dBW
Receiver Noise Power (4)	-141	dBW
Resultant E/No	9	dB
Required E/No (5)	5	dB
Power Margin	4	dB
Diversity Gain (6)	12	dB
Effective Multipath Margin	16	dB

(1) At edge of coverage area
(2) Geosynchronous orbit/2335 MHz
(3) Worst orientation; includes polarization loss
(4) Effective G/T= -19 dB/ K; includes uplink contribution and losses; B_N=3.9 MHz
(5) B.E.R. of 10^{-6}
(6) Satellite spatial, frequency and time diversity provides 9-15 dB multipath mitigation

Fig. 4 Satellite-To-Vehicle Link Budget

The satellite retransmissions (i.e., the downlink with a worst case power flux density of -139 $dBW/m^2/4kHz$) are received by the mobile platforms, particularly passenger automobiles. The mobile platform G/T at worst operational aspect angle is -19 dB/K. The antenna is designed to provide 3 dBi gain within a 20°-60° elevation angle range at all azimuths. The antenna is physically 2.5 cm in radius and 0.4 cm thick, designed for embedment in automobile rooftops. After radio frequency reception, amplification and down conversion in the mobile platform receiver, the transmission from each satellite is individually demodulated. The two signals are time phased together using a maximal ratio combiner and then demodulated. The user selects the specific music or voice channel desired which is then routed to the decompressor, the digital-to-analog converter and the audio amplifier-loud speaker subsystem. Figure 5 shows a block diagram of the mobile platform receiver which is capable of selecting AM, FM, satellite DAB and terrestrial DAB. The mobile platform receiver just described enjoys great resistance to multipath fading and blockage for satellite DAB, since its mechanization takes combined advantage of satellite spatial and frequency diversity as well as transmission encoding and interleaving. Estimates of the consumer price[9] when such receivers would be volume manufactured are US $150-200.

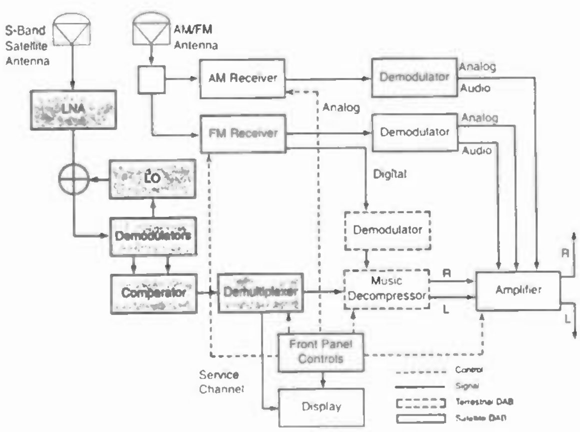

Fig. 5 Vehicle Receiver
Analog AM & FM/Digital Satellite & Terrestrial

CONCLUSION

Satellite DAB systems serving the United States will start implementation in the near future. Such systems will provide new and additional audio services for mobile and fixed users in terms of quality, numbers of channels, programming diversity and availability over large geographical areas. The technology to build satellite DAB systems has been developed which will allow economical construction of the satellites and user receivers, and the techniques for efficient system design are available.

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ADVANCED TELEVISION: PART I

Sunday, April 14, 1996
9:30 am - 12:25 pm

Session Chairperson:
Robert Seidel, CBS, New York, NY

FIELD TESTING THE GRAND ALLIANCE HDTV SYSTEM
Jules Cohen, P.E.
Consulting Engineer
Washington, DC

SUMMARY OF GRAND ALLIANCE VSB TRANSMISSION SYSTEM FIELD TEST
Gary Sgrignoli
Zenith Electronics Corporation
Glenview, IL

SUBJECTIVE EVALUATION OF THE DIGITAL HDTV GRAND ALLIANCE SYSTEM
Robert Leafloor
Communications Research Centre
Ottawa, Ontario, Canada

***SPECTRUM PLANNING AND RF IMPLEMENTATION FOR ATS**
Thomas M. Gurley & Dennis W. Wallace
Advanced Television Test Center
Alexandria, VA

THE HARRIS VSB EXCITER FOR DIGITAL ATV
Robert C. Davis
Harris Corporation, Broadcast Division
Melbourne, FL

COVERAGE CONTOUR OPTIMIZATION OF HDTV AND NTSC ANTENNAS
Oded Bendov
Dielectric Communications
Cherry Hill, NJ

A 2/3-INCH 2 MILLION PIXEL CCD CAMERA FOR HDTV
Hiroshi Kouchi
NHK
Tokyo, Japan

*Paper not available at the time of publication.

FIELD TESTING THE GRAND ALLIANCE HDTV SYSTEM

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ABSTRACT

Following objective and subjective testing by expert viewers at the Advanced Television Test Center (ATTC) and Cable Laboratories, and subjective viewing by non-experts at the Canadian Advanced Television Evaluation Laboratory (ATEL), field testing of the Grand Alliance HDTV system was conducted from a site near Charlotte, North Carolina. Two series of tests were conducted. The first involved only the transmission subsystem. The second series of tests employed the entire prototype system and was conducted with both pictures and sound. In this paper, the procedures followed are described. Results of the tests are being covered in a companion paper.

BACKGROUND

The Grand Alliance, a consortium including AT&T, David Sarnoff Research Center, General Instrument Corporation, Massachusetts Institute of Technology, Philips Electronics North American Corporation, Thomson Consumer Electronics, and Zenith Corporation, had selected the eight-level, vestigial side band (8VSB) transmission subsystem for terrestrial broadcasting after analytical and laboratory comparisons of the use of 8VSB or 32QAM. For cable television distribution, a sixteen-level subsystem (16VSB) was adopted also. In the spring of 1994, prototype 8VSB and 16VSB transmission subsystems were delivered to the Advanced Television Test Site in Charlotte, North Carolina, for testing under actual field conditions of both terrestrial and cable distribution.

Laboratory testing had established that a bit error rate of 3×10^{-6} corresponded to the video threshold of visibility (TOV). In the absence of the ability to deliver pictures and sound employing only the transmission subsystem, bit error rate was employed as the criterion for acceptable HDTV reception.

Laboratory testing of the complete Grand Alliance system prototype was finished in June, 1995. That prototype, and a second complete prototype system certified to be virtually identical to the system tested in the laboratories, were sent to Charlotte for field testing. In this second series of tests, pictures and sound were transmitted permitting judgments to be made as to whether or not impairments in either could be detected.

The field testing program had as a primary objective the evaluation of the Grand Alliance HDTV system under the propagation conditions encountered in both VHF and UHF terrestrial broadcasting, and in cable distribution. Part of that evaluation was a comparison of performance with NTSC, the system that has served well for the past fifty years. Of particular concern was the major difference between the characteristic performance of the digital, Grand Alliance HDTV system and the analog NTSC system. Whereas NTSC deteriorates gradually with decreasing signal strength, digital HDTV maintains its near perfect performance until a signal-to-noise or signal-to-interference ratio reaches a level where the picture and/or sound is no longer useable. The ratio differential between the point where a problem is just detectable and when reception fails can be only a fraction of a decibel.

Secondary objectives of the field testing program involved considerations of both cochannel and adjacent channel interference, particularly for HDTV to NTSC, and the performance of the system when used with a side-mounted UHF antenna.

THE FIELD TESTING SITE FACILITIES

At the field testing site near Charlotte a 1,337-foot tower and transmitter building were made available by Lodestar Towers and by the former tower owner, Journal Broadcasting of Charlotte, licensee of WCNC-TV. An omnidirectional, top-mounted type TFU-24G UHF antenna was provided by Dielectric, and a single-bay omnidirectional VHF antenna, mounted just below the UHF antenna, was provided by Harris. Coaxial transmission lines for antenna feeds were supplied also by Dielectric. A side-mounted, directional UHF antenna and transmission line were provided by Andrew. Comark provided a UHF transmitter with EEV IOT output stage. LDL provided a solid-state VHF transmitter. A field truck was loaned to the project by Harris Allied. A transmitter/translator, used for cochannel and adjacent channel testing, was provided by ITS. Numerous companies made test equipment available to the project.

Studies had shown that VHF channel 6 and UHF channel 53 had the least potential for interference both to and from existing stations. An experimental license was granted by the FCC to operate on those channels. To limit the probability of interference to television broadcast stations, and to avoid the need for a UHF transmitter rated for at least 120 kilowatts, NTSC peak visual power was limited to one-tenth of the maximum permitted by FCC rules. The VHF operation was at an effective radiated power (ERP) of 10 kilowatts. The UHF operation was at an ERP of 500 kilowatts. An analysis made by Planning Subcommittee Working Party 3 of the Advisory Committee for Advanced Television Service (ACATS) had concluded, on the basis of laboratory testing, that the Grand Alliance HDTV system operating at an average power 12 dB below NTSC peak visual should provide service to a distance equivalent to NTSC Grade B. Accordingly, average HDTV ERP was 0.63 kilowatts on channel 6 and 31.6 kilowatts on channel 53.

The field truck was complete with AC generator and extendible mast. Contained within the truck were receivers and monitors, initially for NTSC only during the transmission subsystem testing and for both NTSC and HDTV for the complete prototype testing. The NTSC receiver included a switchable ghost canceler. The antenna employed was an all-band Delhi Model VU-932, with characteristics typical of an antenna of moderate gain likely to be used at a residence. Test equipment included a vector signal analyzer, spectrum analyzer, waveform monitor, channel characteristic analyzer, VM700 video measuring set, VHF and UHF field strength meters, and bit error rate measuring equipment. Included also in the field truck were a personal computer for both control of some functions and for collection of data and a noise generator (for use in determining margin to HDTV failure threshold). Amplifiers, attenuators, splitters and cabling permitted integration of the several units of reception and measuring equipment. The Grand Alliance demodulator and its associated computer and test equipment provided the capability of reading signal-to-noise ratio at both input and output of the equalizer, equalizer tap energy, and segment error rate. For communications with the transmitter operator and for other needs, the field truck was equipped with both a broadcast auxiliary transceiver and a cellular telephone.

The Charlotte facility and field test operations were managed and staffed by the Public Broadcasting Service. Support and guidance were provided by the ACATS Subcommittee 2 Field Testing Task Force, the Association for Maximum Service Television, Inc., and by Cable Television Laboratories, Inc.

TESTING OF THE TRANSMISSION 8VSB SUBSYSTEM

A plan providing the general specifications for field testing was prepared by the Field Testing Task Force and approved by ACATS Subcommittee 2. Detailed procedures and log sheets were developed from that plan by Edmund A. Williams, the full-time manager of the project.

To provide a good statistical sample of performance, eight radials were drawn with the objective of obtaining a variety of terrain conditions, supplemented by grids in Charlotte and Rock Hill, South Carolina, to obtain data under urban conditions of large and medium-sized communities. The radial measurements, 128 in number, 16 per radial, extended to a distance of approximately 55 miles (88.5 kilometers). The main grids within the communities used a spacing of one mile (1.6 kilometers). A cluster with half that spacing was included also within each community grid. A third cluster was located only 2.5 miles (4.0 kilometers) from the transmitter to observe reception characteristics at relatively large depression angles. Data were collected at a total of 199 radial, grid and cluster locations.

At each of the 199 locations, both VHF and UHF field strength were recorded while the field truck was moved over a 100-foot path with antenna elevated to 30 feet (9.1 meters). The field truck was then returned to the midpoint of the path to make the fixed measurements and observations. In strong signal areas, attenuation was required to avoid receiver and test equipment overload (a function comparable to the automatic gain control found in all television receivers).

Measurements were made of the NTSC peak sync level, noise floor, carrier-to-noise level over the 6-MHz band, and video signal-to-noise ratio. Impairments in the NTSC picture were graded using the CCIR five-point grading scale. The threshold for a viewable picture was grade 3, "slightly annoying." Channel characterization data were collected using equipment provided by the Institute for Telecommunication Science of the Department of Commerce National Telecommunications and Information Administration. If cochannel interference was present, its level was recorded. A spectrum analyzer "snapshot" was obtained at each site.

HDTV data collected included input signal level, pilot level, bit error rate, segment error rate, noise floor, signal-to-noise ratio at both the input and output of the equalizer and equalizer tap energy. Margin to failure was determined by adding a measured quantity of white Gaussian noise to the signal until the bit error rate exceeded the 3×10^{-6} threshold.

Collected data were recorded on data sheets in addition to the data storage provided by personal computers in both the NTSC and HDTV locations in the field truck. The computers were used also to record field strength, determine minimum, maximum and median field strength from the 100-foot runs, preserve spectrum analyzer displays, and record both vector analyzer output and channel characterization data. The NTSC signals were recorded on a D3 videotape recorder for future reference if needed.

Supplementary testing during the transmission subsystem phase of the field testing program included UHF cochannel interference, reception on indoor antennas, and reception from a side-mounted, directional UHF antenna.

UHF cochannel interference testing was accomplished by use of a translator located approximately 27 miles (43 kilometers) from the main antenna. Both the main transmitter and the translator were capable of operating in both HDTV and NTSC modes. This made possible the testing of all three interference modes: HDTV to NTSC, NTSC to HDTV and HDTV to HDTV. For greater precision in establishing desired-to-undesired (D/U) ratios, laboratory measurements are more appropriate than field measurements, and such testing had been done at ATTC. The field measurements conducted near Charlotte found D/U ratios of the same magnitude as those determined at ATTC, thus providing support for the validity of the laboratory-determined ratios.

Indoor UHF reception was observed at twelve locations ranging from 4.6 miles (7.4 kilometers) to 24.1 miles (38.8 kilometers) from the transmitter. At each location, outdoor measurements and observations were made also to provide a comparison with the indoor results. Indoor HDTV reception at above the threshold for satisfactory service, and with adequate margin, was found at all twelve locations. Two of the locations required the use of ghost cancellation to obtain satisfactory reception of NTSC.

Tests of the performance of the side-mounted, directional UHF antenna were made at 13 locations on a circle approximately 16 miles (25.7 kilometers) from the antenna, with the greater number of selected locations in directions where the signal was both suppressed and passing through the supporting tower. At two sites, in the most suppressed parts of the directional pattern, satisfactory reception was found for neither NTSC nor HDTV. If the transmitter power were not reduced by 10 dB from the normal maximum, both of those sites would have been expected to receive satisfactory reception of both services. One of the sites showed acceptably low bit error and segment error rates but the margin was considered to be inadequate for satisfactory HDTV reception.

SPECIAL CONSIDERATIONS APPLICABLE TO CHANNEL 6

Upon commencement of the use of channel 6 in Charlotte, complaints were received from cable viewers citing interference to cable reception on channel 6. The station carried on channel 6 by Charlotte area cable systems is WCNC-TV, the NBC television outlet. An investigation by Cable Laboratories showed that some pickup was on the cable systems, some was direct pickup in receivers, but most was likely to be pickup on wiring provided by the viewers to connect a VCR or other device. As a consequence, the use of channel 6 had to be curtailed somewhat. The VHF sample included only 169 of the 199 sites used for UHF.

Channel 6 testing was impacted further by two other factors: impulse noise and interference, particularly from noncommercial FM stations. Power line impulse noise encountered at almost half the locations ranged from moderate to excessive. With respect to FM interference, the absence of an allotment for channel 6 in Charlotte has permitted the noncommercial FM stations to operate at relatively high power. The result was interference to channel 6 television reception. If both NTSC and HDTV had operated at a 10-dB higher power level, as expected for normal operation, neither the impulse noise nor the FM interference would have had as much impact.

TESTING OF THE HDTV SYSTEM PROTOTYPE

When the Grand Alliance HDTV full system prototype became available is 1995, the capability of the testing system was expanded to permit viewing of HDTV as well as NTSC pictures and listening to accompanying sound. Testing of the prototype was conducted in two phases. In the first phase, test locations used were a 40-location subset of the 199 locations used for the transmission subsystem testing plus ten indoor sites. The purposes of the Phase I tests were to provide a check of the validity of the laboratory test results under real world propagation conditions, and to determine whether the results of the 1994 tests on the transmission subsystem, using the bit error rate criterion, were a reliable measure of the system for terrestrial broadcasting. Both questions were answered in the affirmative.

Selection of the Phase I sites was based on Channel 53 performance during the transmission subsystem testing. The sites were selected from radial, grid and clusters on the basis of measured margins to the 3×10^{-6} bit error rate. Ten sites in the selection had margins of 20 dB or more, twenty-two sites had margins of 0 to 10 dB, and eight sites had no margin but were within 6 dB of the threshold. Originally, the plan called for fifteen sites in each of the last two categories, but only eight sites could be found in the third category so the number in the middle group was increased to preserve the objective of forty sites. Channel 6, as well as channel 53, testing was done at all forty of the sites selected by the foregoing criteria, but at two of the indoor sites, testing was confined to channel 53 because of complaints of interference to reception on cable channel 6.

NTSC peak visual ERP and HDTV average ERP were maintained at the same levels as used previously in the 1994 transmission subsystem testing. Procedures followed were similar to those of the earlier tests except that 100-foot runs were not repeated, impairment ratings for HDTV were noted as well as for NTSC, and to provide a viewing experience more like that to be experienced in a home, longer HDTV segments were run.

Impairment rating of the HDTV pictures used the same one-minute sequence that had been used in the laboratory. This was followed with a ten-minute segment of pictures and sound that had been recorded by ATTC from a variety of broadcast material. Commentary was provided relative to whether or not any imperfections in picture or sound were detected during the ten-minute viewing period of the HDTV material. Both NTSC and HDTV pictures and sound were recorded on a D3 machine. In addition to the subjective ratings of the HDTV transmissions, an objective measurement was made as described below.

The Reed-Solomon encoder used in the VSB transmission system inserts redundant bytes in each of the 626 segments constituting a data frame. Those redundant bytes are used to detect segment errors. The VSB prototype receiver includes a display panel providing continuously a showing of segment error rates (SER) in errors per second. During each ten-minute viewing period, an observer watched the display and recorded the number of measurable SER "hits." During steady state conditions, such as are present in the laboratory, a determination had been made that when SER reached 2.5 errors per second, the threshold of visibility was reached along with a bit error rate of 3×10^{-6}. In the field, experience showed that, in most instances of measurable SER hits, no picture or sound impairments were noted. The SER had to reach a count of approximately six before viewers noted impairments to picture or sound.

Phase II field testing was designed to obtain more data relative to HDTV performance under a variety of UHF propagation conditions and, particularly, in locations with substantial multipath. In addition, Phase II testing included visits to ten additional indoor sites, made additional observations of the performance of the side-mounted directional antenna, and explored the matter of interference to NTSC sound from an upper adjacent channel HDTV operation.

To expedite the Phase II testing when the Grand Alliance system prototype would be available, potential locations for testing were visited in the spring of 1995. The plan called for urban clusters of 25 locations each in the communities of Charlotte, Kannapolis and Lincolnton, North Carolina, and Rock Hill, South Carolina. Eighteen locations were selected also on each of two arcs at approximately 45 miles (72.4 kilometers) from the transmitter. The arcs, in the directions of southwest and west-northwest from the transmitter were selected because of the likelihood of encountering irregular terrain conducive to the introduction of multipath effects.

At each of the 136 locations selected as described above, 100-foot field strength measurement runs were made while transmitting NTSC on channel 53. Following the taking of the 100-foot run measurements, the field truck was returned to the central position of the run and observations and measurements such as have been described previously were made on the NTSC transmission.

From the analysis of data from the foregoing tests, 60 locations, 44 in urban clusters and 16 on arcs, were selected for complete testing with HDTV and NTSC. That work was carried out following the completion of the Phase I tests. Ten additional indoor locations were tested. Additional testing of the performance of the side-mounted directional UHF antenna was conducted at 19 locations on a 10-mile arc from the antenna location.

Laboratory testing had indicated that, in only one circumstance, interference to NTSC aural from HDTV may be more critical than to NTSC video. That circumstance occurs when the HDTV transmission is on the first adjacent channel above the NTSC transmission.

Field tests were conducted with NTSC broadcast on channel 52 and HDTV broadcast on channel 53. Output of a nominal one-kilowatt transmitter, operated at 253 watts average HDTV power, was fed to the top-mounted Dielectric antenna. The Comark transmitter output of 5.5 kilowatts was modulated with NTSC and fed to the side-mounted Andrew antenna. Taking into account transmission line losses and antenna gains, ERP for HDTV was 3.16 kilowatts and ERP for NTSC was 50 kilowatts. NTSC aural power was 13 dB down from peak visual power.

Ten measurements and observations were made on each of two arcs. One arc was at a distance of ten miles (16 kilometers). The second arc was at a distance of 20 miles (32 kilometers).

CONCLUSION

The field tests described herein yielded a substantial mass of data supporting the conclusions of the laboratory testing. In combination with the laboratory tests, the results made possible the recommendation by ACATS to the Federal Communications Commission that the Grand Alliance HDTV system, as documented by the Advanced Television Systems Committee, be approved for HDTV broadcasting in the United States.

SUMMARY OF GRAND ALLIANCE VSB TRANSMISSION SYSTEM FIELD TEST

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1. INTRODUCTION

The Grand Alliance, a consortium formed in May 1993 by AT&T, David Sarnoff Research Center, General Instrument Corporation, Massachusetts Institute of Technology, Philips Electronics North America Corporation, Thomson Consumer Electronics, and Zenith Electronics Corporation, proposed the use of the eight-level, vestigial sideband (8-VSB) transmission subsystem for Advanced Television (ATV) broadcast service after extensive comparative testing of 8-VSB and 32-QAM at the laboratories of the Advanced Television Test Center (ATTC) and CableLabs. A 16-VSB transmission subsystem was proposed for cable use. A description of the VSB transmission system can be found in References [1], [2], [3]. The Advisory Committee on Advanced Television Service (ACATS), an all-industry committee originally formed in 1987 to assist the FCC in establishing a new television service to replace the NTSC system currently in use, approved this transmission system recommendation as one component of the high definition television (HDTV) system under development.

In the spring of 1994, a prototype of the 8-VSB terrestrial transmission subsystem (data only) was delivered to the Advanced Television Field Test Site in Charlotte, N.C. for extensive field testing under real world propagation conditions under the sponsorship of the Field Testing Task Force of Systems Subcommittee Working Party 2 (SS/WP2). Additionally, a 16-VSB prototype was made available for testing as well. The absence of a complete system precluded the delivery of pictures and sound; therefore, a signal suitable for bit error rate (BER) measurements was transmitted. Immediately following the completion of final laboratory testing at ATTC in June of 1995, the complete Grand Alliance system (video and audio decoders included) was subsequently tested over-the-air and on cable during the summer of 1995, again in Charlotte, to provide verification of the previous year's subsystem testing. After successful completion of laboratory and field testing, the ACATS committee formally (and unanimously) recommended the Grand Alliance system to the FCC on November 28, 1995 as the next television system for the United States.

An important objective in both field tests was the determination of whether the ATV system would provide satisfactory or superior broadcast service where NTSC service is presently available. This paper will concentrate on both the 1994 and 1995 terrestrial broadcast test results. The 16-VSB digital transmission system was also successfully tested twice in Charlotte over eight technically diverse cable systems. The results of both terrestrial and cable testing are included in References [4], [5], and [6].

The primary purpose of both field tests (1994 and 1995) was to verify the laboratory test results and evaluate the performance of the transmission subsystem under real world conditions, which include multipath, co-channel, adjacent channel, and impulse noise interferences. The 1995 complete system test also provided a verification of the 1994 transmission subsystem test results, which used a BER of $3\text{x}10^{-6}$ as a criterion for threshold performance. A vital part of this field evaluation was a direct performance comparison of the ATV system to that of the 50 year old NTSC system currently being used in the United States. ATV and NTSC system performance was evaluated at a large number of test sites surrounding Charlotte to see if the ATV system provided the same or better reception compared to NTSC. Unlike the analog NTSC system performance which degrades gradually with decreased signal strength or increased interference, the digital ATV system remains essentially error-free above the threshold of visibility/audibility, producing near replica versions of the transmitted video and audio. Only at the threshold of visibility (error threshold), does the signal rapidly degrade and become unusable.

The Charlotte facility (tower and building) and field test operations were managed and staffed by the Public Broadcasting Service (PBS) with direction and data analysis from The Association for Maximum Service Television, Inc. (MSTV), and Cable Television Laboratories (CableLabs). The tests were conducted by the Advanced Television Field Test Project staff, with support from the Grand Alliance in the form of equipment and personnel.

Two field test reports have been issued by the Field Testing Task Force group for the SS/WP-2 working party: one on September 16, 1994 (SSWP2-1354)[4], describing the transmission subsystem (BER) tests, and another on

October 16, 1995 [5], describing the complete Grand Alliance system (video and audio) tests. A complete report on ATTC laboratory testing, ATEL video/audio testing, the 1995 Charlotte terrestrial field testing (1994 test results are not included), and CableLabs cable testing was issued in October 1995 [6].

2. FIELD TEST PLAN

The ACATS System Subcommittee Working Party Two (SS/WP2) established a Field Testing Task Force to develop field test procedures, and report back to the SS/WP2 parent body with the test results and analysis. The following describes the test plan.

The field test was conducted in and around the city of Charlotte, N.C., a city of about 400,000 people. The Charlotte test site was selected due to the varied terrain surrounding the city since the objective was to compare the ATV and NTSC transmission systems under widely varying propagation conditions. The terrain around Charlotte varies from relatively smooth earth to extreme irregularity, providing many obstructed receiving points that have no line of sight to the transmitter. Also considered in the field test plan were test sites in urban, suburban, and rural settings, all of which were found in the Charlotte area. Additionally, there were eight diverse cable systems in the area that provided a good testbed for the 16-VSB cable tests.

It was deemed necessary to assess both the VHF and UHF performance of the 8-VSB system, so the Federal Communications Commission (FCC) granted experimental authority to operate on VHF channel 6 and UHF channel 53. This was based on a study indicating that the use of these channels presented little chance of causing interference to, or receiving substantial interference from, any existing television broadcast operations in this area.

The field test transmitter site, located about 10 miles northeast of downtown Charlotte, had a 1,337 foot tower next to the transmitter building which housed a Channel 53 UHF transmitter with IOT output stage, and a channel 6 solid-state VHF transmitter. Zenith Electronics Corporation provided the Grand Alliance 8-VSB ATV modulator. A pseudo-random bit stream was transmitted during the transmission subsystem testing (1994) while a D3 video tape machine played a pre-recorded tape containing a compressed video/audio/ancillary bit stream during the complete system testing (1995). At the top of the tower was an omnidirectional, top-mounted UHF antenna. Just below the UHF antenna was a single-bay, omnidirectional VHF panel. A sidemounted directional UHF antenna was also present on the tower.

The main over-the-air broadcast tests were conducted at 199 sites surrounding Charlotte. Sites were selected from eight radials (128 sites) extending out to 55 miles, two 5x5 one-mile-spaced grids (urban Charlotte and suburban Rock Hill), two half-mile-spaced clusters (16 sites) contained within each grid, and one group of 5 sites only 2.5 miles from the transmitter. The radial sites provided varying terrain conditions and signal strengths. The 66 urban and suburban test sites typically had strong or moderate signal strengths, respectively, and provided variations in signal levels usually attributed to shadowing and reflections from foreground terrain and clutter near the receiving antenna location. The group of 5 sites near the transmitter provided information on the performance of the antenna at relatively large depression angles.

Testing at the Charlotte site was done at an NTSC transmitter power 10 dB below (one-tenth) the maximum FCC-permitted power for NTSC operations in order to avoid interference to existing NTSC stations, and to limit the need for a very expensive high power (> 60 kW) UHF transmitter. To provide the appropriate comparison of ATV and NTSC reception, the average ATV effective radiated power (ERP) was selected to be 12 dB below (approximately one-sixteenth) the NTSC peak visual ERP. The value of 12 dB had been previously determined by Working Party Three of the Planning Subcommittee (PS/WP-3) to provide an equivalence in ATV and NTSC coverage and service area based on planning factors derived from ATTC laboratory test results. NTSC and ATV power definitions and measurement techniques are described in Reference [7]. NTSC coverage is determined by Grade B contour (C/N≈34 dB/6 MHz) at a 55 mile radius for UHF while ATV coverage is determined by the noise-limited contour (C/N≈15 dB/6 MHz), also at a 55 mile radius for UHF. Figure 1 illustrates this equivalent coverage concept for UHF.

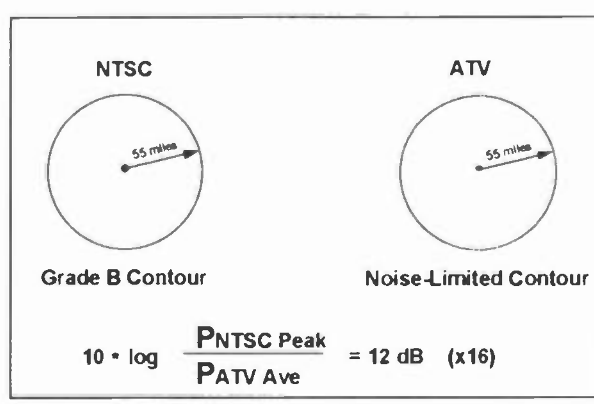

Figure 1 Equivalent UHF coverage between NTSC and ATV

The 10 dB reduction from maximum allowable transmitter power limits the total service for both ATV and NTSC; however, by keeping the 12 dB relationship between the two transmitted signals, a valid comparison was made of the ATV system relative to NTSC. If both NTSC and ATV power were increased 10 dB (10 times) to the maximum FCC-allowed power for a full power station, as is expected in practice, the useful service area would increase for both services, and interfering sources would be substantially less effective in producing impairments. Throughout the field tests, the NTSC peak sync ERP was 10 kW (instead of 100 kW) on VHF channel 6 and 500 kW (instead of 5 MW) on UHF channel 53. In keeping with the 12 dB (1/16) ratio, the ATV average ERP was 0.63 kW on VHF channel 6 and 31.6 kW on UHF channel 53. Figure 2 and Figure 3 contain spectral plots of the transmitted VHF and UHF spectrum, respectively, typical for transmitters with at least 6 dB of backoff (greater than ATV peak/average ratio) and without non-linear compensation.

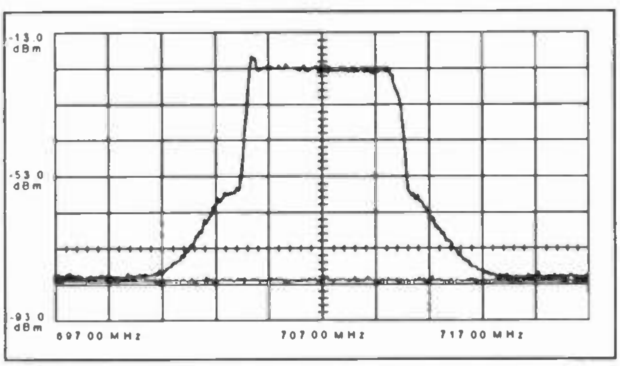

Figure 2 UHF channel 53 transmitted spectrum

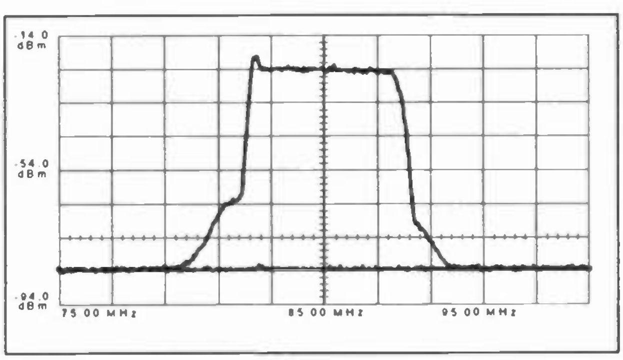

Figure 3 VHF channel 6 transmitted spectrum

The field test truck that transported the NTSC and ATV hardware around the Charlotte area was equipped with a 15 kW generator and a 30 foot extendible mast with rotor control of an all-band consumer antenna. The NTSC signals were viewed on NTSC television monitors and rated on a CCIR 5-point impairment scale for a 1-minute live-action sequence. The audio was also evaluated using expert observation and comments (EO&C). Acceptable NTSC reception was deemed to be a rating of CCIR grade 3 or higher. The ATV signal was evaluated with HDTV monitors and audio equipment. ATV margin (how much the signal level could degrade towards white noise before data errors occurred) was determined, as well as certain transmission characteristics derived from the ATV receiver's equalizer. Both objective bit error rate (BER) measurements (1-minute duration) and subjective video and audio evaluations (10-minute duration) were made. Acceptable ATV performance was deemed to be a BER of less than $3x10^{-6}$ (equivalent to 2.5 segment errors/second), during the 1-minute sequence, or 5 or less noticeable video/audio impairments during the 10-minute sequence. The longer subjective viewing time of 10-minutes provided enough confidence in the measurement of site margin during periods of interference from non-stationary impulse noise (VHF) or complex, time-varying multipath (UHF). Since the digital ATV signal does not experience gradual degradation with reduced signal level, but rather has a sharp transition from perfect to unusable ("cliff effect"), it is important to determine that there is enough margin in the ATV signal at a given site.

During the Charlotte field test, it became apparent that unanticipated results were obtained from the use of channel 6. First, although the FCC had granted authority to use both UHF channel 53 and VHF channel 6, interference from the broadcast CH 6 (primarily NTSC) into a local cable system curtailed testing at a minority of sites. This is why CH 53 had 199 test sites, but CH 6 had only 169 test sites. A study by CableLabs indicated that there was some pickup on the cable system itself, some pickup with the television receiver, and, most commonly, pickup on the householder's own cabling. The interference could be eliminated in many instances by using improved household cabling or installing a set top converter (using a fixed RF channel 3 or 4 output).

Another surprise observed on channel 6 was that almost half the test sites had significant impulse noise that had varying characteristics, ranging from fairly broad pulses to narrow spikes. All the observed impulse noise demonstrated the 120-Hz repetition characteristic of the positive and negative peaks of a 60-Hz power line signal. The fact that the transmitted signals were 10 dB below maximum allowable power may have contributed to this atypical noise problem in Charlotte.

Finally, with no regularly assigned channel 6 television broadcast station in the Charlotte vicinity, there are no restrictions to placing noncommercial, educational FM radio broadcast stations in the 88.1 - 91.9 MHz band.

Strong adjacent interferors had to be attenuated in the truck to protect the antenna preamplifier from overload, thus penalizing channel 6 (82-88 MHz) performance, particularly NTSC. Consequently, at a few locations, adjacent channel interference was sufficiently high that channel 6 parameters (particularly NTSC) could not be adequately measured due to antenna preamplifier overload.

3. ANALYSIS OF TEST RESULTS

A total of 199 sites were visited in the 1994 Charlotte field test. UHF was tested at all 199 sites, but only 169 sites had VHF measurements taken due to the leakage into a local cable system. Test site distances from the transmitter varied from 1.8 miles to 55 miles, and signal strength varied from strong to moderate to weak.

3.1 OVERALL ATV AND NTSC SERVICE AVAILABILITY

In order to compare analog NTSC service availability with that of digital ATV, the NTSC data from all the sites (radials, grids, clusters) was grouped into two categories: those that had a NTSC rating of CCIR grade 3 or greater (satisfactory), and those that did not (unsatisfactory). To assess the overall service availability of the ATV system, the site results were also categorized into two groups: those below the $3x10^{-6}$ BER threshold of visibility (satisfactory), and those above the $3x10^{-6}$ BER threshold of visibility (unsatisfactory). Table 1 contains the percentages of locations that provided satisfactory reception for NTSC and ATV.

Service Performance	NTSC CH 6 169 Sites (%)	ATV CH 6 169 Sites (%)	NTSC CH 53 199 Sites (%)	ATV CH 53 199 Sites (%)
% of Locations With Satisfactory Reception	39.6	81.7	76.4	91.5

Table 1 Relative performance of NTSC and ATV

The statistics in Table 1 reveal that NTSC UHF channel 53 picture performance was significantly better than VHF channel 6. The median NTSC rating was CCIR grade 4.0 for UHF and only 2.5 for VHF. This was attributable to the prevalence of picture impairments due to high impulse noise, co-channel interference from distant stations and cable leakage, and/or the interference of adjacent channel non-commercial (educational) FM stations within the VHF service area. It should also be remembered that the tests were performed with the NTSC and ATV transmitters 10 dB below maximum FCC-permitted power levels.

As was the case for NTSC, the overall ATV service availability was higher for UHF channel 53 than for VHF channel 6. However, the difference was not nearly so significant. The reasons for these unexpected VHF channel 6 ATV results are believed to be the same as described above for NTSC.

In comparing ATV with NTSC, the statistics on service performance in Table 1 also indicate that the ATV system performed much better than NTSC on UHF channel 53, and substantially better than NTSC on VHF channel 6. On channel 53, there were almost three times as many sites that had unsatisfactory NTSC reception compared to ATV. On channel 6, almost twice as many sites had satisfactory ATV reception compared to NTSC. When the test site data was grouped into 10 mile distance increments to compare service availability at various signal level conditions (strong, moderate, weak), ATV was found to outperform NTSC in every distance group, for both VHF and UHF. As the distance from the transmitter increased, the difference between ATV and NTSC became greater, and since the data included both the urban and suburban grids and cluster sites, the ATV system was shown to readily handle signal level variations due to location variability on both VHF and UHF.

White noise margin is an important parameter for ATV reception that measures how much a signal level can drop before the picture and sound are lost to the viewer. The limiting factor is the white noise floor of the ATV receiver. Since signal level varies with location, time of day, seasons, and weather factors, margin is a good indicator of service reliability. ATV margin is examined below as a function of NTSC CCIR impairment, distance from the transmitter, and field strength.

To determine ATV margin statistics compared with various NTSC CCIR impairment ratings, those sites that had satisfactory ATV reception (BER < $3.0x10^{-6}$ and margins > 0) were grouped according to their corresponding NTSC CCIR impairment ratings at the same locations. Median margins were determined for each CCIR rating category, as shown in Table 2 and Table 3 .

At the just acceptable NTSC sites (CCIR grade = 3), ATV had a median margin of over 25 dB on VHF, and about 20 dB on UHF. These values, which include the urban and suburban grid test sites, demonstrate sufficient ATV margin for signal variability at locations where NTSC reception was only marginal. Also of interest is the fact that even for NTSC CCIR ratings of less than 3, there was still ATV margin available for signal fading, indicating that the

ATV signal does not have its "cliff effect" just beyond the NTSC limit of service, but rather extends much farther out than the NTSC service area.

NTSC CCIR Grade Impairment Rating	# of Test Sites	Satisfactory ATV Sites (%)	Median of ATV Margin (dB)
5 Imperceptible	4	100	43.7
4 Perceptible, but not annoying	22	100	38.2
3 Slightly annoying	41	100	25.6
2 Annoying	46	93	16.9
1 Very Annoying	56	50	15.0

Table 2 ATV margin statistics for CH 6 (169 sites)

NTSC CCIR Grade Impairment Rating	# of Test Sites	Satisfactory ATV Sites (%)	Median of ATV Margin (dB)
5 Imperceptible	7	100	41.2
4 Perceptible, but not annoying	104	100	32.4
3 Slightly annoying	41	98	19.9
2 Annoying	29	90	7.7
1 Very Annoying	18	28	19.7

Table 3 ATV margin statistics for CH 53 (199 sites)

In summary, both NTSC and ATV service availability was discovered to be better on UHF than on VHF (primarily due to impulse noise and NTSC co-channel interference on VHF), and digital ATV service availability was found to be better than that of analog NTSC. The conclusion is that ATV service would be available everywhere NTSC reception is presently acceptable, and at many locations where NTSC reception is unacceptable.

3.2 ATV SYSTEM PERFORMANCE

This section describes measurements specifically relevant to digital transmission system design. Various data was collected at each site to evaluate service availability and also to determine how well the ATV system performed under various interference and impairment conditions, including co-channel, adjacent channel, impulse noise, and multipath. The amount of ATV margin present at each test site was also tabulated.

3.2.1 ATV Service Availability

Since the tests were performed at power levels 10 dB below maximally allowed by the FCC, the field test plan allowed for an increase in ATV power by 6 dB at those locations where there was unsatisfactory reception. Only ATV levels were increased. This increase in power was not done for NTSC since it was already at the maximum allowed power for the Charlotte test, limited by both transmitter hardware and existing co-channel stations. However, with the nominal average ATV transmitter power 12 dB below NTSC peak power, a +6 dB increase in ATV power was possible.

Figure 4 and Figure 5 illustrate the change in service availability versus distance due to the 6 dB increase in ATV transmitted power. In comparing channel 6 with channel 53 at these increased power levels, one can see that the service areas for UHF and VHF are identical (94%). By transmitting at the full maximum allowable power (10 dB above where the Charlotte tests were run, rather than 6 dB), the VHF and UHF service areas would remain identical, but both with a slightly larger service availability (i.e., > 94%). This evidence leads one to believe that both VHF and UHF can be reliably used for ATV transmission.

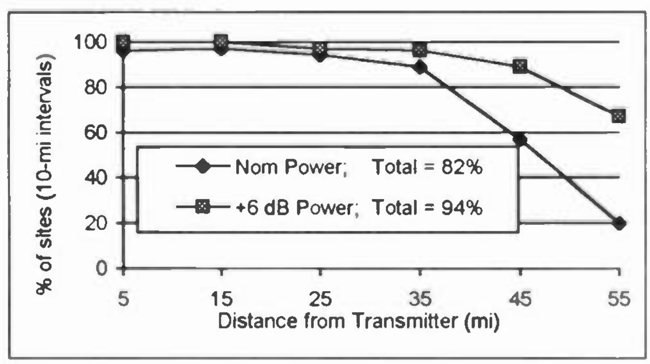

Figure 4 ATV service performance for Ch 6 (169 sites)

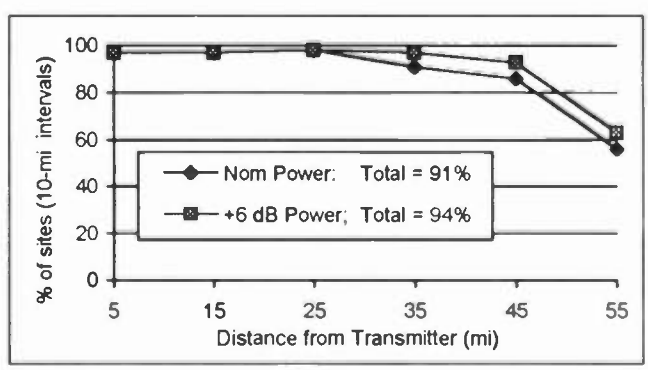

Figure 5 ATV service performance for Ch 53 (199 sites)

3.2.2 ATV Margin

ATV margin was tabulated at all test sites, and grouped into 10-mile increments. As expected, the median value of ATV margin decreased with distance from the transmitter as the signal level progressively decreased from strong to moderate to weak.

At the 50-55 mile interval, there was 10 dB of margin (median value) on VHF and 14 dB on UHF. This is a significant result, especially in light of the 10 dB lower than maximum allowable transmitter power levels. The impulse noise, cochannel, and adjacent channel FM station interference were again the probable cause for the slightly poorer performance on channel 6. Margin in the urban and suburban grids was also good. The median margins for the urban Charlotte (≈10 miles) and suburban Rock Hill (≈30 miles) grids was about 30 dB and 20 dB, respectively, indicating that sufficient margin exists to handle location-to-location signal level variations.

ATV margin was found to be reasonably predictable in white-noise-limited areas by using field strength. Margin is proportional to field strength since the threshold of error-free operation is dependent on the received signal level being about 15 dB above the receiver noise floor (in a 6 MHz bandwidth). Using agreed upon planning factors (see Reference 2), the measured threshold of error-free operation was found to agree with the calculated values of 27.4 dBuv/m for low VHF and 43.4 dBuv/m for UHF. Of course, any other interference (e.g. impulse noise, co-channel, multipath) present in addition to white noise will affect the actual threshold.

3.2.3 VHF Impulse Noise

Power line impulse noise was prevalent on the VHF channel 6 band in Charlotte. The field test expert observers characterized 46% of the 169 test sites as having significant impulse noise present in 1994. Similar results were obtained during the 1995 field test. Some of the radials were fairly clean, others had impulse noise only at the ends of the radials where signal levels were low, and some radials had significant impulse noise throughout. The fact that the transmitted signal was 10 dB below maximum allowable power contributed to the widespread occurrence of substantial impulse noise.

The character of the impulse noise was <u>not</u> consistent throughout all the test sites. Figure 6 illustrates a site where the impulse noise consisted of broad pulses within the band, while Figure 7 depicts a site with large, narrow spikes. Both these figures were recorded from the on-board spectrum analyzer. All the impulse noise was 60/120 Hz related, often appearing to come from power lines in the vicinity, but was often not confined to the typical two narrow bands rolling through the NTSC picture. Despite this severe impulse noise, over 80% of the sites had acceptable ATV reception, and 94% had acceptable reception when the ATV transmitter power was increased by 6 dB.

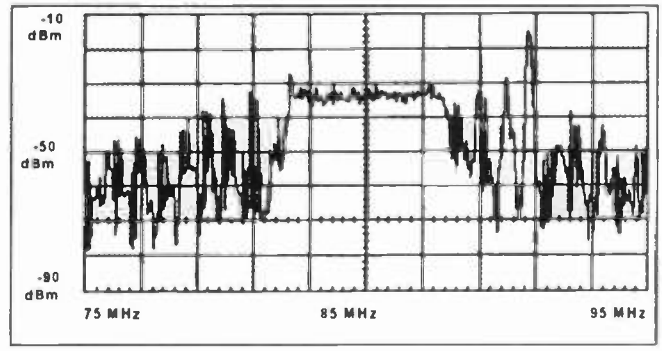

Figure 7 Ch 6 ATV signal with narrow impulse spikes

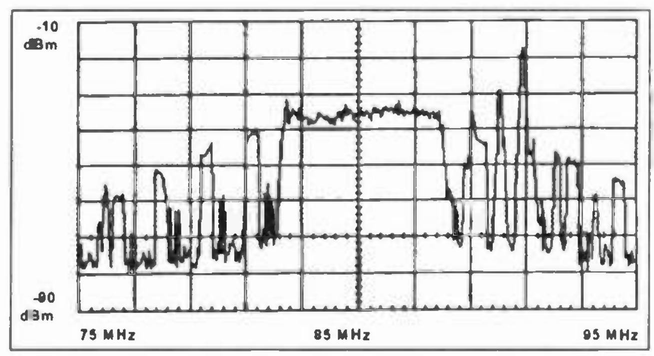

Figure 6 Ch 6 ATV signal with broad impulse spikes.

3.2.4 UHF Complex Multipath

Most of the multipath observed in Charlotte was dispersed rather than discrete. Discrete ghost are often associated with multipath conditions of large, flat surfaces, such as tall steel buildings. However, multipath conditions can also occur when there are hills and dense foliage in fringe areas, and the signal can travel through many paths to the receiver. These complex ghosts require the receiver's equalizer to "work hard" to reduce them to acceptable levels, and often have a time-varying (dynamic) nature to them, especially at UHF frequencies where the wavelengths are shorter. It is the dynamic behavior of the multipath that is the most challenging for the equalizer.

The ATV receiver's equalizer (ghost canceler) provided an indication in the form of tap energy of the amount of channel distortion that was being corrected. The larger the tap energy (the total energy in all equalizer taps other than the main tap, compared to the energy in the main tap), the larger the amount of multipath. Both channel 6 and 53 exhibited multipath, but channel 53 had the most complex multipath (50% of the sites). Some of the sites had dynamic multipath with a periodicity of less than a minute. A vast majority of the echo time spreads observed in Charlotte were less than 10 usecs. Results indicate that the multipath conditions in Charlotte were well within the performance range of the ATV prototype equalizer, even at sites that had complex multipath equal to 60% ghosting. Figure 8 illustrates one case of UHF complex multipath.

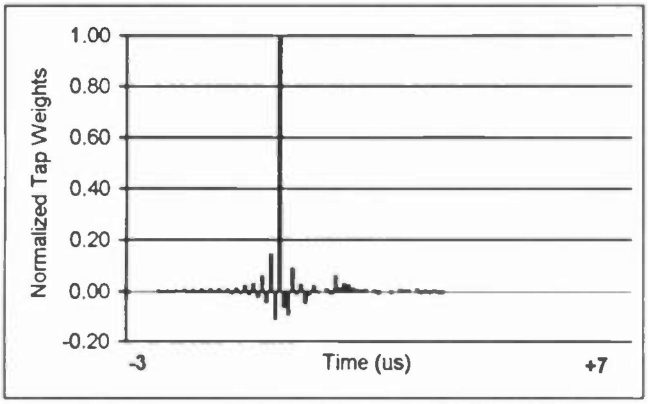

Figure 8 Example of equalizer taps for UHF multipath

Another method of trying to characterize multipath at a given test site was to move the field test truck, with antenna mast extended to 30 feet, over a distance of 100 feet while measuring the NTSC signal level (picture carrier) with a narrowband field strength meter. If multipath was present, the signal level would vary as all the additional multipath signals would change phase, and add or subtract accordingly.

Figure 9 shows the results of one 100 foot run on channel 53 that had moderate ghosting. The corresponding 100 foot run on channel 6 at the <u>same</u> site showed minimal VHF multipath. The disparity between the two was probably due to the very different wavelengths between the VHF and UHF channels. ATV had satisfactory reception at this site while NTSC did not. Figure 10 illustrates the opposite case

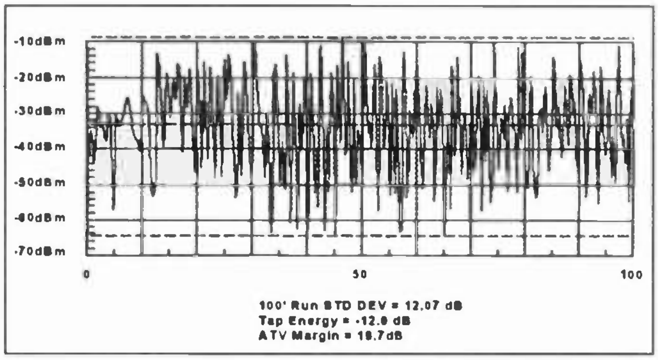

Figure 9 Ch 53 100 foot signal level run

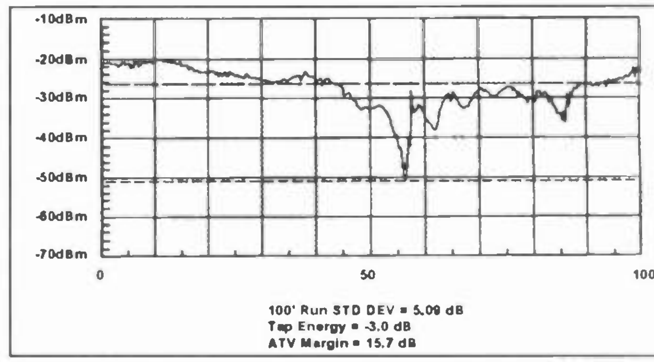

Figure 10 Ch 6 100 foot signal level run

at another site where the VHF signal was found to have more multipath than the UHF signal. Signal variations in the 100 foot runs were found to correlate with tap weight multipath indications, and are useful in predicting the presence of multipath.

3.2.5 NTSC Co-Channel Interference

NTSC co-channel signals of measurable level were recorded at 95% of the test locations on channel 6, and at only 15% of the locations on channel 53. On VHF, about 25% of the locations exhibited D/U ratios less than 20 dB for ATV. Ratios less than 20 dB would cause noticeable interference if the desired station was NTSC rather than ATV. However, the ATV receiver minimizes the effect of NTSC co-channel interference through its NTSC rejection filter, and therefore operates in strong NTSC co-channel conditions, e.g. 2 dB D/U ratios or worse.

Channel 6 in Charlotte has two existing co-channel stations: Wilmington, NC to the east and Augusta, GA to the southwest. However, these two distant stations could <u>not</u> account for 95% of the locations having measurable co-channel. The distribution of the NTSC co-channel interference in the coverage area was found to be weak to moderate and atypical for distant station interference. It is believed that much of the co-channel interference was due to cable television system egress leakage corresponding to the known cable ingress situation in the Charlotte area.

Figure 11 shows an example of a distant NTSC station's signal rising above the flat ATV spectrum. The ATV signal experienced a D/U ratio of 0 dB at this site and yet provided a satisfactory ATV picture. Given the observed co-channel interference distribution, the 10 dB reduction in transmitter power from that of a full power station, and the 82% satisfactory reception statistics on channel 6, the conclusion is that the ATV system performed superbly in the presence of NTSC co-channel interference.

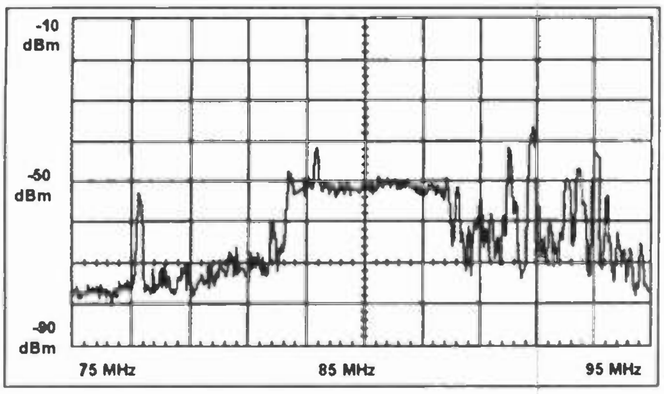

Figure 11 Ch 6 ATV spectrum with co-channel and adjacent channel present.

Since there are no existing full power channel 53 stations close to Charlotte, additional co-channel tests were performed on UHF channel 53 by transmitting a signal from a site about 27 miles northeast of the Charlotte transmitter. Since both the main transmitter and the smaller test transmitter could be operated in either NTSC or ATV mode, three types of tests were performed: ATV into NTSC, NTSC into ATV, and ATV into ATV. By adjusting the transmitted powers of the main and interfering signals, the various thresholds could be determined. For NTSC, the threshold of visibility (TOV) in the desired NTSC picture was used, while a BER of 3.0 $\times 10^{-6}$ was used for the desired ATV signal.

The UHF cochannel test results were: ATV into NTSC D/U ratio at TOV was about 50 dB, NTSC into ATV D/U ratio averaged about 2 dB, and ATV into ATV D/U ratio averaged approximately 16 dB. All of these results confirmed laboratory testing at the ATTC, and support the validity of using laboratory results for the development of planning factors.

3.2.6 Adjacent Channel Interference

Figure 11 illustrates a lower adjacent channel interference signal experienced in the field on VHF. The NTSC channel 5 peak sync power was measured at 10 dB below the average ATV signal power, and produced no impairment effects. Also shown in the spectrum plot are upper adjacent non-commercial FM station signals that are larger than the ATV signal, and also caused no detrimental effects on ATV reception.

An ATV/NTSC upper adjacent channel test was performed during the Charlotte tests by generating an NTSC signal on channel 52 and an ATV signal on channel 53 whose average power was 12 dB below the peak NTSC sync power. This is the typical method that will be used in the future for co-located adjacent channel transmissions.

Due to the spare UHF transmitter power limitations, both signals were transmitted at a lower power (4 kW peak NTSC and 250 watts average ATV). Locations on two arcs were visited (10 and 20 miles), each with 10 test sites. Two NTSC sets were used in the field test truck for NTSC reception, one from the ATTC laboratory having been previously measured as a "median set" in upper adjacent channel interference tests.

At the nominal 12 dB power ratio between peak NTSC sync and average ATV, there was no video or audio interference detected by the expert observers. The most sensitive NTSC set was found to have a threshold of visibility at received D/U ratios of about +4 dB (median), while the other set was observed to have a threshold of visibility at about -1 dB (median). Threshold of audibility was better than -10 dB D/U ratios, and often could not be measured. During all these tests, the ATV signal was error-free, and not affected by the larger adjacent channel NTSC. An example of the received nominal NTSC and ATV spectrums is shown in Figure 12.

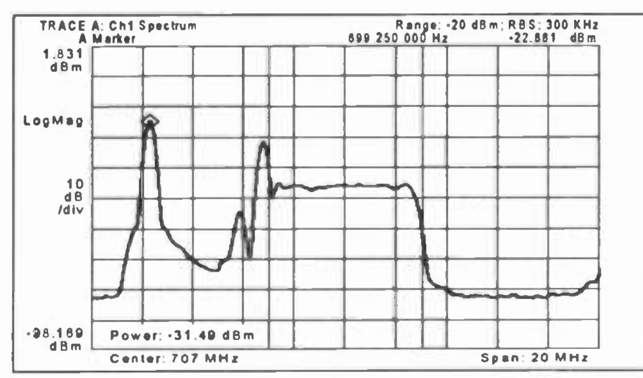

Figure 12 ATV-to-NTSC (12 dB ratio) adjacent channel UHF spectrum

3.2.7 Supplemental RF Tests

Supplemental tests were conducted during both the Charlotte field trials. Considering that broadcasters likely will use sidemounted, directional antennas in the transmission of HDTV, tests were performed using such an antenna to verify acceptable performance around the entire antenna pattern, particularly the back side near the nulls. The channel 53 UHF sidemounted antenna was mounted just below the top-mounted omni-directional antennas used for the main channel 6 and channel 53 tests. It has a typical cardioid radiation pattern. The NTSC peak visual ERP was 500 kW, and the ATV average power was 12 dB below NTSC peak visual. Measurements were made at a total of 32 sites, most of which were on the "backside" of the antenna radiation pattern. At the sites where NTSC reception was observed to be acceptable, the ATV signal reception was also acceptable. The results indicate that ATV transmission using directional transmitting antennas should have no problems.

Additional field tests were conducted in Charlotte to verify that indoor reception of channel 6 and 53 is feasible in areas of reasonably strong signals. After receiving permission from 32 homeowners in the Charlotte area to conduct tests within and outside their homes, NTSC and ATV tests were performed employing a standard VHF "rabbit ears" antenna and a standard UHF "bow tie" antenna, each supplied by the field test project to provide uniformity of observation. The effects of persons moving in the room in the vicinity of the receiving antenna was also noted. The received in-door signals were amplified and sent over 200 feet of quad-shielded coaxial cable to the field test truck for processing and display. These homes varied in distance from the transmitter from 1.2 miles to 25.6 miles.

Satisfactory ATV reception with adequate margin was obtained at all but two of the 32 test sites where NTSC was rated a CCIR 3 or better. Some of the locations had less than acceptable NTSC pictures using the indoor bow tie antenna, while the ATV signal had 10 dB or more of margin. The range of ATV margins, varying from 7.8 to 33.5 dB, was adequate to cover signal variations that might occur over time. The two sites that had marginal ATV and NTSC (CCIR 3) signal coverage had severe time-varying multipath within the home (not observed at the street location using the standard outdoor antenna at a height of 30 feet off the ground). Short ghosts that cause a spectral null in-between the picture and chroma/sound carriers of the NTSC signal do not degrade the picture (or sound) as severely as one might expect. However, the flat spectrum ATV signal is affected by nulls anywhere in the band. It should also be pointed out that it was the dynamic aspect of the severe multipath that caused difficulty in reception. Due to the severity of the multipath (20-30 dB in-band notches), any repositioning of the antenna to another position in the room would cause the null to land near one of the NTSC carriers rendering NTSC picture or sound unacceptable or unusable.

At one of the two marginal ATV sites, a brief experiment with an additional indoor antenna was performed. This dual bow tie antenna with a rear screen reflector, bought at local electronic retailer for $17, was found to greatly decrease the multipath in the home due to its directivity and additional gain (+6 dB). It was a vast improvement to indoor reception for both NTSC and ATV, and reduced sensitivity to both antenna positioning and adjustment. ATV was found to have significant margin over NTSC for this unique type of multipath situation. A new indoor antenna design, currently being evaluated by the NAB, also has promise of much improved television reception within the home.

3.2.8 ATV Video and Audio Performance

The 1994 tests, using bit error rate as the criteria for satisfactory ATV reception, provided the statistical base for the VSB system analysis. As described earlier, the 1995 field tests were performed with the complete Grand Alliance system (video and audio included). The main purpose of the test was to verify the 1994 BER tests, and gather video and audio performance data at marginal reception sites. It was for this last reason that the selected 40 sites in the 1995 test included a majority (75%) of sites that had little or no noise margin. No statistical service availability data was meant to be obtained from the 1995 data.

The use of both a one minute objective measurement period and a 10-minute subjective measurement period during the complete system tests provided not only a good verification of the previous year's BER testing, but also a better understanding of dynamic signal conditions often encountered in the field. It was determined that if no or few (5 or less) ATV impairments (measurable segment errors, or "SER hits") occurred during a 10-minute observation period, there was a strong probability that the site was acceptable for long term viewing. Subjective evaluation of the ATV pictures and sound in the field has shown that occasional, low value SER hits were often masked by the video and audio decoders well enough that no video or audio impairments were observed. Field observations also indicated that major ATV video and audio impairments do appear to correlate, and that loss of satisfactory sound before picture does not occur, which is similar to the NTSC experience. In some instances, minor visual impairments occurred with no accompanying audio impairments.

From the Charlotte field tests, an important guide to the reliability of ATV signal reception is provided by the recorded receiver equalizer tap energy in conjunction with the margin-to-failure value. High tap energy, which indicates the presence of strong multipath, coupled with low margin indicates the possibility of inconsistent long-term ATV reception. On the other hand, even if the margin is not high, if little multipath exists, reliable ATV reception is likely. Strong multipath indicates the possibility of dynamic multipath, which is often a limiting factor for good ATV reception, especially on UHF.

The results from the complete system test again showed ATV outperforming NTSC in both the VHF and UHF bands. As expected, NTSC fared much worse than ATV on channel 6 for the same reasons of impulse noise and co-channel interference. The 10-minute ATV subjective test results were affected more than the 1-minute objective test results by the presence of VHF impulse noise rather than by complex UHF multipath. The VHF impulse noise interference is extremely non-stationary, being episodic in bursts having intervals of occurrence much larger than one minute, but still with the 10-minute testing time. On the other hand, the UHF complex dynamic multipath that was experienced in Charlotte was episodic in time intervals less than one minute, which means that if dynamic multipath existed at a given site, it was surely going to be observed within the one minute testing period as well as the 10-minute testing interval. The effects of impulse noise on the ATV signal on VHF would be significantly lessened with another 10 dB of transmitter power.

The 1995 complete system findings suggest that the results of the 1994 transmission subsystem tests, using bit error rate (BER) as the criterion of performance, were a reliable measure of UHF statistical performance. On VHF, the results are not as conclusive due to the combination of limited sample size (33 sites), reduced transmitter power (-10 dB), and severe impulse noise (in the Charlotte area). More VHF testing and impulse noise evaluation in other cities is necessary for absolute confirmation. However, there is strong evidence from the 1994 tests, using the +6 dB (not even 10 dB) increased transmitter power results, that impulse noise can be essentially overcome with the maximum allowed transmitter power. Also, the successful use of VHF channels 3 and 9 in Charlotte for commercial NTSC broadcast, coupled with the test results of improved service availability of ATV over NTSC, provides compelling evidence for successful ATV transmission on VHF channels.

4. CONCLUSION

The Charlotte field tests verified that the VSB transmission system had better service availability than NTSC, even when transmitted with its average power 12 dB below NTSC peak power. ATV outperformed NTSC at every distance group tested, especially at worst case sites. While UHF reception was better than VHF for both NTSC and ATV due to the reduced (10 dB) transmitter power, the use of a full power VHF station will allow comparable performance to UHF. Even with 10 dB reduced transmitter power, there was more than enough margin at marginal NTSC reception sites and at sites greater than 50 miles from the transmitter. This also confirmed that the calculated 12 dB ATV/NTSC power ratio for equivalent coverage is a conservative value with safety margin built-in. Significant NTSC co-channel and adjacent channel interference into ATV were shown to be surmountable during the tests. Also, the use of sidemounted directional transmitter antennas and indoor receiver antennas was verified.

The complete system test verified that the previous year's BER tests were valid, and that BER test results can be used for proper statistical analysis of service availability. ATV video and audio impairments were found to correlate, and for minor data errors, were often not objectionable or even noticed.

Finally, the 8-VSB digital transmission system performed significantly better than today's analog NTSC system, reliably working in real-world conditions such as complex multipath and impulse noise. The limiting factors in field operations is not white noise, but rather severe dynamic multipath on UHF and severe impulse noise on VHF. After thorough laboratory and field testing, the results indicate that ATV service will be available where NTSC service is presently available, and in many instances where NTSC service is unavailable. With testing complete, and unanimous ACATS Blue Ribbon Committee approval to the FCC, the Grand Alliance 8-VSB system is now ready for implementation as this nation's new television transmission standard.

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SUBJECTIVE EVALUATION OF THE DIGITAL HDTV GRAND ALLIANCE SYSTEM

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Abstract:

This paper describes the background, methodology, procedures and results of the subjective evaluation of the digital HDTV Grand Alliance system. Tests were conducted by the Advanced Television Evaluation Laboratory of the Communications Research Centre in Ottawa, Canada. The overall quality of the Grand Alliance System in both interlaced and progressive modes was examined. Additional tests were conducted to assess the ability of the system to co-exist with NTSC and to support scan conversion at the receiver. In all cases the system met or exceeded the specified requirements.

Introduction:

The Advisory Committee on Advanced Television Service (ACATS) was formed in 1987 to advise the Federal Communications Commission (FCC) on various aspects of advanced television. In 1989 in response to a call from the ACATS for qualified laboratories to carry out its advanced television test program the Advanced Television Evaluation Laboratory (ATEL), a facility of the Communications Research Centre (CRC) in the Department of Industry, Government of Canada offered to undertake the video subjective test program. The substance of this offer was to conduct video subjective tests of ATV systems using pre-recorded digital tapes to illustrate the performance of Systems in simulated terrestrial and cable broadcast environments. This offer was accepted by the Advisory Committee in 1990.

A first round of subjective tests was completed at the ATEL in 1992, in collaboration the Advanced Television Test Center (ATTC) in Alexandria VA and the CableLabs facility at ATTC, to evaluate the performance of six candidate systems. Of these, two systems used analog transmission (Advanced Compatible Television and Narrow-MUSE), and four used digital transmission (Advanced Digital HDTV, Channel Compatible DigiCipher, HD-DigiCipher and Digital Spectrum Compatible HDTV). Based on the work of the ATTC, CableLabs and the ATEL, it was decided that none of the systems were ready for standardization and that improvements should be implemented. A further decision was made to proceed only with digital systems. The four digital proponents were encouraged to collaborate and combine the best parts of their respective systems.

Prior to the second round, the four digital proponents formed the Grand Alliance. The Grand Alliance represents the joint effort of American Telephone and Telegraph Company, David Sarnoff Research Center, General Instrument Corporation, Massachusetts Institute of Technology, North American Philips Corporation, Thomson Consumer Electronics, and Zenith Electronics Corporation. The prototype Grand Alliance hardware, henceforth referred to as the *digital* HDTV Grand Alliance System, included two main format variations 1080 active lines interlaced (1080I) and 720 active lines progressive (720P).

Subjective tests described in this report evaluated the performance of the *digital* HDTV Grand Alliance System in terms of basic received quality, impairment/interference into NTSC, and receiver scan conversion between 1080I and 720P formats. All tests were done using non-expert viewers.

The ATEL conducted its evaluation in accordance with all the requirements and test procedures developed and approved by the FCC Advisory Committee, collectively abbreviated as the Test Plan. The *digital* HDTV Grand Alliance System was tested against the projected prototype performance defined in chapter 9 of the Grand Alliance HDTV System Specification, henceforth referred to as the Target Specification.

The Basic Received Quality tests of the *digital* HDTV Grand Alliance System operating in both the 1080I and 720P modes were conducted during the period June 23 to August 3, 1995. The basic received quality performance of the System exceeded the Advisory Committee's Target Specification.

Interference into NTSC tests were performed for both the Lower-Adjacent and Co-Channels during the period of May 29 to June 5, 1995. In both cases the System exceeded the Advisory Committee's Target Specification. Upper-Adjacent Channel subjective tests could not be completed because no single receiver was adequately representative of the nature of the impairments resulting from the introduction of the *digital* HDTV Grand Alliance System into the Upper-Adjacent channel.

Receiver Scan Conversion tests were done for both 1080I to 720P conversion and for 720P to 1080I conversion during the period August 14 to August 16, 1995. In both cases, the overall performance of the System exceeded the Advisory Committee's Target Specification.

Viewing Conditions/Viewers:

The conditions in the ATEL viewing room were carefully controlled in terms of monitor setup; room luminance, color temperature, and noise level; viewing angle, distance and seating position. The layout of the viewing room is shown in Figure 1. The lightwall provided uniform illumination of the background surrounding the video display screen.

Viewers were recruited through a local university and were screened for visual acuity (normal or corrected-to-normal), contrast sensitivity (normal), color vision (normal), and English comprehension. Those who met the screening criteria were permitted to participate in the tests. Separate groups of viewers were used in each test. Each test consisted of six sessions. A maximum of five viewers participated in any given session.

Basic Received Quality:

The Basic Received Quality tests were conducted to assess the subjective quality of image sequences that were encoded, modulated, transmitted, demodulated and decoded by the *digital* HDTV Grand Alliance System. Two modes of operation of the *digital* HDTV Grand Alliance System were tested: 1080I and 720P.

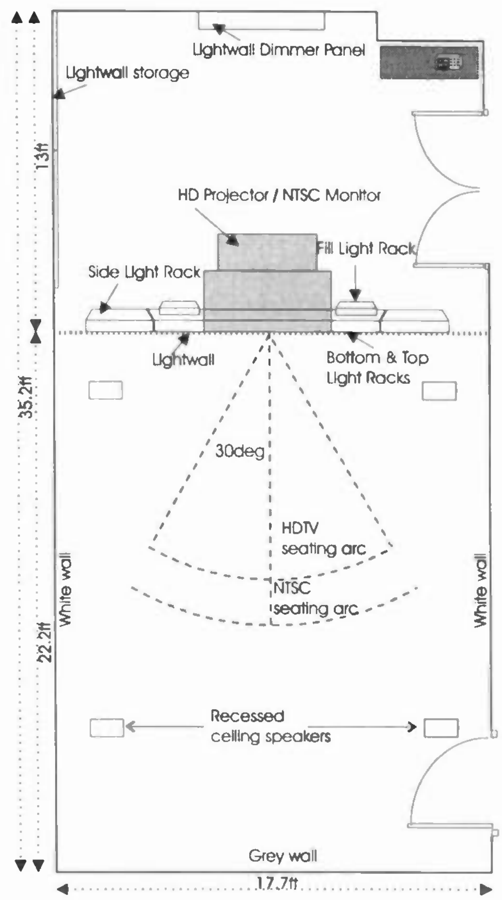

FIGURE 1 : VIEWING ROOM

The Basic Received Quality test sequences were selected by a panel of experts (FCC-ACATS, Planning Subcommittee-Working Party 6) to ensure that a broad range of image attributes were represented. A list of the 26 sequences used in the tests is presented in Table 1 along with a brief description and the image attributes for which they were selected. Of these 26 sequences, ten were retained from the first round of ATV testing, and are referred to as "core" sequences. The core sequences are marked with an asterisk in Table 1.

Reference sequences for both the 1080I and 720P tests of the *digital* HDTV Grand Alliance System were always displayed in the 1035I source format. Test sequences were generated by processing (encoding, modulating, transmitting, demodulating and decoding) source sequences through the *digital* HDTV Grand Alliance System. For the 1080I mode of testing, the 1035I source sequences were used as input to the *digital* HDTV Grand Alliance System.

The extra 45 lines were obtained by duplicating the top 40 lines of the 1035I image frame and adding 5 black lines at the bottom (i.e. 1035 + 40 + 5 = 1080). It was not possible to record test sequences with 1080I lines using the Sony HDD-1000 VTR, the only available recorder at the time of testing. Therefore, for storage, display and subjective evaluation, the extra 45 lines were stripped away, such that the 1080I mode of the *digital* HDTV Grand Alliance System was represented with 1035I lines. For the 720P mode of testing, the 720P source sequences were used as input to the *digital* HDTV Grand Alliance System. Test sequences were displayed in 720P.

The layout of a Basic Received Quality assessment trial is shown schematically in Figure 3, and is based on the double-stimulus continuous quality scale method described in ITU-R (International Telecommunications Union Radiocommunications Sector) Recommendation 500. Each trial consisted of a pair of Reference and Test sequences presented twice in succession. When sequence "A" was a Reference, sequence "B" was a Test, and vice versa.

Viewers were not informed whether "A" or "B" was the Reference or the Test sequence. Viewers were instructed to rate the perceived image quality of the "A" and the "B" sequences using 10 centimeter judgment scales like those shown (not to scale) in Figure 2.

The labels "Excellent", "Good", "Fair", "Poor" and "Bad" were printed at the locations shown in Figure 2. Numerical values in brackets are presented for the readers convenience only, and were not provided to the viewers.

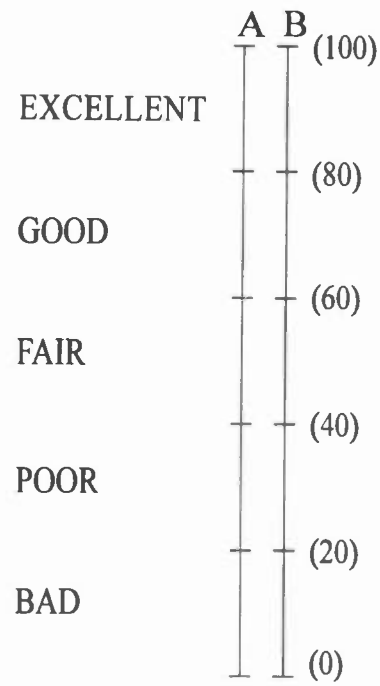

FIGURE 2: QUALITY RATING SCALES

Each participant completed 52 trials (26 Pictures x 2 Replicates). Five practice trials were completed at the start of testing, and 3 were completed after a 30 minute rest-break midway through the session. Two tape orders, varied between subjects, were used. Tape Order refers to the random presentation of the experimental trials.

Difference scores (Test - Reference) for 1080I and 720P are presented in graphical form in Figure 4. A rating greater than 0 indicates that the Test sequence was rated higher than the Reference sequence. A rating less than 0 indicates that the Test sequence was rated lower than the Reference sequence.

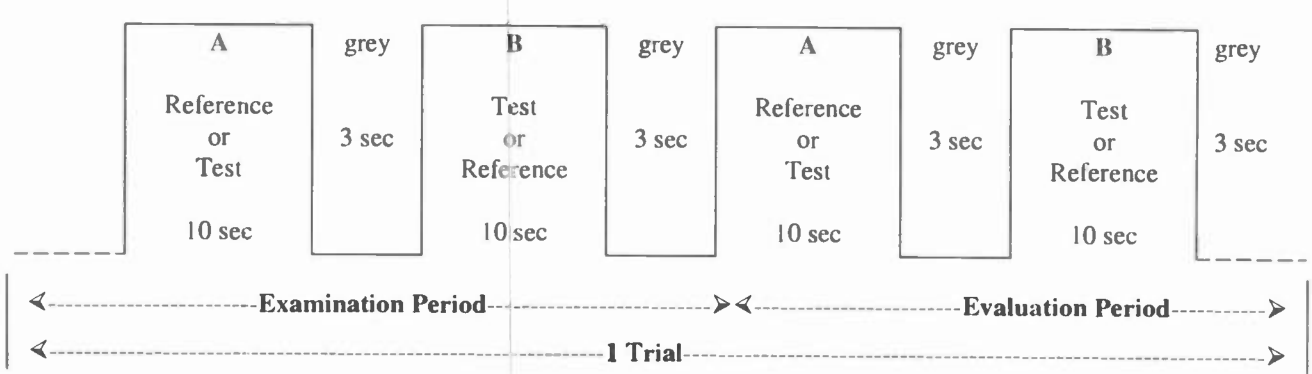

FIGURE 3: LAYOUT OF A BASIC QUALITY ASSESSMENT TRIAL

TABLE 1 BASIC QUALITY TEST SEQUENCES

NO	ID	PICTURE	IMAGE ATTRIBUTES	IMAGE DESCRIPTION
1	S1*	Metal Table and Chairs	Luminance Resolution	Overhead view of round tables and chairs on concrete patio stones.
2	S5*	Tulips	Chrominance Resolution	Purple, yellow, pink and white tulips.
3	S6	Sculptures	Chrominance Artifacts	Colorful dinosaurs on wall.
4	S8*	Toys	Chrominance Dynamic Range and Compression Noise	Checkers, Chinese checkers, plastic blocks, and paint trays.
5	S9*	Girls With Toys	Peripheral Performance and Texture	Two girls playing with toys and books, on a rug.
6	S14A	Cheshire Cat	Spatial Resolution	Graphic. Three potted plants with Christmas balls, on black and white checkered floor.
7	M1*	Window	Dynamic Luminance Resolution	Blowing curtains over kitchen sink.
8	M2*	Fax Machine	Dynamic Luminance Resolution	Fax machine sending Quarterly report.
9	M4*	Mannequins	Chrominance Artifacts	Mannequins rotating in display window with stars background.
10	M5*	Living Room	Motion Rendition	Panning/zooming of camera, from book shelves to stack of art books on table.
11	M6*	Den	Motion Rendition In-Scene-Movement	Man getting book off shelf, and sitting in sofa.
12	M10*	Woman and Room	Motion Rendition and In-Scene-Movement	Woman in green, blue and black argyle sweater getting book, and sitting on sofa.
13	M16A	Rotating Pyramids	Motion Graphics	Graphic. Silver mobile with 4 diamonds spinning, on marble background, with scrolling text.
14	M35	Crosswalk	Motion, Multiple Object	Lots of people walking through busy intersection.
15	M36	Georgetown Loop	Motion With Texture	Sightseeing train traveling along a mountain track.
16	M37	Buckingham Palace	Chrominance Saturation	Marching guards in red uniforms.
17	M38	Snow Tires / Trees	Luminance Resolution	Creek lined with snowy trees.
18	M39	End Zone	Crowd Scene, Slow Zoom	Olympic procession into stadium.
19	M40	Dream Team	Fast Motion, Scene Cuts	Basketball game.
20	M41	Golf	Dissolve, Chrominance Saturation, Zoom	Man teeing off on golf course.
21	M44	Mirror	Film Transfer 24fps	Woman putting on make-up, class reunion invitation on dresser.
22	M45	Christa	Film Transfer 24fps	Woman walking down sidewalk.
23	M46	Fountain	Film Transfer 30fps	Man and woman splashing at fountain.
24	M47	Clock #1	Computer Graphics, Pan and Zoom	Graphic. Internal mechanics of a clock.
25	M48	Connections	Motion Graphics	Flight details given by man in striped jacket.
26	M49	Picnic With Ants	Noise Impairment	Still of four people at table with food and drink, with encroaching noise from all four sides.

* Denotes Core Sequences

Target specifications for the *digital* HDTV Grand Alliance System are presented in Table 2. Note that the Target Specification groups sequences into four categories. In all four categories the observed performance exceeded the Target Specification for both the 1080I and 720P cases.

Additional statistical analyses were done, on a sequence by sequence basis, to discover which differences between Reference and Test were *statistically significant*. For those sequences, observed differences could be attributed to actual differences in rated quality rather than to chance variation. These analyses showed that sequences M2, M4, M5, M37, M38, M40 and M49 for 1080I, and that sequences M35, M38, M39, M40 and M49 for 720P were actually rated below reference.

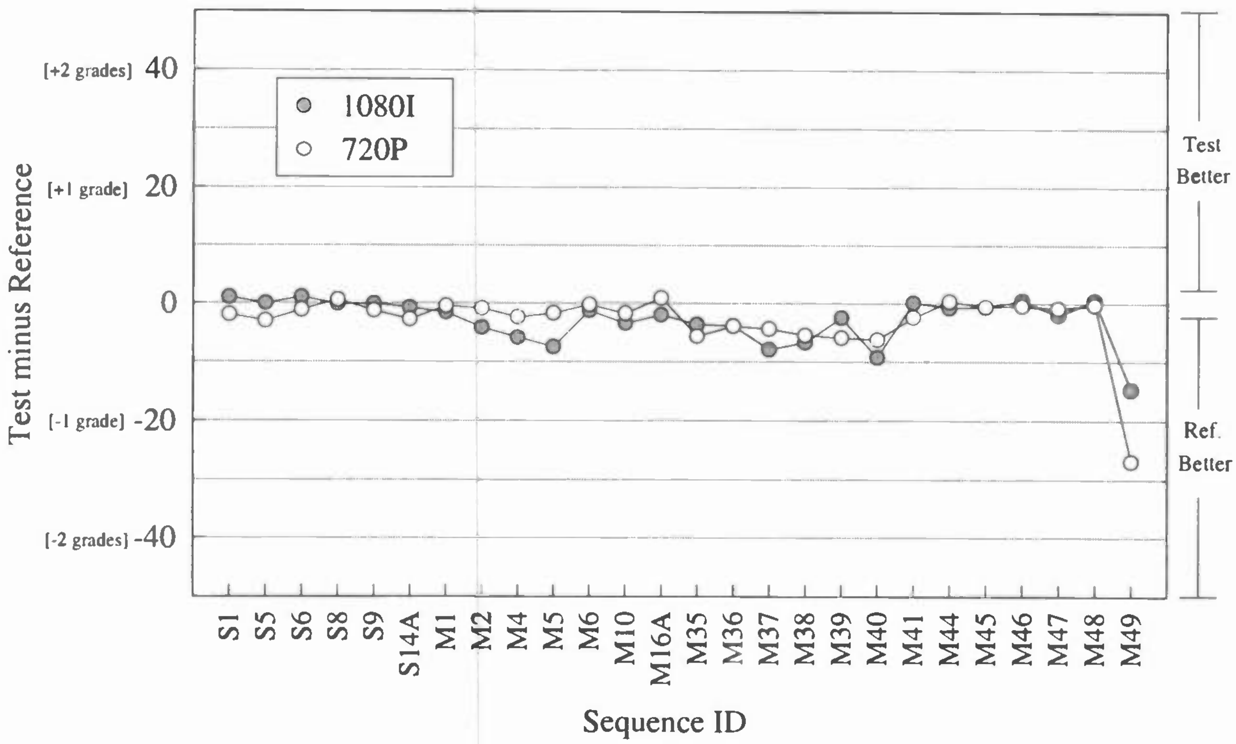

FIGURE 4: BASIC RECEIVED QUALITY DIFFERENCE SCORES

TABLE 2 TARGET SPECIFICATIONS

TEST SEQUENCE	TARGET SPECIFICATION	OBSERVED PERFORMANCE	
		1080I MEAN (Test minus Ref.)	720P MEAN (Test minus Ref.)
Basic Material[1]	≤ 0.3 Grade below reference	-0.12	-0.11
Noise & cuts (M37, M40, M41, M49)	≤ 1.0 Grade below reference	-0.40	-0.50
Graphics (S14A, M16A, M47, M48)	≤ 1.0 Grade below reference	-0.06	-0.04
Film 24fps (M44, M45)	≤ 0.25 Grade below reference	-0.04	-0.01

[1] Basic Material consisted of the following 16 sequences; S1, S5, S6, S8, S9, M1, M2, M4, M5, M6, M10, M35, M36, M38, M39 and M40.

Impairment/Interference:

The Impairment/Interference tests were conducted to assess the degradation in perceived quality of an NTSC picture when a *digital* HDTV Grand Alliance signal was present on an adjacent channel (Upper or Lower) or on the same channel (Co-Channel). Upper-Adjacent Channel subjective tests could not be completed because no single receiver was adequately representative of the nature of the impairments resulting from the introduction of the *digital* HDTV Grand Alliance System into the Upper-Adjacent channel.

Three NTSC sequences were used in the Impairment/Interference tests: Girls with Toys, Co-Channel and Woman with Roses[2]. Reference sequences consisted of these three sequences received at a given signal level in NTSC, but with no interference. Test sequences consisted of the same three sequences received at the same NTSC signal level but with the *digital* HDTV Grand Alliance signal present in either the Lower-Adjacent Channel or Co-Channel.

Signal Level refers to the level of the desired NTSC signal. The Signal Level was varied at -35 dBm (moderate)[3] and -55 dBm (weak) for the Lower-Adjacent Channel Interference test, and was fixed at -55 dBm (weak) for the Co-Channel Interference test. Impairment Level refers to the level of the undesired (interfering) ATV signal. The Impairment Level was varied at the six levels shown in Tables 3 and 4. These levels were selected by a panel of "expert" observers at ATTC to ensure that the severity of perceived impairments ranged from "imperceptible" to "very annoying." To enhance the visibility of noise in the NTSC sequences, the content of the undesired ATV signal was switched (gated) between black and the sequence Rotating Pyramids (M16).

For the Lower-Adjacent Channel Interference test, viewers completed 72 experimental trials (2 Signal Levels x 3 Pictures x 6 Impairment Levels x 2 Replicates) plus 8 practice trials. Five practice trials were completed at the start of testing and 3 were completed after a 30 minute rest-break midway through the session. In the Co-Channel test, viewers completed 36 experimental trials (3 Pictures x 6 Impairment Levels x 2 Replicates) plus 5 practice trials at the start of testing. There was no rest-break in the Co-Channel test. There were two tape orders, varied between subjects. Tape Order refers to the random presentation of the experimental trials.

The five-grade impairment scale method described in ITU-R Recommendation 500 was used to assess perceived impairment of the Test sequences. The layout of a trial is shown schematically in Figure 5. Each trial consisted of a Reference sequence followed by a Test sequence.

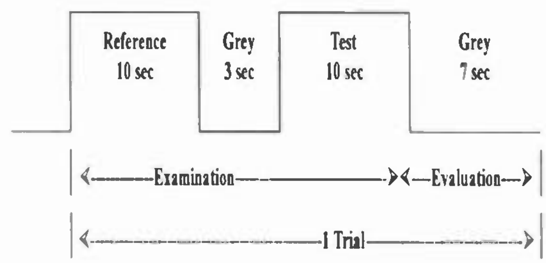

FIGURE 5 LAYOUT OF IMPAIRMENT ASSESSMENT TRIAL

The five-grade impairment scale is shown in Figure 6. Numerical values in brackets are presented for the readers convenience only, and were not provided to the viewers.

IMPERCEPTIBLE ☐ (5)

PERCEPTIBLE, BUT NOT ANNOYING ☐ (4)

SLIGHTLY ANNOYING ☐ (3)

ANNOYING ☐ (2)

VERY ANNOYING ☐ (1)

FIGURE 6 FIVE-GRADE IMPAIRMENT SCALE

For each trial, viewers were instructed to rate the impairment of the Test sequence relative to the Reference sequence. For example, if the Test and Reference sequences contained impairments of equal magnitude, subjects were instructed to rate the impairment as imperceptible. If the Test sequence contained impairments of greater magnitude than the

[2] Girls with Toys is a still image of two girls with toys playing on a rug. Co-Channel (also known as Texas Sign Dude) is a man checking signs in an office; this sequence was used in all Impairment/Interference tests reported here. Woman with Roses is a still image of a woman with a bouquet of flowers.

[3] For purposes of this paper the results for -35 dBm (Moderate) are not reported.

Reference sequence, subjects were instructed to rate the incremental impairment accordingly. Note, that the Reference sequences could contain visible impairments, especially at the weak signal level.

The results are presented in graphical form in Figures 7 and 8 collapsed (averaged) across all three test pictures, along with the target D/U (Desired-to-Undesired) power ratio at level 3. Tables 3 and 4 show the same data numerically.

It is clear that the *digital* HDTV Grand Alliance System exceeded the Target Specification.

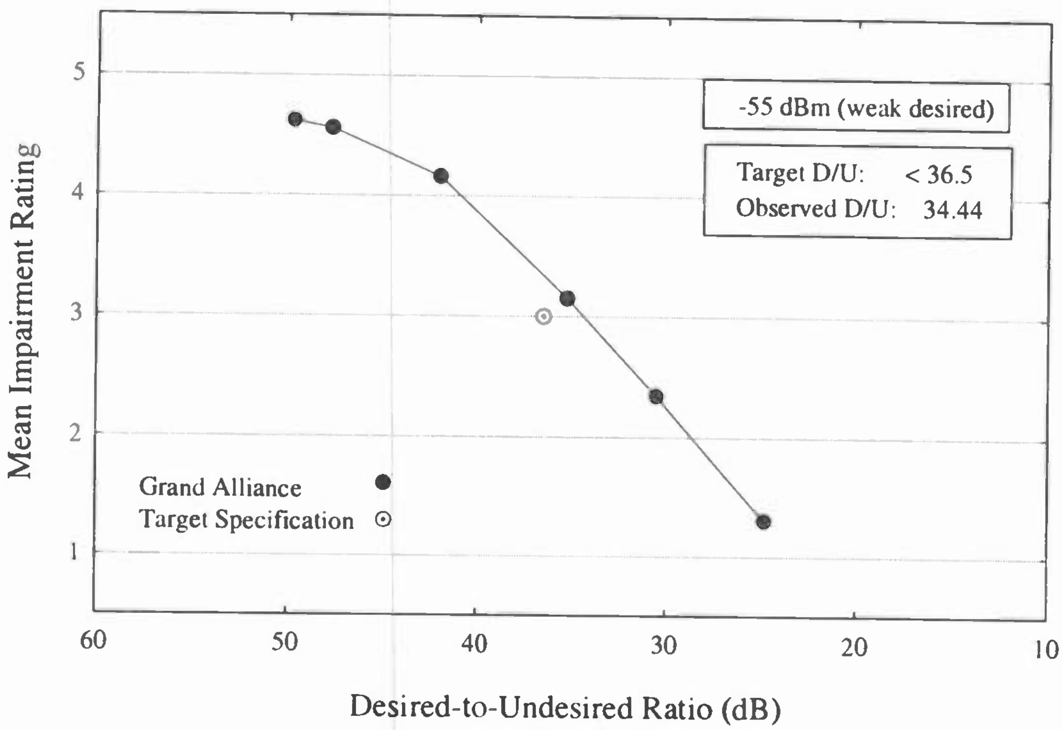

FIGURE 7 CO-CHANNEL INTERFERENCE (ATV into NTSC)

TABLE 3 CO-CHANNEL INTERFERENCE (ATV into NTSC)

PICTURE	UNDESIRED LEVEL (dBm)		D/U (dB)	MEAN RATING
S09, M14, S11	SUB TOV	-104.74	49.74	4.62
	TOV[4]	-102.74	47.74	4.56
	TOV+ 1	-97.01	42.01	4.16
	TOV + 2	-90.26	35.26	3.16
	TOV + 3	-85.51	30.51	2.34
	TOV + 4	-79.80	24.80	1.30

[4] Threshold of Visibility

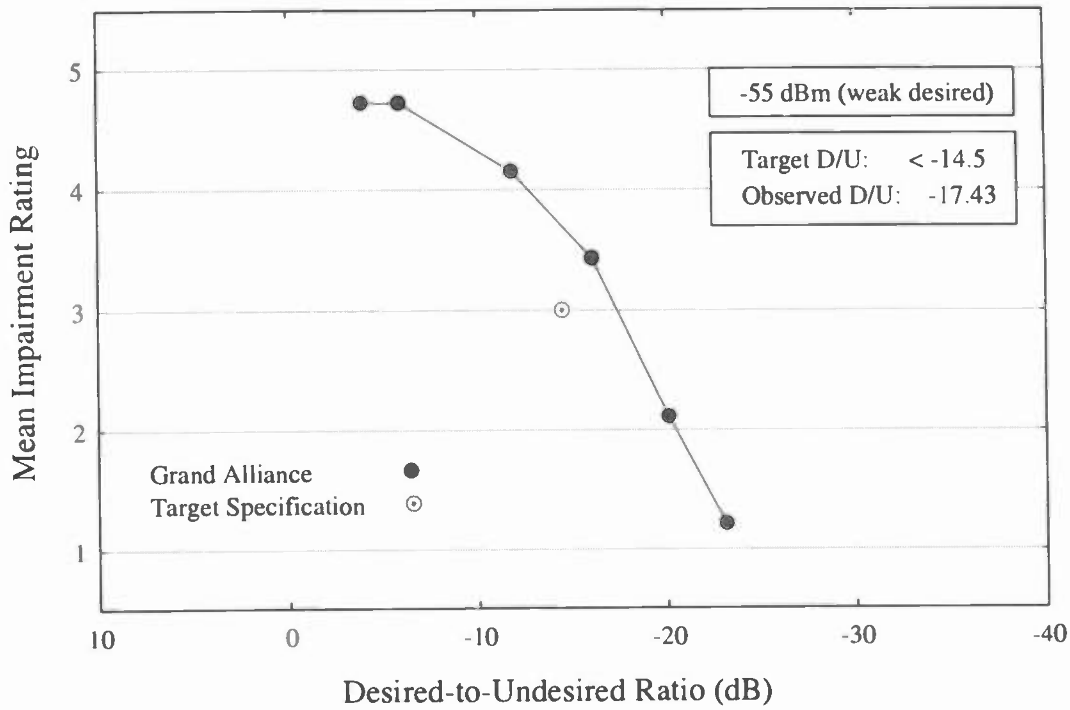

FIGURE 8 LOWER-ADJACENT CHANNEL INTERFERENCE (ATV into NTSC)

TABLE 4 LOWER-ADJACENT CHANNEL INTERFERENCE (ATV into NTSC)

PICTURE	UNDESIRED LEVEL (dBm)		D/U (dB)	MEAN RATING
S09, M14, S11	SUB TOV	-51.08	-3.92	4.72
	TOV	-49.08	-5.92	4.72
	TOV+ 1	-43.14	-11.86	4.15
	TOV + 2	-38.89	-16.11	3.43
	TOV + 3	-34.89	-20.11	2.11
	TOV + 4	-31.89	-23.11	1.20

Receiver Scan Conversion:

The Receiver Scan Conversion tests were conducted to assess the impact on quality of converting, at the receiver, between the two modes of operation (1080I and 720P) of the *digital* HDTV Grand Alliance System. Subjective image quality of unconverted (1080I and 720P) and converted (1080I to 720P, and 720P to 1080I) sequences was assessed.

For this test, the expert group selected only critical sequences in which conversion artifacts were observed by experts. Five sequences were selected for both 1080I to 720P, and 720P to 1080I conversion. One additional sequence was selected for 1080I to 720P and two for 720P to 1080I conversion. Tape capacity restrictions precluded doing a full factorial design where all sequences were converted from 1080I to 720P and from 720P to 1080I.

The eight sequences used in the Receiver Scan Conversion tests were: Metal Table and Chairs (S1), Rotating Pyramids (M16A), Dream Team (M40), Ducks (M43), Picnic with Ants (M49), Den (M6), Woman and Room (M10) and Cheshire Cat (S14A). The attributes of these sequences are listed in, Table 1, with the exception of Ducks (M43)[5]. As in the Basic Received Quality tests, each sequence was a central 10-second portion of a 15-second video clip. However, for Rotating Pyramids (M16A) and Picnic with Ants (M49), the 10-second portion was selected from the end of the clip instead of from the center (M16A comprised the period 4-14 seconds and M49 comprised the period 5-15 seconds). This was done to make these sequences more challenging for compression and conversion.

As in all quality tests described in this paper, Reference sequences were displayed in the 1035I format.

Four types of Test sequences were defined in terms of the input - output combinations: 1080I - 1080I unconverted, 720P - 720P unconverted, 1080I - 720P converted, and 720P - 1080I converted. All four input - output combinations were generated for S1, M16A, M40, M43 and M49. For M6 and M10, Test sequences were generated for 720P - 720P unconverted and 720P - 1080I converted. For S14A, Test sequences were generated for 1080I - 1080I unconverted and 1080I - 720P converted.

There were four factors in the design of the test: Picture, InOut, Replicate and Tape Order. Picture refers to the sequence name. InOut refers to the input - output combination (1080I - 1080I, 720P - 720P, 1080I - 720P, and 720P - 1080I). Replicate refers to the number of times a condition occurred during a session; each condition occurred twice per session. Tape Order refers to the random order of the experimental trials; two Tape Orders were used. Picture, InOut, and Replicate were varied within subjects, and Tape Order was varied between subjects.

Viewers completed 52 experimental trials plus 8 practice trials. Five practice trials were completed at the start of testing, and 3 were completed after a 30 minute rest-break midway through the session. The 52 experimental trials were defined by the factorial combination of Replicate, Picture and InOut; Five Pictures (S1, M16A, M40, M43, M49) were varied at all four levels of InOut, and three Pictures (M6, M10, S14A) were varied at only two levels of InOut.

The results are presented in Figure 9 in graphical form as Difference scores (Test - Reference). A rating greater than 0 indicates that the Test sequence was rated higher than the Reference sequence. A rating less than 0 indicates that the Test sequence was rated lower than the Reference sequence. The Target Specification for the *digital* HDTV Grand Alliance System is presented in Table 5. The observed performance exceeded the Target Specification for both 1080I to 720P and 720P to 1080I conversion. Note that the Target Specification refers to the average performance over a set of sequences. Taken individually, certain sequences were particularly difficult to convert because they had image features that resulted in enhancement of digital coding artifacts (e.g. in M49 large parts of the image consisted of white noise which overloaded the coder and resulted in visible distortion of the central static portion of the image).

Conclusions:

The FCC Advisory Committee developed performance objectives (Target Specification) for the *digital* HDTV Grand Alliance System using information obtained in the first round of HDTV testing completed in 1992.

The ATEL conducted video subjective tests using non-expert viewers as part of an overall assessment of the subjective quality of the *digital* HDTV Grand Alliance System with respect to the Target Specification.

In the Basic Received Quality Tests the *digital* HDTV Grand Alliance System exceeded the Target Specification for all four identified categories of material (Basic Material, Noise & Cuts, Graphics, and 24fps Film).

The Interference Tests for both the Co-Channel and Lower-Adjacent Channel confirmed that the degree of impairment introduced into the NTSC signal by the *digital* HDTV Grand Alliance System was less than that allowed by the Target Specification.

Similarly, in the Scan Conversion Tests the Target Specification was exceeded for both 1080I to 702P, and 720P to 1080I conversions.

[5] This sequence which consisted of a pond area crowded with moving ducks and swans, was chosen to demonstrate complex motion. The 1035I video camera source was converted to 720P.

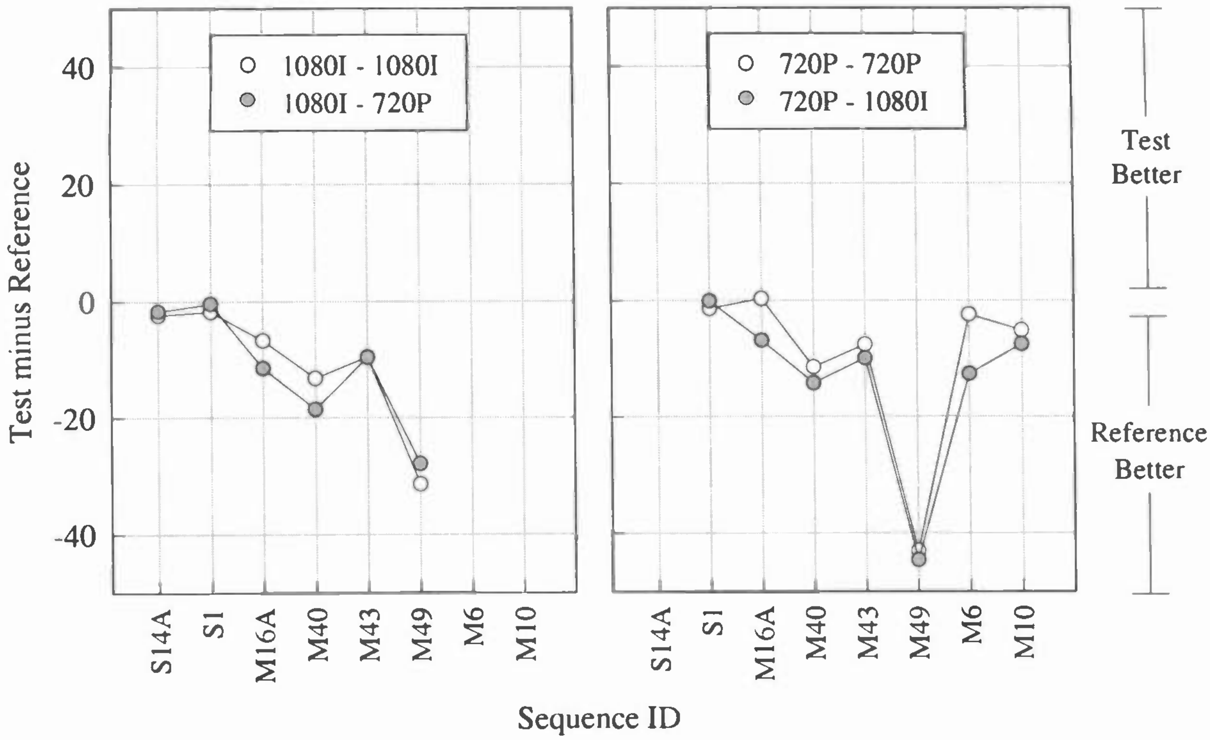

FIGURE 9 RECEIVER SCAN CONVERSION DIFFERENCE SCORES

TABLE 5 RECEIVER SCAN CONVERSION TARGET SPECIFICATIONS

TEST SEQUENCE	1080I - 720P (Test minus Ref.)	720P - 1080I (Test minus Ref.)	TARGET SPECIFICA-TION
S1	-0.02	-0.03	
S14A	-0.08	----	
M6	----	-0.63	
M10	----	-0.37	
M16A	-0.57	-0.34	
M40	-0.93	-0.71	
M43	-0.48	-0.50	
M49	-1.39	-2.23	
MEAN	-0.58	-0.69	≤ 1.0 Grade

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THE HARRIS VSB EXCITER FOR DIGITAL ATV

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ABSTRACT

A brief description of the Harris 8-VSB digital ATV exciter is presented. This exciter implements the Grand Alliance signalling scheme. Various system, performance, and hardware aspects are discussed.

INTRODUCTION

Harris Broadcast Division has designed a Digital ATV Vestigial Sideband (VSB) exciter which implements the Advanced Television Standards Committee (ATSC) specification for the Grand Alliance (GA) proposed American terrestrial transmission system for digital television. This exciter represents the first implementation of the full GA VSB exciter by a transmitter manufacturer. A prototype of this exciter was exhibited by Harris at the 1995 NAB convention. This paper discusses various aspects of the design of this exciter. Results are presented to show various performance parameters including: Spectrum, Eyepattern, Signal Constellation, and Error Vector Magnitude.

8-VSB MODULATION

The modulation used in the Grand Alliance terrestrial broadcasting standard is 8-level Vestigial Sideband Modulation (8-VSB). Figure 1 shows the most basic conceptual interpretation of 8-VSB modulation. As shown, data bits are taken in 3 bits at a time and converted to 8-level pulses which are applied to a baseband filter which is offset in frequency by one-fourth symbol rate to produce a baseband version of 8-level VSB. There are many ways to implement the 8-VSB modulator, but the objective of all those ways is to produce the 8-VSB signal as in the conceptual representation of Figure 1. Harris uses a proprietary digital filter technique to implement the baseband Nyquist filter. The implementation offers perfect cancellation of the lower sideband without the need for a SAW filter to control out-of-band spectral content. The filter is designed so that finite filter resolution and finite tap length effects cause negligible degradations. After upconversion, the first IF is centered at 10.76 MHZ, placing the pilot carrier at 8.07 MHz.

Modulator Functional Block Diagram

Figure 2 shows the basic functional block diagram for the Harris 8-VSB modulator. Data formatted and encoded in accordance with Grand Alliance specifications ultimately emerges from the Transport-to-Transmit Layer Conversion and is presented 3 bits at a time at the 10.76 MHz symbol rate to the Baseband Digital Nyquist filter. After the digital filtering, the data is Digital-to-IF converted to a 10.76 MHz analog IF signal using a high speed Digital-Analog Converter (DAC). The analog IF output is filtered by an x/sinx correction filter whose purpose is to compensate for the sinx/x response inherent in the square pulses emerging from the DAC. A low pass filter is also required to attenuate the image frequency before the IF is upconverted to the second IF at 45 MHz. From the second IF the 6 MHz bandwidth

8-VSB signal is upconverted to carrier.

The Signal Envelope of 8-VSB

Figure 3 shows the typical signal envelope for 8-VSB modulation. Notice that there is great variability in the envelope peaks. This time-varying peaking is the source of the large peak-to-average ratio of 8-VSB. One often hears that the peak-to-average power ratio is around 6 dB. This would indicate that the largest peaks of the envelope in Figure 3 are around twice the voltage of the average envelope. In any case, the variability in the envelope shown in Figure 3 for 8-VSB is the root cause for the requirement of very linear power amplification for this modulation.

Polar Plot and Constellation for 8-VSB

Figure 4 shows another viewpoint of the envelope fluctuation of 8-VSB. This figure is called a polar plot and it leads to the idea of the signal constellation. The plot is of the two legs of the quadrature detector used to detect 8-VSB at the receiver. The instantaneous Q channel detector voltage is plotted against the instantaneous I channel detector voltage. A vector drawn from the polar plot's center to a point on the curve shows the instantaneous received RF amplitude by the length of the vector and the instantaneous RF phase by the phase of the vector. Thus one can visualize the fluctuations of the RF envelope and phase as the tip of the vector travels along the tangled curve of Figure 4.

The dark points on Figure 4 represent the time "snapshots" taken every symbol time of the location of the time-varying vector tip. In the digital data sense, these are the only points of importance in the received waveform. Figure 5 shows a case where only the snapshot points have been retained and the visual clutter of the various meanderings of the vector tip between points has been eliminated. This plot is widely known as the signal "constellation" for 8-VSB.

Note that for 8-VSB all the snapshot points in Figure 5 taken at the symbol rate, all line up on one of eight values on the I Channel axis. This corresponds to the 8-level value originally transmitted. Thus the receiver decides the transmitted value simply by quantizing the I channel at the receiver into one of 8 levels, which is tantamount to deciding which 3-bit group was originally transmitted. This is the way the conversion is made at the receiver from the analog domain back into the digital data domain.

Further note that the Q channel in Figure 5 shows no corresponding digital grouping and is therefore of no use in decoding the digital data transmitted. The Q channel is, however, very vital in causing the suppression of the sideband in VSB. It just has no data-determining usefulness.

8-VSB Eyepattern

Figure 6 shows another well-known representation of an 8-VSB signal characteristic called the eyepattern. The eyepattern is most conveniently thought of as the display on an oscilloscope of the data-carrying I channel at the receiver quadrature demodulator using the symbol rate clock as a trigger signal. The point in the center of the eyepattern of Figure 6 represents the periodically recurring time at which the I channel voltage arrives precisely upon one of those eight vertical lines in Figure 4 or 5. One can see from the eyepattern of Figure 6 that the receiver must very accurately control the symbol rate sampling phase and stability to insure that the eyepattern waveform is sampled precisely at the moment when the eight levels become very distinct and the "eyes" in the center of the diagram are maximally "open." This is to insure that the decision about which one of the eight levels was transmitted is made with greatest confidence and with maximum immunity to noises of various sorts.

Measured 8-VSB Modulator Characteristics

Figure 7 shows several characteristics of the Harris 8-VSB modulator measured by a Hewlett-Packard 89440 Vector Signal Analyzer. In the upper left of the figure one can see the signal constellation we previously discussed in conjunction with Figures 4 and 5. In the upper right of Figure 7 one can see the measured eyepattern for comparison with the ideal of Figure 6. In the bottom left, the Error Vector Magnitude (EVM) is plotted versus time. EVM is the length of the vector defining the error between an ideal 8-VSB signal sample and the actual signal sample. From the numerical display in the bottom right of Figure 7, the EVM has an RMS value of 2.5457%. As a calibration point, the distance between any two adjacent lines in the constellation is 28.6% on this same scale. Thus the EVM value is small enough not to cause any significant degradation in modem performance.

Figure 8 shows the preliminary spectrum measured from the modem. Note the spike in the spectrum on the left side indicating the presence of the 8-VSB pilot carrier.

HARDWARE DESIGN ASPECTS

The design of the VSB exciter consists of 4 major functions, namely, 1) the Transport to Transmission Layer converter, 2) The Nyquist filter, 3) IF conversion, and 4) signal upconversion. (See Figure 9.) Each of these functions is described below.

Transport/Transmission Layer Conversion:

The Grand Alliance has defined the data transport layer to be a 19.39..Mb/s data stream. The data is formatted into 8 bit bytes, and 188 byte frames. Each frame consists of 187 bytes of data and one byte which is reserved as a sync byte. The resulting payload is then 19.289...Mb/s. The transport layer does not have the overhead of error-correction coding and therefore is optimum for distributing the signal at the studio and for communication over the Studio Transmitter Microwave Link (STL). The transmission layer, on the other hand, is less desirable for hardwire and STL distribution because of it's significantly higher data rates. Assuming STLs are unidirectional, there would be no ability to accommodate a separate clock at the transmitter. This would require that the transmitter be synchronized to the Transport layer clock at the studio. The STL link, therefore, represents a particular challenge. It does not allow burst mode operation between the studio and transmitter, but instead a continuous data stream and clock is required. This also implies that the ultimate transmitter frequency will depend on the Transport Layer clock at the studio. This of course means that the Transport Layer clock must be at least as accurate as the channel frequency specification. Provisions for an external reference input at the transmitter would be a possible solution in case the Transport clock is inferior to the system specification. In addition to Transport clock accuracy, the additional problem of clock recovery at the transmitter site must also be addressed. These issues and others are topics of current standards discussions.

Assuming that the Transport clock and data stream can be delivered reliably to the transmitter, the signal must be converted to the Grand Alliance Transmission format before it can be transmitted. This is one of the main functions of the new ATV exciter. The exciter performs the Transport to Transmission Layer conversion by implementing the following functions (see Figure 10): 1) Frame synchronization, 2) Data randomization, 3) Reed-Solomon Encoding, 4) Data Interleaving, 5) Trellis encoding, and 6) Field and Frame sync insertion.

Data Synchronization and Randomizer

The frame synchronizer is required to locate the Transport layer sync bytes, and to align the serial data stream into bytes. Since the sync byte is not a unique number, a false indication may result for any 8 contiguous data bits identical to the sync byte. To overcome this problem, a confidence counter is employed which counts the number of sync bytes over a long period of time. Should the sync byte not be detected, then the confidence of having sync (or frame) lock will be reduced. Should several frames occur without a sync byte detection at a specific byte location, then the exciter will indicate an out of lock condition and begin a new frame search and acquire mode.

The data randomizer is fairly straightforward in its implementation. The only real concern is ensuring that the polynomial generator is initialized at the appropriate time. This must be done before the first data segment occurs. When the randomizer is initialized before the first data byte of 312 segments of data, it sets the starting data byte which must be known for other system functions including the Reed-Solomon encoder, and the field sync inserter.

Reed-Solomon Encoder

The Reed-Solomon encoder requires 20 modulo-256 multiplies and 20 exclusive-or (XOR) adds for each Transport layer data byte. At the relatively high data rates of the ATV signal, a processor capable of 97 MOPS (Million Operations Per Second) is required. This level of processing and the non-standard arithmetic functions make a general purpose DSP approach rather difficult. The encoder was therefore designed using an FPGA in conjunction with a ROM look-up table. This approach easily accommodated the processing requirement of the encoder and offered a reduction in overall cost. The tricky part was dealing with the burst of 20 parity bytes after each 187 data bytes. Fortunately the long memory length required by the data interleaver allowed for plenty of memory storage to accommodate the burst parity bytes.

Data Interleaver

The data interleaver function allows for 4.48ms of data diversity, requiring 5.3k bytes of data storage space. A much larger data storage space was implemented to accommodate the inherent data rate expansion caused by the Reed-Solomon encoder and the Trellis encoder. The larger storage space was needed to incorporate a data buffer of about 32kbytes. This introduced a fixed time delay of about 13ms. The Reed-Solomon encoder writes to the buffer at the Transport layer data rate, while the interleaver reads the data buffer at the Transmission layer data rate. This allows for the system to operate on two different clocks without asynchronous handshaking.

Trellis Encoder

Before the data undergoes Trellis coding the 8 bit data bytes are split into four 2-bit data words. The Trellis coder is actually a comb filter in conjunction with a rate 2/3 coder. The 4-state coder does not code the MSB bit but uses a rate 1/2 coder on the LSB. The output of the coder is mapped into a one-dimensional constellation defined as 8-level VSB.

Sync Insertion

The final step in converting the Transport layer to a transmission layer format is insertion of data segment syncs and field syncs. Each Transmission segment sync must correspond with every Transport layer segment sync. The only exception is during the field sync interval. During the field sync interval, an additional segment sync is added along with a segment-wide field sync. After the Field sync has been added, the final output is a 10.76...MHZ 8-level VSB signal ready for Nyquist bandshaping and upconversion.

Nyquist Filter

The Nyquist filter is a root-raised-cosine filter with alpha factor of 0.1152. This filter, if perfectly implemented, will limit the spectrum to 6MHz Bandwidth (5.38MHz + 11.52%). A perfect filter however, is not possible due to filter tap limitations and finite resolution. There is a trade-off between bandwidth and cost. The filter chosen for the ATV exciter is a 95-tap FIR filter with a Hamming window function. This filter requires 16.3% excess bandwidth at 50dB down given a 16-bit coefficient resolution. No spectral mask has been specified for the system, so this filter may change based on future system requirements. The filter must be a complex filter to accommodate the vestigial sideband function. Also, the VSB pilot must be added to the signal.

Careful attention must be given to signal peaks that arise due to the bandlimiting function of the Nyquist filter. Some peaks arise which exceed a 6dB peak-to-average signal ratio. These peaks are essentially clipped. Computer simulations were used to assure that performance degradation and spectral splatter caused by clipping were minimal. The effects of the clipping are minimal mostly because of the rarity of these large peaks.

After the pilot has been added, the digital signal is then converted to analog IF via a high speed DAC.

SUMMARY

The Harris implementation of the 8-VSB exciter according to Grand Alliance specifications represents a robust approach to this new technology. The implementation is largely digital and this represents a stable, reliable way to meet the stringent requirements placed on modern digital advanced television transmitters.

FIGURE 1 - 8-VSB Conceptual Block Diagram

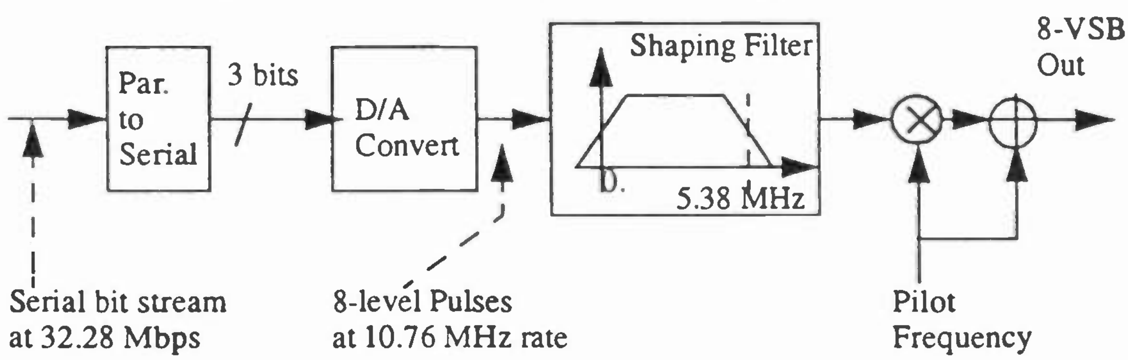


FIGURE 2 - HARRIS DIGITAL ATV 8-VSB MODULATOR

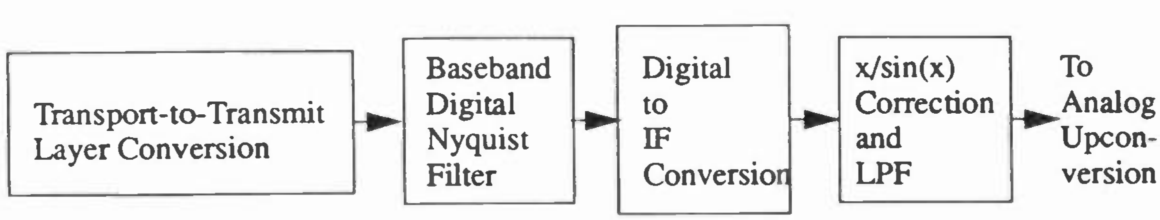


FIGURE 3

8-VSB HDTV RF WAVEFORM

Alpha = 0.11

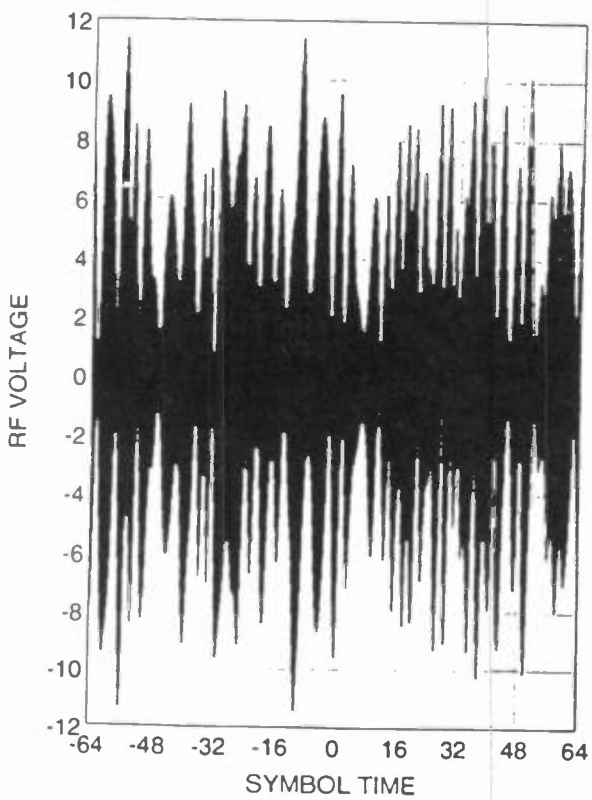


FIGURE 4

8-VSB HDTV "CONSTELLATION"

Alpha = 0.11, Pilot Carrier Offset = +1.
Transitions Between Sample Points Shown

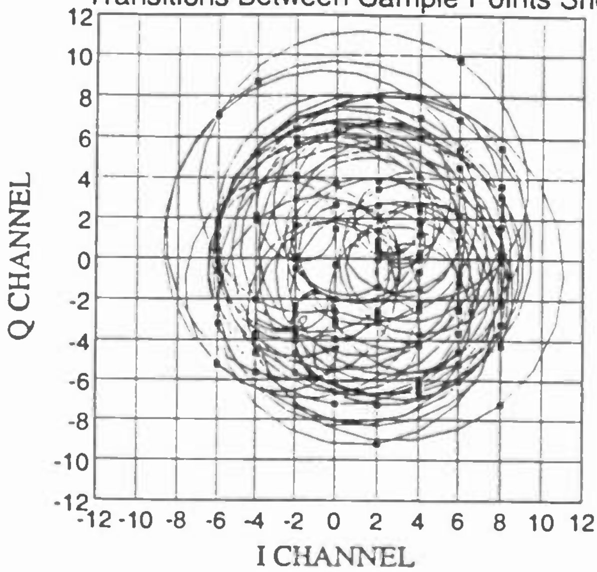


FIGURE 5

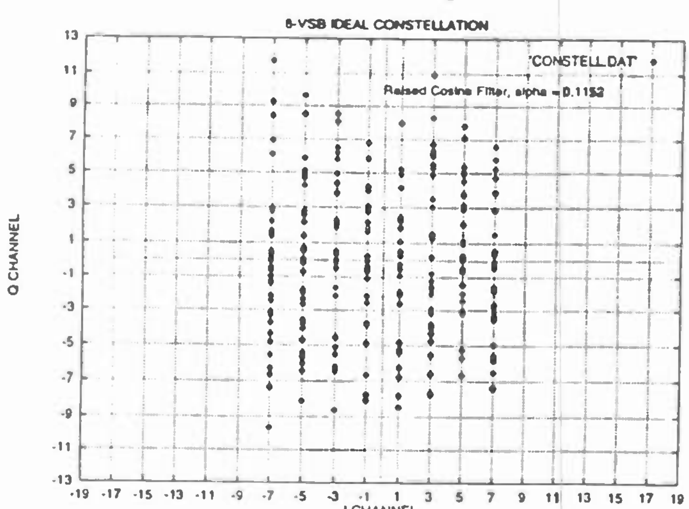


FIGURE 6

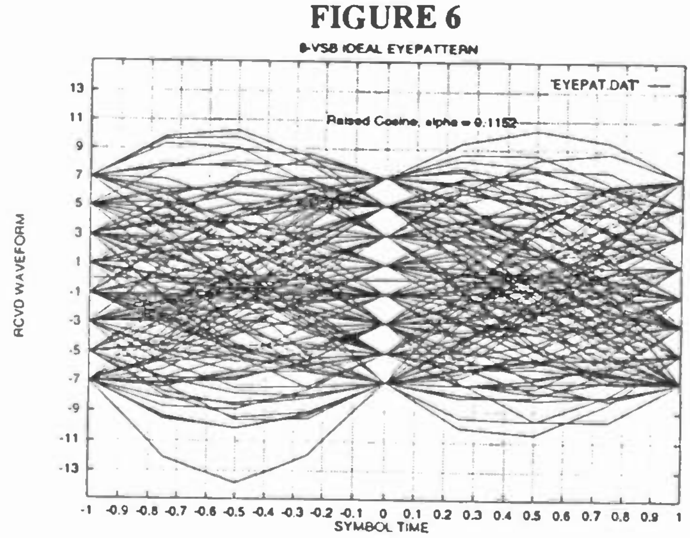


FIGURE 7 - Hewlett-Packard Vector Signal Analyzer Display

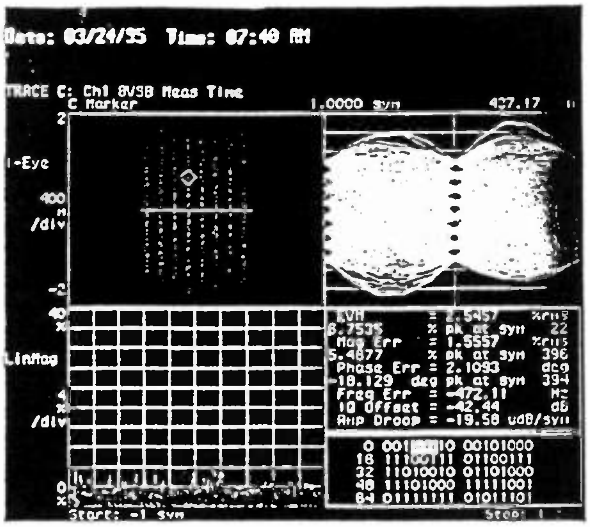


FIGURE 8 - Prototype Measured Spectrum

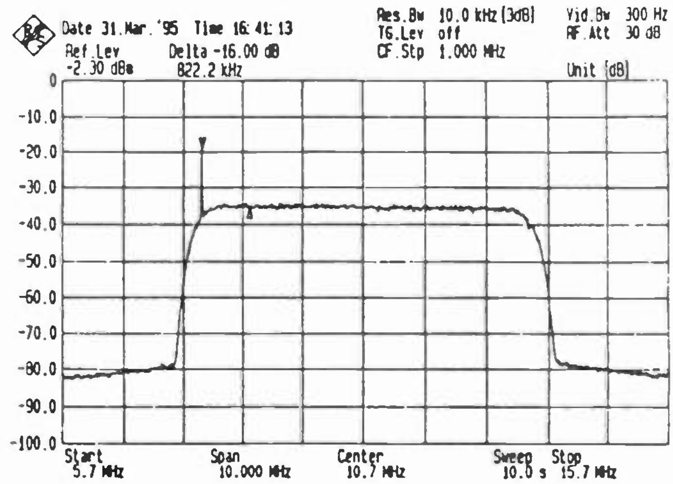


FIGURE 9

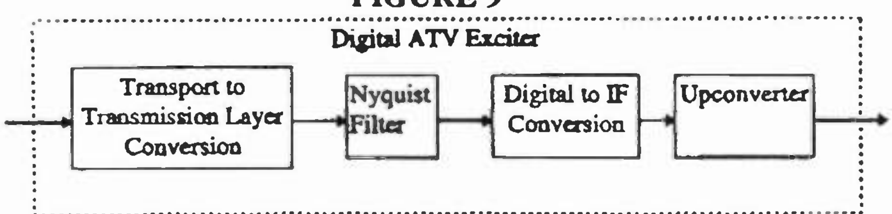
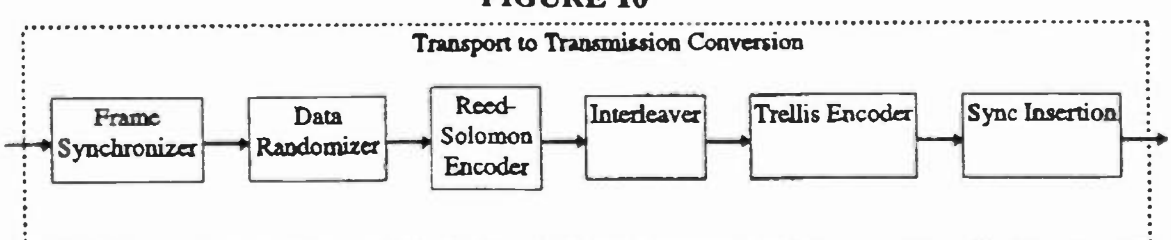


FIGURE 10



COVERAGE CONTOUR OPTIMIZATION OF HDTV AND NTSC ANTENNAS

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INTRODUCTION

The availability of suitable locations for new TV towers is diminishing even in the secondary markets. It is practically non-existent in major markets. Even after the hurdles of zoning variance and suitable tower location are overcome, FAA restrictions and environmental concerns may delay the construction of a new tower for years.

Not surprisingly, many broadcasters are looking at the pros and cons of using existing towers to support their new HDTV antennas even though the prime tower-tops are occupied.

For any given antenna, directional or omnidirectional, the tower will modify the *as-designed* antenna patterns. For optimum coverage, the *as-installed* patterns must be known not just at the carrier frequency but throughout the entire channel before the relative position of the antenna and its azimuthal pattern orientation can be fixed. There is one, and only one, position which would provide the optimum coverage without exceeding the structural limitations of the tower. The optimum position can be calculated.

This paper will describe the application of the optimization process to HDTV and NTSC antennas on a typical triangular tower with a 10' wide face.

TRANSMISSION TRANSFER FUNCTION

Figure 1 shows a side-mounted HDTV antenna positioned 8' from the center of a typical triangular tower with a 10' face. The tower contains two runs of 6-1/8" transmission line and one run of 18" circular waveguide. The tower lattice, the tower legs and the components inside the tower intercept a portion of the energy radiated by the HDTV antenna. The intercepted energy is then scattered in all directions. The scattered energy is combined, constructively in some directions and destructively in other directions, with the primary energy of the HDTV antenna.

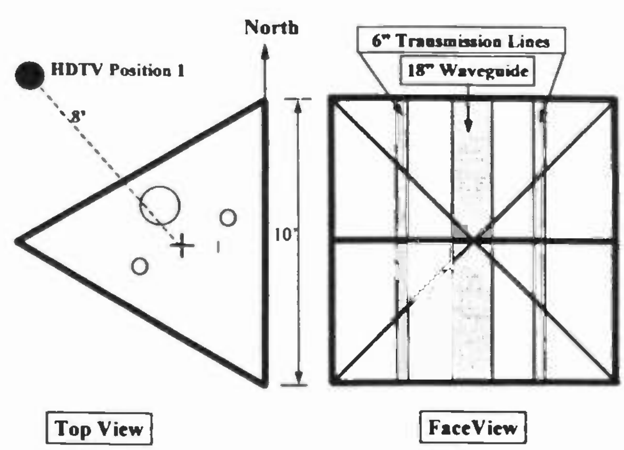


Figure 1: HDTV ANTENNA POSITIONING RELATIVE TO TOWER CONFIGURATION

The total field of the antenna with the tower, relative to the primary field of the omnidirectional antenna without the tower is shown in Figure 2.

The patterns shown in Figure 2 were calculated at the mid-frequency of channel 38. Even if the transmitter and the antenna system are assumed to exhibit a constant amplitude and linear phase transfer function (output/input) across the channel and in all directions, the scatter produced by the tower structure will distort that transfer function, at least in some directions.

As an example consider the transfer functions of the distorted pattern shown in Figure 2 in the azimuthal directions of -71^0 and $+113^0$. The two transfer functions are plotted in Figure 3. In the direction of -71^0, the system's transfer function remains undistorted with an overall power gain of 1.3 dB. In the direction of $+113^0$, the system's transfer function is distorted, with power loss depending on frequency. For simplicity, the phase distortion is not shown. The antenna system is analogous to a network with multiple output ports, one in each direction of interest. Moreover, the transfer function at each output port may be different than that of other ports.

Knowledge of the transfer function of NTSC antenna systems is not necessary for contour optimization because NTSC is a narrow-band transmission. Most of the NTSC picture information is contained within 1.5 Mhz of bandwidth. For this reason, the patterns are calculated at a single frequency, normally the carrier frequency, and are *assumed* constant within the critical bandwidth of 1.5 Mhz.

Because knowledge of the transfer function of NTSC antenna systems is not necessary, the optimization of NTSC contours is based on the calculation of the distorted field pattern such as that shown in Figure 2, at the carrier frequency only. The pattern may be optimized by changing the location of the antenna relative to the interfering structure and by recalculating the distortion at the carrier frequency until an acceptable pattern is achieved. Increasing the distance between the NTSC antenna and the

Figure 2: RELATIVE FIELD OF HDTV ANTENNA AT MIDBAND OF CHANNEL 38

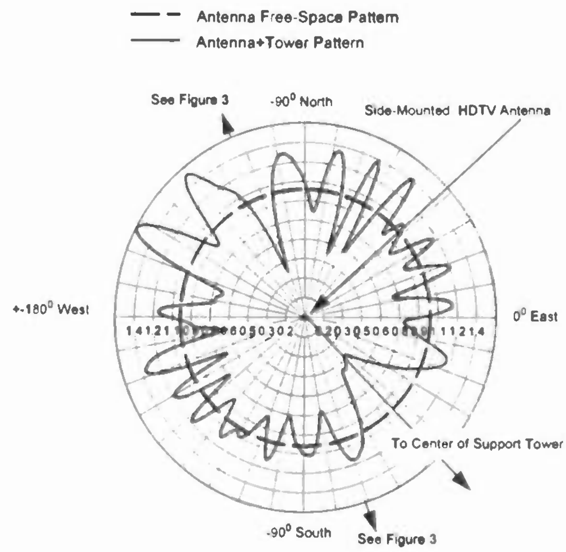


Figure 3: POWER TRANSFER FUNCTION

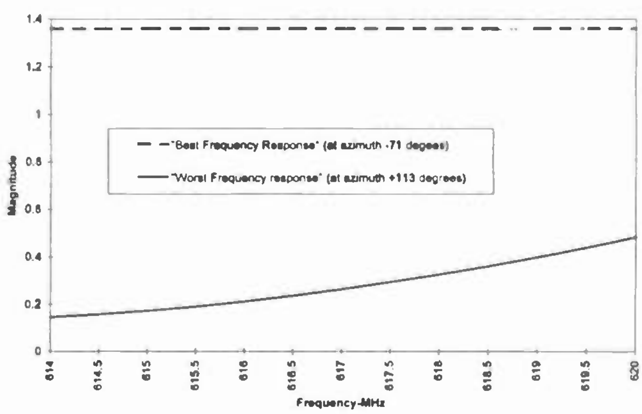
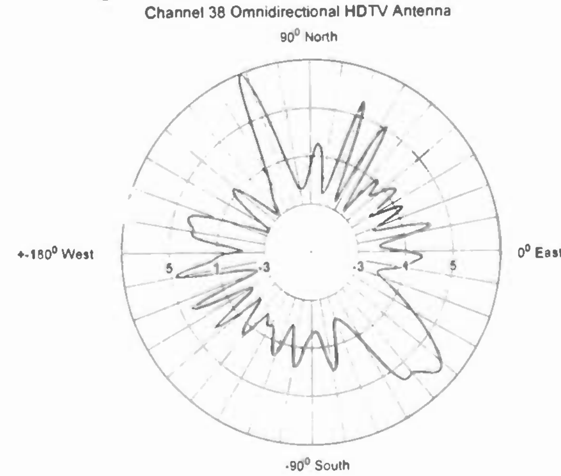


Figure 4: PENALTY AGAINST FREE-SPACE ERP (+-dB)
Channel 38 Omnidirectional HDTV Antenna



interfering structure is generally advantageous as long as the distance remains ≤100'. At larger separations the reflections resolve into visible ghosts. The end point of NTSC optimization is the starting point of HDTV optimization.

Knowledge of the transfer function for HDTV antenna systems optimization is necessary because HDTV is a broad-band transmission. All of the HDTV picture and sound information is spread equally within most of the 6 Mhz of bandwidth. A carrier is not visible in the channel's spectrum, and good response at *all frequencies* within the channel is necessary for optimum transmission. For this reason, the patterns must be calculated throughout the channel. An HDTV pattern calculated at a single frequency is not sufficient to determine the coverage unless the transfer function in each direction is known.

The number of cycles in the transfer function, and the possibility of creating a notch in it, depend on the distance (in wavelengths) between the antenna and the interfering structure. Because the antenna in Figure 1 is only 8' away, less than 1/4 of a cycle is evident in the direction of $+113^0$, as seen in Figure 2. A notch in the frequency response would cause a loss of HDTV picture and sound due to the high penalty, as explained in the next section.

COVERAGE PENALTY

Once known, the transfer function can be processed to determine the total penalty, in each direction, that must be assessed against the undistorted pattern of Figure 1.

The total penalty, expressed in ± dB, is composed of two factors. The first factor expresses the loss (or gain) of the total signal power within the 6 Mhz channel. For example, the penalty due to the loss of power in the direction of $+113^0$ is 5.55 dB even though the power is down 8.4 dB at the low end of the channel and 3.2 dB at the high end of the channel. The second factor expresses the loss of carrier-to-noise ratio margin caused by the equalizer at the HDTV receiver. The equalizer, in attempting to flatten frequency response will increase the system's gain and noise at selected frequencies thereby lowering the carrier-to-noise ratio margin[1]. The equalizer penalty at $+113^0$ is 2.30 dB. The total penalty in that direction is 7.85 dB. The total penalty (± dB), in any direction, against the undistorted pattern of Figure 2, is shown in Figure 4.

Coverage penalty is unique to HDTV because all undesired energies, such as reflections, translate into a loss of coverage, whereas in NTSC the undesired energies translate into a loss of picture quality.

COVERAGE OPTIMIZATION EXAMPLES

The objective is to optimize the coverage of a side-mounted, directional antenna, such as to provide service as similar as possible to that of the omnidirectional antenna mounted on the tower-top. The viewer population is not evenly distributed around the tower. The population density is heavy in the NW direction (city of license) and sparse in the SE direction. The tower and the antenna's location in the NW quadrant are shown in Figure 1.

Two different antennas, a broad-band panel antenna and a single channel slotted-pipe ("pylon") antenna, with cardioid azimuthal patterns shown in Figures 5a and 6a were selected.

The first step is to minimize the distortion of the azimuthal pattern due to the tower effects at the mid-channel frequency for HDTV and at the carrier frequency for NTSC. This is done by

[1] The processing of the transfer function into an overall penalty is detailed in the paper "A New Approach to the Analysis of Adjacent structure Effects on HDTV Antenna Performance," NAB Proceedings, 1995.

changing the antenna's position relative to the tower and by rotating the antenna on its axis to orient the antenna's pattern within the service area.

The results of this first step are shown in Figures 5a and 6a. The position of the antenna is 8' from the tower's center. The direction from the antenna to the tower's center is -80^0. The direction of the center-of-symmetry of the antenna's pattern is $+122^0$.

At this point, if the objective were NTSC coverage optimization, the panel antenna of Figure 5a would be chosen because it is less directional than the pylon antenna, and as such, would provide coverage over a wider sector than the pylon antenna.

For HDTV coverage optimization, the same conclusion cannot be reached without assessing the appropriate penalty. The penalties on the two antennas are shown in Figures 5b and 6b. The sector of no service behind the support tower is twice as wide for the panel antenna as it is for the pylon antenna.

The advantage of the pylon antenna for this HDTV application becomes clearer with the calculation of the FCC contours (flat terrain) for the two antennas after the penalties were assessed against the as-designed antenna patterns. The use of the FCC contours here is not to depict actual service but to compare the relative merits of the two antennas. Actual service determination requires the knowledge of actual terrain and cochannel interference.

CONCLUSION

NTSC coverage optimization is the process of maximizing the radiated power *at the carrier frequency* within the area of critical service. The scattering from nearby structures will affect the picture quality but will otherwise play a secondary role in determining the coverage contours. A notch in the frequency response across the channel, if not at any of the three carriers (picture, color and sound), may not significantly affect program viewability. However, the availability of the NTSC carrier does not guarantee the availability of HDTV signal.

HDTV coverage optimization is the process of reducing the power penalty in the area of interest, possibly at the expense of areas where viewers are sparse. The penalty depends on the *total signal power* within the channel and the *frequency response* across the channel. In particular, a notch (or nearly one) in the frequency response, anywhere within the channel, will cause a loss of HDTV signal.

Figure 5a: RELATIVE FIELD OF CARDIOID PANEL ANTENNA AT MIDBAND OF CHANNEL 38

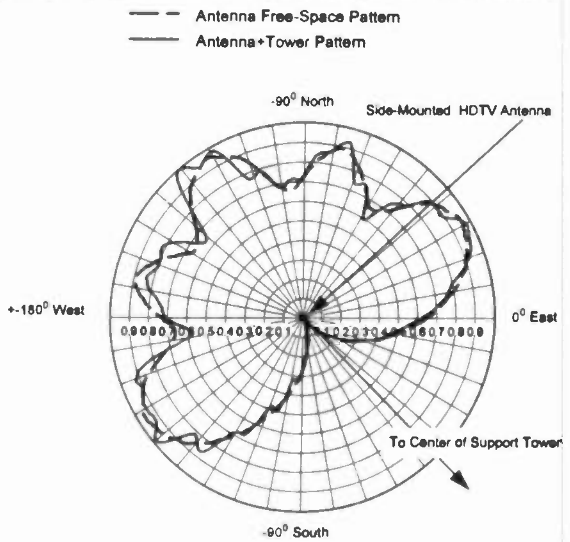


Figure 5b: PENALTY AGAINST FREE-SPACE ERP (+-dB)
Channel 38 Cardioid Panel Antenna

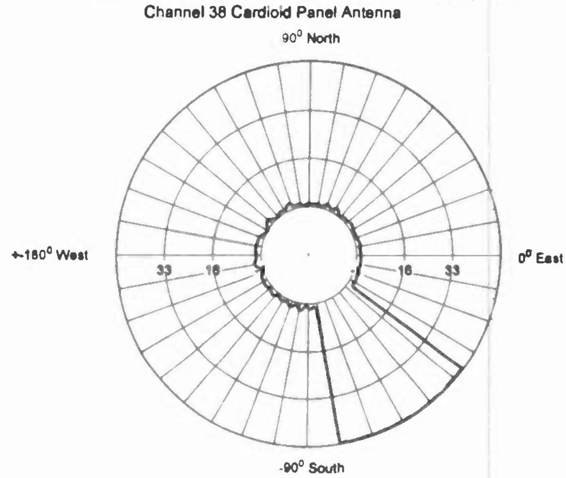


Figure 5c: FCC HDTV CONTOURS OF CARDIOID PANEL ANTENNA
CHANNEL 38; 500 kW HDTV AERP; 1300' HAAT

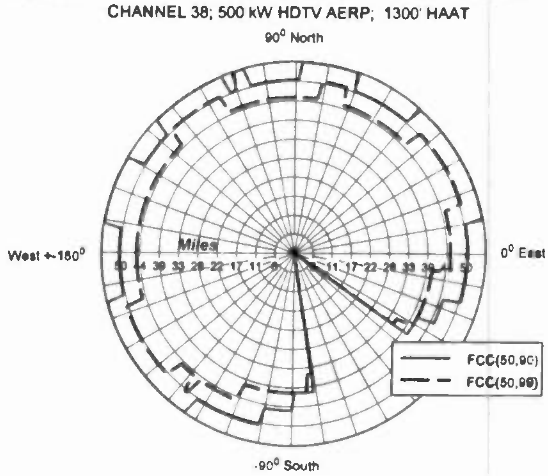


Figure 6a: RELATIVE FIELD OF C170 CARDIOID PYLON ANTENNA AT MIDBAND OF CHANNEL 38

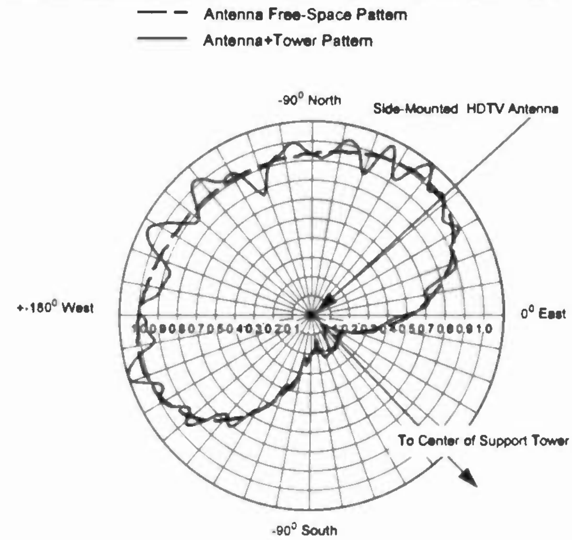


Figure 6b: PENALTY AGAINST FREE-SPACE ERP (+-dB)
Channel 38 C170 Cardioid Pylon Antenna

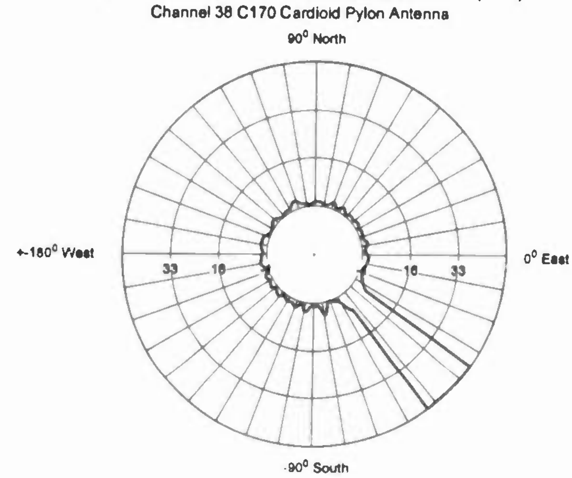
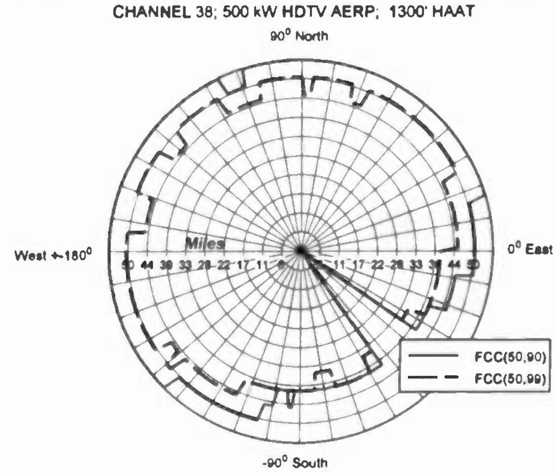


Figure 6C: FCC HDTV CONTOURS OF C170 CARDIOID PYLON ANTENNA
CHANNEL 38; 500 kW HDTV AERP; 1300' HAAT



A 2/3-INCH 2 MILLION PIXEL CCD CAMERA FOR HDTV

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1.Introduction

Some new television systems, namely HDTV(high definition television) and EDTV(extended definition television), are now entering practical use. The trend is towards higher picture quality and wider screen. ATV(advanced television) has also been proposed, and it would appear that we are entering an era of diverse television systems.

In this situation, cameras used for broadcasting must offer to respond to various television systems with keeping high quality picture and holding wider screen. The cameras are also requested greater operating convenience, higher mobility and suitability for various purposes so as to shoot quickly and easily.

We have recently developed a camera that can produce high quality pictures for use in various television systems. It is also the compact, light-weight and power-efficient, with operating convenience and mobility equivalent to those of popular cameras used for CDTV(conventional television) systems. Special effort was made to ensure that the new camera would be available at the lowest possible price through the use of components of CDTV cameras. This camera is expected to play a major role in the television of the future.

2.Fundamental concepts

The functions and performance of the new camera system are based on HDTV cameras, but it was also designed for use with both CDTV and EDTV. The fundamental concepts of the new camera are as follows:

(1)Picture quality --- The camera's basic functions should produce sufficient picture quality and performance for use in HDTV broadcasting.
(2)Versatility --- It should be compact and suitable for various uses, including outside broadcasting, studio shooting and use wherever handy production equipment is necessary.
(3)Compatibility --- To be usable for CDTV, the camera system should have a built-in down-converter.
(4)Operating ease and mobility --- The camera must have the same operational convenience and mobility as a conventional CDTV camera.
(5)Cost --- To minimize the costs, the camera should be designed as simply and with as many commonly usable parts as possible for use in studio and outside broadcasting vans and also as handy equipment. The camera should be designed to fit conventional casing, and to use the peripheral accessories and lenses of current CDTV systems. It should be compatible with conventional peripheral CDTV equipment, including the lens and camera cable, without modification.

3.Usage

Two types of usage were assumed for the new camera, namely, handy application and outside broadcasting/studio application. The camera system we have developed this time is capable of these two types of usage with the same camera. We have thus broadened system usage so as to reduce equipment cost and improve the operating convenience of the camera system.

Figure 1 shows the new camera's handy application. In this application, the camera will be usable as a simplified field-production camera which has minimal functions necessary for field-production and news-gathering purposes. It is connected with a portable VTR with a 26-core analog cable provided in the BTA S-1005A regulations. The maximum length of the cable is 100 meters.

The outside broadcasting/studio application is presented in Figure 2. In this type of usage, the handy camera, with a buildup unit added, can be equipped with a box-type lens and a 7-inch viewfinder so that the camera may be used in the studio or aboard an OB van. The camera system uses optical-fiber cable which can cover the maximum distance of 3,000 meters between the camera head and the CCU(camera control unit).

Figure 1 : Handy Application

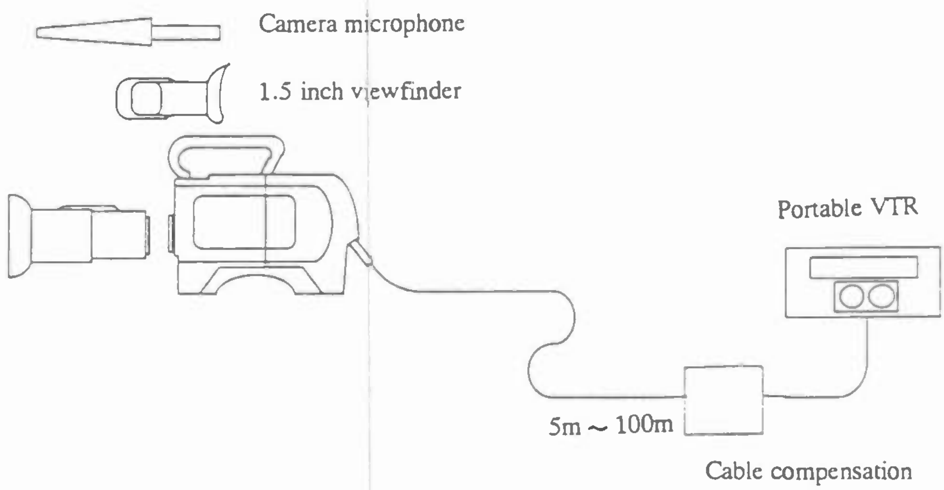
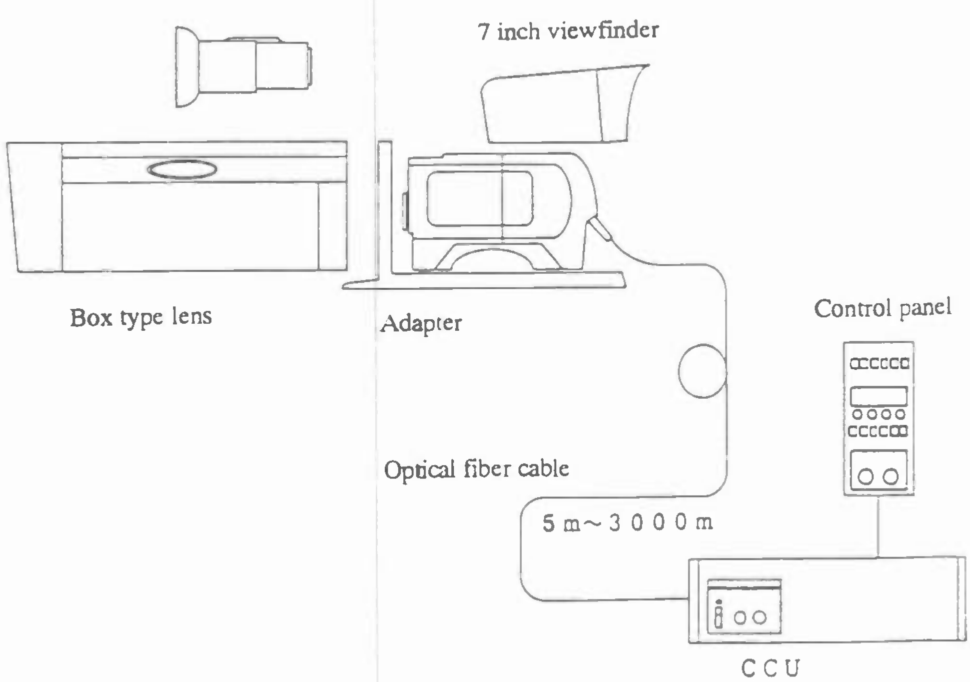


Figure 2 : OB/Studio Application



4. Pickup device

For the image pickup function of the new camera, we have employed newly developed 2/3 inch two-million pixel CCD. The small-size CCD has contributed to reducing the camera size and weight. It is possible to use various 2/3 inch lenses.

4.1 Development of Smaller CCD

To develop a HDTV camera for broadcasting program production that is small, inexpensive and consumes small amount of electricity, it has been necessary to reduce from 1 inch to 2/3 inch the size of the optical system of the CCD pickup device of the camera.

This size reduction from 1inch to 2/3 inch is approximately halves the unit cell area (Figure 3). The CCD performance largely depends on cell area, but 2/3 inch CCD is required the same performance as 1 inch size.

We have developed a 2/3 inch, FIT (frame interline transfer) CCD device with 2 million pixels. Figure 4 shows its elements structure, composed of a 2/3 inch optical system, an image area measuring 9.6 mm (horizontal) x 5.4 mm (vertical). The basic structure is the same as a 1 inch FIT model with 2 million pixels, having effectively 1,920 (horizontal) x 1,036 (vertical) pixels. The horizontal shift register operates at 37.125 MHz for 2-line.

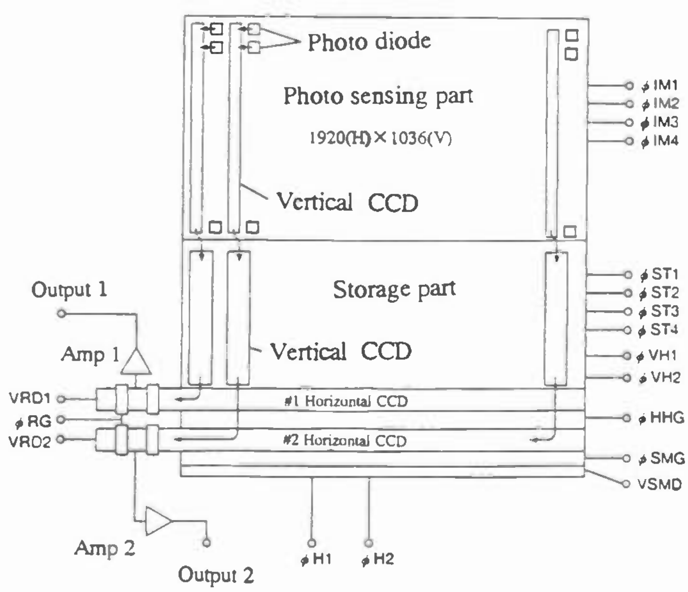


Figure 4 : CCD element structure

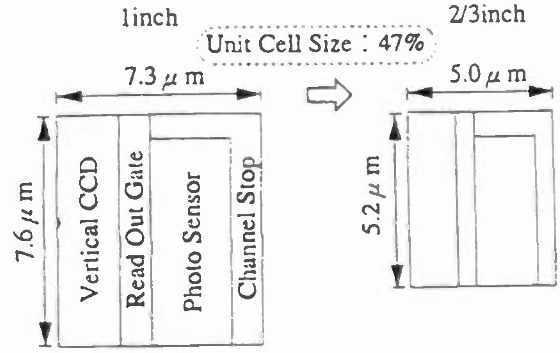


Figure 3 : Comparison of the 2/3 inch CCD unit cell with 1 inch

4.2 Technology for improvement

4.2.1 Smear reduction

For a broadcasting program production camera, it is crucial to hold down smear as much as possible, and the use of an FIT model is a natural choice. A FIT CCD (at a frame shift frequency of 770KHz) reduces smear 27 dB more than an IT type (at a frame shift frequency of 33KHz) because of difference of frame shift frequency. The shortcoming with FIT, however, is that the amplitude of output voltage gets smaller because of from the transmission delay in the high-speed clock, used for frame transfer, which then limits the amount of electron that can be processed.

A solution to this problem is the development of an aluminum lining structure with a poly-silicon buffer for a 1 inch CCD with 2 million pixels (see Figure 5). This technique utilizes high-speed FIT to reduce the transfer delay. The lined wiring is also used to shield light for the vertical shift register, which reduces smear as well as transfer delay. In the poly-silicon buffer, the poly-silicon and the aluminum of the electrode come in direct contact with each other, which causes the channel potential to change and consequently prevents the transfer efficiency of the vertical CCD from deteriorating. Smear presented the largest challenge to the size reduction from 1 inch to 2/3inch. This 2/3inch model has had its structure further improved and the on-chip micro-lens shape optimized. The result is a greatly improved ability to suppress smear level to -120dB at a frame shift frequency of 770 KHz.

4.2.2 Higher sensitivity

As shown in Figure 3, size reduction from 1 inch to 2/3 inch approximately halves the cell area, which seriously lower the level of sensitivity. To retain sensitivity as high as that of a 1 inch model, we have further improved the shape of the on-chip micro-lens. Moreover, by better matching the electrode and using a thinner electrode material, we were able to reduce the degree of eclipse in an area from the on-chip lens to the light converging section, resulting in higher light converging efficiency (Figure 6). Sensitivity deterioration due to the size down of the CCD was improved by means of optimizing the electric charge detector at the CCD output.

4.2.3 Flare reduction

We formed a low reflection film over the light-shielding aluminum in order to reduce flare caused by light reflecting from the surface of the device. This arrangement provides us with a high-quality picture without unnecessary black level floating than ever before even when luminosity is high.

4.3 Performance of the CCD device

Table 1 compares the new 2/3 inch model with conventional 1 inch CCD with 2 million pixels. At 75mV/lx, the new model has about the same sensitivity as the 1 inch version. We achieved the smear level -120dB in 2/3 inch model, it is an improvement of some -20 dB from the 1 inch model.

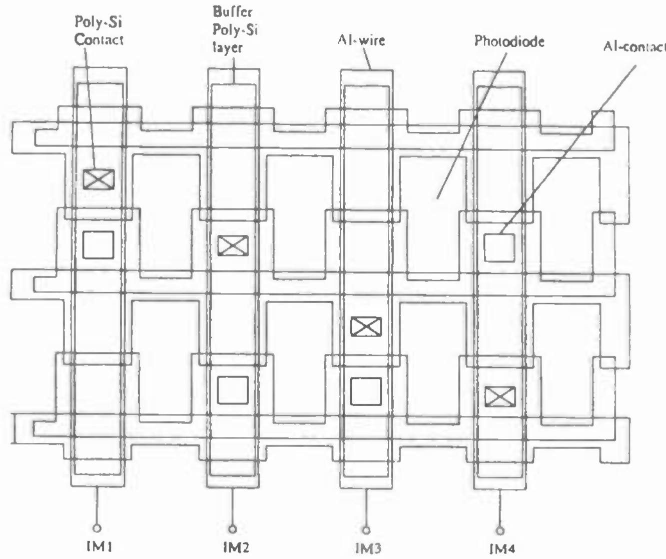


Figure 5 : Aluminum lining structure with a poly-silicon buffer

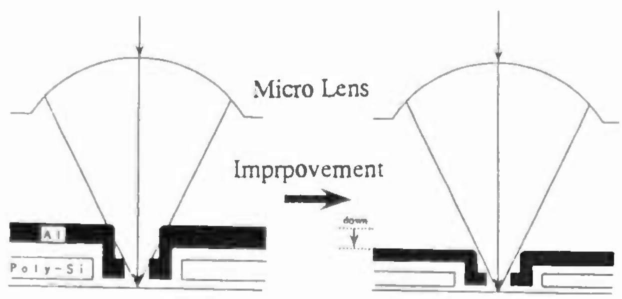


Figure 6 : Improvement of light condensing

	2／3 inch	1 inch
Image area	9 . 7mm (H)×5 . 4mm (V)	14 . 0mm (H)×7 . 9mm (V)
Unit cell size	5 . 0μ m (H)×5 . 2μ m (V)	7 . 3μ m (H)×7 . 6μ m (V)
Sensitivity	75mV／lx	80mV／lx
Saturation voltage	500mV	500mV
Dynamic range	70dB	72dB
Smere	−120dB	−100dB

Table 1 Comparison of the 2/3 inch CCD perfoormance with 1 inch

5.System configuration

Figure 7 shows a block diagram of the new camera. The camera has a self-contained mechanism which includes the image processing circuit in the camera head so that the camera system may be used either for the handy application or for the outside broadcasting /studio application. The camera head also contains circuits for analog-digital, parallel-serial and electrical-optical converters. Using these functions, the camera system performs both operations handy application and the outside broadcasting/studio application on the same camera.

For the outside broadcasting/studio usage, we have equipped the camera with a high-resolution 7-inch viewfinder so that the camera man may focus easily. The viewfinder uses a cathode-ray tube with a 4 by 3 aspect ratio. The CRT screen is fully used for pictures with a 4:3 angle of view, while its top and bottom portions are masked for 16:9 pictures. The viewfinder is thus capable of television systems using pictures with different aspect ratio.

5.1 Camera

We have developed an anti-resonance horizontal drive circuit for the CCD which was able to reduce power consumption at the CCD drive circuit where high frequency of 37MHz and high voltage were required. We have also developed a low-noise high-speed CDS(correlational double sampling) amplifier which was used for detecting the signal from the CCD output. For signal processing, the camera is equipped with a multifunctional process circuit with low power consumption and a digital-delay image enhancer. These features have made this HDTV camera as handy as the present regular TV camera and as functional as the 1 inch 2 million pixels HDTV camera. This portable camera can also be used as a multipurpose camera. It is also equipped with a 26-pin interface (BTA S-1005A) for the connection with portable VTR and an optical interface for the connection between camera head and CCU. The body has the same design as the present camera, which also helps lower the price.

5.2 Signal transmission between camera head and CCU

The CCU is connected to the camera by an electric-optical complex cable comprising two single-mode fibers, four power supply wires, and two control wires. The signal is sent by optical serial digital transmission according to the SMPTE 292M standard (Y Pb Pr 4:4:2, 10bit, 1.5Gbps). It can be transmitted over a distance of 3,000m through serially connected 250m cables. This digital transmission has eliminated a number of problems (signal transmission time skewing, cable equalization, increased noise, electromagnetic interference, etc.) that limited the use of multicore cables, thus the digital optical transmission is enabling transmission of high quality picture and audio over a long distance.

5.3 CCU

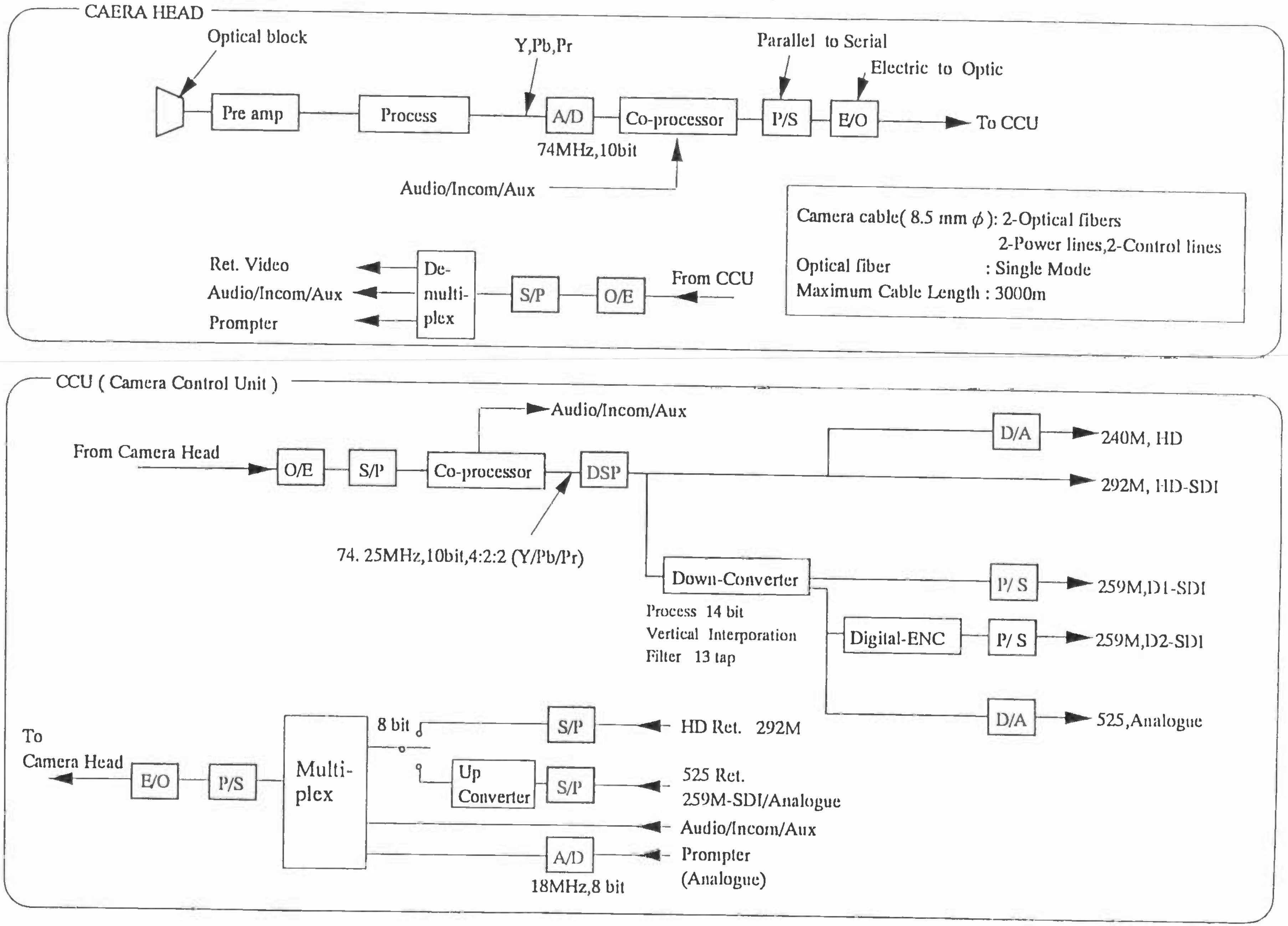
The camera control unit (CCU) of the new camera system can input/output both analog (SMPTE 240M) and digital (SMPTE 259M, 292M) signals so it may be used in various circumstances which include many peripheral equipment. Also the CCU has 4 input ports for serial digital signals (SMPTE 292M) for return picture and all these ports can return 4:2:2 color HDTV signals. Such color return picture can be identified on the camera head.

The CCU is also equipped with a line-change down-converter and a simple up-converter for return pictures. These converters can be used for 16:9 and 4:3 letterbox aspect ratios; the down-converter comes with an image enhancer and a color-correction unit for CDTV color production.

With these builtin converters, this HDTV camera can also be used for program production under conventional TV systems. The output signal of the CCU is serial digital which can be used for both component digital and composite digital(SMPTE 259M).

The camera control panel of the conventional TV camera can also be used for this new HDTV camera, which enables the same operation as that of conventional TV and effects lower price by using common parts of conventional TV.

Figure 7 : Camera Block Diagram



6. Functions and performance

Tables 2 show the basic functions and performance including weight and power consumption of the new camera. Table 3 shows the basic performance of the down-converter used along with the camera system.

7.Peripheral equipment

7.1 Standardization of the optical cable and connector for the camera

Television camera cables are permanently laid at stadiums and ball parks. And, to cover big events, such as the Olympic Games, different production teams are required to jointly use the same apparatus and equipment. For these reasons, it is desirable that future camera cables and connectors will be compatible with cameras made by different manufacturers. We have proposed national common standards for camera cables and connectors. Figure 8 shows the cable specifications we have proposed.

7.2 Standardizing of the box-type lens

The box-type lenses currently used in the conventional TV systems come with the mount with different shapes. The new camera must have a mount which allows the interchanging with CDTV lenses and between new camera lenses. We have determined the shape of the

Table 3 Specification of the Down-converter

Items	Contents
A/D converter	10 bit
Sampling frequency	74.25Mhz (input) 13.5Mhz (output)
Internal calcuration	14 bit
Horizontal filter	25 taps
Vertical filter	13 taps

Table 2 Specification of the camera

Items	Contents
Sensitivity	F8 (at 2000 lux)
S/N	54dB (Y signal, 35 mV)
Resolution	MTF 45% (Y signal, center 800 TVL)
After image	below the limit of measurement (3 field after image)
Dynamic range	600% (R,G,B each channel)
Smere	-120 dB(R,G,B each channel)
weight (camera head)	5.9Kg (1.5inch viewfinder including)
Power consumption (camera head)	30W (1.5inch viewfinder including)

8.Conclusion

We have developed an HDTV camera system with a new 2/3-inch two million pixel CCD. It is easy to use and has mobility equivalent to that of conventional TV cameras. The cost can be brought down to a level comparable with that of CDTV cameras. This camera performs well with F8 sensitivity, 54 dB SN ratio and -120 dB smear.

We employed 1.5 Gbps serial digital technology for the camera-CCU transmission and the CCU video interface to improve the operating efficiency of the new camera system.

The CCU also incorporates a converter so that the new camera can be used in both HDTV and CDTV. The new camera system is epoch-making in that it has eliminated the shortcomings of existing HDTV cameras and is compatible with future television systems. We believe that the new camera will be useful to new broadcasting systems and should contribute to the production of better quality programs.

9. Acknowledgements

The authors wish to thank their many colleagues at Sony Co.Ltd. and Mr.Andou Deputy Director in NHK for their valuable technical contributions to the work described.

10. References

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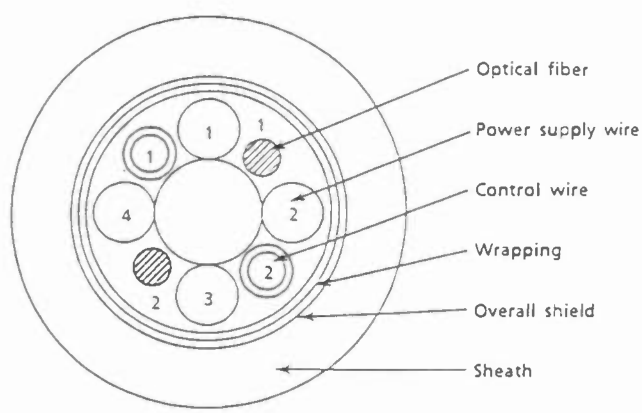
Figure 8 : The proposed cable

Items	Contents
Optical fiber	Single mode
Power supply wire	37.0 /Km
Control wire	113 /Km
Cable diameter	9.2mm
Cable weight	105 Kg/Km



DAB: U.S. AND WORLDWIDE PLANNING: PART II

Sunday, April 14, 1996
1:00 - 5:00 pm

Session Chairperson:
Milford Smith, Greater Media, Inc., East Brunswick, NJ

DIGITAL AUDIO TRANSMISSION ANALYZER
Masaaki Sasada
Tokyo Broadcasting System
Tokyo, Japan

***A REPORT ON DAB FIELD TESTING IN THE UNITED STATES**
Bert Goldman
Patterson Broadcasting
Savannah, GA

A NOVEL PHASE-COMPENSATED PSK MODULATION TECHNIQUE FOR MOBILE DAB
Jin-Soo Lee
Tae gu Mun-Hwa Broadcasting Corporation
Tae gu, Korea

DIGITAL AUDIO BROADCASTING USING COFDM MODULATION
Barry G. Tew
Harris Corporation Broadcast Division
Cambridge, England

BRINGING DAB TO THE CONSUMER
David Witherow
BBC, President of the European DAB Forum
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EUREKA 147 - TOWARDS A DE FACTO WORLD DAB STANDARD
Franc Kozamernik
European Broadcasting Union
Geneva, Switzerland

AN OPEN FORUM ON WORLDWIDE DAB DEVELOPMENTS - ALL PRESENTERS
Moderator: Lucia Cobo
Radio World Newspaper
Falls Church, VA

*Paper not available at the time of publication.

DIGITAL AUDIO TRANSMISSION ANALYZER

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Abstract

Tokyo Broadcasting System has digitalized all the video and audio systems in its new Broadcast Center, which was completed in 1994.(1)

The audio equipment in the new systems requires the coupling of synchronization and timing control to the digital audio signal. This instrument " Digital Audio Transmission Analyzer" was developed to measure serial transmission of the digital audio signal.

Process of Development

So long as an analog base band signal is used in audio, it is not necessary to consider the timing control and the coupling of synchronization of the audio signal. However, timing control means a phase difference in the left and right stéreo channels.

Increasingly, so digital technology has been introduced into the audio area of the broadcasting station. When connecting equipment within and between digitalized systems, the protocols require digital input/output. The establishment of the AES/EBU standards as the serial tranmission standards for two-channel digital audio has contributed to the digitalization of audio equipment. The AES/EBU format signal was adopted by broadcasting stations, and after several improvements, AES3-1992 and EBU Tech. 3250-E are now in use. Much digital equipment is manufactured to conform with these standards. Early inconsistencies in flag or channel satatus data interpretation have gradually been fixed, and measuring instruments corresponding to the AES/EBU standards are now on the market. The instruments now available measure validity, user, parity bits, and channel status data, but they cannot confirm the synchronization of audio signals.

For digital transmission at broadcasting system, synchronization and timing control are necessary. The necessity of timing control is most significant when multiple signals are fed to and mixed at the audio mixing console. Equipment that although an internal buffer for the timing control is widely used, some types of equipment do not have an internal buffer for delay reduction. They require an external buffer(synchronizer) for the unsynchronized input signal to make the same input timing.

An oscilloscope has been used to observe signal timing. A preamble in the subframe does not follow the bi-phase coding rule. Preambles "X" ,"Y", and "Z" have the patterns shown in Table 1:

Table 1.

PREAMBLE	CHANNEL CODING	
	Preceding state:0	Preceding state:1
"X"	11100010	00011101
"Y"	11100100	00011011
"Z"	11101000	00010111

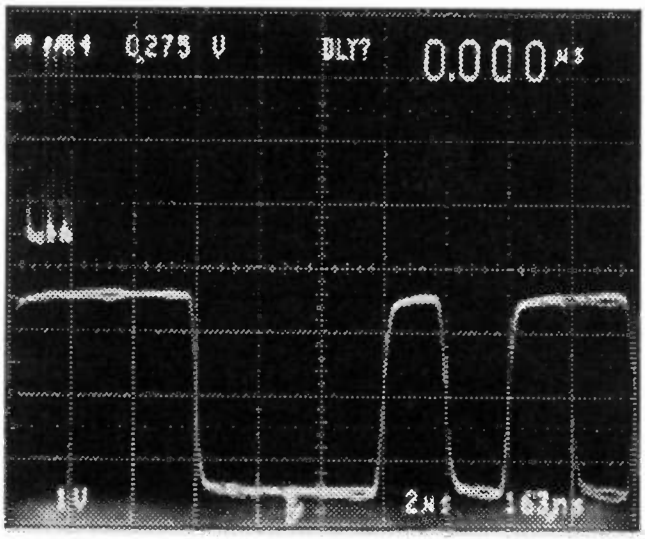

Fig. 1 "X" preamble

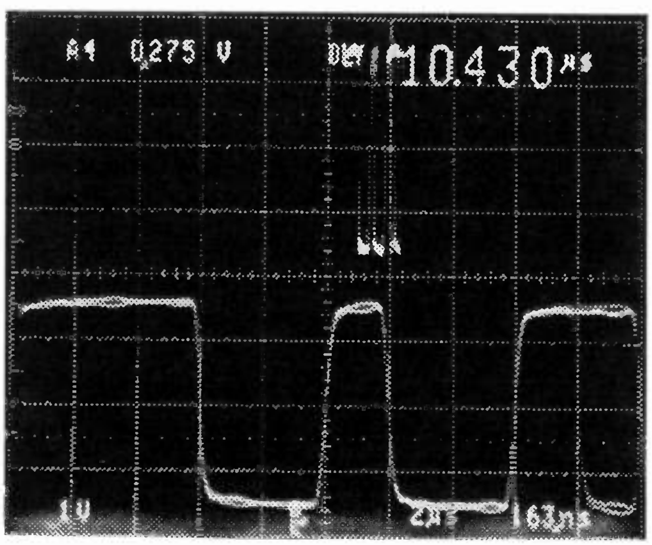

Fig. 2 "Y" preamble

The "X" preamble is shown in Figure 1, the "Y" in Figure 2, and the "Z" in figure 3. The "Z" preamble signal appears in only 1 of 192 frames. The photos were taken from an oscilloscope.

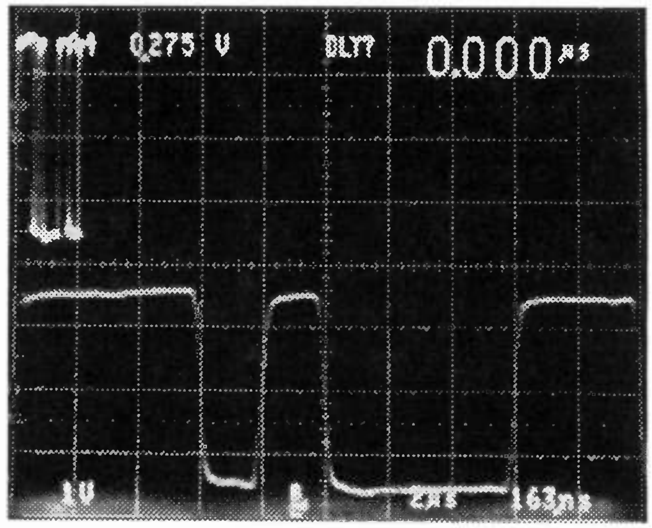

Fig. 3 "Z" preamble; locatehead of a block alternated "X" preamble

An oscilloscope is useful in observing digital signals. With an oscilloscope, such readings as amplitude and jitter can be observed correctly. However, greater skill in the use of an oscilloscope is necessary to locate the "Z" preamble, and location takes time. In addition, to find temporary errors it is necessary to look at the display continuously, and that is inconvenient. When, in 1992, a radio station was built that connected all the digital serial signals, the necessity was recognized of instruments to measure the AES/EBU signals particular timing and digital items.(2) Engineers who constructed the digital radio broadcasting system used oscilloscopes for timing adjustment. In 1992, however, it was impossible to measure directly the coupling of synchronization. Now, with the digital audio transmission analyzer, it is possible, for the first time, to measure the timing between two AES/EBU format signals.

Purpose of development and measurement items

The major purpose of the digital audio transmission analyzer is timing control, using preambles with the AES/EBU format audio signal. Another important purpose is the confirmation of the long-term stability of transmission equipment. The analyzer is effective in the capture of noise, in the synchronization of input/output, and in finding clock jitters, with a low probability of generation or temporary phenomena.

The development of the analyzer focuses on the following points particular to the AES/EBU format:

Measurement

1. Timing between two AES/EBU signals (80 nsec accuracy)
2. Timing to external word clock
3. Timing to external NTSC black burst signal
4. Jitters (max 40.7 nsec—min 407 psec p-p)
5. Amplitude of serial input signal

Display

1. Serial input: ON/OFF
2. Polarity of input
3. Lock: IN/OUT
4. Sampling rate
5. Audio levels

The analyzer can detect and display errors in the frame and subframe format.

Error Detection

1. Confidence erros (eye diagram)
2. Out of Lock error
3. Bi-phase coding error
4. Parity error
5. Validity bit
6. Channel status bit
7. Slipped audio sample

In addition, sound monitoring is possible. Using a computer is convenient for the purpose of long-term observation. If a particular error pattern is found, error information with the data before and after the error is sent to an external computer for analysis through a GP-IB interface.

In the days of analog, noise search and source detection was difficult when noise had a low probability of generation. Digital systems, compared with analog, have a benefit: with them, it is possible to watch clean signal transmission by checking the parity bit or by other ways of inspecting data. Approriate system design should help make clear the benefits of digital systems. And to confirm the installation and maintenence of digital systems, appropriate instruments should be developed.

Block diagram

Figure 4 shows a block diagram depicting principles of measurement. In the past, such measurement was conducted by the combination of two instruments, but at present, only one instrument has the capability of numerical display. The "Digital Audio Tranmission Analyzer" use as reference AES/EBU signal, word clock signal or NTSC signal. The timing between two AES/EBU signals is measured at the location of the “Z” preambles. The “Z” preambles appear periodically at 4 msec in the signal alternated “X” preamble. The maximum number of timing measurement is 4 msec. Using a word clock (fwc =48 kHz, 44.1 Hz, or 32 kHz) for reference, the signal does not have a periodic mark point. The maximum number of timing measurement is 1/ fwc in this case. Using word NTSC video signal for reference, the digital audio transmission analyzer is used to measure for reference horizontal synchronization pulse (fH≈15.73 kHz). The maximum number of timing measurement in this case is 1/ fwc (20.83µsec at fwc=48 kHz). The word clock is generated by fH of the NTSC video signal using the following equation:

$$f_{wc} = \frac{1144}{375} f_H \qquad (1)$$

If using the color subcarrier (fsc≈3.58MHz) of the NTSC signal, direct dividing of the fsc signal is not accurately 48 kHz. Measuring timing to the reference signal, it is possible to confirm the coupling of synchronization. The minimum level of timing measurement (resolution) is 1/(256 x fwc). Relations of fsc, fH and fwc are shown in Fig. 5.

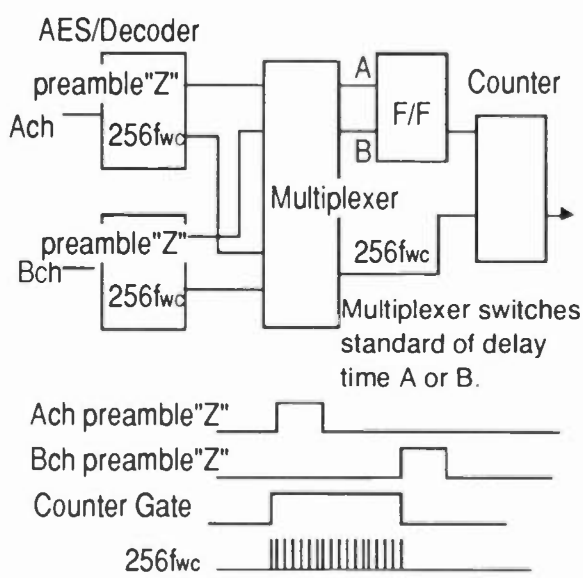

Fig. 4 Block diagram

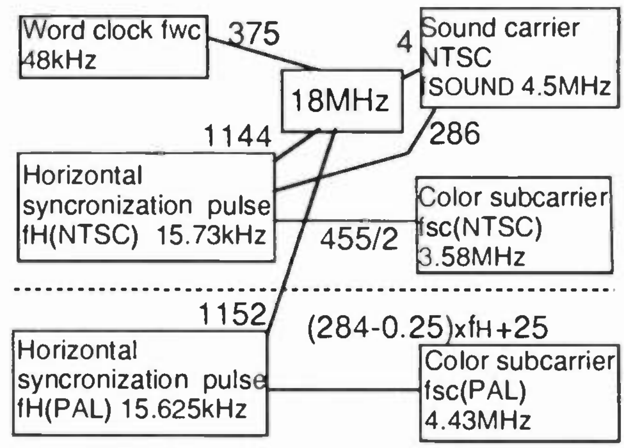

Fig. 5 Relation of Word clock frequency

CONCLUSION

The "Digital Audio Tansmission Analyzer" was developed for two purposes: One, the confirmation of the coupling of synchronization and the measurement of delay in audio systems; and Two, by the utilization of the major merit of digitalization, capture the noise generation of low probabilty in the system.

A measuring instrument that is easy to use will benefit the process of digitalization. The digital audio transmission analyzer can also be used in manufacturing and inspection. Lastly, we would like to acknowledge the efforts of the people at Shibasoku with whom we collaborated in development.

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A NOVEL PHASE-COMPENSATED PSK MODULATION TECHNIQUE FOR MOBILE DAB

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ABSTRACT

This paper describes a new digital modulation technique for mobile DAB, referred to PC-PSK (phase-compensated PSK), with phase compensation property, The basic idea in the proposed method is that time diversity for phase compensation. It has phase error compensation property over the phase uncertainty environments such as motion of the receiver, inhomogeneities in the propagation medium, AM/PM conversion effects at non-linear amplifiers. The maximum range of phase error compensation is from +90 to -90 degrees. We propose that it is suitable for mobile DAB system on slow-fading and time-varying Rician channel.

I. INTRODUCTION

DAB has the potential for delivering programming to the public with no degradation from the original source. However, because of mobile DAB receivers travel through all areas they have limitations in delivering signals to receivers which contain all the original information. The major sources of disruption of mobile DAB signal are propagation problems. Any unintentional phase modulation that is introduced into the received signal by the transmitter, by the propagation medium, or by the receiver is a source of phase uncertainty. This may be due to motion of the transmitter antenna or the receiver antenna that may be due to vibration; to phase errors arising in the local oscillators; to phase errors arising in the carrier recovery circuitry; or to phase errors due to inhomogeneities in the propagation medium [1,2]. The PC-PSK can compensate for phase error occurred due to the motion of a receiver antenna, multipath fading, and nonlinearity of amplifiers. The PC-PSK is based on the PAL TV system. PAL receivers have a high degree of immunity to defects in signal propagation and nonlinearity of transmitter and receiver circuit [3-5]. The constellation diagram of PC-PSK is quite similar to the vectorscope display of PAL chrominance signals. The PC-PSK corrects phase error between +90° and -90° by consecutively sending the phase-inverted image of the quadrature component of the signal with respect to the in-phase axis, where the phase error occurred in transmission is cancelled out in the process of adding the reinverted image of the received quadrature component. We simulate it on adaptive white Gaussian noise channel, slow-fading Rician channel, and time-varying Rician channel each for mobile DAB. The numerical results show that the PC-PSK can have better BER than that of the conventional PSK with the same bandwidth in the range of high SNR. The drawback of PC-PSK is its broad spectrum bandwidth. Thus, poly-ary PC-PSK will be considered for spectrum economy.

II. PRINCIPLES OF PC-PSK

In *M-ary* PC-PSK, the phase of the transmitted signal can take on any pair of M values in a given symbol interval while amplitude is maintained constant.

Thus, the signal set $S_i(t)$ can be defined by

$$S_i(t) = A\cos(\omega_o t \pm \Phi_m(t)),\quad 0 \le t \le T_s$$
$$= \begin{cases} I_1(t)\cos\omega_o t - Q_1(t)\sin\omega_o t, & 0 \le t \le T_s/2 \\ I_2(t)\cos\omega_o t + Q_2(t)\sin\omega_o t, & T_s/2 \le t \le T_s \end{cases} \quad (1)$$

where, A is amplitude, $\Phi_m(t) = \frac{(2i-1)\pi}{M}$, $i = 1, 2, \ldots, M$, and T_s is symbol period.

A block diagram of a PC-QPSK modem is shown in Fig. 1(a). The NRZ data stream entering the modulator is converted by a serial-to-parallel (S/P) converter into separate NRZ streams. One stream is in-phase, $I(t)$, and the other is quadrature phase, $Q(t)$.

One of the S/P converter output streams, the Q stream in the case shown Fig. 1(a), is then alternated by the Manchester-coder. Thus, Q streams have two values. These are opposite values between time intervals $0 \le t \le T_s/2$, and $T_s/2 \le t \le T_s$ each other. The resulting phase states at the modulator output are the same as symmetric twice QPSK with respect to I axis.

The basic circuit configuration of the PC-QPSK coherent demodulator is shown in Fig. 1(b). Under the phase uncertainty environment. the received signal at the detector output $S(t)$ consists of a modulated signal $S_m(t)$, noise $n(t)$ and intersymbol interference(ISI) $i(t)$ [6].

$$\begin{aligned} S(t) &= S_m(t) + n(t) + i(t) \\ &= A(t)\cos[\omega_0 t + \Phi_s(t)]. \end{aligned} \quad (2)$$

$$\Phi_s(t) = \begin{cases} \Phi_m(t) + \Phi_c(t), & 0 \le t \le T_s/2 \\ -\Phi_m(t) + \Phi_c(t), & T_s/2 \le t \le T_s \end{cases}$$

where ω_o is a carrier angular frequency, $A(t)$ is amplitude, $\Phi_m(t)$ is a modulated phase, and $\Phi_c(t)$ which includes $n(t)$ and $i(t)$. The product detector, which consists of multipliers as shown in Fig. 1(b), multiplies the received signal $S(t)$ and the reference signals $r_c(t)$ and $r_s(t)$ generated by the carrier recovery circuit with a phase 0^0 and 90^0, respectively.

$$\begin{aligned} r_c(t) &= \cos[\omega_0 t + \Phi_r(t)] \\ r_s(t) &= -\sin[\omega_0 t + \Phi_r(t)] \end{aligned} \quad (3)$$

The PC-QPSK demodulator detects baseband components from the multiplied signals.

$$\begin{aligned} \hat{I}_1(t) &= A(t)\cos[\ \Phi_m(t) + \Phi_c(t) - \Phi_r(t)] \\ \hat{I}_2(t) &= A(t)\cos[-\Phi_m(t) + \Phi_c(t) - \Phi_r(t)] \end{aligned} \quad (4)$$

$$\begin{aligned} \hat{Q}_1(t) &= A(t)\sin[\ \Phi_m(t) + \Phi_c(t) - \Phi_r(t)] \\ \hat{Q}_2(t) &= A(t)\sin[-\Phi_m(t) + \Phi_c(t) - \Phi_r(t)] \end{aligned} \quad (5)$$

If the carrier recovery circuit operates perfectly, i.e., $\Phi_r(t) = \Phi_c(t)$, the detected signals represent envelopes of the modulated original signal. In actual radio transmission, fading occurs frequently, and, in addition to the noise and ISI influences, $\Phi_c(t)$ varies randomly between 0^0 and 360^0. A conventional PSK demodulator cannot perform perfectly, i.e., $\Phi_r(t) \neq \Phi_c(t)$, which results in overall performance degradation.

In [7], it is assumed that the fading is slow enough so that at least over a few bits, the channel transfer function remains constant, in addition, it is assumed that the amplitude and phase fluctuations on a received carrier have the same statistical characteristic as that of narrow-band Gaussian noise.

(1) For $0 \le t \le T_s/2$, the half symbol time switch Si and Sq in Fig. 1(b) are in position ①. If there is a phase error $\Phi(t)$, the detected signal phase is changed into error region as shown in Fig. 2(d). These are called $\hat{I}_1(t)$ and $\hat{Q}_1(t)$ signals respectively.

$$\begin{pmatrix} \hat{I}_1(t) \\ \hat{Q}_1(t) \end{pmatrix} = \begin{pmatrix} \cos\Phi(t) & -\sin\Phi(t) \\ \sin\Phi(t) & \cos\Phi(t) \end{pmatrix} \begin{pmatrix} I_1(t) \\ Q_1(t) \end{pmatrix} \quad (6)$$

There are two memory devices M_I and Mq in the receiver that memorize $\hat{I}_1(t)$ and $\hat{Q}_2(t)$ signals as shown in Fig. 1(b).

(2) For $T_s/2 \le t \le T_s$, the half symbol time switch Si and Sq in Fig. 1(b) are in position ②. If there is also same phase error $\Phi(t)$, the detected signal phase is also changed into error region as shown in Fig. 2(e). These are called $\hat{I}_2(t)$ and $\hat{Q}_2(t)$ signals respectively.

For error compensation, it is necessary to re-reflect that return the $\hat{Q}_2(t)$ signal into $-\hat{Q}_2(t)$ signal with respect to I-axis. This is done by means of a subtraction as shown in Fig. 2(f). The combined $\hat{I}(t)$ signal will be $\hat{I}_1(t)+\hat{I}_2(t)$ and $\hat{Q}(t)$ signal will be $\hat{Q}_1(t)-\hat{Q}_2(t)$.

$$\begin{pmatrix} \hat{I}_2(t) \\ -\hat{Q}_2(t) \end{pmatrix} = \begin{pmatrix} \cos\Phi(t) & \sin\Phi(t) \\ -\sin\Phi(t) & \cos\Phi(t) \end{pmatrix} \begin{pmatrix} I_1(t) \\ Q_1(t) \end{pmatrix} \quad (7)$$

The combined signal of demodulator is as follows that

$$\hat{I}(t) = 2 \cdot I_1(t)\cos\Phi(t) \quad (8)$$

$$\hat{Q}(t) = 2 \cdot Q_1(t)\cos\Phi(t) \quad (9)$$

$$\hat{\Phi}_m(t) = \tan^{-1}\left(\frac{\hat{Q}(t)}{\hat{I}(t)}\right) = \tan^{-1}\left(\frac{Q_1(t)}{I_1(t)}\right) = \begin{cases} \Phi_m(t) & |\Phi(t)| \le \pi/2 \\ \Phi_m(t)+\pi & |\Phi(t)| \ge \pi/2 \end{cases} \quad (10)$$

The PC-QPSK modulation technique improves BER performance in phase uncertainty environments for $|\Phi(t)| \le \pi/2$ as shown in Fig. 2(g).

III. PERFORMANCE

The drawback of the PC-PSK is its broad spectral bandwidth owing to using Manchester coder. Thus, we compared the BER performance of QPSK with that of PC-16PSK. The applied channels are AWGN channel, slow fading Rician channel, and time varying Rician channel.

1) Slow-fading Rician Channel

It is instructive to consider the resulting BER performance for QPSK under phase uncertainty environment. Assuming a constant phase error Φ in the tracking loop, the conditional BER given both the value Φ and the constant fading amplitude A is clearly shown as [8-10].

$$P_b(\Phi, A)_{QPSK} = \frac{1}{2} Q[\sqrt{\frac{2 \cdot E_b}{N_o}} A \cdot (\cos\Phi + \sin\Phi)] + \frac{1}{2} Q[\sqrt{\frac{2 \cdot E_b}{N_o}} A \cdot (\cos\Phi - \sin\Phi)] \quad (11)$$

In determining upper bounds on BER performance with PC-16PSK, it is useful to recall the modified generating function in [11]

$$P_b(\phi, A)_{PC-16PSK} < \frac{1}{2}[Q[\sqrt{\frac{8 \cdot E_b}{N_o}} A \cdot \cos\phi \cdot \sin(\frac{\pi}{16})] + Q[\sqrt{\frac{8 \cdot E_b}{N_o}} A \cdot \cos\phi \cdot \sin(\frac{3\pi}{16})] + Q[\sqrt{\frac{8 \cdot E_b}{N_o}} A \cdot \cos\phi \cdot \sin(\frac{7\pi}{16})]]$$

....(12)

The QPSK and PC-16PSK BER performance can then be obtained by averaging with respect to both Φ and A according to

$$P_b = \int_0^{\infty} \int_{-\pi}^{\pi} P_b(\phi, A)\ p(\phi|A)\ f(A)\ d\phi dA \quad (13)$$

where $f(A)$ is given channel and $P(\Phi \mid A)$ is the conditional pdf of the phase tracking error given the constant value A of the fading amplitude. The pdf of the steady-state phase error for a first-order phase-locked loop is given [8-10]

$$P(\Phi) = \frac{\exp\ [\alpha_o \cos \Phi]}{2\pi\ I_o(\alpha_o)};\quad -\pi \le \Phi \le \pi \quad (14)$$

It follows that the BER performance of QPSK

$$\begin{aligned} P_{b_{QPSK}} = \frac{e^{-\zeta}}{2\pi} \int_0^{\infty} \int_{-\pi}^{\pi} [\ & Q\left(\sqrt{\frac{2E_b y}{N_o(1+\zeta)}}(\cos\phi + \sin\phi)\right) \\ & - Q\left(\sqrt{\frac{2E_b y}{N_o(1+\zeta)}}(\cos\phi - \sin\phi)\right)] \\ & \cdot \frac{1}{2} \frac{\exp\left(\sqrt{\frac{y}{1+\zeta}}\alpha_o \cos\phi\right)}{I_o\left(\sqrt{\frac{y}{1+\zeta}}\alpha_o\right)} \\ & \cdot e^{-y} I_o(2\sqrt{y\zeta})\, d\phi\, dy \end{aligned} \quad (15)$$

where we have made use of the normalization constraint together with the definition of $\zeta = \gamma^2/\sigma_a^2$ as the ratio of specular to diffuse energy, the fact $\sigma_a^2 = 2\sigma^2$ is the common variance of the inphase and quadrature component of diffuse input $a(t)$, and $I_o(\cdot)$ is the modified Bessel function of the first kind of order zero.

Upper bounds on the BER performance of PC-16PSK

$$\begin{aligned} P_{b_{PC-16PSK}} = \frac{e^{-\zeta}}{2\pi} & \int_0^{\infty} \int_{-\pi}^{\pi} P_b(\phi, y)_{PC-16PSK} \\ & \cdot \frac{\exp\left(\sqrt{\frac{y}{1+\zeta}}\alpha_o \cos\phi\right)}{I_o\left(\sqrt{\frac{y}{1+\zeta}}\alpha_o\right)} \\ & \cdot e^{-y} I_o(2\sqrt{y\zeta})\, d\phi\, dy \end{aligned} \quad (16)$$

which can be evaluated as a function of E_b/N_o for specified values of ζ and α_o as shown in Fig. 3 and 4.

2) Time-Varying Rician Channel

Assuming that the fading amplitude is constant at the value A throughout a specified signaling interval.

By using [7-10], it follows by averaging the conditional error probability over both Φ and A. The unconditional bit error probabilities of QPSK and PC-16PSK are given.

$$\begin{aligned} P_{b_{QPSK}} = \frac{e^{-\zeta}}{2\pi} \int_0^{\infty} \int_{-\pi}^{\pi} [\ & Q\left(\sqrt{\frac{2E_b y}{N_o(1+\zeta)}}(\cos\phi + \sin\phi)\right) \\ & - Q\left(\sqrt{\frac{2E_b y}{N_o(1+\zeta)}}(\cos\phi - \sin\phi)\right)] \\ & \cdot \frac{1}{2} \frac{\exp(\alpha \cos\phi)}{I_o(\alpha)} e^{-y} I_o(2\sqrt{y\zeta})\, d\phi\, dy \end{aligned} \quad (17)$$

$$\begin{aligned} P_{b_{PC-16PSK}} = \frac{e^{-\zeta}}{2\pi} & \int_0^{\infty} \int_{-\pi}^{\pi} P_b(\phi, y)_{PC-16PSK} \\ & \cdot \frac{1}{2} \frac{\exp(\alpha \cos\phi)}{I_o(\alpha)} e^{-y} I_o(2\sqrt{y\zeta})\, d\phi\, dy \end{aligned} \quad (18)$$

A typical situation is $B_o \ll B_{Lo}$ so that the tracking loop is able to track variation in signal component phase. Figs 5 and 6 illustrate bit probability performance as a function of E_b/N_o for selected values of ζ and $\alpha_o = 10$ and 15 dB for all $B_o/B_{Lo} = 0.1$.

IV. RESULTS AND CONCLUSIONS

A novel phase-compensated PSK modulation scheme named PC-PSK has been studied for mobile DAB. These results also show that PC-16PSK is effective for obtaining excellent BER performance on slow-fading and time-varying Rician channel for mobile DAB. Fig. 3 illustrates that BER of PC-16PSK(solid lines) is better than that of QPSK (dotted lines) in the high E_b/N_o regions for the case $\zeta = \infty$ corresponding to the AWGN channel. Fig. 4 shows that the BER performance of PC-16PSK is better than that of QPSK for various choices of ζ on slow-fading Rician channel. In Figs. 5 and 6 we illustrate the BER performance of QPSK (dotted lines) and PC-16PSK (solid line) on time-varying Rician channel with 10 dB, 15 dB E_b/N_o . For large E_b/N_o, it was seen that the better BER performance is obtained by using PC-16PSK than that of QPSK. The numerical results show that the PC-PSK scheme can be a potential candidate as a digital modulation technique for mobile DAB.

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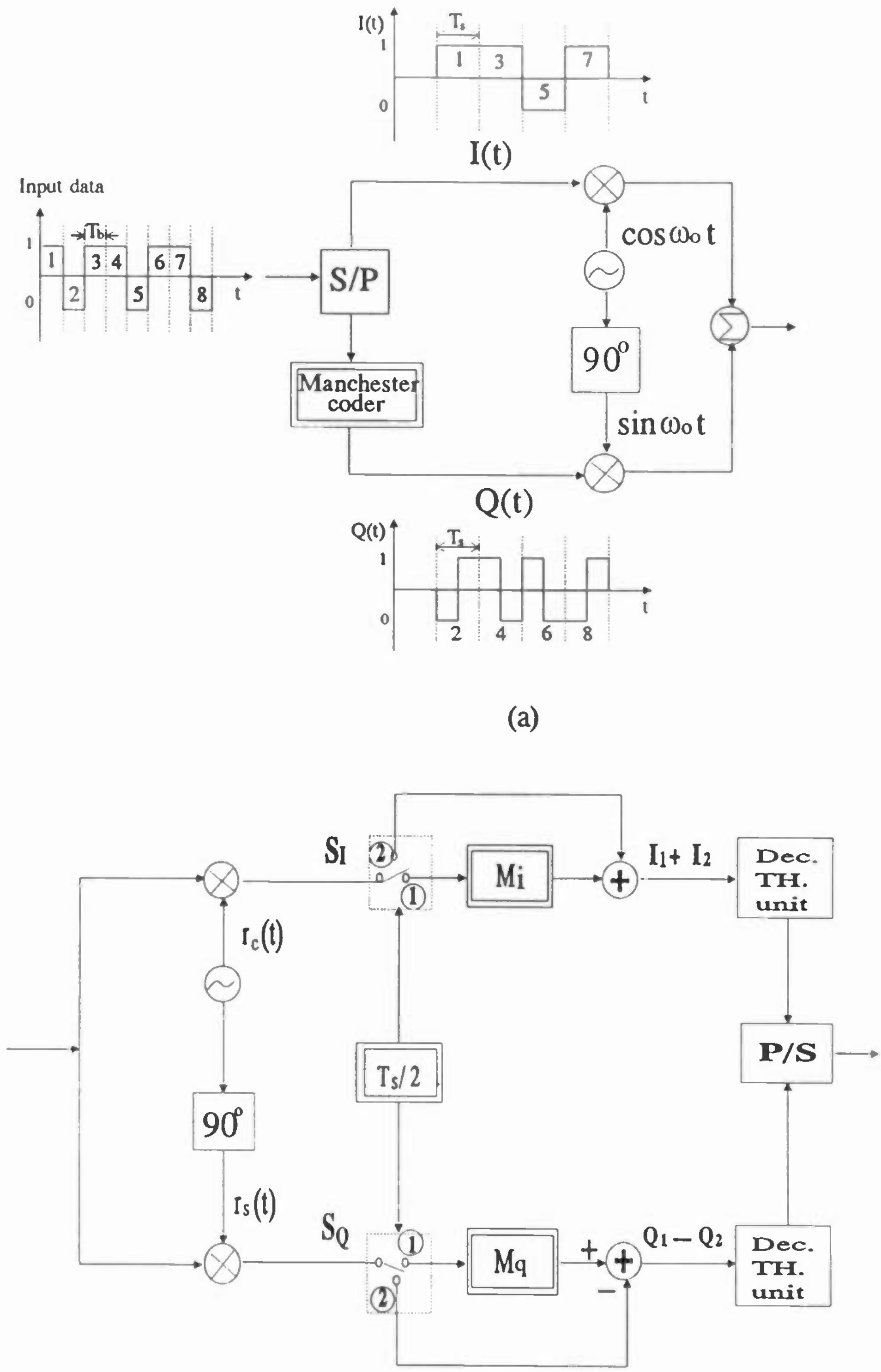
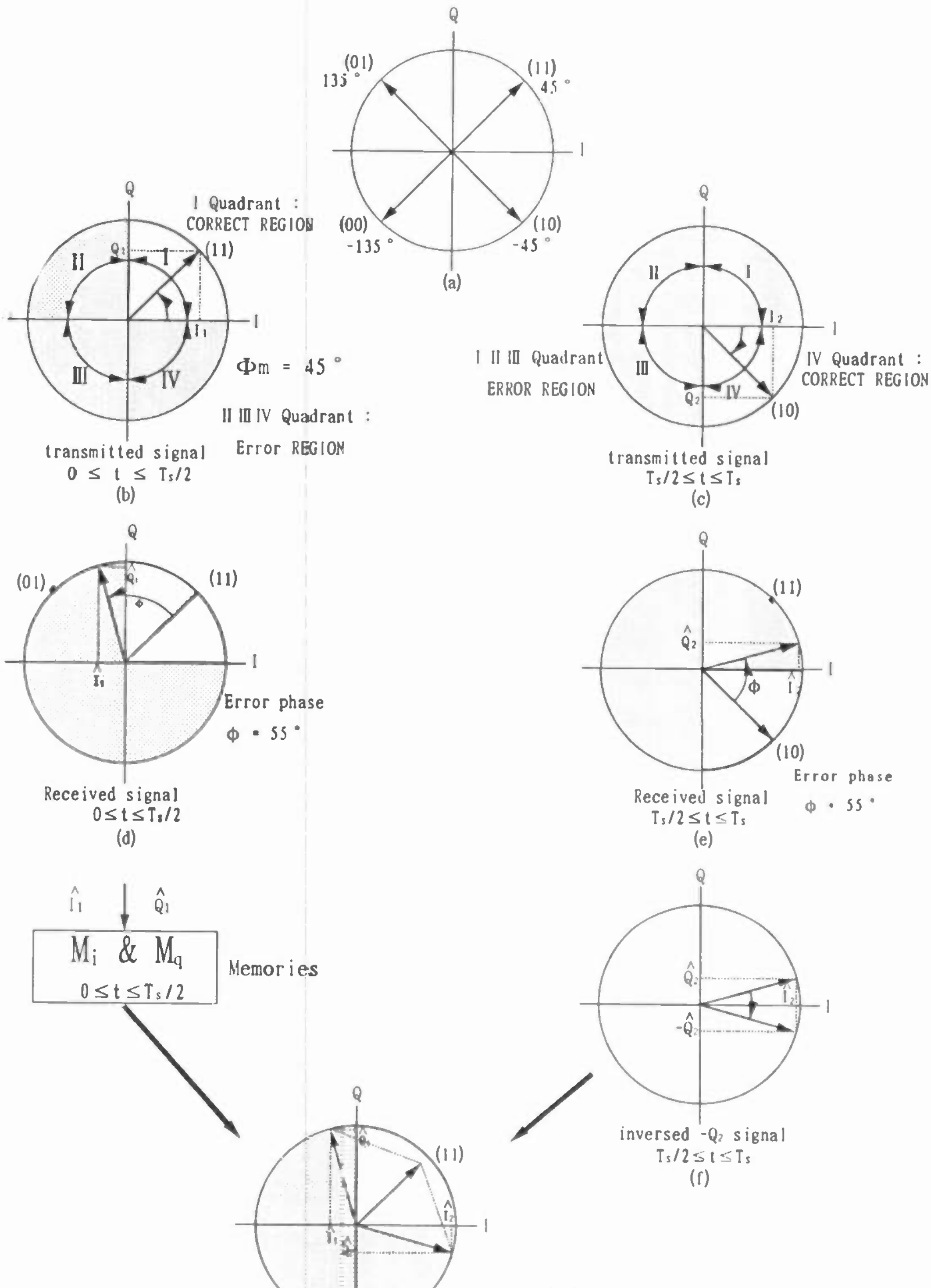


(a)

(b)

Fig. 1. Block diagram of PC-QPSK.
(a) Modulator of PC-QPSK.
(b) Demodulator of PC-QPSK.



$$\hat{\phi}_m = \tan^{-1}\left[\frac{\hat{Q}_1 - \hat{Q}_2}{\hat{I}_1 + \hat{I}_2}\right] = \tan^{-1}\left[\frac{Q_1 - Q_2}{I_1 + I_2}\right] = \begin{cases} \phi_m & , |\phi| \le \pi/2 \\ \phi_m + \pi & , |\phi| \ge \pi/2 \end{cases}$$

Fig. 2. Principles of Phase Compensation of PC-QPSK.

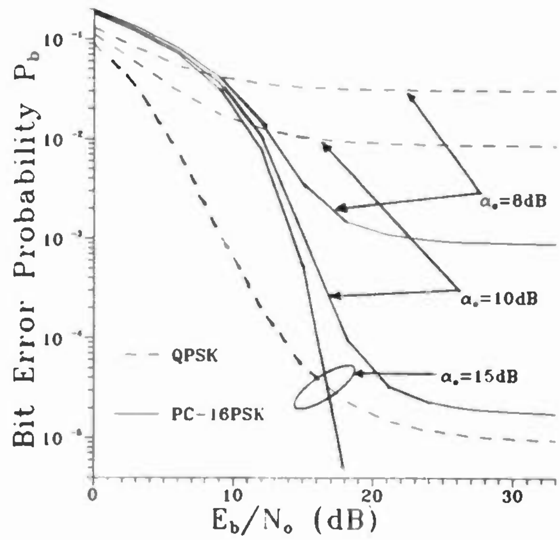


Fig. 3. Comparison of a bit error probability of QPSK and PC-16PSK on an AWGN channel.

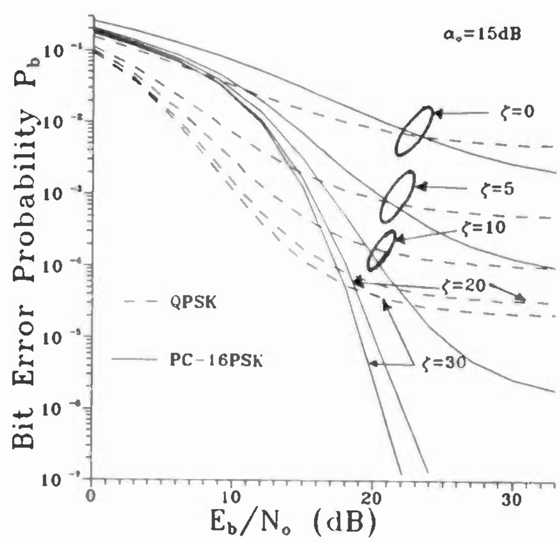


Fig. 4. Comparison of a bit error probability of QPSK and PC-16PSK on a slow-fading Rician channel with α_o = 15dB.

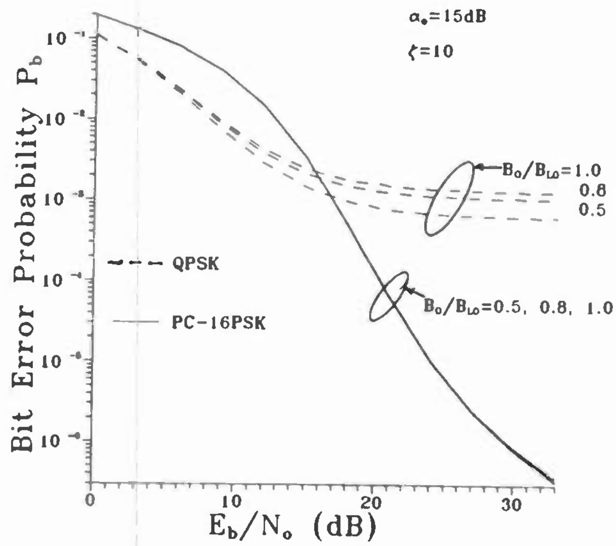


Fig. 5. Comparison of a bit error probability of QPSK and PC-16PSK on a time-varying Rician channel with ζ = 10 and α_o = 10dB.

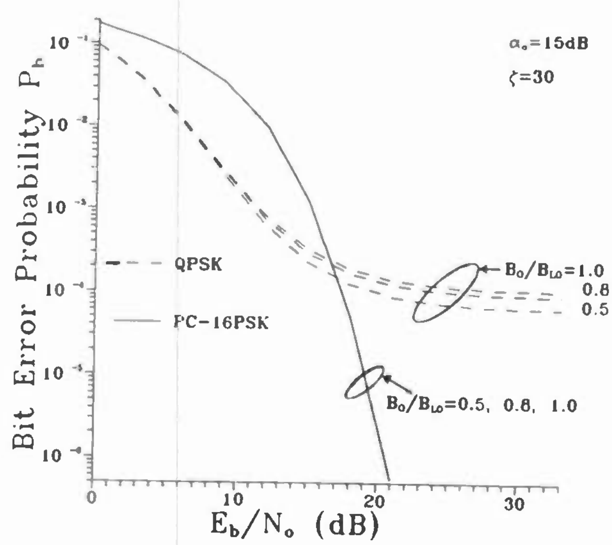


Fig. 6. Comparison of a bit error probability of QPSK and PC-16PSK on a time-varying Rician channel with ζ = 30 and α_o = 15dB.

DIGITAL AUDIO BROADCASTING USING COFDM MODULATION

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ABSTRACT

Many countries in the world have started trials and service transmissions using Coded Orthogonal Frequency Division Multiplex (COFDM) modulation for digital audio transmissions. Digital Audio Broadcast (DAB) transmissions have been allocated spectrum mainly around 230MHz and 1.5GHz for this purpose.

This paper will describe COFDM and its advantages and disadvantages as a modulation method both for national single frequency networks and local broadcasting.

The design concepts and operating requirements of digital transmitters for this type of service will also be described.

The paper is intended to provide the broadcasting engineer with a working knowledge of COFDM modulation and the requirements of transmission systems to broadcast it.

INTRODUCTION

The desire to improve the quality of terrestrial sound broadcasting has brought forward many proposals for systems which bring CD quality audio to the home and the car. In Europe the Eureka 147 DAB system has been adopted and is now being used for Public Service Broadcasting trials in both the United Kingdom and Sweden. Many other countries are operating pilot trials and plan to commence full service broadcasting soon. Indeed it is expected that by the time of the full European launch in 1997 , one hundred million people in eight countries will be able to receive DAB transmissions.

COFDM AND DAB

Why COFDM ?

The present FM stereo broadcast system was designed primarily to provide quality reception at fixed locations. Although receivable in mobile installations, problems are encountered due to fading and multipath propagation. This is especially true where the terrain is hilly or in cities were multiple reflections produce constructive and destructive interference to the received signal.

Compact disc's have raised the audio quality expectations of the consumer both for home and mobile entertainment.

Digital Audio Broadcasting using COFDM fulfils these requirements by providing near- CD quality reception in both mobile and fixed locations.

What is COFDM ?

COFDM is a new digital transmission system which can provide rugged reception , even in a fading channel. Work on this system was initiated by CCETT in France and developed in to a new broadcasting standard by a European collaborative project, Eureka 147.

A major problem for the reception of radio signals is selective fading caused by multipath propagation. This phenomenon produces delayed reflections which appear at the receiver input together with the wanted signal. The resulting demodulated signal can be noisy and distorted.

This type of fading is frequency selective, with some frequencies being enhanced and others attenuated.
With a narrowband signal there is very little that can be done to solve this problem. However if the signal is wideband with redundancy added, then the loss of some parts of the signal may not matter, and the problem can be solved. This is just the case with COFDM.

In COFDM modulation the data is divided between many closely spaced carriers (approximately 1500 in Mode I), so that only a small part of the data is carried on each of them. Additional coding is applied so that the loss of data on some carriers, due to frequency selective fading or impulse noise, will not prevent the signal being reconstructed by the receiver.
In a conventional system, each carrier would need to be positioned far enough apart, so that filters could be used to separate individual carriers before demodulation. COFDM uses a special technique to reduce the bandwidth required to transmit many carriers. Each COFDM carrier has a spectrum shape in the form of a sinx/x function. There are repetitive points, either side of the carrier, where the energy transmitted is zero (zero - crossings). By carefully selecting the carrier spacing, it is possible to arrange for the null points of alternate carriers to coincide.
In this way the many carriers used in OFDM can be compressed into a much smaller bandwidth than by conventional methods. The orthogonal structure of the carriers enables them to be separated from each other during the demodulation process.

The phase of the carrier is used to transmit the data by using quadrature phase shift keying (QPSK).
To demodulate QPSK, a coherent phase reference is required for unambiguous demodulation of the exact phase state.
The demodulated QPSK signal does not contain enough information for this coherent reference to be derived. To overcome this limitation the data is differentially encoded before modulation so that the phases of the reference signal are incremented 45 degrees each symbol. By this method the information is no longer carried solely by the exact phase state of the carrier but by its relative value compared with that of the previous transmitted symbol. The correct term for this type of modulation is $\Pi/4$ D-QPSK.

In the original proposals for OFDM , the data was split and used to modulate a series of individual oscillators, which were then summed to produce the final output Because of the complexity of this system new techniques had to be devised before it could be practically implemented. Fortunately it is now possible to perform this function using digital signal processing. A mathematical derivation for this can be found in reference 1.
The signal is defined in the frequency domain using software, and by using an inverse Fourier transform, the OFDM signal is produced in the form of a series of equally spaced sinx/x functions or carriers. In this way the OFDM spectra are formed. Although basically rectangular in shape, the overall output spectrum has shoulders that fall away gradually, due to the fact that sinx/x functions have an infinite but amplitude reducing bandwidth[3].

Synchronisation , Guard Interval and Single Frequency Networks

Due to different signal path lengths, reflected signals will be time delayed with respect to the main signal.

COFDM is designed to make use of short delay echoes but to ignore long delays which would corrupt data decoding. It does this by having a long symbol period and by the use of a guard band in front of the valid symbol period.

In the receiver it is necessary to sample the incoming signal and perform a Fourier transform to recover the data

To achieve this, the receiver must be frequency and phase locked to the incoming signal.

This ideal situation is difficult to achieve, at receiver switch on, because of the delay occurring whilst timing lock is acquired

Two methods are used to establish synchronisation. Coarse synchronisation is achieved by switching all of the carriers off and on, on a regular basis. This is called the null symbol. After each null symbol, a reference signal is transmitted. By comparing the received symbol with the known transmitted symbol the impulse response of the channel can be measured. This allows a more accurate timing signal to be obtained and also allows automatic frequency control to be implemented.

The receiver signal must be decoded by sampling with the required accuracy within each symbol period, and transforming these sample values into phase information for demodulation. To avoid sample timing problems, more than one complete symbol is transmitted ; the part of the symbol that is repeated being called the Guard Band or interval.(fig.1)

The wider the interval is, the more rugged the system will be to loss of receiver synchronisation However, the time to transmit the data and also the power required to do so will increase.

The guard interval is important as it also gives ruggedness in the presence of echoes. These echoes can either be from reflections or, transmissions from other transmitters.

It can be shown that if another transmitter is transmitting the same data at the same time, an enhancement of the received signal will occur if the signals when received are not delayed by more than the guard interval.

Experiments with single frequency networks have shown that signal enhancements of up to 10dB can occur in some instances. The single frequency network is clearly very spectrum efficient as it is possible to provide both large area coverage and to extend that area just by adding additional transmitters on the same frequency as the existing network. However it is necessary to synchronise all these transmitters together for the benefits to be realised.

The guard interval required will depend on the frequency range used and the spacing of transmitters used in the single frequency network. For a system with transmitters fifty miles apart a guard interval of two hundred and fifty micro seconds has been found to be satisfactory at Band III.

It is also possible to use on-frequency repeaters consisting only of a high gain amplifier and two separated antennas for reception and re-transmission, as simple cost effective 'gap fillers'.

Mobile Reception

As we have discussed, the COFDM system has been designed to minimise multipath fading and also to use echoes to enhance the operation of the system. The mobile application is the most stringent, testing the system to its extremes, not only due to reflections, but also Doppler shift effects, produced by the velocity of the receiving vehicle.

To provide acceptable results at all frequency ranges and design parameters, three modes of system operation are defined by Eureka 147[6] and a fourth mode is under consideration. A feature of these modes is the speed limitation incurred by the requirement for an acceptable mobile system performance.

Each system is designed to provide mobile reception at speeds of at least 100 mph. In Mode I the upper frequency limit is 375MHz. For frequencies below this higher speeds will provide the same system performance. Therefore the selection of mode must take in to account the maximum speed of the mobile receiver.

Services

The COFDM system used for DAB proposes a digital system with a data transport capability of 2.3Mbits/s in a complete ensemble. The term 'ensemble' is used because it is envisaged that many multiple audio sources will be transmitted together at the same time. Audio bit-rates from 348kbits/s down to 32kbits/s can be accommodated.

Typically six high quality stereo audio programmes or 20 moderate audio channels could be transmitted together with programme associated data and other data services, all from the same transmitter.

It is even possible to transmit still pictures not only to a fixed location but also to mobile installations.

The ensemble of services is multiplexed together and the resultant bit stream used to provide input for the COFDM coder.

The subject of who controls the multiplex and the fact that the transmitter owner now has a system with spare capacity that can be rented out to others is a topical issue not covered in this paper.

PRACTICAL SYSTEMS

Overall system

Let us consider the components required to transmit six high quality stereo programmes from a single transmitter:

Audio compression and coding

Each source audio is first sampled, coded and compressed to reduce the amount of data to be transmitted. Many compression systems have been designed and used for signal distribution and for compact discs.

Dab uses MPEG-1 Audio layer 2 coding which is identical to 'Musicam'[2]. This system has the capability of reducing the required bit rate from 768kbit/s to 100kbits/s in a monophonic channel, while preserving the subjective quality of the digital audio for any critical signal. Many bit rates can be handled within the DAB standard.

A half sampling option is also being considered using a down sampling filter in the encoder and an up sampling filter in the receiver.

Programme Associated Data (PAD) is also added to the data frame at this stage, dependent on the audio mode, for voice drama or music reproduction. PAD is also used to provide the user with additional data for display on the receiver, as well as providing dynamic range control signals and signals to allow changes of processing for speech or music. Special control signals can be added to the PAD so that video pictures can be displayed at the correct time within a programme. The coder further processes the signal to provide error protection and time interleaving.

Signal Multiplexing

The six coded and processed stereo signals are then multiplexed together to form an ensemble of signals. The multiplexer interleaves the signals in time and also adds an additional fast information channel (FIC) after interleaving. This channel carries information about the structure and sampling rate of each part of the ensemble and is needed so that the receiver can decode the data automatically

COFDM coding

The multiplexer output, approximately 2.4Mbits/s now has to be interleaved in frequency, this is done in the COFDM coder.

The coder uses a mathematical process to modulate the data on to a large number of carriers so that the data rate carried by each carrier is small. The make up of the coding is determined by the transmission mode required, three modes being available as discussed earlier.

The mapped signal is then passed to a Π/4 D-QPSK modulator which produces the final COFDM signal at a centre frequency of 2MHz.

The output signal consists of many carriers which overlap due to their modulation. The resulting overall spectrum looks like filtered noise, is rectangular in shape and occupies a bandwidth of 1.5MHz. The peaks of the signal contain little energy but can be up to 12dB higher in amplitude than the displayed average level. The peak amplitude of the modulated carriers would in fact be much higher if it were not limited by the coding and scrambling of the signal.

Signal Transmission

The 2MHz signal from the coder is upconverted to 35MHz, filtered and processed by the intermediate frequency (IF) corrector. This corrector provides pre-distortions for amplitude, and phase with amplitude, to correct the distortions in the following amplifier stages.

The IF signal is upconverted again to its final frequency and amplified by Class A amplifiers before being split to drive each Class AB power amplifier. Such a combination of amplifiers classes provides a good compromise between gain, linearity and efficiency.

The resulting output is bandpass filtered before entering the antenna transmission line.

TRANSMITTER DESIGN

As we have seen COFDM signals contain both amplitude and phase information , so unlike an FM transmission , it requires linear amplification.

COFDM is a system of equal spaced carriers. Intermodulation signals are produced between them and also cause the signal to spread into adjacent channels. The amplitude of these intermodulation products is a function of the system non-linearity.

At first sight it would seem that a whole new technology has to be created to transmit these signals. Fortunately this is not the case; high quality conventional analogue television transmitters already use adequate linearisation methods.

Linearity correction

FET and transistor amplifiers produce both amplitude non linearities and phase distortions which vary according to the amplifier transfer curves.

As these distortions are predictable it is possible to provide a pre-corrector which, when correctly adjusted, provides equal and opposite phase and amplitude corrections, to cancel out those of the power amplifier. In this way, an overall corrected transfer curve is obtained from the transmitter amplifier.

Corrections for amplitude are provided by routing the signals through a series of parallel diodes. The bias point of each of these diodes can be adjusted to set the point at which they conduct. In this way, as the input signal level increases more diodes turn on, lowering the signal path resistance and altering the shape of the transfer curve to cancel out the opposite curve produced by the non-linearity in the final amplifier.

Phase is corrected using a 'Quadrature' corrector . This corrector uses the same type of expansion circuits as the amplitude linearity corrector. Signals from the amplitude linearity corrector are summed with those from the quadrature corrector which operates with a phase shift of 90 degrees. The resulting output is the vector sum of the two signals. Using this method it is also possible to produce the correcting curve for the final amplifier. These techniques have been used on television transmitters for many years.

COFDM Power amplification

The ideal solution for amplifying COFDM would be to use a perfectly linear amplifier, however in practice this is not necessary and conventional amplifiers can be used.

As we discussed earlier the peak to average ratio of the DAB signal is in the order of 12dB. Amplifiers used to transmit DAB would be very large and expensive if they had to reproduce these peak signals. In practice a peak to average ratio of 6dB produces acceptable results. In some experiments a peak to average ratio of 4.8dB has been used in working systems.

The peak to average ratio transmitted is a function of the saturated power output of the amplifier used and the average power it is backed off to. A transmitter with a saturated power output of 1kW would produce 250Watts DAB power if the backoff was 6dB.

DAB power therefore is expressed as the average power output and not the peak capability of the amplifier.

Operating with a backoff less than 12dB clips the peaks of the amplified signal which contributes to both the in-band and out of band intermodulation products.

In practice this means that even a totally linear amplifier will produce intermodulation products (IP's) due to the clipping effect.

These intermodulation products cause the creation of shoulders on either side of the transmitted spectrum.

This is clearly not acceptable as the signal will interfere with other adjacent channels.

Stringent spectrum masks have been produced as part of the ETS specification[7] for DAB and this requirement is the dominant factor determining the linearity requirements of the transmitter.

Two masks have been produced one for non-critical applications and the other for use where DAB channel occupancy is expected to be heavy. A mask similar to the non-critical mask is also being used at L-band, mainly because of the practical limitations of designing filters with sharp enough sides at these frequencies.

In a practical transmitter for DAB it is necessary to have a linearity which produces unfiltered shoulders at a level of 30-35dB. This will require an amplifier with an uncorrected linearity of -20 dB, with 10-15dB of correction.

The 35dB figure is not enough to meet the critical spectrum mask. To do this with the transmitter alone would require nearly 80dB of linearity correction. Whilst there are techniques such as feed-forward and Cartesian feedback which can be used to linearise amplifiers, corrections of this magnitude are not possible.

It is necessary in all DAB transmitters to have an output bandpass filter to provide the additional attenuation of IP,s. The filter required has a very tight specification as it has to produce large out of band attenuation without excessive in band loss.

Clearly the output filter is a critical item and it can be more cost effective to have a very good filter than buy a much larger and more linear transmitter to obtain the same system performance.

The Harris DAB 2000 system has been designed to successfully meet these requirements providing the most cost effective solution for DAB transmitters.

CONCLUSION

Digital Audio Broadcasting is now a reality in many countries and by its official consumer launch in 1997, at least 100 million people will be able to receive it.

It will not only bring near CD quality transmissions to the car and the home but will also provide the possibility of additional data services and picture transmission and provide more efficient use of the frequency spectrum.

The system has proved to be very rugged in use and can be amplified and transmitted using existing techniques and products with proven reliability in existing Band III television services.

In this paper it has only been possible to give an outline of COFDM and associated systems.

A single paper of this length can only give a simple overview of this very exciting topic. The

references at the back of this paper will give a much more detailed understanding of the subject matter.
I would like to thank my colleagues both in Quincy Illinois and Cambridge England for their help in producing this paper.

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CONCEPTUAL DAB SIGNAL BLOCK DIAGRAM

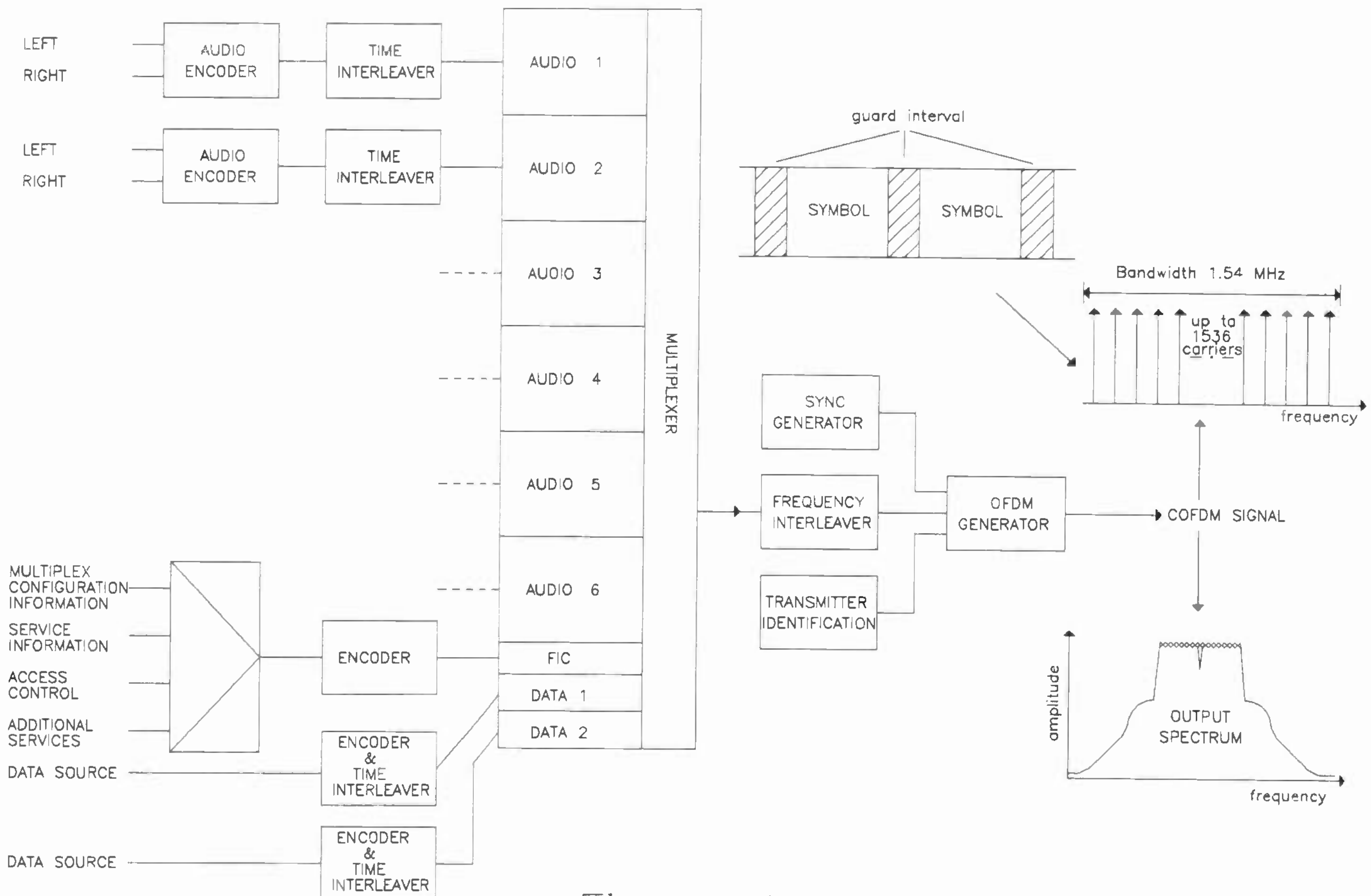


Figure 1

1000 WATT DAB TRANSMITTER BLOCK DIAGRAM

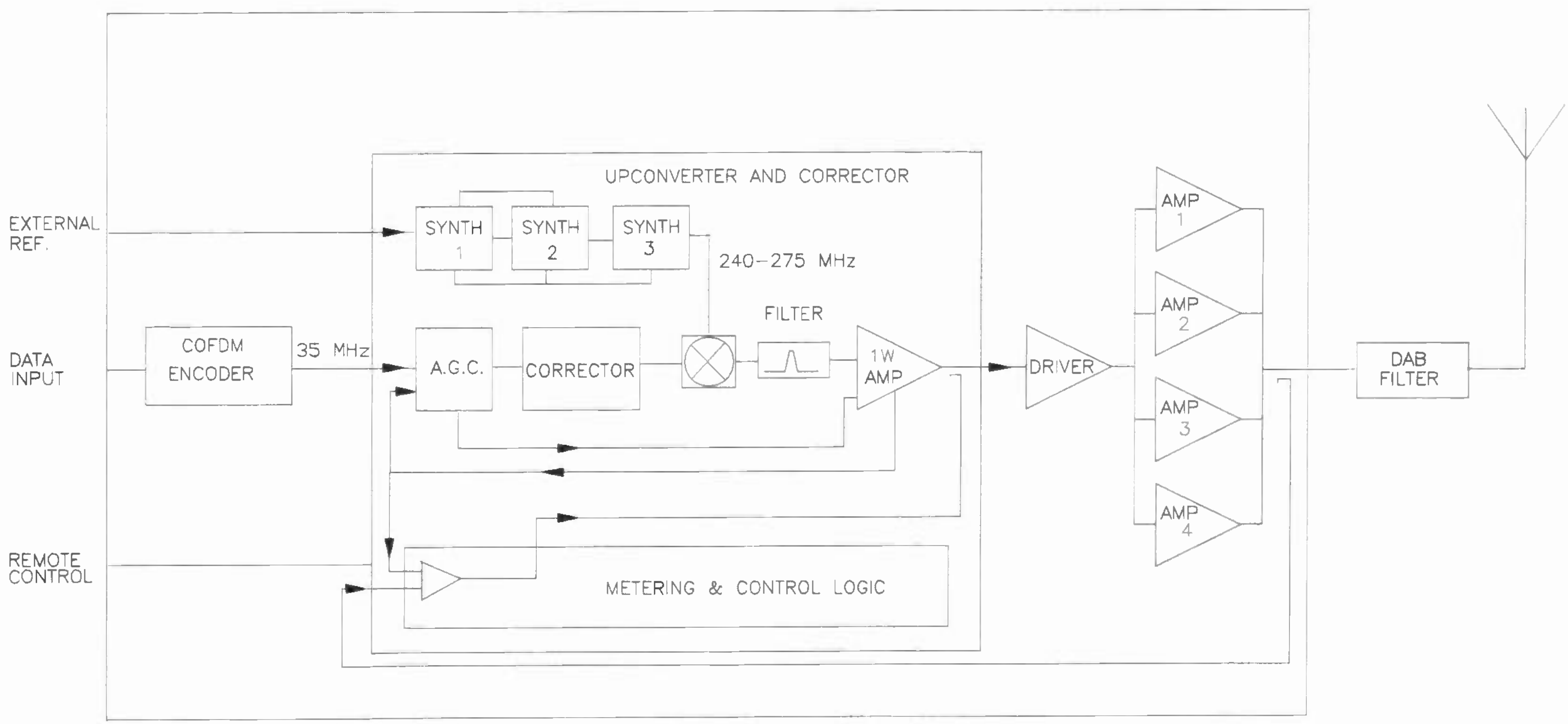


Figure 2

BRINGING DAB TO THE CONSUMER

David Witherow
BBC, President of the European DAB Forum
London, United Kingdom

ABSTRACT

Digital radio using the Eureka 147 DAB (Digital Audio Broadcasting) system has now reached take-off point in Europe and Canada and is being tested in many other parts of the world. By the Autumn of 1997 - when consumer receivers start to become more widely available - at least seven European countries will have started operational DAB services, with pilots taking place in many other nations. The primary focus has now moved from technological development to implementation. Through the European DAB Forum (EuroDab), the EBU, the Eureka 147 Project and national groupings, all sides of the broadcasting industry are working together in a productive partnership to ensure the successful consumer launch of DAB. This paper outlines the growth of Eureka 147 DAB in Europe and beyond, looks at some of the planned services, and describes how practical issues of implementation are being addressed, particularly through EuroDab. While much work remains to be done, and great commitment will be required from all sections of the industry, the author foresees continuing progress towards the long-term goal of DAB as the future of radio in many parts of the world.

INTRODUCTION: INTO THE COMMERCIAL PHASE

Digital Audio Broadcasting (DAB) using the Eureka 147 system is now a reality. It has moved out of the technological development phase into the implementation or commercial phase. Operational services have already started in some countries, many more are planned, and by the end of 1997 in excess of 100 million people in Europe will be within reach of DAB. But it is not only in Europe that DAB is taking off: Canada has of course long been one of the pioneering nations, but what we are seeing now is a rapid growth of interest in many other parts of the world. Among them are China and India, with their vast populations and where radio remains one of the prime means of communication.

Of course, the development or enhancement of the Eureka 147 technology continues and is described elsewhere by Franc Kozamernik of the EBU. But the decision- making processes have now switched from research and development engineers - to company boards, to managers, editors, programme makers and other service makers, marketing departments etc. in a wide range of companies and organisations now involved in the service implementation aspects of DAB. A critical mass has been formed. There is now no question of DAB not happening, the only real question is at what pace.

POTENTIAL WINNERS ALL

As with any new product or technological advance, decisions to commit company resources and enter markets will depend on hard-headed decisions about the risks involved, and the commercial or other benefits. While some seed-corn money is being provided by some governments for building infrastructure or for pilots, that is not the case in many others - including mine. We in the BBC are having to find the considerable sums needed from within current resources. The commercial side will also have to be prepared to put up the risk capital, though at least the Government is proposing some incentives in terms of franchise periods.

The fact is that DAB offers something for everyone, for all the players.

- for governments: more efficient use of radio spectrum, the possibility of reusing (or reselling) FM/AM spectrum, the political plus of facilitating a better deal for the consumer.

- for manufacturers of transmission, receiver and other equipment: potentially large profits, especially if DAB ultimately replaces analogue, and as it grows across the world.

- for current broadcasters: scope for additional programming, text/data as well as audio, and a place in the digital future which will help keep radio alive and thriving in the next century.

- for new service/product providers: new business opportunities, especially in data products

- and for the consumer: all those benefits which DAB offers - terrific sound quality, ruggedness of signal, ease of use, mobility, more choice from more and varied types of product and service.

Yes, of course there are risks. But it is clear that the main interest groups now see DAB as the future. Well, perhaps not yet the consumers, because at this stage the general public has little awareness of this great and marvellous new information and entertainment medium gathering force and coming its way.

DAB GATHERS SPEED IN EUROPE

Eureka 147 was originally developed as a European co-operative venture, and not surprisingly it is in Europe that a great deal is now going on to introduce DAB services. One key short-term milestone will be the Internationale Funkausstellung in Berlin in August 1997, when many manufacturers are expected to display their first generation of consumer receivers, and by that time it can be expected that seven or eight countries will have operational services running providing a large potential consumer base of at least 100 million people. A dozen or more are currently piloting or testing DAB and I shall briefly describe some of these projects.

Let me start with what is happening in my own country, the **United Kingdom**. The government has allocated 12.5 MHz of spectrum in Band III, sufficient for seven blocks or multiplexes, each of which can carry a number of audio and data services. Two of these blocks are for nationwide transmissions (one BBC, one commercial), the other five to be used to provide services at the local and regional level (both BBC and commercial). Legislation is now before Parliament which will provide the regulatory framework and which offers commercial radio stations incentives in the form of longer franchise periods for investment in DAB. These developments have certainly increased interest in the commercial radio sector, and the transmission company NTL has announced that it will set up a pilot multiplex in the London area this year.

In the meantime the BBC has had a pioneering role in the United Kingdom. Initially using the multiplex assigned for its national radio services, the BBC began DAB broadcasts in the Greater London area last September and is now in the process of building up a single frequency network of 27 transmitters which will provide coverage to 60% of the United Kingdom population by mid-1998 (Figure 1). During the next eighteen months or so, before consumer receivers become generally available, we will be using our transmissions to do a number of things:

- to pilot different types of audio, text and data services, including later this year a new service we are calling BBC Now. This will be a rolling 24-hour information service, which can be received either as a direct service or eventually, with each bit of information labelled, as a selectable audio or text information on demand.

- to gauge the reaction of our licence-fee payers, through qualitative panel research, to these services and to the other features and facilities which DAB offers

- to experiment with "flexing" the multiplex, using different bit rates for different types of programming and increasing or decreasing the DAB offering according to editorial need (for instance, running an Election Special service for a limited period during the coming General Election)

- to investigate the man-machine interface to see how best we can help consumers find their way round the changing services within a multiplex

- to iron out any transmission, coding, multiplexing and delivery issues

- to prepare for the introduction of our local radio services

- and to co-operate with the commercial side of the broadcasting industry to ensure the successful launch and development of DAB in the United Kingdom.

Germany is another key country for the progressive operational development of DAB in Europe. By the end of 1995 three of the German states had initiated DAB pilot projects, all on a substantial scale, with a fourth scheduled to start in a few monthsí time. By the end of 1997 a large proportion of the German population will be within reach (Figure 2) of what by then will have become regular services. Typically, the pilot projects bring together public and private broadcasters, manufacturers, R&D institutions, PTT/Telecom operators, administrations and others, and include transmission in both Band III and L Band.

The on-going commitment to DAB as the radio of the

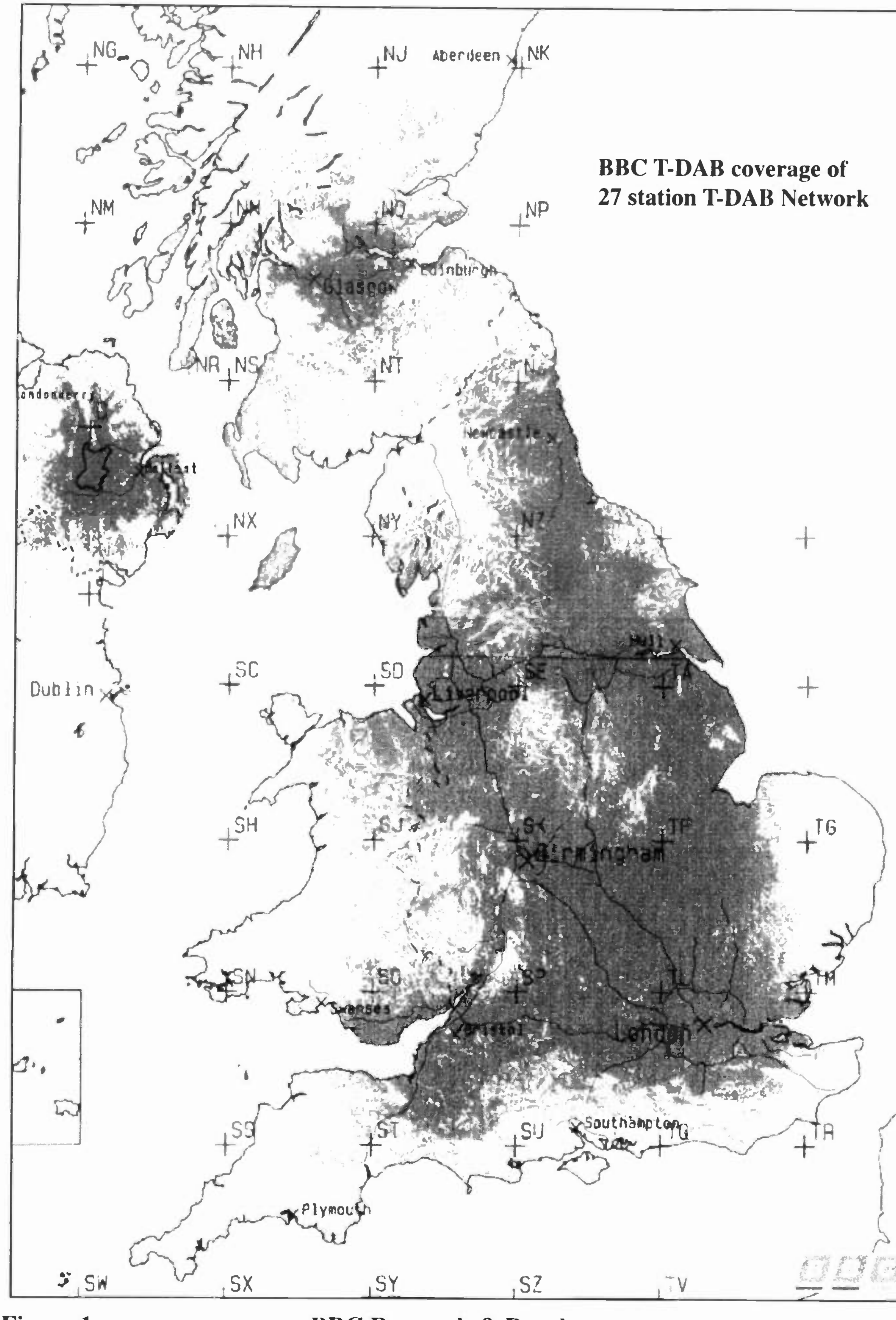


Figure 1 **BBC Research & Development**

Figure 2.

future in Germany was confirmed when the authorities announced that there would be a special supplement added to the licence fee from 1997. This supplement will produce 176 million Deutschmarks (125m $US) over four years to aid the public service broadcasters build up the DAB infrastructure. Additional financial help may come from the Laender governments for the commercial stations.

A key issue in these and other pilots elsewhere is the availability of receivers. By the beginning of this year three European manufacturers - Philips, Bosch/Blaupunkt and Grundig - were making pre-production sets for special order and several other manufacturers around the world (among them Delco) were working on prototypes. It is estimated that more 10,000 of these early receivers will be in use during the German pilots for evaluation by all parties, including the general public, of DAB services and facilities.

Turning to the Nordic countries, <u>Sweden</u> began a DAB service in the Stockholm region in September 1995 and this will be followed this year by single frequency networks in Gothenburg and Malmo, with the likelihood of a further extension to a non-metropolitan area. By the end of 1996 about 36% of the Swedish population will be covered. The Swedish Government is making additional funds available for the development of digital radio and digital television.

In <u>Denmark</u> too some seed-corn money is being proved by the Government. Transmission began last September in the Copenhagen area. Five hundred DAB receivers are being distributed to consumers with the emphasis on home hi-fi rather on car audio as in the German pilots. Research will be carried out the Danish Radio in co-operation with the set manufacturers.

<u>Norway</u> has had test transmission running since 1994, with the Oslo area now being covered by a two transmitter SFN (with a third transmitter due to be added this year). A fourth will be installed on the western coast to test DAB in a very mountainous region. One of the aims of the Norwegian test transmission is to see how much

improvement there is over the very variable FM quality in such difficult topographical conditions. The test network will cover about 30% of the Norwegian population, and its is expected that a formal decision to start regular DAB services will be taken this year.

The **Netherlands** is another country well on the way towards operational services. The driving force is the Netherlands DAB Foundation, and with four transmitters covering the main population areas, about seven and a half million people or about 50% of the population are within reach of DAB. Some of the Dutch experiments are centred on exploring low bit rates for audio, on different aspects of datacasting and on Differential GPS for car navigation accuracy.

It is not possible within the constraints of space to describe the many other pilots and tests - for instance in **Italy**, **Switzerland**, **Finland**, **Hungary** and **Poland** - but some mention needs to be made of the situation in **France**. France is an important country for the development of DAB in Europe, both for its geographical position and for the size of the potential market in its population of 57 million. The French had an early and enthusiastic involvement in the technological development of DAB, and the French DAB Club was founded as long ago as 1991. However, after initial testing in Band I and L Band (the latter clearly preferable), progress more recently has slowed down due to lack of funds. Yet interest remains high among a number of organisations and companies, including TDF (the transmission provider) and Radio France (the public service broadcaster). The outgoing chairman of Radio France, Jean Maheu, recently described the introduction of digital radio as the most important challenge and the top priority for his successor. Strategic alliances have been formed to provide coverage and services when market conditions allow. TDF has been working with other potential partners to look at innovative multimedia applications of the DAB system and electronic newspaper publishing. It is expected that a new pilot project in the L Band will get under way later this year.

THE GLOBAL ADVANCE

In an ideal radio world, there would be a single standard for digital broadcasting, delivered either terrestrially or by satellite, and a vast market for receivers with common functionality and world-wide operability. The Eureka 147 system is in many ways best candidate. Whether or not the decision makers in the United States will be convinced remains to be seen, and I am not qualified to enter into the technical and other arguments. But whichever way the USA goes there are growing signs that Eureka 147 DAB is being taken very seriously by more and more countries outside Europe. There are a number of inherent reasons - the sheer range of benefits, from quality and reliability of sound through spectrum efficiency to system flexibility - but also to large countries, and to international broadcasters, there is another attraction. That is the potential compatibility of terrestrial with satellite-delivered DAB to provide large area coverage.

Some tests of satellite-delivered DAB have already been carried out, and others are planned. An L Band satellite traial was conducted in July 1995 in Mexico using the Solidaridad 2 satellite, and showed encouraging potential even for mobile reception. A trial has also been conducted in Australia using the OPTUS 3 satellite.

Canada is of course the most immediate example of the outward progression from Europe of DAB. Canada was an early adopter of the system and had done much pioneering work in its development, and last November Eureka was formally confirmed as the Canadian standard At the same time, the regulatory authority announced that it was now accepting applications for transitional digital radio licences, to remain in effect until the development of a long-term policy. Current AM and FM licensees will automatically qualify for transitional licenses, and other applications will be considered on a case by case basis. Broadcasters will be able to use part of their allocated spectrum for datacasting. With pilot transmissions already operating in Toronto, Ottawa and Montreal, a fourth will start up this year in Vancouver. Between them these pilots will put 35% of the Canadian population within reach of DAB.

Australia is also starting test transmissions in three major cities - Sydney, Melbourne and Brisbane. A government-appointed commission is currently looking at the policy issues involved in going digital and is due to report with its recommendations in July of this year.

And the worldís two most populous countries, **China** and **India** are now demonstrating significant interest in the Eureka 147 system. In **China**, under a European Commission sponsored co-operation agreement, several European countries have been involved in helping to get experiment DAB transmission under way in Beijing and Guongdong Province. In **India**, the public service broadcaster, All India Radio, has developed a three stage plan to implement DAB, starting with terrestrial transmissions in at least four cities in 1998/99 using Band II. Satellite-delivered DAB services in the L Band are planned for around the year 2003.

Finally on this brief worldwide tour to **Japan**, where no decisions have yet been taken on what digital system to adopt. NHK, the public service broadcasting organisation, has been developing its own in-band proposal. But with all the major Japanese maunfacturers now partners in the Eureka 147 Project, and with much interest among some commercial broadcasters, the system question remains an open one.

CREATING "OPTIMUM CONDITIONS"

In December 1994, in response to requests from various National DAB Platforms, the European Broadcasting Union launched an initiative to create an umbrella organisation which would bring together all the interested parties as DAB entered its implementation stage. That initiative culminated in the formal setting-up last year of the European DAB Forum (EuroDab). The idea was not to duplicate work already being done through other international bodies (such as the Eureka 147 Project) or national groupings, but to co-operate on service and implementation issues and on marketing. EuroDab already has more than 100 member organisations from all sectors - public and private broadcasters, other service providers, transmission companies, electronics industries, receiver manufacturers, regulators, international broadcasters, etc. They come form some 25 countries, and as indicated earlier there is a growing membership from countries outside Europe. EuroDab now has a Steering Board, a Project Manager and office support based at the EBU, and a number of working groups or "modules" (Figure 3).

From the outset the members agreed that EuroDab should be more than a talking shop, more even than a forum for the exchange of information, important though that is. It was agreed it should be a positive force to get things done, to iron out the inevitable creases, to help drive DAB to success. And in order to concentrate mind and effort, we focussed our planning initially on the Autumn of 1997, because that is when we can expect a range of consumer receivers to be available in the market place. Hence the overall objective "to create the optimum conditions in Europe and other parts of the world for the successful consumer launch of DAB in the Autumn of 1997, and for its growth and progress after that".

SETTING SIGHTS

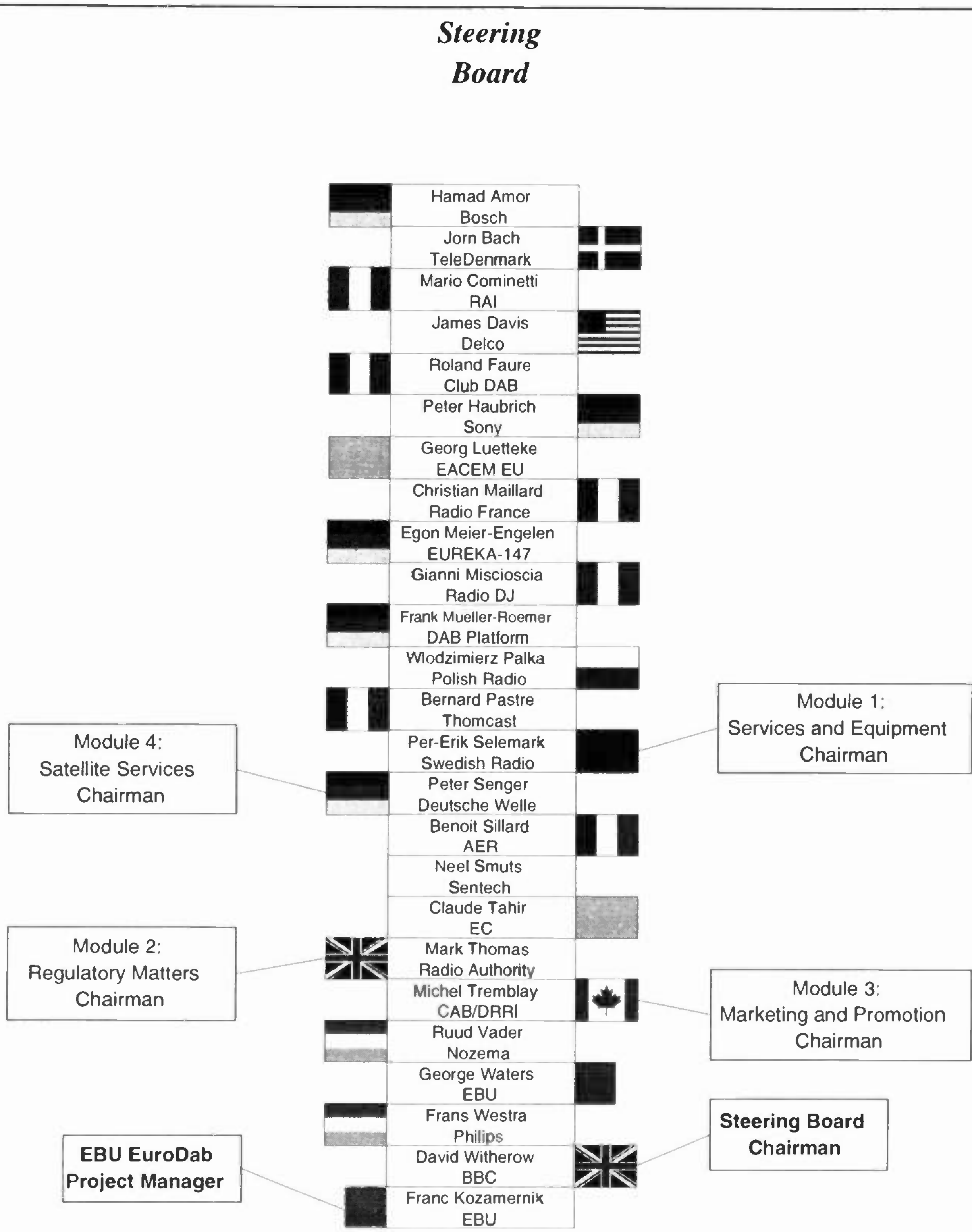
The Steering Board adopted a series of targets to be achieved by this date. The following partial list gives a flavour of them:

- operational services in at least eight countries in Europe reaching a potential audience of at least 100 million people
- a co-ordinated national/international marketing strategy informed by sufficient and relevant market research
- high awareness of DAB both among the general public and within particular interest groups
- a range of receivers in the market place at prices comparable with or better than other consumer electronic products at the same stage of entry into the market
- common systems, and receivers with common basic features and functionality
- retailers full informed, trained and motivated to sell DAB products
- an attractive range of audio and data services which are specific to DAB
- regulatory regimes which aid rather than hinder the development of DAB
- progress towards a consensus on a satellite DAB system which is compatible with the terrestrial one.

These targets are simple to state but clearly more difficult to achieve, and much hard work will be needed over the next eighteen months. The engine rooms are the four work modules: Services and Equipment, Regulatory Matters, Marketing/Promotion and Satellite Services. Each has its own terms of reference and work tasks. In this section I will concentrate on the Services and Equipment Module because this has largest, most complex and also most immediate remit. Indeed, it was originally intended to divide the work between two separate modules, but in practice it became clear that if we went down that route there would be a huge overlap both in tasks and in membership.

SERVICES AND EQUIPMENT

In broad terms, the task of this module is to marry the requirements of the service or product providers (whether audio, data or multimedia) with the delivery and receiving equipment which will allow the consumer access to those services and products. And time is very short before the first consumer receivers must be ready. While there is a certain amount of agreement among the service providers as to the range and type of audio material envisaged, there is much less clarity about text and data requirements. So the broadcasters, private and public, and the commercial data providers have to get their act together, particularly for the initial services. They also have to be realistic as to what the first generation of receiver equipment will be able to do. On the equipment manufacturing side, the parties have to seek to meet at least the minimum service functions and facilities.


Steering Board
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Jorn Bach
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Mario Cominetti
RAI
James Davis
Delco
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Club DAB
Peter Haubrich
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DAB Platform
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Thomcast
Per-Erik Selemark
Swedish Radio
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Claude Tahir
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Radio Authority
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George Waters
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Services and Equipment
Chairman
Module 4:
Satellite Services
Chairman
Module 2:
Regulatory Matters
Chairman
Module 3:
Marketing and Promotion
Chairman
Steering Board
Chairman
EBU EuroDab
Project Manager

Figure 3

A matrix was developed (Figure 4) to facilitate discussion of the various system features that could be implemented in the different types of receivers in the three phases identified (i.e. by 1997, before 2000 and after 2000). Five types of receiver are being considered: hi-fi, portable, car, PC and walkman. It seems unlikely at this point that chip development will have reached the stage for portables and walkmans to be available by 1997 but the expectation is that all the other three will be purchasable by the end of that year.

So far, agreement has been reached between manufacturers, service providers and network operators on some 90% of all the features but some important questions still need to be resolved, especially in relation to the first generation receivers. These include: the range of audio bit rates; the ability to reconfigure a multiplex without muting or glitches; harmonisation of PTY and other coding between RDS and DAB; linking to FM; whether TII (Transmitter Identification Information) is required, and if so how it will be implemented; and minimum display size for text. Another issue which has emerged with greater priority than before is conditional access. Many commercial interests see a need for pay services of one kind or another, especially for data applications, if investment costs are to be recovered and profitability made possible. But what kind of a CA system? And indeed, what sort of data applications?

These are the issues that the Services and Equipment Module has been grappling with, and each of the other Modules has a similar list of work tasks to be accomplished within given timescales. It is proving an effective way of taking DAB forward from technological development to application in a real, and highly competitive, world.

THIS IS THE FUTURE AND IT WORKS

DAB will be a success. The only question is how rapidly and how widely it will progress. It is increasingly accepted that the future of radio is a digital one and that digital delivery will in time largely supplant AM and FM. Increasingly also Eureka 147 is seen as the system to adopt. A vast potential market for DAB is already assured within the next two years.

There are of course still plenty of possible pitfalls, difficulties, questionmarks, uncertainities. But as we enter the implementation or commercial stage of DAB, all sides of the broadcasting industry are working together in a productive partnership to maximise the opportunities, minimise the risks, sort out in-service issues, and really start to bring DAB to the consumer.

Module 1: Services and Equipment: Features Matrix

(Key: PP=Programme provider, SP=Service provider, MP=Multiplex provider, NP=Network provider, RM=Receiver manufacturer)

No	Feature	Provider					Hi-F receiver			Portable receiver			Car receiver			PC card receiver			"Walkman" receiver			Conditional access		
		PP	SP	MP	NP	RM	1997	2000	2000+	1997	2000	2000+	1997	2000	2000+	1997	2000	2000+	1997	2000	2000+	1997	2000	2000+
14	Audio 224 kbits/s																							
15	Audio 256 kbits/s																							
16	Audio 320 kbits/s																							
17	Audio 384 kbits/s																							
35	TII coding																							
36	TII database																							
	● main id																							
	● sub id																							

Figure 4

EUREKA 147 - TOWARDS A DE FACTO WORLD DAB STANDARD

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Abstract

It has taken almost ten years for the Eureka 147 system to achieve world-wide support and confidence among public and private broadcasters, network operators, administrations and manufacturers all over the world. The technical excellence of the system, in particular for portable and mobile radio receivers, has been a prerequisite, but operational advantages, possible cost and spectrum savings, universal use in the different delivery media as well as the system design flexibility for future extensions towards multimedia have been decisive factors for world-wide acceptance and general recognition of the system. The Eureka 147 system offers a myriad of business opportunities in audio and data broadcasting and is becoming a major business success from which many existing and new broadcasters and manufacturers will benefit in the future.

This paper puts the Eureka 147 system in perspective and discusses its principal advantages and assets vis-à-vis other digital technologies, such as DVB, GSM and other digital audio broadcasting systems. The recent system technical refinements are outlined and some outstanding frequency planning and frequency management issues are highlighted. The paper describes the Eureka system in terms of a generic multimedia transport mechanism.

1 Introduction

Since the 1920's radio technology has achieved tremendous growth: today there are over 2000 million radio receivers in use on the planet. Radio is ubiquitous throughout the world. The power of sound radio to inform, educate and entertain at home and on the move is unmatched. Digital technology is now capable of modernizing this old medium. A very successful European collaborative project - Eureka 147 - has developed a new digital transmission system proposal which has received world-wide acceptance and general recognition from all broadcasters, network providers, professional and consumer manufacturers, administrations and regulators for its technical excellence and operational flexibility. The Eureka 147 DAB system - now a worldwide and a European Standard - has all the ingredients to be converted from a brilliant engineering achievement into a commercially successful product in the marketplace.

The world's first official DAB services were inaugurated in the UK by BBC Radio, and in Sweden by the Swedish Broadcasting Corporation, in September 1995. Several pilot services are on air in Germany. These extensive broadcasts should encourage manufacturers to bring their DAB receiver products to the marketplace at an affordable price as soon as possible.

In Europe, DAB is seen as a universal delivery mechanism for all broadcasting media including future multimedia services where mobile reception is a primary requirement.

2 Competitive digital audio technologies

In Europe, DAB services are coming into a mature consumer-electronic market in which radio programmes are being delivered by several digital broadcast systems such as NICAM 728, Digital Satellite Radio (DSR), Astra Digital Radio (ADR) and Digital Video Broadcasting (DVB). These systems use relatively simple modulation and channel-coding techniques and have been designed for specific purposes where immunity to frequency-selective fading is not required and where reception is only via a static receiver. Contrary to the Eureka 147 System, they do not provide reliable reception in a multipath propagation environment.

Internet is becoming increasingly important. Also, radio is witnessing increasingly strong competition from non-broadcast media which use digital techniques to produce the optimum performance, at a cost that is acceptable to large consumer markets.

The compact disc was the first mass-storage digital medium to offer superior sound quality in the domestic marketplace. The CD has now been joined by various other tape and disk storage formats, such as R-DAT and S-DAT, digital compact cassette (DCC), MiniDisc (MD) and CD-I.

3 Principal characteristics of the Eureka 147 System

3.1 Mobile reception

Mobile reception is increasingly important for a transmission system that claims to be ubiquitous, such as radio. There are more than 150 million cars in Europe most of which are equipped with at least a simple radio set.

The principal strength of the Eureka 147 system is that it enables an interruption- and interference-free reception in adverse receiving conditions where no line-of-sight between the transmitter and the receiver is possible and where multipath and shadowing loss are the dominant factors that affect the signal transmission. Eureka is designed to overcome In the Eureka 147 system, a transmission technique called *coded orthogonal frequency division multiplex* (COFDM). With COFDM, problems of multipath reception encountered with conventional analogue transmission systems are practically eliminated. Due to the low data rate of each RF carrier, any delayed reflections of the signal (i.e. "passive echoes") add in a constructive manner to the direct signal already received. The only situation where passive echoes do not contribute in a constructive manner is when the delays are much greater than the time guard interval of the DAB signal, i.e. greater than 300 μs at VHF. Delays of this magnitude are extremely rare in most types of terrain where multipath reflections are apparent.

3.2 Multiplexing

Currently, individual broadcasters are allocated a frequency to transmit a given programme. A whole frequency channel bandwidth is needed for one radio programme. Unlike conventional analogue broadcasting, the DAB system enables several sound programmes and data services to be multiplexed together and broadcast on the same radio-frequency channel to an identical coverage area. These programmes may emanate from different programme sources, but they need to be synchronized.

Different programme/service providers will have to agree on how to share the data capacity available in the multiplex. It may well be that an appropriate regulatory regime will need to be established on a national level or on an international level consisting of a group of countries. Different degrees of flexible allocation of bit rates to broadcasters is possible. Broadcasters will be able to compete on the basis of the programme quality and attractiveness to the listener, rather than on technical quality grounds.

The number of sound programmes in an "ensemble" (i.e. a multiplex) depends on the trade-off implemented between the encoded bit rate per audio programme, the channel protection that is provided against errors occurring on the propagation path, and the data capacity required for the various programme-associated and independent data services that are included in the ensemble. The system allows for flexible changes of the multiplex structure; these reconfigurations may occur on the fly and are automatically interpreted by the receiver.

3.3 CD quality

One of the main benefits of the Eureka 147 is that the high sound quality, normally indistinguishable from that of the CD is effectively available in the receiver. Subjective tests have shown that CD quality can normally be achieved at 224 kbit/s for stereophonic signal. However, broadcasters may exploit the built-in flexibility and use lower audio bit rates when true CD audio quality is not required.

3.4 Spectrum efficiency

A further advantage of DAB is that it is spectrum-efficient. This means that it will be possible to increase the number of radio stations - initially by a factor of at least three when compared with FM - without congesting the radio waves. As more efficient audio coding (compression) methods are introduced, it will be possible to carry even more radio programmes with no degradation to existing services, and without needing to modify existing receivers. A radio set of the future will thus make it possible to choose, for example, a favourite type of music station from among hundreds of music stations.

3.5 "Active echoes"

The Eureka 147 DAB system is able to use "passive echoes" such that they add in a constructive manner to the direct signals already received. The Eureka system is also able to use "active echoes" constructively - i.e. delayed signals generated by other co-channel transmitters. This leads to two important concepts:

- single frequency networks (SFNs);
- co-channel gap-fillers.

The SFN concept enables all transmitters covering a particular area with the same set of sound programmes to operate on the same nominal radio-frequency channel, i.e. within the same frequency block. Although the signals emitted by the various transmitters are received with different time delays, the receiver integrates the powers of the direct signal and the "active echoes" received within the guard interval.

Gap-filling represents the second type of application which makes full use of the "active echo" concept. A gap-filler acts rather like a mirror; it receives the signals from the main transmitter and retransmits them at low power on the same frequencies to provide coverage in an area where the main transmitter is not received satisfactorily. Although the listener receives signals from both the main transmitter and the gap-filler at slightly different times, the two sets of signals add together constructively to enhance the reception of the programme. The gap-filling concept is useful both for terrestrial and satellite broadcasting systems.

As a result of these two concepts, DAB eliminates the problem of having to re-tune car radios at frequent intervals.

3.6 Flexible bit rates

The Eureka 147 DAB system is a highly flexible and dynamically re-configurable system. It can accommodate a range of bit rates between 8 and 384 kbit/s, with a range of channel protection mechanisms. Extremely low audio bit rates, such as 8, 16 and 24 kbit/s per monophonic audio programme are made possible by using a sampling frequency of 24 kHz, instead of 48 kHz, in conformance with ISO/IEC MPEG 2 Layer II audio coding.

Some broadcasters are interested in using low audio bit rates per audio channel, say between 16 and 64 kbit/s, in order to transmit more channels at slightly reduced quality. With a bit rate of 32 kbit/s per audio channel, the Eureka multiplex of 1.5 MHz can accommodate as many as thirty-six channels, with ½ channel protection coding.

3.7 DAB transmission modes

Technically, the Eureka 147 DAB system can be used at any frequency between 30 MHz and 3 GHz. This wide range of frequencies includes VHF Bands I, II and III, UHF Bands IV and V, and L-Band (which is around 1.5 GHz). Since the propagation conditions vary with frequency, four DAB transmission modes are used.

These modes are detected automatically by the receiver and are transparent to the user. *Mode I* is suitable for SFNs operating at frequencies below 300 MHz. *Mode II* has been designed for local and regional services at frequencies below 1.5 GHz and *Mode III* is available for satellite broadcasting below 3 GHz. *Mode IV* has recently been introduced to enable existing transmitter sites to provide optimum and seamless coverage of large areas by means of SFNs operating in L-Band. The parameters of Mode IV lie between those of Mode I and Mode II.

3.8 Data services

Although audio has been its primary *raison d'être*, the Eureka 147 transmission system can also be used to carry a large variety of programme-associated and independent data services. Many data services of the programme-associated category will probably be transmitted from the outset and will be received by the first generation of DAB consumer-type receivers. Later on, independent data services may also appear. These would be received by dedicated data receivers, including those incorporated in desk-top and lap-top computers. Two examples of this application are the electronic delivery of newspapers and the transmission of compressed video images such as weather maps.

The Eureka system's immunity to multipath and other reception impairments will guarantee error-free data reception in the mobile environment . Hence, the Eureka 147 system is an ideal complement to the wired Information Highway distribution system now being established worldwide.

3.9 Future-proofing

The DAB system is future-proof. Once the receiver has been purchased, it will not become obsolete as the digital technology develops, nor as new services and applications emerge. Receivers will respond to those types of services for which they are designed and will simply ignore newer types of services in the DAB multiplex. Most importantly, as new types of services are introduced, older DAB receivers will not cease to work on continuing service types.

In Europe, for example, DAB delivery will commence via terrestrial networks. Nevertheless, the receivers designed for use with these terrestrial services should, in principle, also be able to receive future DAB services delivered via satellite and cable. In other words, the Eureka 147 System will become a universal means to deliver sound programmes and data, irrespective of the transmission medium used-[1].

4 System constraints

The design of a new system is inherently a trade-off between different technical and operational choices. Thus, when introducing DAB services, one must be aware of the technical constraints of the Eureka 147 system, which may generally be overcome when designing its implementation and operational practice.

4.1 System processing delay

The DAB system chain includes several blocks which introduce a significant processing delay. For example, the time interleaver introduces a delay of 384 ms, and the audio coder/decoder introduces a delay of several tens of milliseconds. The total delay in the system may vary from one implementation to the next.

The system delay should be taken into account when the receiver switches between DAB and FM "simulcast" programmes, so that a seamless transition is obtained. It will become necessary for simulcast FM transmissions to be delayed by nominally the same amount, typically between a half and one second, regardless of the receiver design. This nominal delay should be taken into account when signalling the current time information.

1 Studies are now being conducted on the use of the Eureka 147 system as a digital television delivery medium for mobile reception on small screens.

4.2 Frequency accuracy in SFNs

In order not to reduce the performance of the DAB system, the difference in frequency between geographically-adjacent transmitters must be kept to an absolute minimum - of the order of a few hertz in 10^8. Consequently, the local oscillators of all transmitters should be locked to a rubidium oscillator, or to a common reference which is distributed to all the transmitters using, for example, GPS.

4.3 Time accuracy in SFNs

The time difference between geographically-adjacent transmitters will have an implication on the system's capability to cope with "active echoes". Therefore, all the transmitters operating in an SFN should be time-synchronized with an accuracy of more than 10 % of the guard interval (i.e. approximately 25 µs in Transmission Mode I). In fact the tolerance selected should be taken into account when planning coverage; it is not difficult to achieve an accuracy of 1 µs or less.

4.4 Bit by bit compliance in SFNs

In principle, the bit-streams emitted from all transmitters operating in an SFN should be identical. If this condition is not fulfilled, there will be a "mush area" (i.e. interference zone) between the transmitters where the DAB receiver may be confused. Tests are being undertaken to assess the size of the mush area in the case where a local transmitter "opts out" from an SFN, thus emitting a different bit stream, or part of bit stream, to the other transmitters in the SFN.

4.5 Receiver speed limit

As the speed of a vehicle increases, the performance of an on-board DAB receiver progressively degrades, due to the Doppler effect. The "receiver speed limit" is a soft limit and may be considered as the vehicular speed at which the RF signal-to-noise ratio degrades by 4 dB, due to the worst-case Doppler effect (i.e. equal signals from opposite directions). While this does not affect the audio quality, it may reduce the DAB coverage area slightly - but only in a fast-moving vehicle. In the case of an SFN operating in Mode I at VHF, the receiver speed limit is upwards 250 km/h[2]. When the receiver operates at 1.5 GHz and Transmission Mode II is used, the speed limit is about 160 km/h.

5 Recent technical studies and developments

Several compatible refinements to the Eureka 147 DAB System have recently been performed and agreed by the Eureka 147 Consortium, as follows: a) additional transmission mode (i.e. transmission mode IV) for optimum transmission of DAB signals in Single Frequency Networks in L-Band, and b) audio half-sampling rate and extension to very low audio bit rates (i.e. 8, 16, 24 kbit/s). The former amendment to the ETS was felt appropriate in order to increase the separation distance of the transmitters operating in a SFN beyond that allowed in transmission mode II. Thus, transmitter networks in L-band may become substantially more economical, since the existing transmitter infrastructure used for TV and FM networks could generally be reused for DAB. The latter amendment was required by commercial broadcasters who would like to use the System with extremely low bit rates, thus increasing the number of audio programmes within the multiplex. The subjective audio quality achievable and the sensitivity to transmission errors at these low rates is under study.

Progress has been made recently on defining the interfaces of the Eureka 147 System. Two interface-related DAB standards, produced within the framework of the Eureka 147 DAB project, are very important for broadcasters. The first one, *Ensemble Transport Interface (ETI)* [7], has already been issued by the EBU. The second one, entitled *Service Transport Interface*, will be completed by the Eureka 147 Consortium this autumn. In addition, a common interface in the receiver, termed *the Receiver Data Interface (RDI)*, is under consideration to allow the connection of the various dedicated application decoders, computers, printers and dedicated decoders for data applications, as

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[2] Below 250 MHz the limit rises to about 700 km/h at the bottom end of Band II.

well as devices for audio post-processing and recording, and also to implement a return channel for receiver control from an application terminal.

Several refinements to the Standard were necessary to readjust some important service features, in particular the conditional access, service identification, data service component types, service linking, time and country identifier, programme type and announcements.

Based on the work carried out by the Eureka 147 Project Group, the EBU is planning to publish *Guidelines for Implementation and Operation* [6] as a companion to the ETS 300 401 Standard. This document - which is extremely important for EBU Members who wish to start DAB services - is intended to provide additional information on the system, to aid interpretation of on-air signals conforming to the ETSI Standard, and to assist the broadcasters and manufacturers to implement systems using the specification features as intended.

6 System standardization

A common transmission standard for DAB, as opposed to a multitude of proprietary standard, has always been preferred by EBU Members. A single standard will readily lead to the mass production of DAB receivers, bringing their prices down to an affordable level. It will open the door to free market competition, resulting in a wide variety of receiver brands offering a range of qualities and features. A unique DAB standard will mean that the same core electronic circuitry in the receiver could be used in all parts of the world, as is the case today with AM and FM radio, and maintain global portability of radio receivers. It will also reduce the need to perform standards conversion with its inherent degradation of the signal. A single DAB standard will introduce stability in the market and DAB technology would last for a long time.

In pursuing the above objectives, the EBU has been instrumental in establishing a unique DAB standard on both European and worldwide levels.

6.1 ETSI

In late 1994, the Eureka 147 DAB system was adopted by the European Telecommunication Standards Institute (ETSI) as the European Standard. ETSI then published the standard - *ETS 300 401* - in February 1995. The ETSI Standard describes the technical details of the broadcast on-air signal and is applicable to terrestrial, satellite and cable delivery, in all the frequency bands that are available for broadcasting above 30 MHz. The concept of the Standard is such that it includes both mandatory and optional features of the system, and it allows for future functional refinements and additions by the application of appropriate software tools. The Standard permits different levels of implementation to meet a variety of market requirements, production costs and receiver types.

6.2 ITU

The global DAB standardization process is being conducted within the International Telecommunication Union (ITU) which, among other things, considers new developments in broadcasting technology and agrees the technical standards of broadcasting systems - for both radio and television - on a worldwide basis. Over the years, EBU Members have contributed extensively to different ITU working parties on the results of R&D work carried out in their own laboratories.

At the late 1994 meetings of ITU-R Working Parties 10B and 10-11S, it was decided unanimously to adopt two Draft Recommendations, BS.1114 [1] and BO.1130. The first of these drafts recommends to ITU members to use Digital System A for terrestrial delivery in the frequency range 30 - 3000 MHz. The second one recommends that administrations wishing, in the near future, to implement BSS (sound) which meets some or all of the requirements stated in ITU-R Recommendation BO.789 should consider the use of Digital System A.

Both these Draft Recommendations include a Note which, in principle, opens the door to other systems as well - when they are sufficiently developed and tested, and when they have shown that they would meet the agreed and approved ITU requirements given in Recommendations 774 or 789 (for terrestrial and satellite systems, respectively). However, the global market is set to become established with Eureka 147, given the commitment of the receiver industry world-wide.

The achievement of a common worldwide Standard is rare in the history of broadcasting. In the case of the Eureka 147 system, it was only possible due to the joint efforts of, and extensive cooperation between, European and Canadian broadcasters, research institutes and the radio manufacturing industry. The Eureka system also had the support of many administrations outside Europe, particularly from the developing countries.

6.3 CENELEC

The European Committee for Electrotechnical Standardization (CENELEC) is planning to release a receiver standard for Eureka 147 DAB, by the end of 1995. Based on a draft technical report already prepared by European Association of Consumer Electronics Manufacturers (EACEM), the CENELEC Standard will define only those mandatory parameters which are necessary for Eureka 147 DAB receivers to interpret correctly the received signals; non-mandatory parameters will not be specified and may be open to competition in the marketplace.

7 Eureka 147 results in EIA tests

The EIA tests were divided into three categories as follows:

a) subjective quality tests on the source coding system, operating in a clear channel (i.e. with no transmission errors);

b) objective digital tests on the overall system performance;

c) objective and subjective compatibility tests carried out to determine the interaction between the digital audio broadcasting system and the analogue transmission system within the FM band.

The quality assessment results show that the Eureka 147 system - using ISO MPEG Layer II Musicam at 224 kbit/s - had the highest overall rank and the most consistent ratings across the whole range of audio material which was used for the tests. Eureka 147 was the only system that never fell below the "perceptible but not annoying" range. Out of nine critical audio passages that were evaluated, four were judged to be transparent.

The published test results show that, in general, the "in-band" digital systems as currently considered in the USA may cause intolerably high interference to, and suffer interference from, the analogue services that are overlaid - particularly in a multipath environment. Therefore, those broadcasters who wish to preserve the high broadcasting standards of their existing FM services should probably be very careful when opting for such a digital solution, given the present stage of its development.

The test results on the Eureka 147 DAB system published by the EIA are very favourable. They confirm the conclusions of prior extensive laboratory and field tests conducted in Europe, Canada, Australia and elsewhere - that the Eureka system eliminates problems such as FM multipath and signal failure (dropout). It also enables digital radio to coexist with AM and FM services with no interference.

8 Frequency planning aspects

The Eureka 147 System is capable of operation in any frequency band between 30 and 3000 MHz. The required spectrum planning parameters for the Eureka System are well defined and are available in the public domain (see for example the ITU Special Publication on Terrestrial and Satellite Digital Sound Broadcasting to vehicular, portable and fixed receivers in the VHF/UHF bands, Geneva, 1995). The System has been implemented in various parts of the world at frequencies in the region of 50, 80, 100, 210-240, 600, 800, and 1452-1492 MHz.

Generally, the higher the frequency, the higher the cost of the coverage. Also, building attenuation increases with the frequency. Nevertheless, all the experiments have shown that the Eureka System performs very well at higher frequencies, including in L-Band. Lower frequencies (e.g. Band I) are characterized by a high level of man-made noise and should be avoided.

The introduction of DAB requires a forward-thinking long-term plan. To this end, sufficient radio-frequency spectrum is required to prevent significant co-channel and adjacent-channel interference between DAB services. Clear spectrum is required in blocks of about 1.75 MHz. In Europe, such spectrum slots are normally not available in Band II which is extensively used by FM services. Most of the countries opted to start their DAB services in Band III or in L-Band.

A three-week Planning Meeting was convened in July 1995 by the European Conference of Postal and Telecommunications Administrations (CEPT). The aim of this Meeting was to produce a Special Arrangement for the introduction of terrestrial transmissions of the Eureka 147 DAB system in the frequency bands 46 - 68 MHz, 174 - 240 MHz and 1452 - 1467.5 MHz, as well as to prepare an associated Frequency Block Allotment Plan, taking into account the final requirements of the CEPT member countries.

The Allotment Plan drawn up at the Meeting provides practically all the member countries of the CEPT with two sets of frequency blocks, each of width 1.536 MHz. This is a vital prerequisite to the wide-scale launch of terrestrial DAB services in Europe. Most of the CEPT countries opted for frequency block allotments in VHF Band III and in L-Band.

Eighty-five frequency blocks can, potentially, be used for current and future DAB services in Europe. The distribution of these frequency blocks is as follows:

- 12 blocks in VHF Band I (47 - 68 MHz);
- 38 blocks in VHF Band III (174 MHz - 240 MHz);
- 23 blocks in L-Band (1452 - 1492 MHz);
- 12 blocks in VHF Band II (87 - 108 MHz).

Each frequency block carries a two- or three-character label, which is easier to remember than the centre frequency of the block, and which is convenient for receiver manufacturers and consumers to use when initially programming their receivers. The labelling system of the frequency blocks in VHF Band I and Band III is fully compatible with the existing VHF television channel numbers (i.e. Channels 2 to 13). Each of these television channels can accommodate four DAB blocks; six blocks in the case of Channel 13.

One of the important results of the Meeting was a definition of the centre frequency of each ensemble (i.e. frequency block). This information is very important for receiver manufacturers and may help substantially to simplify the receiver design; before the Meeting, any frequency in the 16 kHz roster could be used as the centre frequency, resulting in a very large number of possibilities. The number of defined centre frequencies has now been reduced to match the total number of ensembles allocated in Band III and in L-Band (i.e. 61). These centre frequencies are likely to be implemented in the first-generation DAB receivers manufactured for the European market.

9 Eureka 147 - a system for satellite broadcasting

The chosen satellite system (designed to cover large areas) will ideally have the same modulation/coding system parameters as the ground-based system (designed to cover regional/national territories), such that the same receiver could be used. An essential requirement for any new satellite system is that it should be able to provide for mobile and portable reception in all types of propagation environments (rural, urban, etc.).

Although the Eureka 147 DAB system has been developed as a terrestrial system, there is no technical reason why it could not be used for satellite delivery as well. Many computer simulations have shown that this assumption may be true, but real experiments are needed to demonstrate that satellite delivery is both a technically viable and an economically attractive proposal.

Two such experiments have been conducted recently - one in Australia, the other in Mexico. The Australian test was carried out using the Optus B3 satellite at 1552 MHz. The trial in Mexico, carried out by the BBC, used the Solidaridad satellite. Both satellites were originally launched to provide mobile phone services; they were not specifically designed for multi-carrier systems such as Eureka 147. Even so, the results showed that fixed and portable reception of DAB signals via a satellite is technically feasible. Due to the low transmitting power of the test satellites, mobile reception was possible only under line-of-sight conditions.

A satellite simulation - using a helicopter - is being carried out jointly by the ESA, the IRT and the BBC in Munich, to determine the service-availability performance (i.e. percentage of coverage) for different elevation angles.

For international broadcasting, all WARC-92 bands (i.e. bands located at 1.5, 2.3 and 2.6 GHz) should be considered. Preference should clearly be given to the 1.5 GHz band for technical and economic reasons (the best trade-off between the size of satellite transmit antenna and its transmit power). Preliminary studies have shown that, at 2.6 GHz, considerably larger transponder output power would be required (of the order of four times greater than that required at 1.5 GHz). DAB Transmission Modes II, III or IV are suitable for use at these frequencies.

10 Eureka 147 - a multimedia carrier

Further developments are underway to study the use of the Eureka 147 System as a multimedia and data broadcasting system. These studies are aimed at expanding the future use of the Eureka 147 DAB system beyond the provision of excellent sound reception in adverse mobile and portable environments. In addition to the conventional audio services, the system is opening up many new opportunities to carry a number of non-audio services, such as text, still pictures, moving images, etc.

Examples of DAB data services currently being implemented are given below. These services may be presented either in the form of textual information (shown, for example, on a simple receiver display of, typically, between 8 and 128 characters), still pictures or even video images on more sophisticated displays.

- *Programme-associated services* such as current song title, interpreter and performer, lyrics, news headlines, CD covers, etc.;
- *News* including events, traffic messages, weather, sport, stock market, travel and tourist information;
- *Traffic navigation* by means of transmitted digitized road maps, combined with positional information provided via the GPS or TII system.
- *Advertisements and sales* including sales catalogues, purchase offers, etc.;
- *Entertainment* including games and non-commercial bulletin boards;
- *Closed user group services* such as banking information, electronic newspapers, fax print-outs and remote teaching.

Ideally, multimedia services should be fully interactive, in which case the consumer can communicate with the service provider's database. Since broadcasting services are one-way only, the return channel could be provided by GSM telephone (in the case of mobile DAB receivers) or via a telephone line (in the case of a fixed receiver). Nevertheless, a semi-interactive mode is also possible. In this instance, information is downloaded by the service provider to the user's data terminal and stored there as a database. All interactivity is then handled within the user's data terminal, but the database contents have to be updated regularly by the data service provider. The storage capacity of the user's terminal is a trade-off between the service transmission rate, the repetition rate and the cost of the memory.

A key factor for the success of DAB will be its ability to address each receiver individually. This will allow service providers to customize the "bouquet" of services provided to each user, and even to identify the user in an interactive transaction. This feature has some far-reaching implications, particularly for privately-funded radio.

Studies are continuing on the suitable presentation of DAB data services. Currently, data services specified in the ETSI Standard have a text-based presentation. In order to improve the man-machine interface, the Eureka 147 System will be enhanced to support a graphical user interface, such as, for example, Microsoft Windows. This will be of importance for screen-based services which seem to be more relevant for stationary and portable receivers. For mobile receivers, synthesized speech-based interfaces are a better alternative, as they would be less distracting to drivers. Clearly there are many opportunities for DAB services to develop and make the radio medium grow into a new age in the years to come.

For the user's data terminal, a unified transmission protocol will be very helpful, as no distinction between different transport mechanisms would be necessary. A software-based language for object-oriented page description is being developed to define a communication and a presentation layer. Such a unified protocol for the multimedia transport mechanism could be used not only with DAB services, but also in other communication systems.

11 Manufacturers

In order to receive DAB services, consumers will need to buy a new kind of receiver. The first generation of consumer DAB receivers will also contain FM and AM circuits which, initially, will be analogue. However, it will not be long before the AM and FM circuits in a DAB receiver become digital. These all-digital AM/FM/DAB receivers will be based on advanced computer technology, which will allow the downloading of large quantities of information to program the radio set and its associated equipment (PC, DCC, MiniDisc recorder, etc.)

At a recent IFA fair in Berlin, several manufacturers displayed their current DAB receivers. The most advanced are those from Bosch, Grundig, Philips and Kenwood. Current receivers look more like semi-professional equipment; the DAB part is in a separate box, mounted in the boot with a link to an ordinary FM/RDS receiver in the dash-board. These first generation car receivers are not generally available yet; they can only be purchased on special order and in limited quantities for evaluation purposes. Nevertheless, several tens of thousands of such test receivers will be available for evaluations in Europe over the next two years at a price of about 2000 to 4000 DM.

The purpose of these tests is to estimate the prospective interest of European audience in, and demand for, DAB services. The tests will be useful to identify the key elements of a range of DAB features and applications that will be most attractive to the general public. The attraction may vary, of course, from country to country, it may be different for in-home radios and in-car radios, and it may depend on the price range of the receiver. The study will show how DAB will be accepted in different demographic sectors (age, sex, socio-economic groups, etc.). Also, the study will demonstrate which DAB programmes and data services could be specifically interesting for DAB compared to existing FM/AM broadcasts and to what extent interactivity would be important for radio medium. This information will be highly valuable both for broadcasters and receiver manufacturers.

Concerning the receiver prices, it is quite normal that, initially, they may be relatively high, but it is expected that they are going to drop very rapidly, as the market starts to pick up. It is of crucial importance for a successful market penetration that the initial incremental costs of the addition of DAB to FM/AM radio is not too high. Therefore, it may be sensible that receiver manufacturers spread their research, development and production cost over a longer period of time than they usually do with other electronic consumer products. It is likely that the competition on the DAB receiver market will be very big; more than 15 manufacturers have already started to design their DAB receiver, and three or four announced that they will develop their own integrated circuits. The market will probably require a range of receivers, offering a range of receiver functions and capabilities. Nevertheless, it is important that all receivers are easy-to-operate and user friendly, so that the users will not be frustrated when using them.

12 Conclusions

It is becoming clear that DAB will revolutionize the broadcasting structure and operational practices. For example, many questions are being posed over the control of the multiplexer, the value-added data services and the conditional access. The broadcasters, together with manufacturers and network providers, are continuing their cooperation to investigate how DAB can be used optimally for new applications which will be attractive for all listeners. The recently-formed EuroDab Forum will be instrumental in pursuing those objectives.

The Eureka 147 System is likely to become a generic audio/multimedia broadcasting system using ground-based transmitter networks where high signal quality, uninterrupted service quality and mobile/portable reception are important requirements. The potential of the Eureka System for the satellite delivery to mobile receivers at 1.5 GHz is being investigated.

Several tens of pilot trials using terrestrial and satellite transmitters are being carried out in Europe and world-wide. By Autumn 1997, more than 100 million people in Europe will be covered by the DAB signal terrestrially. At the same time, first consumer DAB receivers, both for home and the car, will be commercially available from several consumer electronics manufacturers, hopefully at an affordable price.

The big unknown that remains is the programmes. The production of attractive, interesting and entertaining programmes, specifically designed for the DAB medium, is yet to become the reality. So far, only a few new programming ideas have been put forward, but it is hoped that the EuroDab Forum could help to generate some ideas on this matter.

ADVANCED TELEVISION: PART II

Sunday, April 14, 1996
1:00 - 5:00 pm

Session Chairperson:
Robert Seidel, CBS, New York, NY

*A TECHNICAL REVIEW OF THE PERFORMANCE OF SIDE MOUNTED UHF ANTENNAS FOR ATV SERVICE
Kerry W. Cozad
Andrew Corporation
Orland Park, IL

THREE MPEG MYTHS
Christopher D. Bennett
Hewlett Packard Company
Santa Clara, CA

FREQUENTLY ASKED QUESTIONS FOR STATION ATV PLANNING
Bruce Jacobs
Prairie Public Broadcasting
Fargo, ND

A UNIQUE SOLUTION TO THE DESIGN OF AN ATV TRANSMITTER
Nat S. Ostroff
Comark
Colmar, PA

*COVERAGE EXTENSION WITH DIGITAL VSB TRANSMISSION
Richard Citta
Zenith Electronics Corporation
Glenview, IL

CAN ATV COVERAGE BE IMPROVED WITH CIRCULAR, ELLIPTICAL, OR VERTICAL POLARIZED ANTENNAS?
Robert J. Plonka
Harris Corporation, Broadcast Division
Quincy, IL

CHANNEL COMBINING IN AN NTSC/ATV ENVIRONMENT
Dennis Heymans
Micro Communications, Inc.
Manchester, NH

*SOLID STATE HIGH POWER UHF ADVANCED TELEVISION TRANSMITTER CARLTON DAVIS
Carlton Davis
Westinghouse Wireless Solutions
Baltimore, MD

DESIGN AND IMPLEMENTATION OF A 3-CCD, STATE OF THE ART, 750-LINE HDTV PROGRESSIVE SCAN BROADCAST CAMERA

Stuart M. Spitzer
Polaroid Corporation
Cambridge, MA

Anton Moelands
Broadcast Television Systems
Breda, The Netherlands

525-LINE PROGRESSIVE SCAN SIGNAL DIGITAL INTERFACE STANDARD

Akihiro Hori
NTV
Tokyo, Japan

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*Paper not available at the time of publication.

THREE MPEG MYTHS

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ABSTRACT

As disk-based video migrates from a variety of incompatible JPEG formats, the clear choice is the standards-based MPEG. While alternative compression schemes will continue to coexist, MPEG compression has emerged as a major technical force in the broadcast community. Because MPEG is relatively new, there is much confusion and misinformation about the advantages this technology brings to broadcasters. Compounding this confusion is MPEG's variety of different -- but compatible -- flavors for different applications. This paper explains the basic workings and terminology of MPEG, and explores the differences between the MPEG variations. Building on this base, we will provide a framework for evaluating compression techniques in server applications, and challenge three current myths: 1) that higher bit rates are always better, 2) that 4:2:2 is always superior to 4:2:0, and 3) that it is not practicable to edit MPEG-2.

WHY COMPRESS?

The transformation of analog video information into the digital world has many benefits. Quality is more consistent. Copies suffer no generation losses. Images are easier to manipulate.

Images are by nature filled with redundant information. The statistical likelihood that adjacent pixels share similar characteristics is excellent for any image except white noise. Consequently, images are obvious candidates for compression. Virtually every video standard employs compression in some fashion, including analog composite standards (for example NTSC). Even "perfect" CCIR-601 reduces chroma information 50%.

Compression performed in the analog world is necessarily crude and marginally effective. Digital manipulation, combined with the vast computational power now available, enables far more sophisticated and effective compression techniques. As a result, digitally compressed images are much easier and cheaper to manipulate, store and move. This enables far more cost-effective server solutions with no noticeable degradation in image quality.

WHAT IS DIFFERENT ABOUT MPEG?

There are several differences between MPEG and most other compression techniques.

First, MPEG narrowly focuses on -- and is tuned for -- video. MPEG uses a single color space (Y, C_r, C_b), a limited range of resolutions and compression ratios, and has built-in mechanisms for handling audio. In contrast, JPEG adapts to the widest range of still picture applications, offering tremendous flexibility: a limitless range of image resolutions, color spaces, and compression ratios, but no concept of audio.

Second, MPEG efficiently guarantees a target bit rate. Simple control of a single variable (combined with a small picture buffer) ensures a constant bit rate. This makes MPEG extremely predictable, and allows very efficient use of server bandwidth. Motion JPEG includes some similar features.

Third, and most fundamental, MPEG takes advantage of two characteristics of video: the high degree of commonality between pictures, and the usually predictable nature of movement. The feature capitalizing on these characteristics is "inter-picture" encoding.

To accommodate inter-picture encoding, MPEG creates three types of pictures. Intra-pictures are completely self-contained and analogous to ordinary JPEG pictures. Predicted pictures and Bi-directional pictures, though they contain fewer bits, are functional supersets of I-pictures. P-pictures contain forward interpolation or "prediction." B-pictures contain forward and backward prediction. P and B pictures also include compensation for any errors during prediction, so they are just as accurate as I-pictures.

MPEG motion estimation

Forward and backward prediction greatly reduce the data required to describe complex changes (Figure 1). The MPEG standard does not specify the algorithms, which are usually very complex.

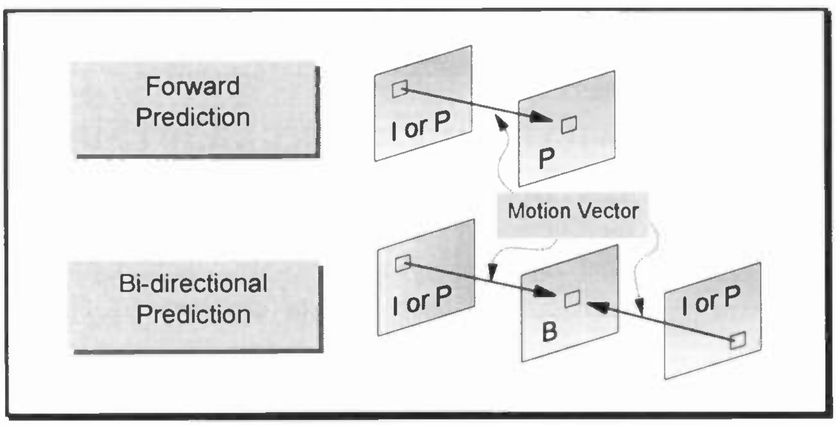

Figure 1. MPEG Motion Estimation

The computational power required to compute the prediction real-time is very high. For example, a single Hewlett Packard MPEG-2 encoder employs twelve high-performance RISC processors.

Once calculated, very simple "motion vectors" represent the prediction. These vectors simply describe the relative position of a particular block compared to the reference I or B pictures. After calculating the motion vectors, the encoder subtracts the "predicted" picture from the actual picture. Usually very little information remains, and compresses easily.

The result: a P or B picture with all the accuracy of a self-contained I-picture, but consuming up to six times less storage space. P & B pictures are what enable MPEG to be so much more efficient than JPEG.

MPEG picture structure

The I, P and B pictures organize into arbitrary sequences called Groups Of Pictures, or "GOP" (Figure 2). Officially, the only defined rule for a GOP is that it must contain at least one I-picture, but GOP has become synonymous with I-P-B sequence. Longer GOP's are more complex, but because they take better advantage of video's characteristics, they are more efficient. Shorter GOP's are generally much easier to implement, but they consume more space. I-only GOP's are little more efficient than JPEG, and eliminate the efficiency advantages of MPEG. Hewlett Packard's MPEG-2 encoders use a 15 picture GOP and are *very* efficient.

MPEG standards

Three active MPEG standards currently exist.

MPEG-1 is the original MPEG implementation, targeted at multimedia applications with SIF (352x240) resolution.

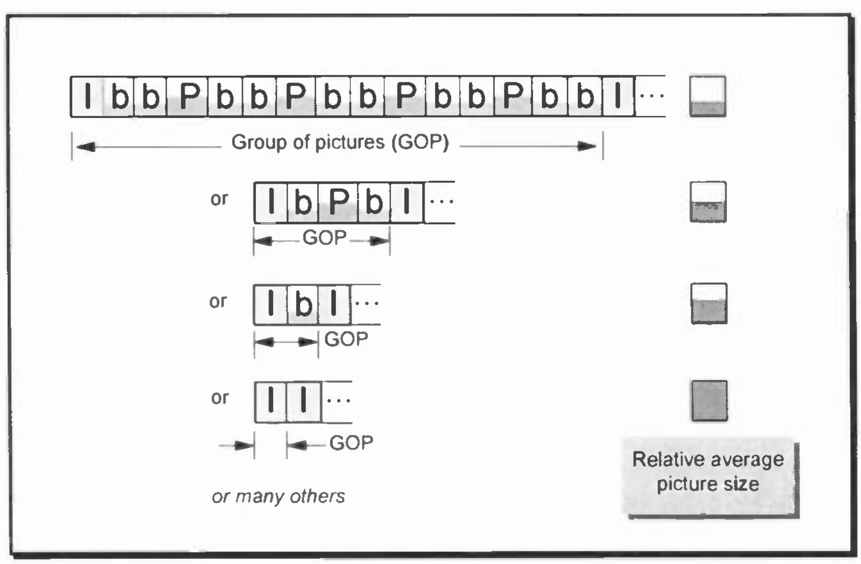

Figure 2. GOP Structures

MPEG-2 offers full CCIR-601 resolution for broadcast applications. Most MPEG development focuses on MPEG-2, which adds capabilities for interlaced video.

MPEG-3 was originally targeted at HDTV applications, but since MPEG-2 absorbed the key specifications, MPEG-3 is no longer in use.

MPEG-4 is a new standard using quarter-SIF resolution and very low bit rates for teleconferencing applications.

MPEG-2 profiles and levels

Five "profiles" and four "levels" describe the MPEG-2 organization (Table 1). Profiles are analogous to features, describing the available characteristics. Levels describe the upper bounds for a given feature, and are analogous to performance specifications.

By far the most popular portion of the MPEG-2 standard is Main Profile/Main Level, which is the "common language" of the MPEG world. All higher profiles are capable of decoding Main Profile/Main Level streams.

The Simple profile offers advantages for some applications involving inexpensive set-top boxes, and Main/Low is replacing MPEG-1 in many applications.

The SNR and Spatial profiles offer some interesting features, but are generally not expected to see common use. The High profile targets HDTV applications, and along with Spatial offers the potential for dual layers of pictures within a single data stream. This feature enables a high-performance receiver to use both layers to display a HDTV picture, while an inexpensive receiver uses only the low bit-rate layer to display a standard picture.

A new "4:2:2" profile is in development that is similar to Main Profile/Main Level, but offers 4:2:2 chroma resolution (Figure 3), higher bit rates, and increased vertical resolution. Sometimes referred to as "Studio" or "Professional," this profile should be finalized by mid-1996.

The three *key* flavors of MPEG-2 are SIF (Main Profile/Low Level), Main (Main Profile/Main Level) and 4:2:2 (4:2:2 Profile/Main Level).

Levels (specifications)		Profiles (features)				
		Simple	Main	SNR / Spatial	High	4:2:2
	Pictures	I, P	I, P, B	I, P, B	I, P, B	I, P, B
	Chroma Format	4:2:0	4:2:0	4:2:0	4:2:2	4:2:2
High	Max Bit Rate		80 Mb/s		100 Mb/s (25 base layer)	
	Samples/line		1920		1920	
	Lines/frame		1152		1152	
	Frames/sec		60		60	
High 1440	Max Bit Rate		100 Mb/s	60 Mb/s (15 base layer)	80 Mb/s (20 base layer)	
	Samples/line		1440	1440	1440	
	Lines/frame		1152	1152	1152	
	Frames/sec		60	60	60	
Main	Max Bit Rate	15 Mb/s	15 Mb/s	15 Mb/s	20 Mb/s (4 base layer)	50 Mb/s
	Samples/line	720	720	720	720	720
	Lines/frame	576	576	576	576	608
	Frames/sec	30	30	30	30	30
Low	Max Bit Rate		4 Mb/s			
	Samples/line		352			
	Lines/frame		288			
	Frames/sec		30			

Table 1. MPEG-2 profile and level summary. For simplicity, the SNR and Scaleable profiles are condensed into a single column.

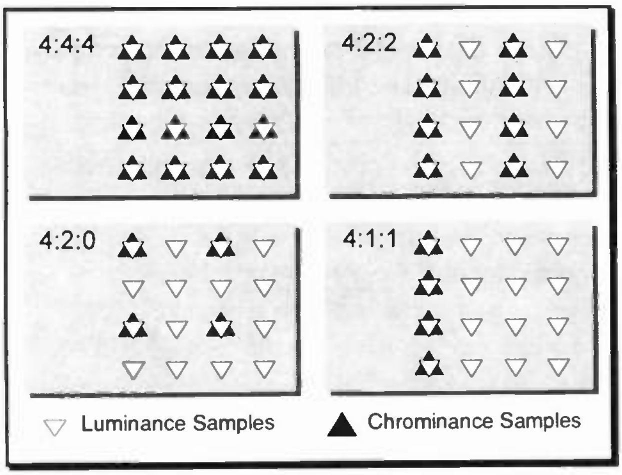

Figure 3. Common chrominance formats. Note that 4:2:2 truncates horizontal chroma 50%.

MPEG-2 "stream" hierarchy

One of the key features of the MPEG standard is the establishment of protocols for handling various forms of video. Accordingly, MPEG defines a hierarchy of three data "streams": elementary, program & transport.

The most simple, elementary streams simply describe individual audio, video or data sequences. Program streams multiplex elementary streams in a relatively error-free environment, typically a local network. Among other features, the stream specifications describe how to combine the three data types within a single set of variable-length packets. Labeled with timing and buffering information, the packets also include program description information such as frame and audio sampling rates.

Transport streams are designed for multiplexing multiple program streams for delivery on distributed digital networks. The stream specifications include sophisticated timing, synchronization, and jitter-correction mechanisms crucial for reliable long distance transmission. By design, the fixed data packet size maps perfectly with ATM.

Combined, these three transportation standards provide a means of realizing the promise of widespread video data interchangeability on a local, national, and global scale.

WHY IS MPEG SUPERIOR?

As a compression format, MPEG-2 has several compelling advantages. First, it is a true international standard, independent of any one manufacturer. No more crazy tape format wars designed to lock out competitors. Furthermore, the specifications allow for wide latitude in the means manufacturers use to create compressed signals, but very little flexibility in what the data structure must look like upon completion. This provides manufacturers the ability to continually refine the efficiency and effectiveness of their compression techniques, but guarantees backward compatibility and continued interoperability.

Second, MPEG-2 is scaleable. The standard provides for a wide range of applications, but is carefully crafted so that the higher Profiles and Levels are supersets of the lower. This provides great flexibility in matching a compression scheme to an application, but ensures a significant level of interoperability even among different applications.

Third, MPEG-2 is transportable. Designed from the start for video delivery, it incorporates a range of powerful data movement mechanisms.

Fourth, and probably most important to broadcasters with vast archives of spot, program and news material, MPEG-2 is efficient -- *extremely* efficient.

MPEG-2 is extremely efficient

Because of inter-picture encoding, full IPB MPEG-2 is typically three to five times more efficient than JPEG or I-picture MPEG, and two times more efficient than IB MPEG. This efficiency has many benefits.

For a given video quality, the number of channels available from a single server increases substantially, since each channel uses much less bandwidth. The storage costs drop dramatically. The network bandwidth required to move programming greatly reduces, increasing the speed of material movement. Off-line archive access increases significantly, changing the dynamics of archive from "cold storage" to true near-term access.

Key MPEG-2 flavors

Each of the three key MPEG flavors targets a specific application, optimizing picture quality for different bit rates (Figure 4). The SIF resolution Main Profile/Low Level offers the best picture quality for bit rates below about 5 Mb/s. This provides acceptable quality for interactive and multimedia applications. Main Profile/Main Level offers the best picture quality at rates from about 5 to 15 Mb/s. This provides excellent quality for play-to-air applications, where more than four generations is rare. Main Profile/Main Level also benefits from the support of virtually all available chip-sets, which is a key reason it is so interchangeable.

When finalized, the new 4:2:2 Profile will offer high quality for multiple-generation applications. However, until server manufacturers incorporate full IPB chip-sets, the bit rates required to achieve image quality comparable to Main Profile/Main Level are two to three times higher. This is because I or IB MPEG is so much less efficient than IPB.

High quality full-IPB chip-sets for 4:2:2 Profile are due in 1997. Hewlett Packard expects to be among the first to commercialize this technology, which will be an excellent fit for high-end multiple-generation applications.

Even when 4:2:2 Profile is widely available, Main Profile/Main Level will still be the best choice for mainstream play-to-air applications. This is because the subjective picture quality of Main Profile/Main Level is excellent for this application. At lower data rates, the extra chroma and lines make the 4:2:2 Profile up to 35% less efficient than Main Profile/ Main Level. This is a crucial distinction often lost in the cacophony of competitive claims. The eye is less sensitive to chroma than luma. Allocating precious bits to extra chroma is a poor trade-off at lower bit rates. For play-to-air applications, 4:2:0 usually provides *better* subjective picture quality than 4:2:2.

WHAT'S THE CATCH?

The benefits of full IPB MPEG-2 are compelling, but there must be a catch for anything that sounds so good. Actually there are three.

First, the complexity of real-time forward and backward prediction require vast computational power. As a result, implementing MPEG encoding is more expensive, but this is rapidly changing. Computational power gets cheaper every day. Volumes and competition are increasing. Hewlett Packard's newest server features 35% lower cost per channel than the previous model. Looking to the future, substantially reduced-cost IPB encoders are on the horizon.

Second, IPB MPEG is technically much more difficult to implement than traditional JPEG schemes. Hewlett Packard invested many engineering years to achieve the high reliability and quality level

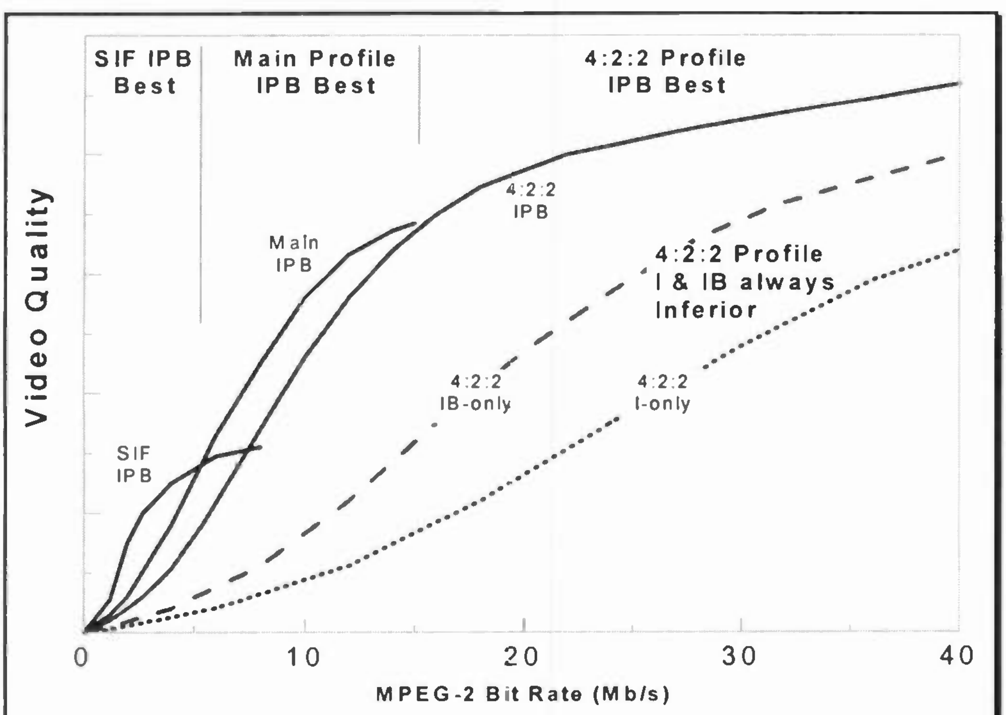

Figure 4. Different MPEG Flavors are Targeted at Different Applications

available today. Many server manufacturers are converting to MPEG, which is a great benefit to the broadcast industry, but these newcomers will take some time to refine their designs.

Last, IPB MPEG is more complex to edit. Most manufacturers are not willing to even try, but editing *is* possible.

Playing random clip sequence

To understand how IPB editing is possible, we need to be clear how non-linear editing happens today.

First consider the case of several clips stored on disk, and played at random (Figure 5).

This capability is useful for spot insertion, and is a basic form of non-linear access: the server must access the clips in a different sequence than the record sequence. Hewlett Packard's first-generation MPEG servers have this functionality, which conventional wisdom suggests should not be possible with IPB MPEG.

Creating effects

Next consider the creation of effects (Figure 6).

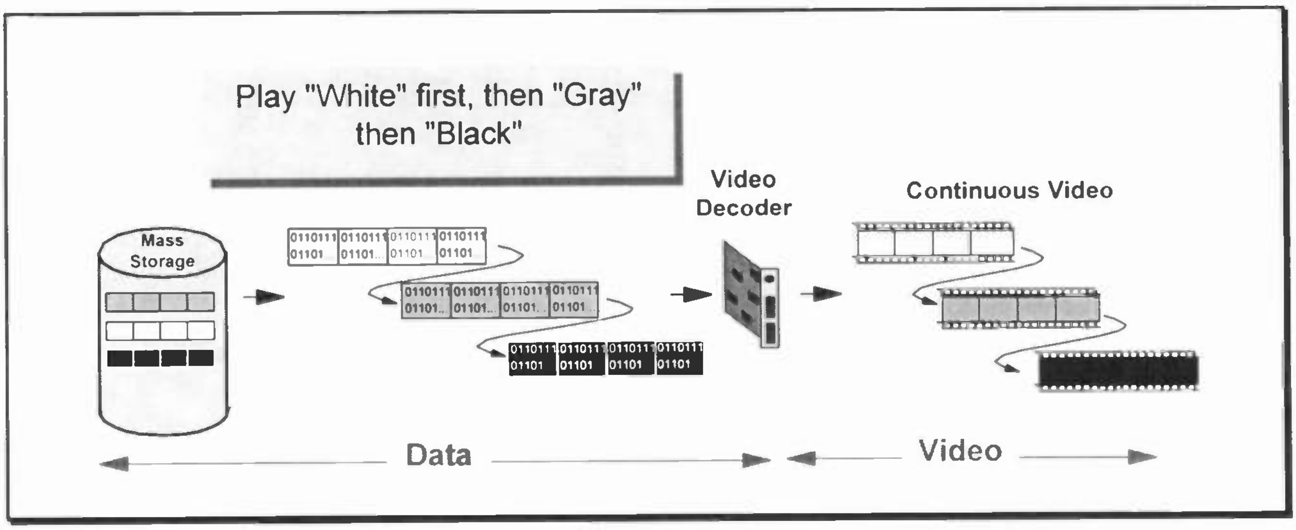

Figure 5. Playing Random Sequence of Clips

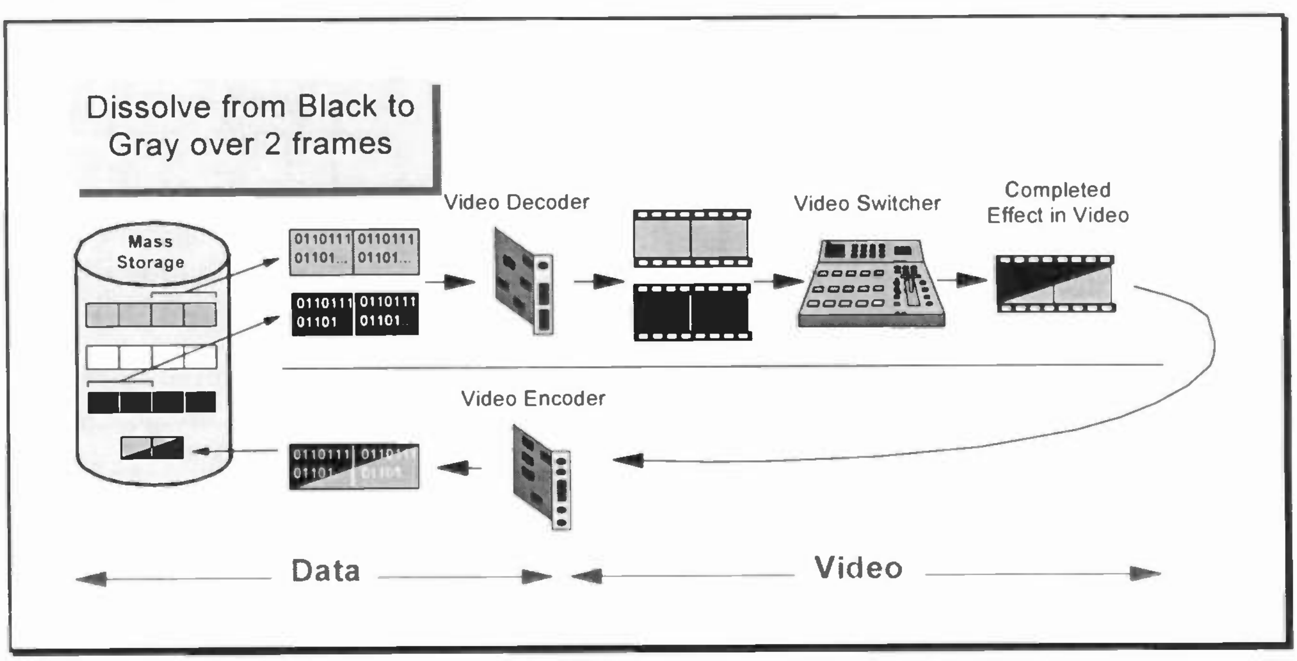

Figure 6. Creating effects. Note that the effect is almost always done in video.

Almost universally, the effect is created by decoding the stored data into video for two or more clips, manipulating the clips in video form to create the completed effect, and re-encoding the new clip to store on the disk. Note that hardware or software rendering produces the effect with video, not data.

Cuts editing

Last, consider the requirements for cuts editing (Figure 7). This is simply an extension of the previous two examples. To perform cuts editing, the server must rapidly supply random sequences of pictures (including any stored effects) to the decoder to provide an arbitrary but continuous video output.

The key requirement for a server to support this editing is access ("indexing") to individual pictures within a clip. Indexing is the requirement that is more difficult to meet in IPB MPEG than JPEG.

Indexing to random pictures.

The reason that indexing is more difficult with IPB MPEG is that usually the desired picture does not correspond with an I-picture. With a 15 picture GOP, the odds are only 1 in 15 that the desired picture is an I-picture! Consequently, depending on GOP structure, up to six other I and P pictures must be decoded to re-create the desired picture (Figure 8).

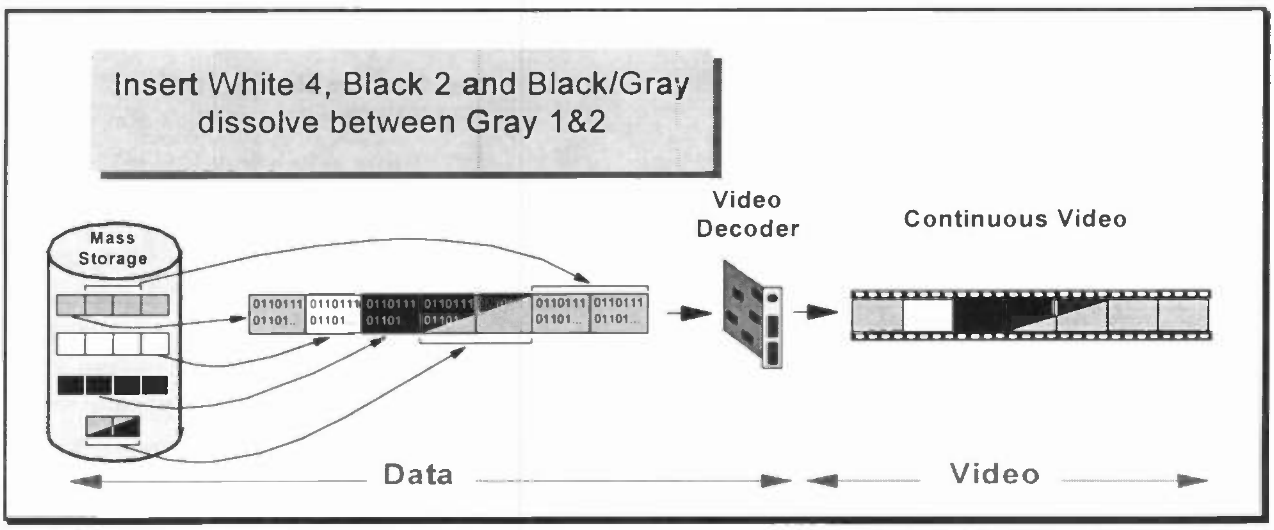

Figure 7. Cuts editing. This is a special case of playing random clip sequences.

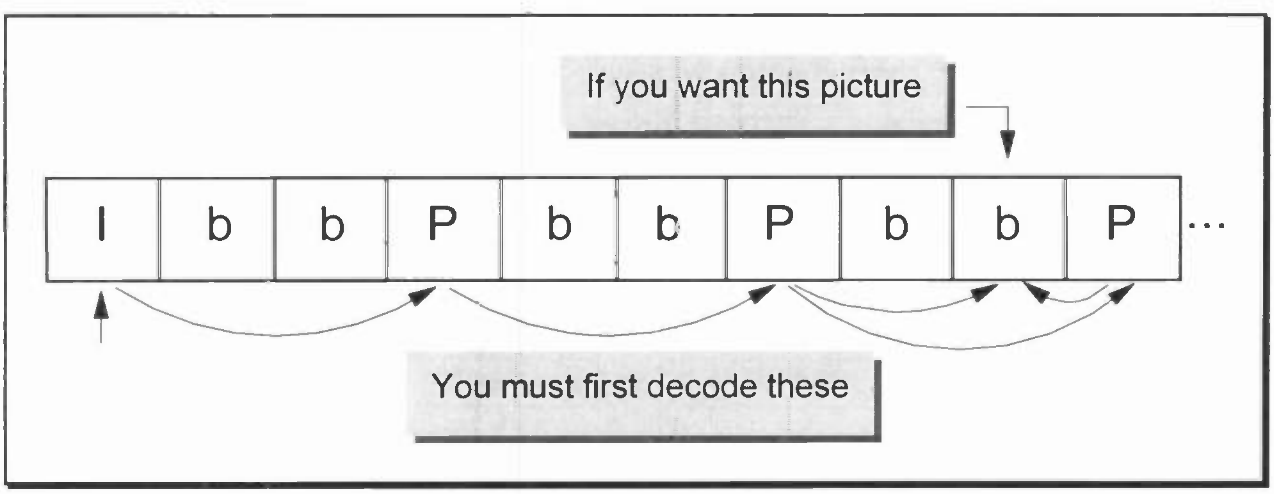

Figure 8. Indexing to Random MPEG Pictures is More Difficult

This requires that the server have substantially more processing power. The server must rapidly pre-charge the decoder with the required picture data to keep the output running smoothly.

Solutions?

The "orthodox" solution to this technical challenge is to stick with I or IB-only GOP's. This approach mimics JPEG, and largely eliminates any indexing problem. Unfortunately, this approach also eliminates the efficiency benefits gained by using MPEG!

Hewlett Packard's approach is to invest the engineering effort to create a server capable of rapid decoder pre-charge. This requires several clever software and hardware processing techniques, and approaches JPEG or IB-MPEG in responsiveness. Best of all, this approach retains the efficiency of full IPB MPEG-2 but provides editing features for broadcast applications.

UNCONVENTIONAL WISDOM

At Hewlett Packard we work to keep the so-called conventional wisdom from dampening our pursuit of creative solutions to clients' problems. As is true with many issues, the conventional wisdom is wrong about MPEG-2.

The myth is that higher bit rates are always better. The reality is that you really want higher quality pictures at lower, more efficient bit rates.

The myth is that 4:2:2 is always superior to 4:2:0. The reality is that for most play-to-air applications, 4:2:0 provides better subjective picture quality than 4:2:2.

The myth is that it is not practicable to edit MPEG-2. In reality, editing MPEG is quite possible, but it does take more engineering.

MPEG-2 does not solve all of a broadcaster's format or storage problems, but does provide many benefits. MPEG-2 is extremely efficient, providing broadcasters with cost-effective solutions today for on-air applications. MPEG-2 is a real standard; not something cooked-up in one company's lab and foisted on the world, but a compression format with unprecedented international activity, and tuned specifically for video applications. As a result, MPEG-2 is independent of any one manufacturer, and scaleable across applications. This gives broadcasters a compression format with a future. MPEG-2 is designed for transportation, providing broadcasters with a key building-block towards a future of shared digital media. MPEG-2 is a compression format for today, and for tomorrow.

FOR FURTHER READING

An excellent primer on compression is: “Image and Video Compression Standards, Algorithms and Architectures” by Vasudev Baskaran and Konstantinos Konstantinides, Hewlett Packard Laboratories, 1995. ISBN 0-7923-9591-3

FREQUENTLY ASKED QUESTIONS FOR STATION ATV PLANNING

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Washington, DC

ABSTRACT

Advanced Television represents the largest single change ever faced by Broadcast Television. As with any change of this magnitude, many of us struggle to even know what questions to ask. This paper provides questions and answers of most importance to broadcasters at this early stage before ATV implementation.

I - DEFINITIONS

What is "HDTV", exactly?

Here is the FCC's definition: "High Definition Television offers approximately twice the vertical and horizontal resolution of NTSC, which is a picture quality approaching 35 mm film and has a sound quality approaching that of a compact disc."

What is "ATV"?

"Advanced Television refers to any television technology that provides improved audio and video quality or enhances the current NTSC television system." As first used, ATV meant an HDTV program compressed to fit within a current NTSC broadcast channel. Recently, the definition has been broadened to include the concept of multiple "SDTV" video programs and other data simultaneously carried within one channel.

What is "SDTV"?

"Standard definition television (SDTV) is a digital television system in which picture quality is approximately equivalent to the current NTSC television system." The quality will be better than NTSC, with no "chroma crawl" or noise. Depending on the number of SDTV programs in the ATV channel and their degree of motion, there may be visible artifacts that NTSC does not have.

What is the "Broadcaster's Caucus"?

This is a group made up of ABC, CBS, FOX, NBC, PBS, NAB and MSTV with the goal to protect the broadcaster's interest as rule making is developed for ATV. It has been working to develop a way to assign channels for ATV that will treat existing broadcasters fairly and prevent interference to NTSC stations during the transition period to ATV.

What is "MSTV"?

The Association for Maximum Service Television is an organization with a history of involvement in issues of channel allocation and broadcast spectrum protection.

What is "COFDM"?

COFDM is a modulation method for digital signals that is favored in Europe. It has some presumed but undemonstrated advantages for on-channel repeaters and immunity to multi-path. However, it has some performance and implementation disadvantages over 8VSB modulation. The US Advisory Committee decided not to recommend it for further testing.

II - YOUR FCC ATV LICENSE

What is the earliest date that the FCC will accept ATV license applications?

First, the FCC needs to issue rule making establishing the transmission standard, channel allotments and assignments. There is a possibility this will be complete by mid 1996.

Can we use ATV for multiple programs?
The recent FCC Notice of Proposed Rule Making provides a clear indication that ATV can be used for multiple "standard definition" programs. However, the FCC indicates that they will still exercise control over all services to insure that they are in the public interest. They have requested comment regarding how much actual HDTV programming should be required.

Will we be required to simulcast the same program on our NTSC and ATV channel?
The FCC is reconsidering what amount of simulcasting they will require.

I've heard various comments that there will be a "use it or lose it" clause in the regulation. Would the FCC really force an NTSC station off the air if it means a loss of free TV service?
There are strong forces acting to release spectrum for non-broadcast uses which might force some loss of service in the interest of other gains. This will be a very political issue.

What deadlines will the FCC mandate for conversion?
The FCC wants to move quickly, perhaps completing the transition within eight years of final rule making! They think the pressure from cable, DBS and telephone entry will cause commercial stations to transition soon. They have suggested that stations be required to show evidence of progress within three years, to keep their ATV allocation.

Will the FCC allow us to initially build a low-power ATV facility, and if so, would our full-power allocation be protected?
The FCC has acknowledged the need to address this issue. Perhaps the example of the low NTSC minimum power requirement will be followed.

If we can't afford an ATV transmitter during the transition period, will the FCC let us instead use our existing UHF channel, antenna, line, and transmitter for ATV operation when NTSC is shut down?
The FCC recognizes this question, and has requested comment on what special assistance should be provided to rural and public broadcasters in order to facilitate the transition.

III - Channel Allocation

Years ago, the FCC issued a list of channels allocated to each market, but no indication which of those channels we would get. How would this work?
This preliminary planning model did not have the benefit of test data and was not intended to be final. It allocated enough UHF ATV channels to accommodate existing broadcasters but did not consider many interference factors.

Later, the Broadcaster's Caucus issued a much different and more specific proposal. How did they assign us an ATV channel and power?
The Caucus proposal attempted to replicate each station's current coverage and minimize interference between new and existing stations. To accomplish this, each existing broadcaster was assigned a specific ATV channel and power level, assumed the identical location, radiation pattern and height as the existing facility. Most assignments were UHF. Some VHF channels had to be used in congested markets.

What does the Caucus mean by "replication"?
The Caucus first studied how your Grade B contour is limited by terrain using a sophisticated computer model. Then, they calculated how much your coverage is further reduced by co-channel, adjacent-channel and taboo-channel interference. (This may further significantly reduce the area served.) Finally, they calculated what ATV power will be required to produce a service contour with an equal area.

Our NTSC coverage is currently limited by the cost to build and operate a larger facility. Would we end up getting locked into this limited service if we are only allowed to replicate our current coverage?
Since publishing the table, the Caucus has successfully tested the feasibility of improving the coverage of many small UHF NTSC broadcasters without upsetting the model. This issue will be determined by the nature of the FCC rule making.

Why did the Caucus assign us a high UHF channel, when we are in a rural area with no UHF stations?
Their first model chose the channel for each location with the least possible interference, which resulted in many high channel assignments. It may be possible in many cases to utilize a lower channel.

How will the FCC determine our ATV channel?
This is the most complex issue in the conversion to ATV. The FCC has not responded to the Caucus proposal, but it appears that they are recognizing the need to consider replication as a guiding factor. Recently, they have even asked whether ATV should be an all-UHF service. Although low-band VHF is not suitable for ATV due to impulse noise, hi-band VHF would work well and could accommodate more than the existing number of channels due to ATV's superior interference characteristics. The drive to reclaim spectrum for non-broadcast uses will be a strong force in the channel allocation and assignment process.

IV- Reception, Translators

In what ways will ATV deliver a superior picture?
The most obvious quality improvement provided by ATV is that it can deliver HDTV programming with double the resolution of NTSC and a wider aspect ratio. But there are other advantages that are realized whether ATV is used for one HDTV or multiple SDTV programs. First, ATV eliminates the interference between brightness and color (chroma crawl) that is inherent in NTSC. Second, ATV eliminates noise in the picture that most viewers experience to various degrees due to poor off-air reception or noisy cable systems. Finally, ATV eliminates minor ghosting. With ATV, the quality of the picture will be equally superior for all viewers.

What problems might viewers experience with ATV reception?
Some viewers may be annoyed by the longer time needed for the receiver to display a new picture when changing channels. People who view poor quality NTSC reception due to low signal strength, severe multi-path or noise may not be able to receive ATV at all. It may be possible for some viewers to overcome this barrier by improving their antenna system. This a complex phenomena with many varying situations.

How will existing cable systems handle ATV?
Results of the field tests suggest that cable systems can carry the ATV signal without difficulty. The laboratory test subjected the ATV signal to numerous mismatch impairments, which the receiver compensated for.

What will happen to existing translators when ATV stations go on the air?
The FCC classifies TV translators and low-power TV (LPTV) stations as "secondary service". If this policy is continued, a new ATV station can require turning off an existing translator if they think it is interfering with their new service.

What will the rules be for ATV translators?
The FCC has not addressed this issue. ATV does allow channels to be spaced more closely, which is a great advantage for translators. It may turn out that ATV translators can be accommodated, at least after NTSC is eliminated.

V - ERP Requirements

(Based on the Broadcaster's Caucus Model)

FUTURE VHF ATV STATIONS:

What ERP will we need if we are given a VHF ATV channel?
Theoretically, ATV operation in the high-band VHF channels should require far less power than VHF NTSC operation. However, low- band VHF is prone to impulse noise and would not have as much advantage. More study is needed if the FCC allocates VHF ATV channels.

FUTURE UHF ATV STATIONS:

Please show some system examples and associated coverage for flat terrain, based on 1000 feet HAAT and channel 37.

ERP [KW]	Transmitter Power [KW]	Approx. Coverage [miles]	Antenna Gain	Rigid Line [inches]
10	2.5 SS	39	15	1-5/8
20	2.5 SS	42	15	3-1/8
50	2.5 SS	45	30	6-1/8
100	15 IOT	48	15	3-1/8
200	15 IOT	50	30	3-1/8
500	30 IOT	54	30	6-1/8

CURRENT UHF STATIONS:

How will our ATV UHF ERP compare to our current ERP, for similar coverage?

If you are one of the current UHF stations, you can use a much **lower** ATV ERP for equal coverage area. (Many of the existing stations can use less than 5% of their current ERP to replicate coverage.) Hopefully, you will be allowed to use an ERP closer to your current level and thus significantly improve your coverage.

CURRENT VHF STATIONS:

How will our ATV UHF ERP compare to our current ERP, for similar coverage.

If you are one of the current VHF stations and will convert to UHF, you will need a much **higher** ERP for an equal coverage area. If you now have a low-band VHF channel, you are probably one of the stations that would need an ERP of more than 2MW! Half of current VHF stations would need an ERP of roughly 1.2 MW. The rest would need less than 1 MW.

Will a large ATV ERP require an expensive facility like large NTSC UHF systems?

The cost of an ATV facility is higher than you might think since an ATV transmitter needs to be sized for peaks of at least 4 times the average rating. For example, a 5 megawatt ATV ERP could require a transmitter with 200 KW average rating, or a peak rating of 880 KW. This would require a transmitter with 15 parallel IOT's!

Will anyone pay to buy and operate such a huge transmitter?

We think the largest ATV transmitter broadcasters will purchase is a four-tube IOT, having an average rating of about 60 KW (240 KW peak). This would produce about 1,200 KW ERP on channel 37 with 6" line, an antenna gain of 30 and a tower of 1,000 ft. Many will choose a two tube 30 KW transmitter, which offers the redundancy of a parallel design.

VI - COVERAGE PREDICTION

Will we still have "City Grade", "Grade A" and "Grade B" coverage circles?

Since ATV reception is either perfect, gone, or intermittent, there is no value in showing various "grades" of signal quality, so the FCC will use just one coverage contour. However, there is still value in showing areas in which the signal level is higher and simpler receive antenna systems can be used.

I'm new to UHF broadcasting. Why are the high UHF channels less desirable?

Aside from the higher loss in the transmission line, there are reception issues. A good receive antenna has uniform gain over a dipole throughout the UHF band. However, because antenna elements are shorter for higher frequencies, they "capture" 5 dB less of the available field at channel 69 as they do at channel 14. In addition, downlead loss is greater, there is more attenuation due to foliage, and less diffraction around objects and terrain. The FCC takes none of these factors into account. Also, the signal is more readily reflected at higher frequencies– resulting in worse multi-path.

How does changing the tower height affect UHF ATV coverage?

As with NTSC, ATV UHF propagation is primarily line-of-site, so go for all the height you can get. For example, at 50 KW ERP, every extra 200 feet extends coverage almost as much as doubling the power!

HAAT [feet]	Distance [Miles]
200	32
400	37
600	39
800	42
1,000	45
1,200	47

How does changing the ERP affect UHF ATV coverage?

The good news is that with a HAAT of 1,000 feet, cutting your UHF ATV power in half only reduces your coverage by 3 miles. But that also infers that you must double your power for each three miles of coverage you want to gain!

ERP [KW]	Distance [Miles]
10	39
20	42
50	45
100	48
200	50
500	54
1,000	58
2,000	62
5,000	67

If there are no people to serve at the edge of our coverage, should we save money and use less power?

As with NTSC, there will be people well-within your coverage area with marginal antenna systems who will suffer if you lower your power. A 2 dB drop may take them from perfect pictures to regular total loss of programming.

What receiver performance did the Broadcaster's Caucus presume, and can receivers be made better than that?

The Caucus model planned for 43 dBu field strength, assuming receivers with a 10 dB noise figure. (NTSC rules specify a noise figure of 12 dB.) It would be fairly easy to achieve a noise figure of 7 dB with today's technology. It will help broadcasters a great deal if the FCC mandates minimum receiver performance, especially with regard to receiver sensitivity and selectivity.

What parameters do we plug into a Longley-Rice terrain-sensitive model to predict ATV noise-limited coverage?

More field experience is needed to answer this question.

VII - TRANSMITTER

We understand that ATV ERP is specified as the average power, whereas NTSC ERP is specified as the peak power during sync. How would a current NTSC transmitter be rated for ATV?

An ATV transmitter needs to handle peaks of roughly four times it's average rating. For example, a transmitter rated for 60 KW NTSC will be rated for about 15 KW ATV.

What kind of transmitter should we buy for NTSC, if we want it to be compatible with ATV?

Either IOT(klystrode) or solid-state transmitters are good choices for future ATV operation.

Can a current transmitter be converted for ATV operation?

At a minimum, a new exciter will be needed. Compared to NTSC performance, the transmitter will need to provide a full 6 MHz bandwidth and will probably need better out-of-band rejection. Additional filters and other modifications may be required. For a Klystron or Tetrode transmitter, poor linearity may cause the power rating, efficiency and error performance to be poorer than you would expect. More data is needed in this area.

What will be the approximate ATV transmitter cost, by average power rating?

1.2 KW Solid State	$300K
2.5 KW Solid State	$400K
15 KW IOT (One Tube)	$400K
30 KW IOT (Two 60 KW Tubes)	$560K

What will be the tube life and cost?

A 60KW IOT costs about $36,000. The latest IOT tubes last about 20,000 hours. Compared to klystrons, IOT life is degraded more by power-line disturbances and overloads.

What power efficiency will an IOT ATV transmitter have?

Manufacturers suggest that the minimum transmitter efficiency will be 25%, and are working on designs that might reach 40%.

I've heard that a high-powered solid-state UHF transmitter is being developed that would use silicon carbide transistors. How much power will be practical, and how soon will such a transmitter be available?

This technology is some years away, and information is still pretty sketchy.

VIII - Tower, Line, Antenna

How should we begin to plan for the addition of ATV equipment on our tower?

You should conduct a tower study to determine the cost of modifications to allow addition of an ATV antenna and line. Ideally, have the study performed by the tower manufacturer. For a 1,000 foot tower, the minimum cost is about $5,000. If accurate, up-to-date prints are not available or the study is complex, the cost can double or triple. Be prepared, most towers were designed under requirements that were less stringent than the current EIA standard. Even your current facility will likely now be considered "overloaded". If you lease space on your tower to others, you may want to plan for limits on renewals in order to reduce tower loading.

What kind of transmission line is suitable?

It appears that the current guidelines will continue to apply: Waveguide is more reliable than coax, has lower loss (especially at high channels), and can handle higher power levels. However, it has higher wind-load and may not work for multiplexing widely spaced channels. Rigid line has lower wind load, but more signal loss and has frequency limits: 6-1/8" line can be used at all channels, 8-3/16" line can be used through channel 56, and 9-3/16" line can be used through channel 40. Semi-rigid line has more loss than rigid of the same diameter, but is less expensive to install and maintain. Andrew Corporation suggests that their line's rated power can be used as the ATV average power rating.

Is there any alternative to running a second line up the tower?

There are custom "band splitters" available that reliably combine VHF and UHF transmitter outputs into one line, and split the signals out again at the top. This may be a good solution for towers with existing VHF or FM antennas.

What transmit antennas will be appropriate?

Most of guidelines for NTSC will continue to apply. Omni-directional UHF antennas with a gain of up to 30 will be used. The best solution for omni-directional coverage is a top mounted antenna. If you must be side mounted, a wrap-around panel antenna is the best solution. Directional antennas will continue to be appropriate, and many solutions exist to accomplish this. The effect of antenna radiation pattern on coverage may be of more concern with ATV and needs further study.

What are antenna bandwidth, group delay, gain uniformity and VSWR requirements?

Because the ATV signal uses the entire 6 MHz bandwidth, performance requirements will be more stringent than current NTSC antennas.

What are the possibilities of combining several ATV channels into one antenna and transmission line?

The feasibility of shared facilities depends on many factors. European experience has proven that this is often a good solution, but cooperative planning and solid operating agreements are required.

Is it possible to install a single transmit antenna for both FM and UHF?

This "shared-aperture" concept is possible, but requires a custom design.

IX - MICROWAVE

Will there be new microwave channels set aside for ATV?

There is little possibility of additional spectrum for broadcast microwave. In fact, there is significant pressure on the FCC to release portions of the 2 GHz band for non-broadcast uses in large markets.

What is the cheapest way to convert our STL to digital?

The cheapest solution requires you to totally simulcast. We expect that a manufacturer will produce equipment specifically for the 20 Mbps ATV data stream. Using 16 QAM modulation, it would fit within either 2 or 7 GHz channels. At your transmitter site, you would feed the data directly into your ATV transmitter and down-convert one of the ATV programs to feed your simulcast NTSC transmitter.

What if we want to feed our ATV and NTSC transmitters separately but within the same microwave channel?

A 7 GHz channel is large enough for a typical 45 Mbps (DS3) digital radio. You can multiplex an ATV 20 Mbps signal and a slightly compressed 20 Mbps NTSC program within this data stream. You will need to re-license the channel for this use. Manufacturers may need to have their equipment type accepted in this band.

Can we convert our existing analog microwave radio to digital?

Digital transmission has performance requirements that are very different from analog. The local oscillators must have higher phase stability. Each hop must have an adaptive group delay equalizer and must regenerate the digital signal. By all indications, it would be difficult to convert existing analog radio to digital.

Are our existing microwave path designs going to be adequate for digital?

If you have paths that regularly fade into the noise floor, you will want to upgrade them using space diversity or other techniques. Otherwise, your viewers will experience the abrupt loss of service at threshold that is characteristic of digital transmission.

X - REFERENCE

More information can be found in "Advanced Television Transmission - Planning Your Station's Transition", by Tom Vaughn and Associates, published by PBS and NAB.

A UNIQUE SOLUTION TO THE DESIGN OF AN ATV TRANSMITTER

Nat S. Ostroff
Comark
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1. INTRODUCTION

The requirements for transmitting the Grand Alliance ATV signal are significantly different from that of today's NTSC standard. This difference creates the opportunity to design a transmitter that is uniquely matched to the new standard.

The application of an ATV transmitter may also be different from today's systems. The ATV transmitter will most likely be installed in an existing site with the intent of minimizing costs. Even in a new site, floor space and installation customization will be desired to be kept at a minimum to keep cost down.

Product cost considerations will be more significant with the ATV transmitter than normal since the revenue stream that it will produce for the station may be years away from the initial installation of the system. Therefore, keeping product cost and support cost in mind, it is also desirable to design a system that is self monitoring with a strong emphasis on user ease of operation. This implies the need for advanced remote control and automated fault analysis.

This paper will explore the options that are available with today's technology to meet the objectives discussed above. It will also look to the future and discuss the possible alternatives that may emerge for transmitter design.

Finally, this paper will describe a system that is being created at COMARK based on the realities discussed.

2. SIGNAL DIFFERENCES: NTSC TO ATV

Today's UHF NTSC transmitter is required to deliver high peak and average power. The power levels in the market today are generally between 60kW and 240kW peak. The average power is about 60% of the peak. The average power varies with the picture content which places a varying load on both the AC power line and the internal components of the transmitter.

This power hungry NTSC system typically demands large water cooled tubes with the attendant support systems and large RF transmission line structures. The floor space that these systems occupy can be significant and expensive. Some modern transmitters that use IOT tubes are capable of providing good ATV service even if they have not been optimized for the parameters discussed earlier. Older transmitters using klystrons may not be reasonable solutions for ATV even on a short term basis. This is true based on their cost of operation and the lack of a revenue stream from ATV transmissions.

The ATV signal is significantly different from that of NTSC. Specifically, the ATV signal operates at a constant average power at a level that is about 12 dB below the peak power of NTSC, for equivalent coverage. Recent industry analysis, taking into account fade and noise margins, estimates the ATV power requirements to be 12kW average in many station situations. This is well below the 80kW average power levels found in a 120kW peak NTSC transmitter. The peak power of this 12kW average ATV signal would then be about 50kW. (It should be noted that this peak power occurs less than 1% of the time. The NTSC peaks occur about 10% of the time.)

The differences in the two systems, when addressed, present the designer with some very interesting alternatives.

3. TECHNOLOGY CHOICES FOR THE POWER AMPLIFIER

A. Solid State?

The lower average power and duty cycle for ATV suggest that a solid state solution may be possible. The constant signal power of the ATV signal eliminates one of the biggest problems in NTSC applications of Class AB solid state amplifiers. The constant level changes in NTSC vary the temperature of the Class AB amplifier's junctions and thus its bias point. This, in turn, varies all of the transistors parameters including gain and linearity. Sophisticated adaptive bias circuit designs have been applied to this problem in NTSC and have managed to provide some degree of relief. Class A amplifiers don't suffer from this problem and have better linearity capabilities. Unfortunately, Class A requires more amplifiers for the same power level as Class AB and it is thus more expensive to implement. Nevertheless, some manufacturers have chosen to use Class A for it's simplicity and linearity.

In ATV it should be possible to use Class AB amplifiers since there are no signal variations in average power and new advanced linearity correction systems are about to be made available.

Today's Class AB amplifying transistors are capable of up to 150 watts peak in UHF. Could combining these devices in an array be a solution for a 50kW peak transmitter? The answer lies in the economics, not in the technology.

The typical UHF transistor has a gain of about 8dB. This means that there is a requirement for significant drive power as well as output power. It is also necessary to remember that, while the average power of the ATV signal is lower, the peak power must be reasonably maintained. Thus, any transmitter must be able to support the peak signal demands in a reasonably linear fashion. This means that a solid state transmitter, rated for 12kW of average power, must contain enough transistor junctions to meet a 50kW peak demand.

The number of transistors required in a 50kW peak/12kW average ATV transmitter could climb to as much as 400 devices. This estimate is based on the 150 watt and 8dB rating of today's available devices. It also takes into account practical combining losses. 400 devices and their associated support circuits and hardware will require a large and costly cabinet footprint. In addition the cost of the transistors alone could reach the magnitude of $100,000. This is not conducive to the manufacture of a low cost ATV transmitter.

Is there a new technology in solid state that might change these economics? Recent industry announcements have suggested that Silicon Carbide (SiC) devices would be available for ATV applications. These devices may have the capability to provide four (4)times (or more) the output power of the conventional silicon transistors today. This would certainly change the economics if they were reasonably priced. A price that is 4 times the cost of today's devices would not change the picture very much. Unfortunately the SiC devices are still in the laboratory or available only in small quantity hand made prototype amplifiers. Commercial production is not here today. The demand for ATV transmitters is just around the corner (hopefully) but SiC is not available for such systems.

The decision to chose a technology is an economic as well as technical one, and today's solid state devices are not ready for the task of providing a low cost ATV transmitter. Perhaps in the future SiC or another device technology will make solid state a reasonable solution for a 50kW peak transmitter for ATV.

Certainly any solid state transmitter designed with today's available technology will be too expensive and obsolete when compared to the possibilities in the near future. It is therefore reasonable to pursue a vacuum tube solution with the idea that such a system should provide a low cost reliable answer to the needs for the early implementation of ATV.

B. Vacuum Tube Technologies?

There are several Vacuum tube technologies that are contenders for ATV service. These are the tetrode, Diacrode and IOT. The klystron is not considered here since it is not a low cost device that can benefit from the low average power of the ATV signal. The Klystron is essentially a Class A device, if it is not pulsed.

The criteria that should be used to evaluate these tube technologies are as follows;

1. Power capabilities
2. Linearity
3. Proven reliability record in the field
4. Support requirements
5. Cost
6. Availability

I. Tetrode?

The tetrode is a conventional vacuum tube that places two grids into the electron beam that is supplied by the cathode. In the U.S. this insertion of grids into the electron stream has been taken as having a negative impact based on the more familiar and reliable structure of a Klystron. Secondly, the output circuit of the tetrode appears electrically in series with the input circuit and the load. The parasitic reactance of the tube elements is a part of the input and output tuned circuits. This has the effect of making any tube element parameter change to cause a change in the critical output tuning response. The result is the need to periodically retune the output circuit (cavity) as the tube ages, thus creating an additional maintenance requirement. The series nature of the tetrode places stringent limitations on internal element spacings and sizes in order to reduce transit time through the tube's vacuum space. Thus, tube elements must be minimized and be placed very close together. This requirement has the effect, at high power, of reducing the robustness of the device and makes its protection a critical matter in any design.

The tetrode relies on a screen grid to accelerate the electrons to the tube's plate circuit. This grid must be very thin to meet the internal capacitance limitations of the structure and thus cannot dissipate a lot of power. Typically the screen is rated for a tiny fraction of the total tube's output even though it is in the main stream of the output electron flow. The voltage placed on the screen is adjusted to be less positive than that on the plate. The plate element is behind the screen and thus most of the electrons will flow through the screen to the plate. This minimizes the screen dissipation. Under ideal conditions the tetrode will provide good service as long as none of the internal elements are over dissipated.

In actual field service the tetrode is connected to an antenna and transmission line. The transmission line may be 1000 feet long or longer and the antenna will not be a perfect match. Under these practical conditions the load impedance presented to the tetrode can be varying over the broadcast day by as much as 15%. Such a load change will not only effect the tube's output response, but it will also change the RF voltage swing on the tube's plate circuit and may cause the plate to momentarily swing less positive than the screen. In low power applications this phenomenon will cause large screen currents to flow but will not destroy the screen. In high power applications (greater than 20kW), the effect, if not protected against, can destroy the tube. The only true solution is to derate the tetrode to allow for this VSWR variation. Present manufacturers data sheets do not reflect this effect based on the assumption of a perfect resistive load being placed on the tube. In the field this is clearly not the case.

Even with the above limitations, the tetrode has had a wide level of acceptance as a TV amplifier at power levels from low power to 25-30kW peak. The highest rated tetrode (TH563) is capable of 50kW in visual service and 25kW in combined aural/visual service at 10dB aural/visual ratios. The TH563 is used in some NTSC systems at 30kW combined at a 13 dB aural/visual ratio. While the number of systems is small compared to other tube technologies (less than a few percent of the number of IOT sockets) the TH563 has provided average lifetimes that approach 10,000 to 15,000 hours.

The linearity of the Tetrode is excellent and is a strong point for ATV consideration. While not as good as other technologies, the tetrode has been used at 25kW in combined NTSC service and provides a manufacturers published specification of -52 to -54dB third order intermodulation performance at the overall system level. Perhaps some manufacturers consider this to be good enough for correction to ATV requirements, but it is not as good as IOT transmitter system levels that consistently demonstrate -60dB or better in NTSC service.

The tetrode is provided at the power levels required for ATV as a liquid cooled device. Air cooling at these power levels is not possible since the plate of the tube is part of the output RF circuit and size must be minimized. In addition, the tetrode has the full plate voltage (8kV) on the plate that must be liquid cooled. In order to facilitate tube cooling, the tetrode is supplied with a flow of deionized, distilled water that must maintain a resistivity of more than 50,000 ohms per square cm., since it is in contact with the high voltage. This requires a filter and purification system. No glycol can be used in this cooling loop. If freezing temperatures are likely to be encountered during the year, a second water/glycol loop and another liquid to liquid heat exchanger is required in order to be able to vent the heat to the outside world.

This cooling problem is a major deterrent to choosing the tetrode as an ATV device for an ATV transmitter since maintenance and reliability are major considerations.

These problems with the tetrode, along with the fact that it is a low gain device providing only 15dB of gain, does not make it a prime candidate for ATV service at the required 50kW peak power level. The low gain places additional requirements on the driver. The driver in the ATV transmitter must be solid state to minimize maintenance, and the low gain of the tetrode can make the drive power expensive to produce.

The TH563 is available only from Thomson Tubes Electronic in France and thus there is no second source.

II. Diacrode?

The Diacrode is a clever modification to the basic tetrode. By placing an electrical extension onto the output circuit structure of the cavity it became possible to double the tube's internal elements and thus its power output. Thus, the Diacrode is rated for upwards of 60kW in combined service. This is slightly more than twice the demonstrated capability of the TH563 which is the design base for the Diacrode. Since its introduction, the Diacrode has been used in several transmitters. There have been some early problems with the tube/cavity/system interface. Thus, the reliability data for this device is still quite thin and its early problems might be misleading with respect to its long term performance possibilities.

The Diacrode is still basically a tetrode. It suffers from all of the limitations discussed for the tetrode. The fundamental reasons for not choosing the Diacrode as an ATV device is that it has low gain (15dB to 17dB) and that it must be liquid cooled with deionized, distilled water. Finally, the Diacrode, with its initial factory and field operational service, has not provided strong evidence of being a robust device. A number of tubes were lost in both the factory and in the field before any consistent service was obtained.

III. IOT?

The current field installed versions of the IOT are rated for 60kW visual only service and 40kW combined service at a 10dB aural/visual ratio. A new version of the tube may be introduced shortly that is rated at 60kW combined at 13 dB and 55kW at 10dB, with ongoing discussions of even higher ratings. A compact tube is also under development and it is based upon a unit made available that is rated for 20kW in combined 10dB aural/visual NTSC service. This tube is very interesting since testing has confirmed that it will support 50kW peak and 12kW average powers when amplifying ATV signal formats. Most importantly, it is AIR COOLED.

The linearity of the IOT is well proven in NTSC service where it consistently provides -60dB or better intermodulation performance in combined 10dB aural/visual service. This is fully 6dB better than any other current technology at the system level. The IOT, in a COMARK transmitter, is the device that has been used for over two years in Charlotte NC to prove the Grand Alliance ATV system.

The IOT was introduced to the industry in 1986 by COMARK and Varian as the Klystrode®. COMARK and Varian received an EMMY award for their work on inventing the Klystrode® and its transmitter in 1991. Early success with the tube was later marred by production problems at Varian that slowed its industry acceptance. The first Klystrode® transmitters were put into full time broadcast service at Georgia Public Television in 1989 and are still providing reliable operation today. In 1991 EEV introduced an improved IOT which rapidly gained industry wide acceptance. Early problems with the EEV IOT were traced to gun stability and input cavity dielectric strength issues. By 1994 these problems were largely solved. Today (2/96) there are over 200 tubes in service and 100 under contract in over 20 countries. Tubes manufactured after mid 1993 show a Weibull life analysis of 37,500 hours. This means that by 37,500 hours 2/3 of the tubes shall have died. From the same data, the hours per failure were 28,500 hours. By December 1995 there were at least 61 tubes (30.5%) still operating with more than 15,000 hours and more than 22 tubes (11%) still operating with greater than 25,000 hours. To date not a single tube has exhibited a failure from "old age".

From the above data it is clear that the IOT has fulfilled the initial promise of "klystron like life". Thus, the reliability of the IOT is certainly proven in NTSC actual field service. Since we have no way of measuring statistically the ATV life we can only speculate that it will be at least as good as that of its NTSC service.

In fact, there are good reasons to believe that ATV service will be much easier on tubes and systems than NTSC. These reasons include the constant average power of ATV that will eliminate the temperature cycling that NTSC picture changes induce. Another reason is that the peak power duty cycle demands of ATV are less than 1/10 the demands of NTSC. These factors certainly will extend the lifetime of the IOT. The cathode capacity of the IOT is virtually the same as a klystron. Thus we might hope for close to 50,000 hours or more from an IOT in ATV service. 50,000 hours represents almost 6 years of life at 24 hours a day. ATV may not be transmitted daily for 24 hours in the early years so that the initial installed IOT could very well last till a viable solid state solution emerges in the next century.

A new IOT available from EEV is AIR COOLED and this eliminates a vast set of problems that would be part of any tetrode or Diacrode solution. The higher gain (21dB to 24dB) of the IOT makes the driver comparatively much less expensive. The other support systems for an IOT are well understood and have been proven many times in the field.

The one concern that might be raised is the high voltage required for the IOT. COMARK and other major manufacturers have been supporting IOT's and klystrons for many years at their required beam voltage levels without significant problems. Even tetrode voltages at 8kV can be a problem if the designer does not understand the requirements of the task.

IOT's are available from at least two sources with two more sources tooling up to bring a product to the market. This will insure consistent product supply and price stability into the ATV future.

IV. Technology Choice For The Power Amplifier!

The clear choice for the first generation of dedicated ATV transmitters is the IOT in an air-cooled configuration.

4. CONTROL SYSTEM

Conventional wisdom in transmitter design and packaging dictates that all of the transmitter functions be tied together by hardwire harnesses and multipin connectors. Once that is accomplished, a central control logic system (today it's solid state, possibly computer based) is used to direct the sequence of events and protect the transmitter.

The wire harnesses and their associated connectors are one of the major sources of field problems. They are also a major cost center in the manufacturing process. Therefore, if the wire harnesses and connectors could be eliminated the ATV transmitter would be less costly to build and more reliable in the field. It would be a bonus if the new system would be able to provide diagnostics and "work arounds" automatically.

COMARK's new ATV transmitter will NOT use wire harnesses and will provide automatic fault detection and operational fall-back conditions.

DISC™

Commercial building control systems and the telephone network have something in common. Both are adopting a new technology to control their operating functions. They are fitting control nodes that contain computing capability right at the point that is controlled. These control nodes carry discrete digital addresses that are controlled from a central processor. The individual nodes are programmed to fit the specific function that they are to perform and can adjust their parameters to the local conditions. They can also be commanded to change their role as a result of instructions received from the central processor. Each node contains internal self calibration instructions as well as monitoring and reporting capabilities.

COMARK calls it's adaptation of this technology "Distributed Intelligent System Control" or DISC™ for short.

DISC™ uses a pair of central computers at the system level for redundancy, to monitor and instruct the intelligent nodes placed throughout the transmitter {Figure (A)}. The nodes have discrete addresses and are communicated to by a daisy chained pair of fiber optic cables. Thus, there are no longer conventional wire harnesses and a minimum of multipin connectors remain. All parameters are calibrated automatically at their source before being transmitted to the central processor. The use of fiber eliminates any RFI and common mode problems in the control system. It also allows direct control of high voltage functions without the need for isolation transformers or the like.

For example, the bias and filament voltages of the IOT can be monitored and adjusted by simply sending the correct command to the intelligent node that is actually floating at the high voltage level. The central computer and the metering circuits don't have any high voltage involvement.

Other advantages of this DISC™ system are listed below;

1. No complex and unreliable inter-wiring harnesses and connectors.
2. No calibration potentiometers for meters or comparators. Measurements are designed to be accurate since they are developed inside of the intelligent node and are absolute and presented in digital code to the central controller.
3. No common mode signal problems. Each measurement is made at the voltage it is measuring.
4. Absolute records maintained on all measurements and adjustments.
5. Automatic calibrations on command from the central controller.
6. Adjustments to all operating parameters are made as digital commands to the intelligent nodes.

One singular advantage of the DISC™ system is that controlling all the transmitter's functions is accurate because it is not an analog system. The internal intelligence of the control points and their discrete address allows the system to be controlled and adjusted without needing physical access to the actual controlled point. This feature not only allows for a much more precise control platform, but it creates the ability to change operation of the transmitter and all or any of it's internal parameters by "log-on" verified remote control.

With the addition of a sophisticated touch screen display system and diagnostic software, it is possible for the operator to remotely control and analyze all aspects of his system. This meets the objective of reduced operator involvement with the operation of the ATV transmitter. In fact, in many cases, factory support and repair or work arounds can be done without actually traveling to the transmitter site.

It should be noted that critical protection systems are backed up with hard wire controls as an absolute protection for the tube and hardware.

The technology that is behind the DISC™ system is used extensively in commercial building control systems and in telephone network control. It is even incorporated into modern commercial aircraft control systems. It is therefore about time that the transmitter takes advantage of this modern digital solution to control and monitoring.

5. SPECTRUM CONSIDERATIONS

The ATV spectrum will be very crowded. It is expected that the output from an ATV transmitter must be closely controlled for out-of-band energy. {Figure (B)} shows the spectrum mask that may be applied to the ATV transmitter. A critical requirement is the reduction of the out-of-band energy. The nonlinearity inherent in any power amplifier that is operating efficiently creates intermodulation products that fall out-of-band. This is one reason why intermodulation performance of the power amplifying device to so critical. The current analog and/or digital predistortion technology that is applied to NTSC systems will not be able to meet the out-of-band energy requirements of ATV without the addition of a high power output filter {Figure (C)}.

Output filters are large and expensive components. Thus it would be desirable to eliminate the filter if at all possible. In the process of eliminating the filter it would also be a bonus if the in-band performance of the transmitter were enhanced.

COMARK, in conjunction with the David Sarnoff Research Center (SARNOFF), is developing a new kind of digital adaptive predistortion technology that will provide intermodulation product reductions of over 30dB {Figure (D)}. This huge reduction of nonlinearity effects will vastly improve the in-band transmitter performance AND eliminate the need for a large and expensive output filter. In addition, the system is fully automatic/adaptive, and does not require operator intervention. The days of tweaking numbers of linearity control adjustments are eliminated with this system.

Thus, the criteria for minimizing operator involvement with the transmitter is achieved. In addition, the cost in man-hours for setting the transmitter up initially and as an ongoing requirement is eliminated. That meets the low cost criteria set for the new generation of dedicated ATV transmitters.

6. CONCLUSIONS

ATV is a breakthrough technology that will become the standard terrestrial transmission system in the 21st century. The early implementation of ATV may be impeded by the lack of a receiver population that will generate a revenue stream to the station. This paper has described the reasons for choosing technologies that will allow the production and installation of an advanced digital ATV transmitter that meets today's requirements. Those requirements are low purchase cost and operating cost, reduced operator involvement, reliable long term service and reduced risk of early obsolescence through flexibility in design.

The ATV service demands new solutions. This paper describes such a solution for the ATV transmitter.

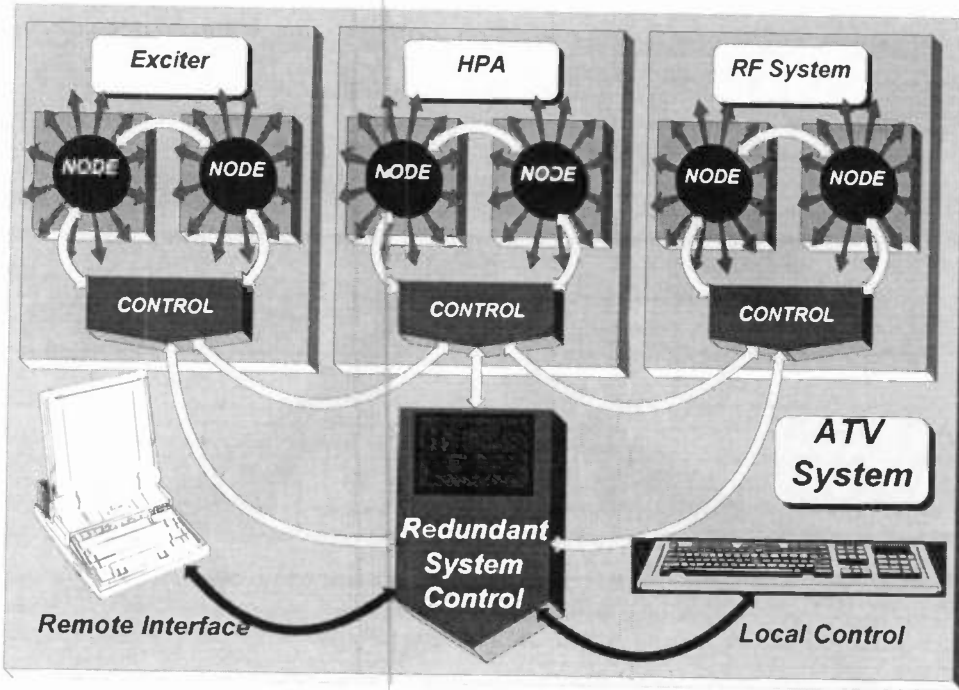

Figure A - DISC™

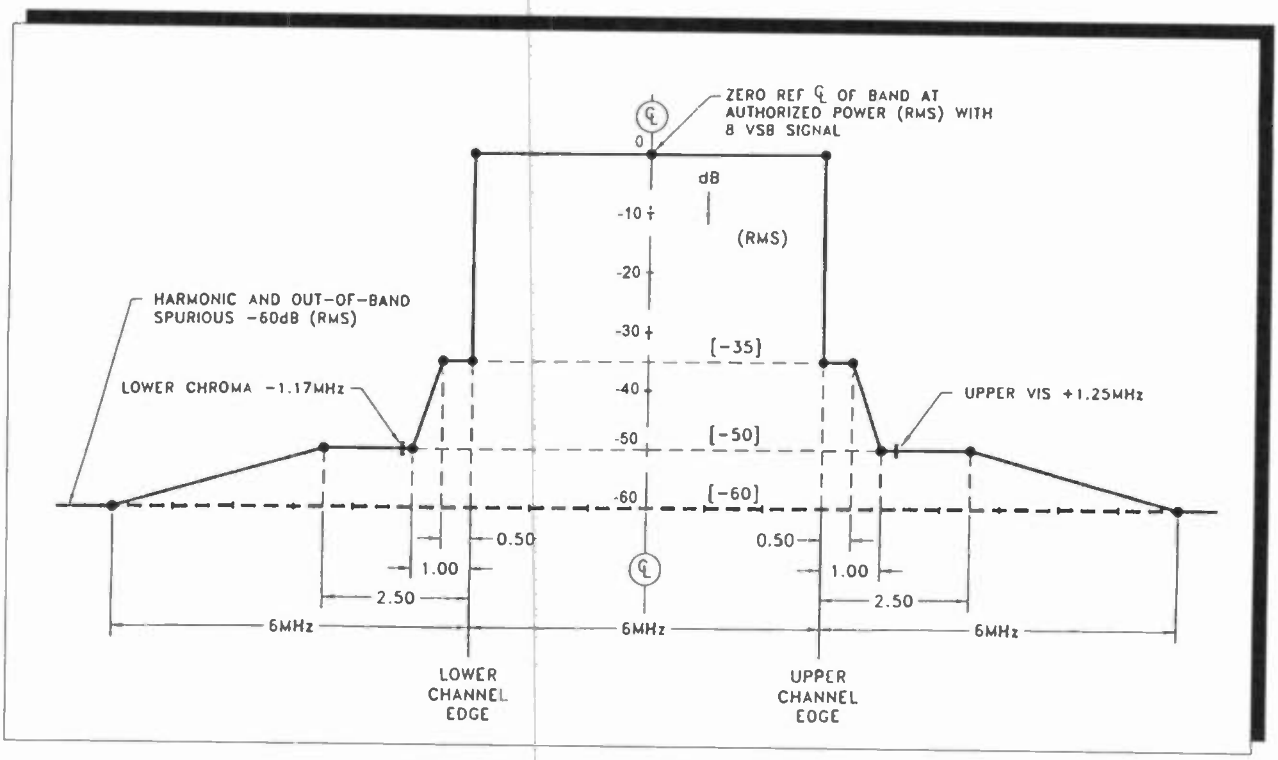

Figure B - ATV Spectrum Mask

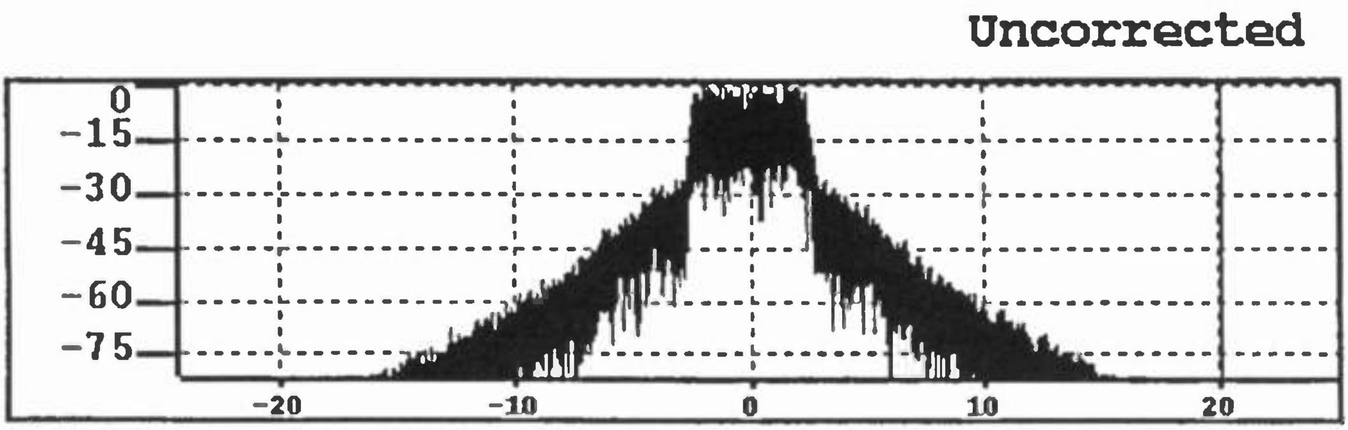

Figure C - Uncorrected ATV Spectrum

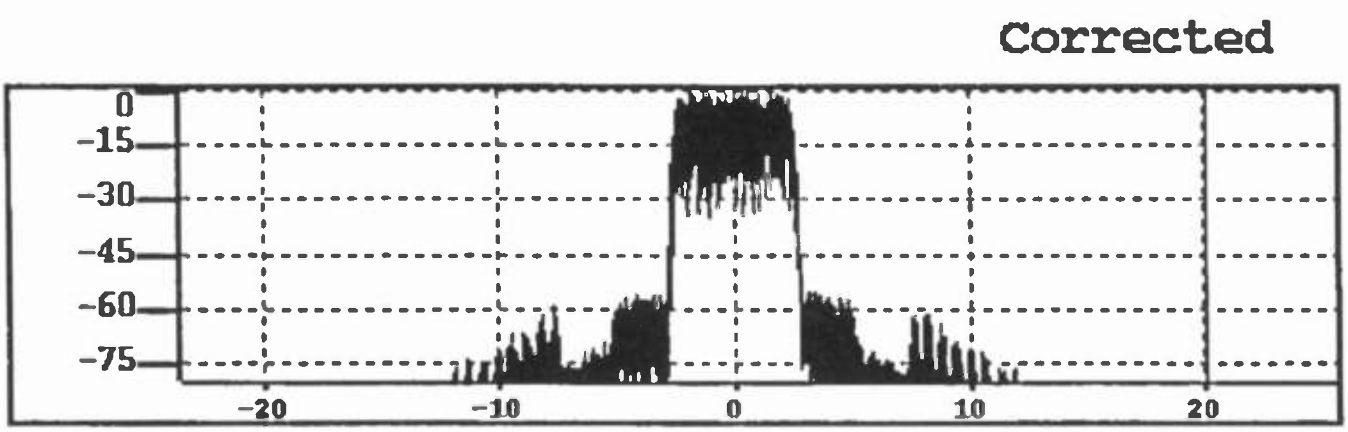

Figure D - Corrected ATV Spectrum

CAN ATV COVERAGE BE IMPROVED WITH CIRCULAR, ELLIPTICAL, OR VERTICAL POLARIZED ANTENNAS?

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Abstract

This is an important question to be answered for effective ATV systems planning that may use various polarization methods to achieve maximum coverage. This paper will examine some of the technical issues behind the various polarization techniques. This will be in the form of computer coverage analysis and available test data.

The ATV "Cliff Edge Effect" phenomenon will also be discussed along with its resulting effect on broadcast service area coverage. The concept of polarization diversity using circular polarization will be introduced as a possible means to reduce the cliff edge effect.

Has CP been a mixed bag of results?

Talking to most broadcasters about circular polarization (CP) apprears to be an uphill battle in the debate as to the coverage effectiveness achieved with CP operation. One of the most prominent objections is the requirement to double the transmitter power which more than doubles the costs. This will be discussed further in this paper with some promising results that may help ameliorate the problem.

Part of the CP acceptance problem was due to the "mixed bag of results" achieved on the VHF band where CP receiving antennas were not available to viewers to take advantage of CP benefits. In addtion, energizing the V-pol during CP operation caused some unwanted coupling on receiver down leads that produced an additional ghost during CP operation that was not present when only H-pol was being transmitted. Without a CP receiving antenna and a new shielded down lead to the reciever, most CP benefits were lost.

Adding to this dilemma was the physical size of the VHF receiving antenna where practical prototype antennas consisted of crossed yagi or log periodic antennas. On low band TV channels, the CP receiving system could be described as "huge". From this observation, it could be seen that consumer acceptance would be low.

However, the current prospect of moving ATV digital transmission to the UHF band will change all of this.

Arrival of ATV

The ATV system to date has demonstrated its excellent system capability of delivering unprecedented picture quality and sound to NTSC grade B areas. The field test coverage effectiveness was found to be equal to or better than NTSC coverage with less power, i.e., -12 dB RMS relative to NTSC peak of sync. This power level, however, is not a one to one comparison where it should be noted that when the 6 dB ATV peak to average ratio is subtracted from the -12 dB transmission ratio, a 6 dB difference exists on a peak to peak basis. This means the ATV trasnmitter peak power rating is 6 dB down from NTSC ratings or about one fourth of the NTSC transmitter power to provide comparable coverage. This assumes all other things are held constant in this comparison, i.e., antenna gain, transmission line loss, tower height and channel of operation.

The much heralded arrival of ATV, however, has brought along with it its own "Achilles Heel", the "Cliff Edge Effect".

Cliff Edge Effect

One of artifacts that accompanies most digital transmission systems is a sudden loss of received data when the input signal fades below the system carrier to noise threshold. The threshold for 8VSB is shown in Figure 1.

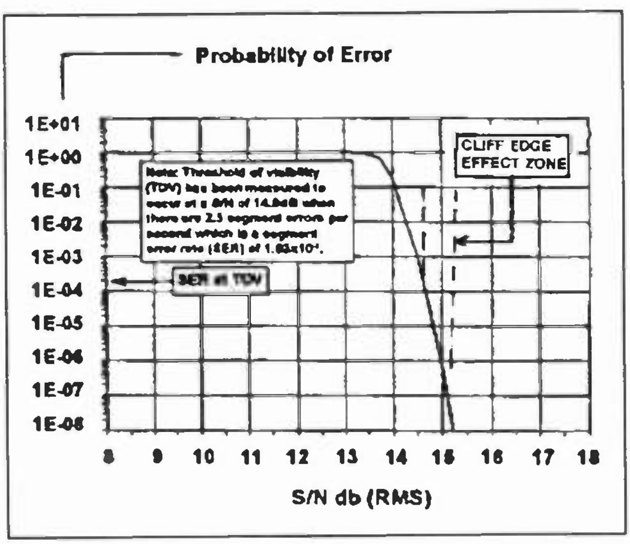

Figure 1. The threshold of visibility (TOV) for received errors in 8VSB, the onset of the cliff edge effect.

Figure 1 was taken from ATSC Doc. T3/259 reference [1]. To get an idea as to how sharp the curve is in terms of received program loss refer to Figure 2 where the digital uncertainty zone is illustrated to be approximately .6 dB wide as read from Figure 1 from 14.6 dB to 15.2 dB.

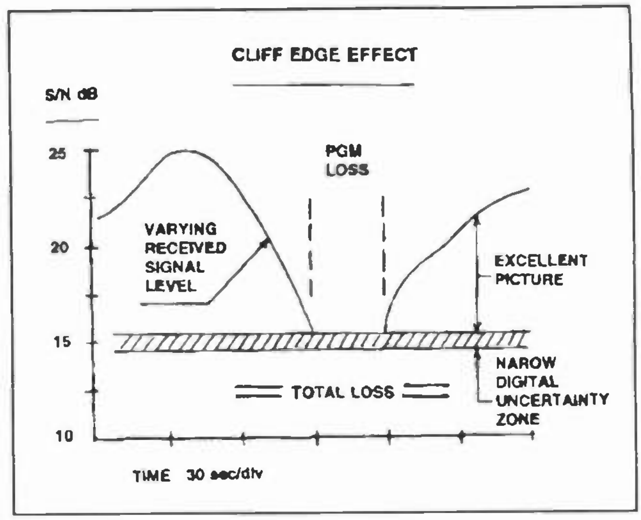

Figure 2. ATV signal shown fading into Cliff Edge Effect zone.

Figure 2 further implies another important issue regarding the length of time an outage can be tolerated by the average viewer before he switches channels. Suppose the program loss interval illustrated in Figure 2 lasts for 30 seconds, will the viewer stay tuned? It is not the intent of this paper to study viewer reactions but rather to analyze propagation effects that cause the varying (fading) signal and to find a practical means to reduce its effect.

Part of the solution would be to increase the transmitter power above the industry suggested -12 dB ratio. This has been indirectly suggested by ATV proponents by moving the FCC (50,50) planning factor to (50,90) to increase the time reliability of the received ATV signal from 50% to 90%. This additional margin helps reduce the fading effect by increasing the transmitter power about 9 dB through calculation of a new ERP when changing from (50,50) to (50,90). This increased power level, however, maybe burdensome and restrictive for ATV channel allocations.

It is proposed in this paper to reexamine the time occurrence of fading TV signals and to look toward other techniques that may help in an equal or better manner, i.e., by using polarization diversity through the transmission of CP.

Past History on Diversity Techniques

It was noted from the literature [2] that some early Cable TV systems experimented with a form of space and polarization diversity to help reduce the frequency of deep signal fades on head end systems. The technique used two identical antennas, one mounted horizontally and the other mounted vertically. The antenna outputs were hybrid combined together with the result that when one signal faded the other signal was still present with sufficient signal strength to maintain service. Refer to Figure 3.

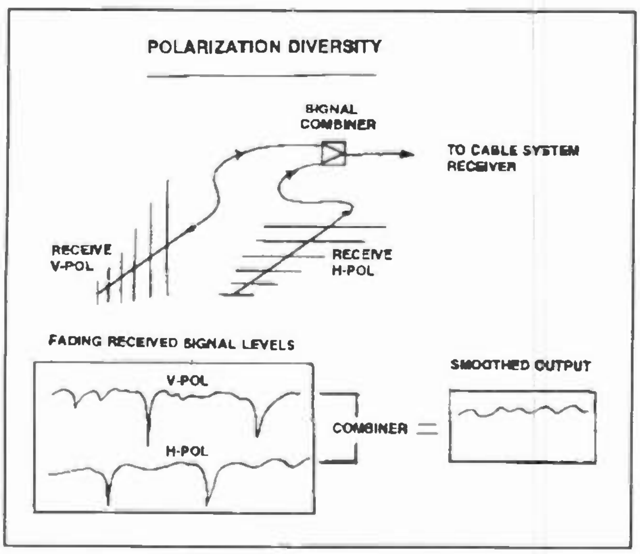

Figure 3. Early Cable TV receiving system using space and polarization diversity.

The interesting thing about the diveristy system shown in Figure 3 is the polarization aspect because at the time when this system was being tested, there were no CP stations on the air in the vicinity. The vertical component of the signal was apparently generated through a diffraction process over obstacles along the path and as a result, the polarization diversity was site specific.

Polarization Diversity

In view of the concepts shown in Figure 3, the idea of polarization diversity looks very promising if the V-pol signal were to be deliberately transmitted through CP. This could provide the missing signal level when one of the H or V signal components takes a deep fade. If inherent diversity effects exist in CP operation then it can be put to good use in reducing the cliff edge effect.

The answer to this suppositon maybe brought into focus by examining the following issues regarding polarization diversity.

1. Do H and V signal components normally fade together or are they independent functions that are uncorrelated?

2. Does a CP receiving antenna combine the H and V signals on a vector sum basis such that if one component H or V is lost, will there be enough output level for effective diversity?

Polarization issue 1 has been studied by various authors [2] [3] [4]. The results strongly indicate that H and V signals fade, to a large extent, independently. The correlation factor between H and V signals was noted to be +/-.3 or less [4] which implies that independent fading characteristics can be counted upon to exist a high percentage of the time. This supports the concept of polarization diversity.

Polarization issue 2 will require further analysis since little information regarding the combining action of CP receiving antennas can be found in the literature. In particular, the actual H and V combining characteristics must be accurate to within 1 dB to match an analysis where the cliff edge effect occurrs within a 1 dB level window. This is an item for further analysis and antenna range testing.

A further requirement of the analysis would be a transmitter power level assesement to determine if effective CP diversity can be obtained with normal H-pol licensed power or if will it be necessary to double the transmitter power by putting the same H-pol power in both H and V polarizations.

Analysis Objectives

The first part of the following analysis will be to show measured test data on CP transmit and receiving antennas. Figure 4 shows the test setup.

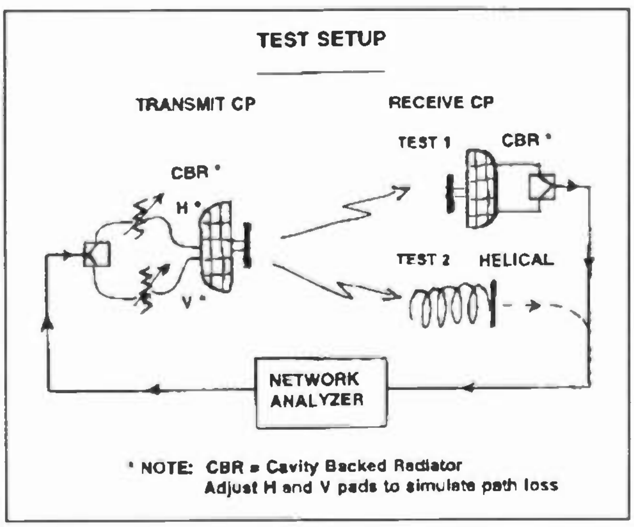

Figure 4. CP transmit and receive test setup to allow fading H and V polarizations independently with pads.

The test setup shown in Figure 4 uses a circularly polarized cavity backed radiator (CBR) as a transmit antenna. Two tests were set up, test 1 that uses the CBR as a receiving antenna and test 2 that uses a helical receiving antenna. The purpose of the test was to determine the output signal after fading either H or V signals.

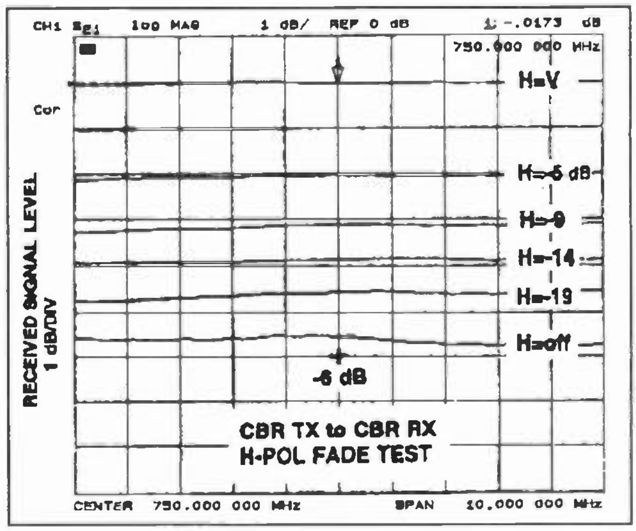

Figure 5. H-pol fade test using CBR to CBR antennas. The H-pol fade is in dB.

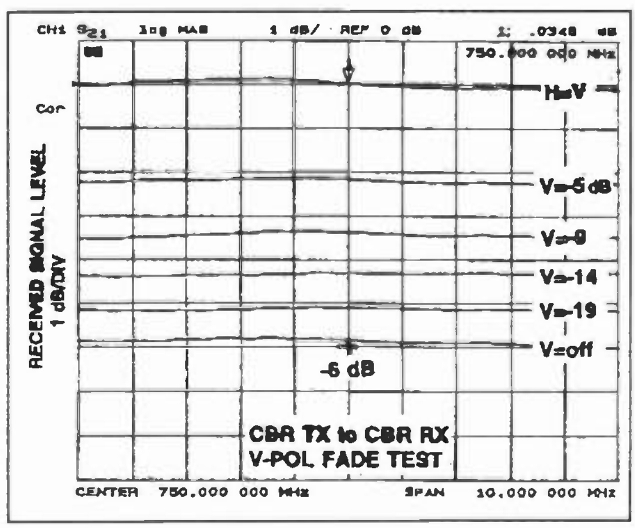

Figure 6. V-pol fade test using CBR to CBR antennas. The V-pol fade level is shown in dB.

The interesting result shown in Figure 5 is that after fading down the H-pol signal by significant amounts, the received output signal did not drop below -6 dB. This implies that polarization diversity will work in those areas where independent fading of H and V signals occurrs.

In Figure 5, the received signal was supported by the presence of the V-pol that was held constant for this test. The reference received level is at the top of the graph where H=V. From this point, the H-pol fading levels are shown in dB.

It can be noted that if this were a distant ATV receiver experiencing this H-pol fade and if the ATV receiver S/N threshold was at -7 dB, then the receiver would have suffered a crash at the H=-9 dB fade point shown in Figure 5. Also note, the received signal level was measured at -4 dB (on the H=-9dB fade line) to provide adequate headroom above the -7dB threshold to prevent a crash. The vector sum of a 0 dB V-pol signal plus a -9dB H-pol signal equals -4 dB in the combining mechanism of a CP receiving antenna.

In other words, the CP antenna maintained TV service while the H-pol antenna failed. This of course is a special situation that may have infrequent occurrences (to be determined) but does show the effectiveness of CP polarization diversity.

Figure 6 is the same setup, however with the V-pol faded down. The V-pol fading characteristics is similar to the H-pol case.

Both Figures 5 and 6 started out with H=V input signal level. This also corresponds to the level on an H-pol only transmission system and represents the same level that would be received on a H-pol to H-pol case.

For comparison purposes the transmitter power is the same in the above test case for CP to CP or H to H. The CP power was not doubled but split equally to each polarization.

The next test was to see if the -6dB fade number is characteristic of non-hybrid combining receiving antenna, i.e., a helical antenna.

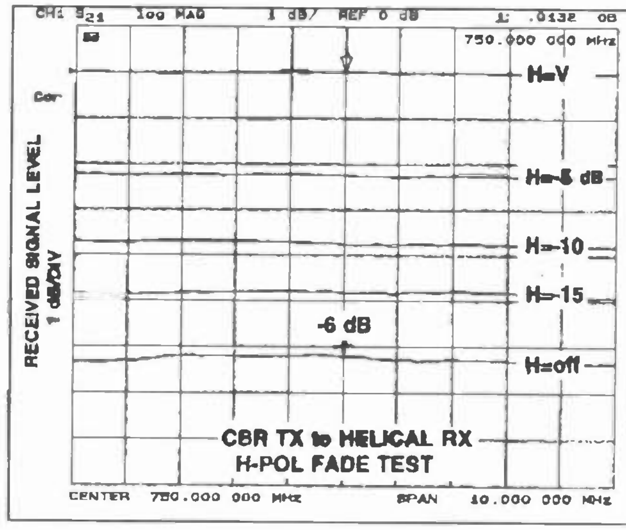

Figure 7. H-pol fade test on a CBR to helical antenna. H-pol fade numbers in dB.

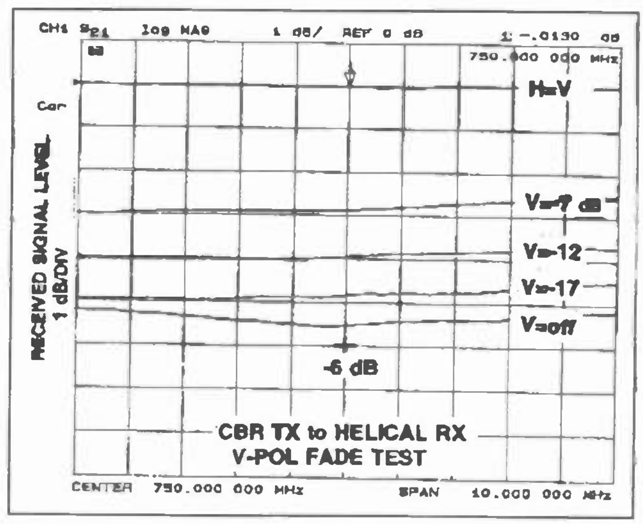

Figure 8. V-pol fade test on a CBR to helical antenna. V-pol fade numbers in dB.

The helical receiving antenna tests shown in Figure 7 and 8 displayed a similar result as the CBR to CBR tests, that is , a -6dB fade level when one polarity is faded down to 0, the "off" condition indicated in the Figures.

From this, it appears both hybrid combined and electromagnetically combined receiving antennas exhibit the same characteristics. Both antenna types will provide a reasonable degree of polarization diversity to enhance ATV reception in fading areas.

Next, a computer CAE program using nodal analysis techniques was be set up to compare the measured results shown in Figures 5,6,7,8 with calculations. Afterwards, the CAE program will be used to examine other reflective path conditions to note other system parameter improvements using CP.

CP Computer Program

A computer program was set up to model the circular polarized path with reflection points. The program was based on standard CAE nodal analysis techniques with special attention paid to additional processing to model the reflection and combiner points [6]. Figure 9 shows a functional block diagram.

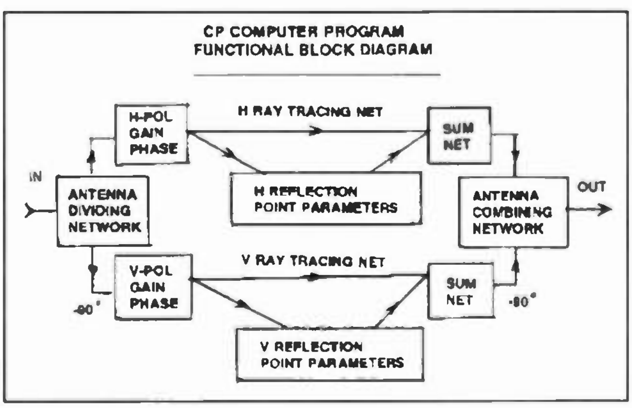

Figure 9. Computer model for a circularly polarized transmission path.

The computer was run using the measured H and V pol attenuation values for comparison. This is shown in Figure 10.

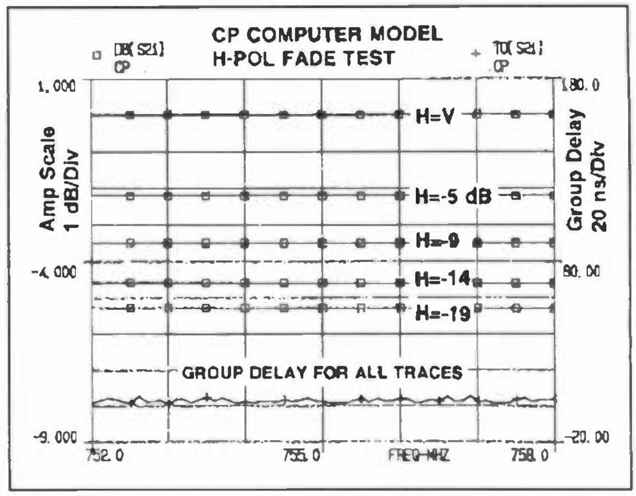

Figure 10. H-pol fade test on CP computer model with group delay.

The computer results shown in Figure 10 benchmarked very well with measured data to prove the program useful for further analysis. This is also shown in the tabular data below.

CP Computer Test Run Comparison

	H-pol Fade		V-pol Fade	
dB	meas.	cal.	meas.	cal.
0	-2.0	-2.2	-2.1	-2.1
-5	-3.2	-3.4	-3.3	-3.2
-9	-3.2	-3.4	-3.3	-3.2
-14	-3.9	-4.5	-4.2	-4.3
-19	-4.6	-5.2	-5.1	-4.9
off	-5.6	-6.2	-5.9	-5.7

Table 1. CP Computer Benchmark Test.

The results in Table 1 show good agreement between the measured and calculated tests. Notice that group delay is available in computer model for CP path analysis.

For a realistic view of the performance down a CP reflective path versus the same path conditions on H-pol only, the following analysis is provided. The computer model was set up to provide 30% reflection conditions for both paths. Figure 11 shows the results on the H-pol only path.

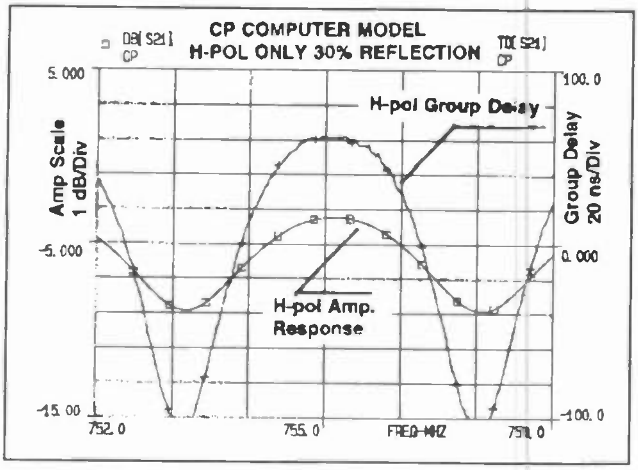

Figure 11. The results of 30% reflection on a H-pol signal over 6 Mhz.

The same 30% reflection situation is now presented to the CP system with the results shown in Figure 12.

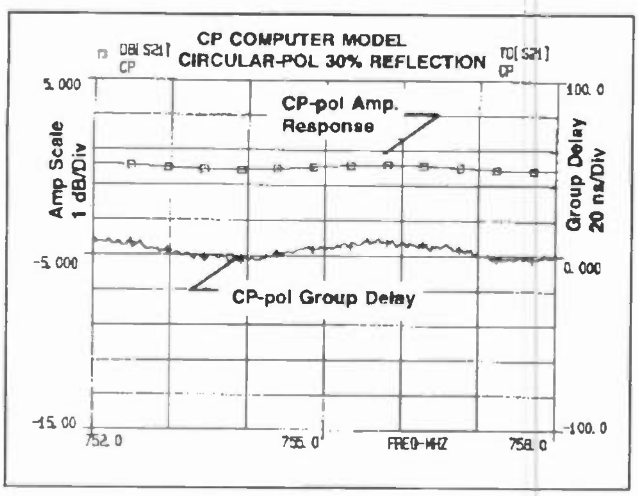

Figure 12. The same 30% reflection on CP.

It is known from field observations that a 30% reflection on a standard H-pol TV is not tolerable as is shown in Figure 11 depicting a 3 dB response change with over 160 ns of group delay variation in the pass band.

The CP signal as shown in Figure 12 has the same 30% reflection but is considerably improved. This is to be expected since this has been the main attribute of CP operation, ghost cancellation. The data presented here has provided another way to look at an old problem.

It may be pointed out that CP operation usually suffers from degraded axial ratio down a reflective path where the V-pol may experience additional attenuation due to possible Brewster angle conditions or vegetation losses.

Even with additional V-pol losses, CP operation still provides better performance as is shown in Figure 13 where the V-pol was attenuated 6 dB.

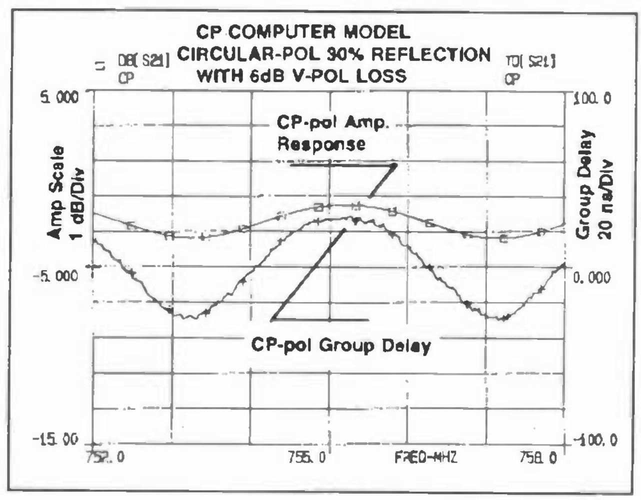

Figure 13. CP operation with 30% reflection and 6 dB attenuated V-pol.

The performance shown in Figure 13 is acceptable for NTSC but marginal for ATV. The overall response improvement is another important feature of CP. This characteristic can help unburden the ATV receiver adaptive equalization requirement.

Coverage Reliability Improvement

The CP polarization diversity improvement demonstrated so far is for a single path situation. The diversity idea, which is really another way of improving the reception reliability, is intended to reduce the occurrence of cliff edge effect drop outs. To assess the impact of this, it is necessary to look at an overall coverage situation.

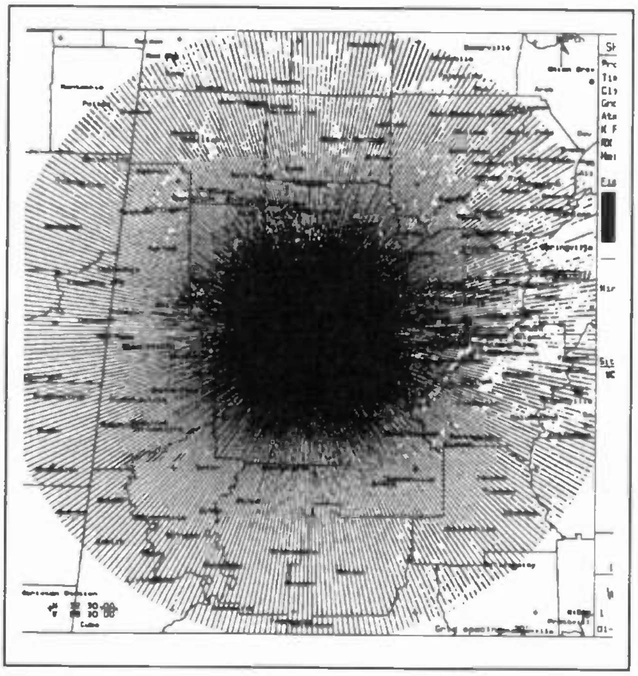

Figure 14. Typical TV coverage map including terrain obstacles. EDX Coverage Software [A1].

In Figure 14, coverage to the right on the map shows light areas that indicate low signal strength. Note, the high quality of the original coverage map in color does not reproduce well here but the general idea of terrain blockage and low signal strength areas can be seen.

A expanded segment scan of the low signal areas to the right in Figure 14 was made to further examine the diversity improvement. This is shown in Figure 15.

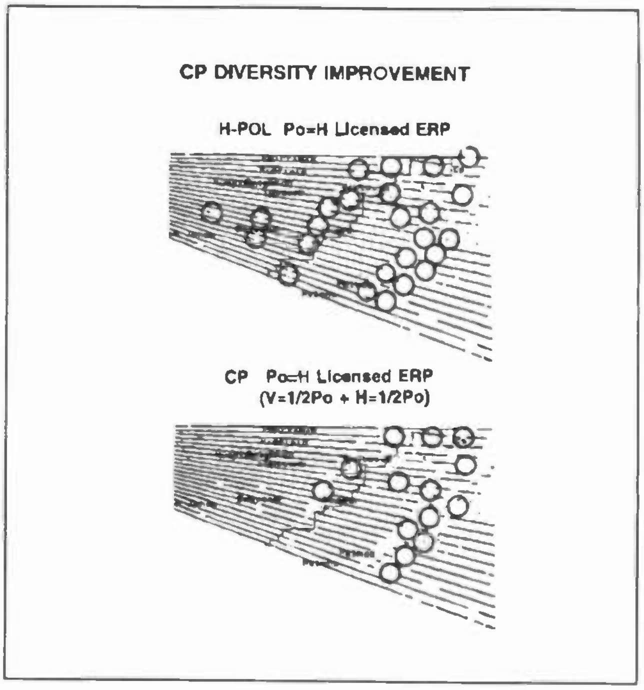

Figure 15. Expanded view of coverage zone with circles to indicate potential ATV cliff edge effect areas and CP improvement.

The general CP improvement effect shown in Figure 15 is to move a number of cliff edge effect areas further out to fringe areas. It appears the CP diversity improvement is approaching the [50,90] planning factor without increasing transmitter power. This is an item for further in-depth studies. The above is just a snap shot view of the diversity effect.

The above coverage estimate was made by running RPATH™ [A1], a path profile program, a number of times over the low signal areas to determine the H and V components. The resulting CP signal was calculated using the computer program described earlier and the CP improvement was estimated by noting the CP level above an assumed receiver S/N threshold versus the H-pol signal.

When a path analysis is first run, little H or V signal variations may be noted ,however, when the K-factor is also changed, then noticeable H and V signal differences of 4 to 8 dB can occur. This is a weather variable effect and is the source of time varying signal levels. Table 2 shows the effect of varying K-factor. These were some of the values used to estimate the CP improvement shown in Figure 15.

Table 2
CP Diversity Improvement

K-factor	V-pol	H-pol	CP1	CP2
1.33	-52	-52	-49	-51
1.20	-52	-56	-51	-54
1.10	-52	-58	-52	-55
1.00	-53	-61*	-54	-57

Note: The V-pol and H-pol levels in dBm .

In Table 2, if the ATV receiver threshold is assumed to be -60 dBm then the H-pol signal has suffered a cliff edge effect crash at the (*) symbol. The CP1 values shown are for CP power H=V (2xH-pol). CP2 is a start up case where one half H-pol power is put into H and V, that is, not doubling the transmitter power. Notice that no cliff effect crash has occurred on either CP1 or CP2. The CP diversity improvement shown in Figure 15 uses CP2 values.

This analysis is just the beginning of a much more in depth study necessary to gain further insight into polarization diversity. The key item here would be the K-factor data which is usually presented as refractive gradient values on a cumulative distribution chart. This can provide a reasonable view of the statistical nature of the diversity effect and would begin to answer important planning questions such as how much improvement can be obtained. The answer will be a time related, statistical function.

Elliptical Polarization

Elliptical polarization is a form of circular polarization with the axial ratio reduced as required. It also has some benefits similar to CP but reduced in effectiveness.

One of the primary reasons for using elliptical polarization is to provide coverage fill in for urban areas where the additional V-pol penetrates better and improves coverage to second set homes using a loop or rabbit ear type antennas.

An example of an elliptical antenna azimuth pattern is showing in Figure 16.

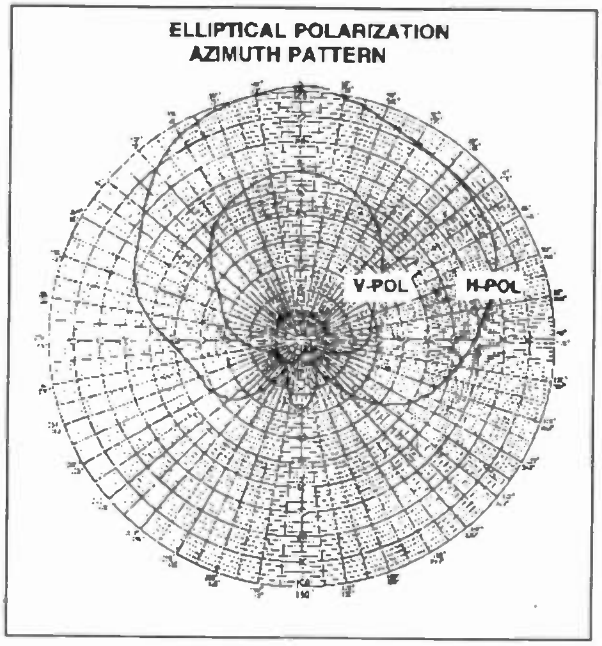

Figure 16. Elliptical polarization azimuth pattern showing 70% H and 30% V power split.

The 70/30 power split shown in Figure 16 closely matches the CP performance shown in Figure 13, when using a CP receiving antenna.

A number of UHF stations are using elliptical polarization with favorable results.

Vertical polarization

Vertical polarization is a new item for TV broadcasting and can be used as another means to reduce interference [5] [8]. Noting that interference is a major issue for ATV allocations, and in situations where no ATV assignment maybe available, vertical polarization can provide service. Also noting that a CP antenna can be set up for V-pol operation to suit the requirements of a tight allocations plan during the transition period and then later be modified for CP operation, makes a flexible type CP antenna a choice for special interference situations.

Closing Comments

Using the vertical component in any format, albeit CP, elliptical or vertical, can provide various ATV coverage improvements. A properly designed CP antenna installed at the start of an ATV project can provide the flexibility to transmit standard H-pol, CP, elliptical or vertical as required and then later be modified after the transition period to operate on full CP for enhanced coverage.

Conclusions

1. CP can be used as a polarization diversity system to enhance ATV coverage by helping to reduce the cliff edge effect.

2. Noteworthy diversity improvement can be obtained without doubling the power.

3. Inherent CP reflection cancellation can reduce the equalization effort required of ATV receiver adaptive equalizers.

4. Various forms of elliptical, vertical polarization and patterns can be accommodated with CP transmit antennas.

5. Planning for CP at the onset of an ATV project can improve coverage during the transition period and prepare the way for overall improved ATV coverage afterward.

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Acknowledgments

A1. Trade mark, EDX engineering, Inc. Eugene, OR.

CHANNEL COMBINING IN AN NTSC/ATV ENVIRONMENT

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ABSTRACT

Many techniques are being offered to the broadcaster on ways to utilize the existing tower for both the NTSC channel and the future ATV channel. Combining TV channels will allow some broadcasters to use the existing structure to transmit NTSC and ATV from a common line and antenna or ,in the case of a VHF and UHF, utilize the same line to feed two separate antennas.

This paper will discuss the design aspects of selected channel combiners, compare measured performance and analyze their applications for present and future terrestrial broadcasting.

BENEFITS

In the upcoming transition period from NTSC to ATV, many broadcasters will choose to utilize their existing tower to transmit both NTSC and ATV channels. Some may choose to add a new line and ATV antenna, others may combine their ATV with NTSC and transmit from a common antenna and line and still others may choose to consolidate to a new structure common to many local channels. For most it will be a choice of cost and feasibility.

TYPES OF COMBINERS

Channel combiners, also known as multiplexers or diplexers, have different designs that are used for different applications.

1. ***Constant impedance*** designs consist of two identical filters placed between two hybrids.

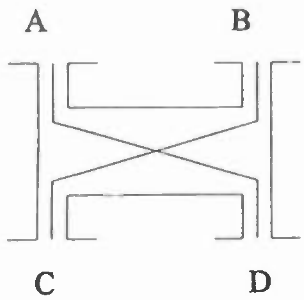
2. ***Starpoint*** designs consist of single bandpass filters phased into a common output tee.

3. ***Resonant Loop*** types utilize two coax lines placed between two hybrids. The coax lines are of a calculated difference length.

4. ***Common Line*** types use a combination of bandstop filters matched into a common output tee.

CONSTANT IMPEDANCE COMBINERS

The makeup of this combiner contains two bandpass (or bandstop) filters placed between two hybrids. The function of the hybrid is to split the input into two signals of equal amplitude and phased 90 degrees apart. Figure 1 shows how the hybrid operates.



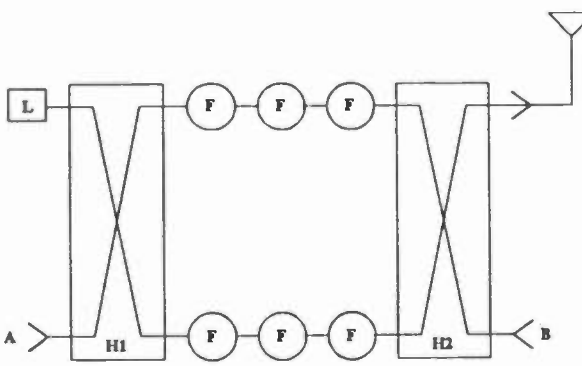
(A/2@ 0^0) (A/2@ -90^0)

Figure 1. 3dB crossover hybrid

A signal fed into Port A splits equally into Port C and D but with a 90 degrees phase difference between the two signals. If a short circuit is placed on C and D the input signal will now appear at Port B with very low loss.

In the constant impedance combiner one input has the narrow band characteristics of the bandpass filter with the filter tuned to pass a specific bandwidth and reject (or short circuit) all others. The second input features the broadband characteristics of the output hybrid.

Figure 2 shows the makeup of the constant impedance channel combiner.



H 3dB hybrid
L Balance load
F Bandpass filters
A Narrow band input
B Broadband input

Figure 2. Constant Impedance Combiner

Input A is split into two signals as it goes through the first hybrid. The bandpass filters are tuned to A allowing the signals to pass. These signals arrive at the second hybrid and because of the phase relationship between them they add at the antenna port. Input B is split in the second hybrid and sees a short circuit (i.e. the skirts of the filters). These signals then combine to the output.

Typical Performance

Isolation: Isolation between the two inputs is not the same for both. The signal that passes through the bandpass filters will have isolation properties of the second hybrid towards ChB. ChB on the other hand will be isolated from ChA primarily by the bandpass filter performance. Isolation numbers in excess of 35dB and 70dB respectively are possible.

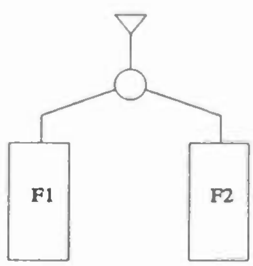
Insertion Loss: Channel loss or efficiency is also different for either input. The signal (ChA) that passes through the bandpass filters will have the loss characteristics of the filter. This number can be as low as 0.10dB or as high as 0.50dB depending on the number of filter elements used and the channel spacing. The signal (ChB) that gets reflected back off the filters will have loss figures relevant to the second hybrid. These figures are much lower and are usually 0.10dB.

Power rating: The power rating for each channel is determined by the power handling of the bandpass filters used as well as the power handling of the output hybrid.

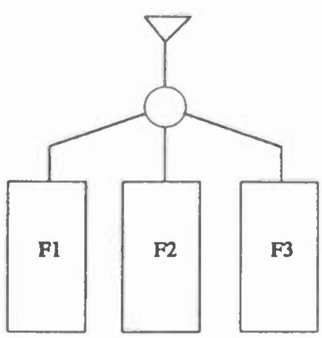
Multi-Channel Operation: Constant impedance combiners are an ideal module for use in multi-channel installations. These combiners allow for additional channels to be added after initial installation. There is no restriction between the channels being combined and variations in channel spacing.

STARPOINT

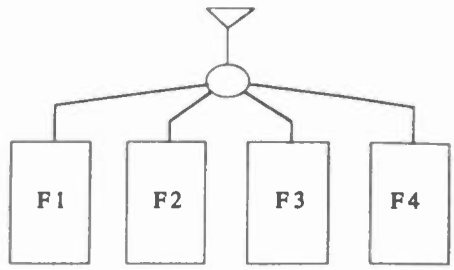
The starpoint combiner consists of bandpass filters with a certain number of tuned circuits. The number of tuned circuits is based on the passband requirements and the required isolation between these channels. The bandpass filters are designed to match the RF output power of the transmitter channels being combined. Filter designs vary from interdigital, iris coupled (capacitive and inductive) and loop coupled. The outputs of each bandpass filter are combined into a common node (star-point) through defined lengths of coax. The lengths are different from each other and are dependent upon the frequencies of the channels being combined. Each length is a common denomination of a half wavelength of the frequencies that are being combined with a particular channel. Figure 3 shows the arrangement of a two, three and four channel starpoint combiners.



2 channel combiner



3 channel combiner



4 channel combiner

Figure 3. Starpoint Channel Combiners

Typical Performance

Isolation: Isolation between all inputs in a starpoint configuration relates directly to the roll-off of each bandpass filter. The slope of the filter is based on the number of elements used. The higher the number of elements the sharper the slope, resulting in higher isolation. Typical isolation figures in a starpoint combiner, with single channel separation, ranges from 40 to greater than 70dB for greater channel spacing.

Insertion Loss: Loss for each channel being combined is related to the loss of the bandpass filter. Channel spacing and filter design play an important role in the insertion loss characteristics. Channel loss can range from 0.10dB and upwards. As the spacing between channels combined gets smaller the loss increases and vice versa.

Power rating: The power rating for each channel is determined by the design and power handling of the bandpass filters used, coupled with the power handling of the output tees.

Multi-Channel Operation: Starpoint combiners can be used to combine up to 6 channels into a single output. Because of the phasing line technique used to combine the channels all inputs are fixed tuned to a specific channel. Expansion from a 2 to 3 (or more) in the field in not practical.

RESONANT LOOP COMBINERS

Resonant loop combiners are best used when the frequency separation between the signals being combined is greater than (24MHz). The frequency separation is taken from the center of each channel being combined. The essential elements are a matched pair of 3dB hybrids interconnected by two transmission lines whose electrical lengths differ by a half wavelength. This half wavelength is related to the difference frequency of the two signals being combined.

The following calculation shows the difference length for two specific channels.

Input A = CH 23 (524-530MHz)
Input B = CH 31 (572-578MHz)

Difference Frequency = 48MHz (575-527 MHz)

λo = wavelength in free space

$$\lambda o(inches) = \frac{11803}{f\,(mhz)}$$

$$\lambda o(inches) = \frac{11803}{48} = 245.89 \text{ inches}$$

Df = difference length

$$Df = \frac{\lambda o}{2} = 122.95 \text{ inches}$$

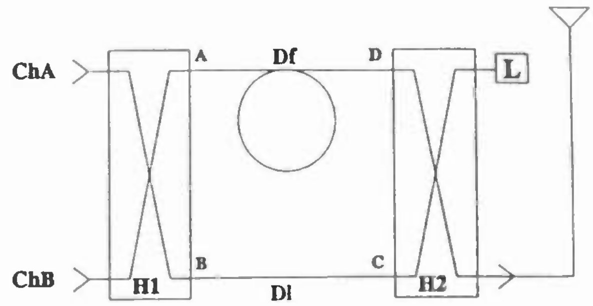
λo = 22.49 inches @ 527 MHz

λo = 20.53 inches @ 575 MHz

$$Df\ @\ 527\ MHz = 5.5\ \lambda o = \text{odd number of } \frac{\lambda o}{2}$$

$$Df\ @\ 575\ MHz = 6.0\ \lambda o = \text{even number of } \frac{\lambda o}{2}$$

Figure 4. Calculation of a difference length for combining two specific UHF channels.



ChA,ChB	inputs
H	3dB hybrids
Dl	"direct" interconnection
Df	"difference line" interconnection
L	balance load

FIGURE 5. Resonant Loop Combiner

The makeup of a resonant loop combiner consists of a number of components shown in Figure 5. The transmitters, ChA and ChB, are connected to the "isolated" ports of the first hybrid. Input ChA will combine at the antenna provided the path difference is electrically an even number of half-wavelengths. Input ChB will combine at the antenna provided the path difference is electrically an odd number of half-wavelengths. The fourth port is for the balanced load.

Typical Performance

Isolation: Isolation between the two channels is determined by the isolation performance of the first hybrid. This is typically better than 30-35 dB and will vary with channel spacing. Higher isolation can be achieved by the use of additional filters at the hybrid inputs.

Insertion Loss: Insertion loss for the resonant loop combiner is dependent on both the channel spacing and the mean frequency. For UHF applications, where the mean frequency is high compared with the channel bandwidth, channel loss varies with spacing. A combiner with three channels of separation will have insertion loss figures around 0.30 to 0.50 dB whereas a separation of four or more channels will result in loss figures of 0.10 to 0.20dB.

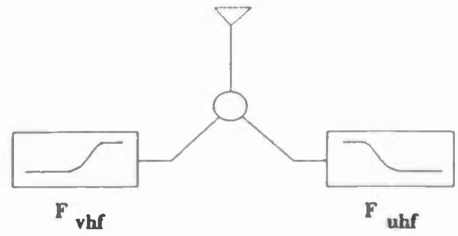
Input VSWR: Input VSWR for either channel is less than 1.10:1 and typically 1.05:1.

Power rating: Combiner designs are available for a wide range of power ratings and based on the ratings of the hybrids chosen. Actual transmitter powers and channels being combined will be used to determine the proper sizes.

Multi-Channel Operation: In general Resonant Loop combiners are only used for combining two channels and only if the channel spacing is wide (>24MHz). Using this design for more than two channels requires equal spacing between channels.

COMMON LINE COUPLERS

The design of this combiner is made up of two coupling units, one for inside the building and the other for the tower. The function of the coupling units is to combine a VHF with a UHF and then de-couple the signals to feed separate antennas. The combining is achieved using the technique similar to that of starpoint combiners. The VHF input uses a harmonic (lo-pass) filter which passes the VHF channel and also rejects the third, fourth, fifth and so on harmonics of the VHF channel. These harmonics appear in the UHF band so the VHF lo-pass filter acts as a short circuit to UHF channels. The UHF input uses waveguide as the lo-pass filter and because UHF waveguide cannot propagate in the VHF bands it acts as a short circuit to the VHF input. These two lo-pass filters are then combined into a common output tee using the line phasing techniques of the starpoint combiner. Figure 6 shows the arrangement of this design.



F_{vhf} lo-pass filter (pass VHF reject UHF)

F_{uhf} hi-pass filter (pass UHF reject VHF)

FIGURE 6. Common Line Combiner

Typical Performance

Isolation: Isolation between the VHF and UHF channels is related to the harmonic performance of the VHF lo-pass filter. This figure is typically > 50dB. Because the UHF waveguide doesn't propagate at VHF, isolation figures from UHF to VHF have been measured at values > 90dB.

Insertion Loss: Insertion loss for either input is very low and is related to the performance of the lo-pass filters used. Low values <0.10 dB are possible.

Input VSWR: Input VSWR for either channel is typically on the order of 1.04:1 or lower.

Multi-Channel Operation: The use of the common line combiner is limited to one VHF and one UHF channel. This design uses lo-pass filters which are insensitive to temperature variations. This allows the de-coupler to be located at the top of the tower.

NTSC/ATV APPLICATIONS

Different channel combiner designs will have their place in the future NTSC/ATV environment. The design chosen will be based on the particular needs of the station. The following summary of the channel combiners discussed compares performance and uses. Following chart assumes all ATV channels will be UHF.

NTSC/ATV	CIBP	STAR	LOOP	DCCL
LoV/UHF	---	---	---	√
HiV/UHF	---	---	---	√
UHF/UHF	√	√	√	---
Multi-CH	√	√	---	---

√ = can be used --- = cannot be used

Figure 7. Channel combining choices for NTSC/ATV conversion period.

	VSWR (in band)	LOSS (dB)	ISOL (dB)
CIBP	1.05	0.10 0.30	40 70
STAR	1.08	0.10 0.60	50 80
LOOP	1.10	0.10 0.50	30 40
DCCL	1.05	0.10 0.15	50 90

Figures reflect low to high values expected.

Figure 8. Performance comparison for different channel combiners.

Efficiency Conversion:

$$\% \text{ Eff.} = 10^{-x}$$

$$-x = \frac{loss(dB)}{10}$$

0.10dB = 97.7%
0.20dB = 95.5%
0.30dB = 93.3%
0.40dB = 91.2%
0.50dB = 89.1%

Figure 9. Conversion of insertion loss in dB to system efficiency in percent.

CONCLUSION

Various types of combiners have been discussed. The performance of the combiner designs discussed in this paper, are dependent on the filters, hybrids and channel spacing. The choice of channel combining during and after the NTSC/ATV transition period will depend upon the individual needs of each station. Tower loading, tower leasing, and tower space will play an important role in choosing to combine or not. The current technology of channel combining will allow each station to benefit economically. We are already seeing the large markets choosing to consolidate onto a common structure. Each station will have its own unique situation and the choices are many.

Comparison of channel combiners has been provided and it is hoped that this paper will act as a guide and allow a proper evaluation to the benefits of co-locating.

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DESIGN AND IMPLEMENTATION OF A 3-CCD, STATE OF THE ART, 750-LINE HDTV PROGRESSIVE SCAN BROADCAST CAMERA

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ABSTRACT

This paper describes an HDTV camera system implementing the 1280 x 720 image format supporting the 750/60/1:1 production standard. An interlaced HDTV camera has been adapted to meet the proposed US HDTV progressive standard. A new 1" video format, 16:9 aspect ratio progressive scan, frame-transfer (FT) CCD sensor with square pixels was designed, and sensor incorporation and camera adaptations were implemented. The new sensor is described and the impact of the 750-line standard is discussed. The first prototype of this camera will be demonstrated at the 1996 NAB show.

INTRODUCTION

The CCD image sensor was introduced into broadcast cameras around 1986. In 1992 the first HDTV CCD camera system to meet the proposed (European) EUREKA HDTV standard was introduced [1]. This LDK 9000 camera system, designed by Broadcast Television Systems (BTS), is based on a 1" frame-transfer sensor that was developed to support the 1250/50/2:1 interlaced European standard as well as the new American 1920 x 1280 format 1125/60/2:1 interlaced production standard. This camera has been used for the production of test material by the Advanced Television Test Center.

The Advisory Committee for Advanced Television Services (ACATS) recently released its proposal for a new U. S. television standard commonly referred to as HDTV [2]. The development of a progressive scan format for HDTV has been driven by concerns such as compatibility with computer systems, ease of compression, freedom from flicker (especially with graphics), and better temporal resolution. As discussed in the ACATS process, the interoperable and extensible HDTV system can serve not only entertainment and television, but can also offer economic and qualitative benefits to education, health care and human services, commercial enterprise, and the information infrastructure.

Several factors were considered critical to achieving interoperability [3]. One of these factors is the use of progressive scan square pixel image formats in capture, transmission, and display. Thereby, the television equipment can be extended to and stimulated by applications in computer communications, high quality imaging, synthetic imaging, animation, motion pictures, and so forth. The information infrastructure needs an image architecture that eases exchange between industries and applications.

Significant technical hurdles have acted as barriers to deploying a progressive scan HDTV system. Nonetheless, the Grand Alliance did incorporate progressive scan among their formats. Most experts agree that a progressive scan system is ultimately desirable and certainly inevitable in the proposed lifetime of HDTV, though the time frame is debated.

The major technical hurdle has been the difficulty in producing a progressive scan camera of comparable sensitivity and specifications to a studio quality interlace scan camera. Existing commercial and prototype cameras have been inadequate. Indeed, the existence of adequate component technology has been in doubt. Herein laid the motivation for our research and development efforts.

Through a cooperative effort between the Polaroid Image Sensor Technology Division and BTS the LDK 9000 HDTV CCD Camera system was recently adapted for the progressive 1280 x 720 standard (750/60/1:1). A new frame-transfer sensor meeting this standard was developed at Polaroid to be optically, electronically, and mechanically compatible with the previous interlaced sensor, although differing in image format and timing. The main adaptations of the camera system performed by BTS included the following elements:

- Camera/sensor pulse generator
- Camera Processing Unit pulse generation
- Vertical contour delay
- 7 inch view finder

Apart from the above-mentioned functions, minor adaptations were made in several areas to meet the timing specification and to optimize sensor performance.

Parameter	Value
Aspect Ratio	16:9
Interlace	1:1 (progressive)
Field frequency	60 Hz
Total number of lines	750
Number of active lines	720
Line frequency	45,000 Hz
Total line time	22.222 μsec (1650 samples)
Active line time	17.239 μsec (1280 samples)
Horizontal blanking	4.983 μsec (370 samples)
Sample frequency	74.25 MHz
Sync pulse	Tri-level

Table 1. Main characteristics of the 750/60/1:1 1280 x 720 production standard

PROPOSED 1280 X 720 PROGRESSIVE TELEVISION STANDARD

Interlace scanning has proven to be an efficient way of sampling pictures. The flicker perception of the human eye demands a refresh rate of the CRT of at least 50 times per second to prevent large area flicker. In order to save bandwidth, it was decided to refresh alternately the odd lines and even lines, thus doubling the vertical resolution for a given signal bandwidth. This means that for a given signal bandwidth the number of pixels in an interlace standard will be twice the number of pixels in a progressive system, resulting in a better static resolution.

But interlace scanning also shows some well-known artifacts, especially with moving pictures:

- It is impossible to combine two fields to one picture for moving objects, as each field comes from a different moment in time. This is a major drawback for creating still pictures from a moving scene, and for video to film transfer.
- While, with proper filtering, the frame can be nearly free of aliasing, each field may contain aliasing since it has only half of the samples in the vertical direction. The human eye has to integrate out aliasing effects per field, to see the full frame resolution of the picture. This results in small area flickering at field rate. The cancelling of aliasing between fields only holds for still pictures.
- Even at slow vertical movement of one line per field vertical aliasing is dramatically increased. This is especially visible on slowly moving almost horizontal lines in the picture.

With the move to digital television, the performance of the compression system becomes critical. Compression of interlaced signals is more complex and performs worse than compression of progressive scanned signals, where the entire image is sampled at the same time.

The 750-line 1280 x 720 format progressive scan standard provides a good, practical solution to the problems of interlacing while obtaining excellent compatibility with interlaced HDTV [4]. The key characteristics of this standard are given in Table 1 [5].

Several features of this standard greatly add to its practicality. Firstly, the picture format uses exactly 2/3's the horizontal and vertical pixel counts of the interlaced standard for ease in resampling. Next, the field and pixel frequencies are identical allowing the use of the same production equipment. In addition the line time is 3/4's that of interlaced potentially easing analog delay designs. Finally, adequate horizontal and vertical retrace intervals are allowed.

CCD SENSOR

The CCD sensor was designed specifically for progressive scan high definition video applications. With square pixels, an active array of 1280 x 720 pixels (1296 x 730 total pixels), a 16 mm diagonal for use with 1" format lenses, and with 60 frames/second operating speed, this sensor is ideal for the proposed 750/60/1:1 progressive-scan HDTV standard (Table 2). The frame-transfer architecture used provides high sensitivity, high fill-factor, no lag, and no smear when used with a mechanical shutter wheel as in the LDK 9000. The imager (shown in Figure 1) consists of the imaging array with both active and dark reference pixels, a full resolution storage section, a dual-channel horizontal register, and two output buffers.

The imaging pixel (Figure 2) is a 3-phase buried channel

Parameter	Value
CCD-type	FT
Optical format	1 inch
Image diagonal	16 mm
Image area width	14.00 mm
Image area height	7.88 mm
Number of lines	730
Pixels/line	1296
Pixel width	10.8 μm
Pixel height	10.8 μm
Chip width	15.29 mm
Chip height	15.25 mm
Chip area	233 sq. mm
Output registers	2
Pixel output rate	74.25 MHz
Frequency H-clocks	37.125 MHz
Swing H-clock	5 V
Frequency V-clocks	2.475 MHz
Swing V-clock	10 V

Table 2. CCD characteristics

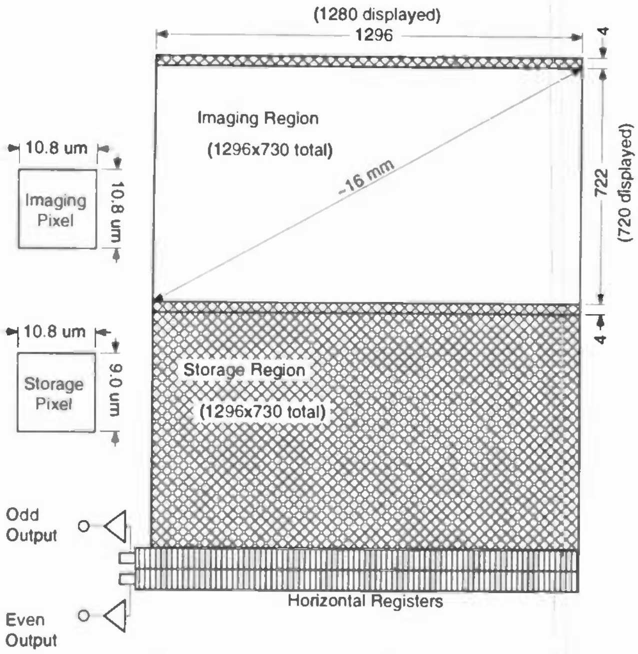

Figure 1. CCD Block Diagram

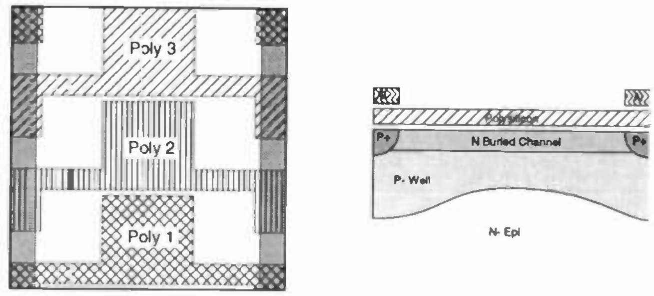

Figure 2. Imaging pixel top view (left) and cross-section (right).

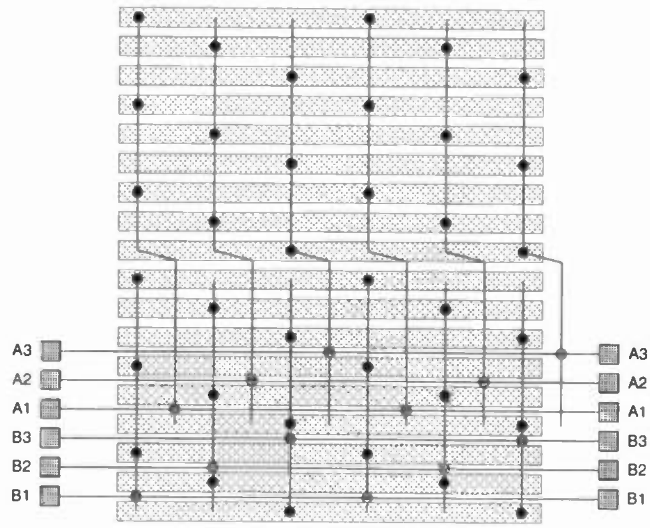

Figure 3. Double-level metal clock interconnect

device with integrated vertical anti-blooming protection. It is 10.8 x 10.8 microns square. The device is formed with three polysilicon layers, one for each phase, with large open areas (>33% of pixel) for enhanced blue light sensitivity. The vertical N-type buried channels are separated by P+ channel stops. The P-Well doping is modulated to form a weak spot in the center of the channel that acts as the anti-blooming barrier, which turns on when the pixel fills up to drain excess photocurrent down into the lightly N- doped epi layer. The storage pixel is configured similarly, although it uses wider poly gates for greater charge storage density. Thus the storage pixel could be made smaller (9.0x10.8 microns).

Since this sensor was designed to be compatible with the already existing camera, the process was carefully adjusted to give proper operation at the supplied clock voltages. This was complicated by the large number of functions that the pixel must implement: light absorption, charge collection, vertical overflow drain, charge transport, and charge reset (frame clear) for exposure control.

High vertical transport shift frequency (2.475 MHz) is required to move the charge from image to storage section during the brief optical blanking period provided by the shutter wheel. This frequency is by necessity higher than that used in the interlaced design because there are more lines to move (730/frame versus 576/field). Two-level aluminum wiring was used to shunt the polysilicon gate resistances resulting in less than 1 ohm equivalent series resistance. This allows the roughly 6 nf capacitive load to be driven at the required speed. Narrow aluminum straps connecting the poly gates run over the channel stops in the imaging section so that they have minimal impact on light sensitivity. Care was taken in their design to ensure that no fixed pattern artifact such as stripes was introduced by the straps. These are connected by busses on the second aluminum layer running across the storage section as shown in Figure 3. These busses are tied to package leads at each end to further lower series resistance.

During readout the charge is shifted one line at a time from the storage section into the horizontal registers. These charge packets are split up into the two registers on an even-odd column basis, using just a single transfer gate. The two registers operate in parallel at one-half the pixel frequency (37.125 MHz). The charge is transported along the 4-phase buried channel register to the matching sense nodes. The camera provides 4 phase clocking with 4-5 volt amplitudes, which posed a greater challenge in this design because the square pixels are wider and hence the horizontal gate length is greater reducing the fringing fields which assist charge movement. Extensive simulation was performed to ensure that excellent horizontal charge transfer would be achieved even at high clock frequencies.

Parameter	Value
Sensitivity (at sense node)	14 uv/e-
Amplifier gain	0.4
Noise after DLP in 30 MHz	33 e-
Bandwidth output	150 MHz
Quantum efficiency (peak)	26%
Sensitivity with BG40	2150 e-/lux
Overexposure	100,000 X
Full well capacity	40K e-
Dynamic range	62 db
Sampling frequency vertical	92.6 line-pairs/mm
Sampling frequency horiz.	92.6 line-pairs/mm
Image lag	none
Smear (incl. camera)	none

Table 3. CCD Performance

The output buffers are fairly conventional three-stage source-follower design. All three drive transistors are surface channel, giving high transconductance for low noise operation. The bandwidth (>120 MHz) is high enough to ensure accurate signal transmission. The layout of the two buffers was arranged to ensure they would match even with layer-to-layer misalignment during fabrication. The two video output signals are combined in the video pre-processor using the delay line principle (DLP).

Measured performance of the initial samples of the CCD sensor is summarized in Table 3.

OPTO-MECHANICAL DESIGN

The opto-mechanical system of the 1" CCD HDTV camera is designed to use lenses with a maximum aperture of f/1.2. The system consists of (from front to back): seal glass, IR-filter, retardation plate, shutter wheel, two 4 position filter wheels (for effect and ND filters), beam-splitter, optical low pass filters and sensors. In the adaptation of the optical system to the progressive format, in addition to the sensors only the optical low pass filter was changed.

Modulation Transfer Function

The modulation transfer function (MTF) of the camera is determined by the lens, optical low-pass filter, aperture of the image cell, and the electrical sample-and-hold. The MTF of the lens at f/4 is mainly diffraction limited. The MTF of the optical low pass filter is cosine shaped. The aperture of the image cell and the sample and hold both have sin(x)/x characteristics. Based on this model one expects a MTF of 47% for a sine wave at 27 MHz, versus a measured value of 50% (Figure 4).

Aliasing

A CCD-camera is a two-dimensional spatial sampler. The kings of fashion do not care about Nyquist nor does Nature! Therefore in everyday life the Nyquist condition - that the maximum frequency of the optical signal must be below half the sampling frequency - will be violated. This will cause Moire, or aliasing, patterns, which will create low-frequency patterns the eye is very sensitive to.

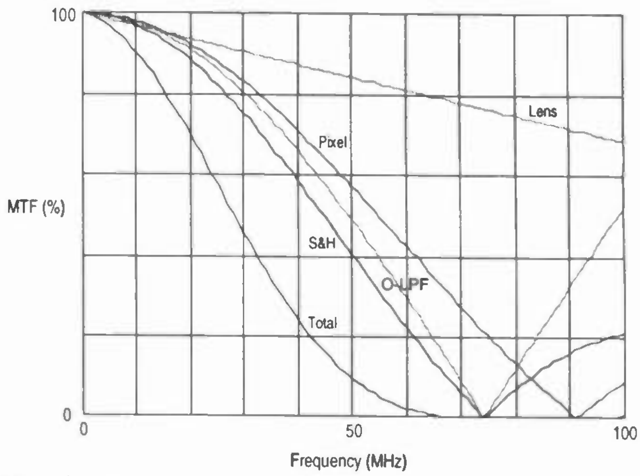

Figure 4. The modulation transfer function (MTF) of the camera. Shown are the separate contributions of lens, optical low-pass filter (O-LPF), aperature of the pixel, and the sample-and-hold.

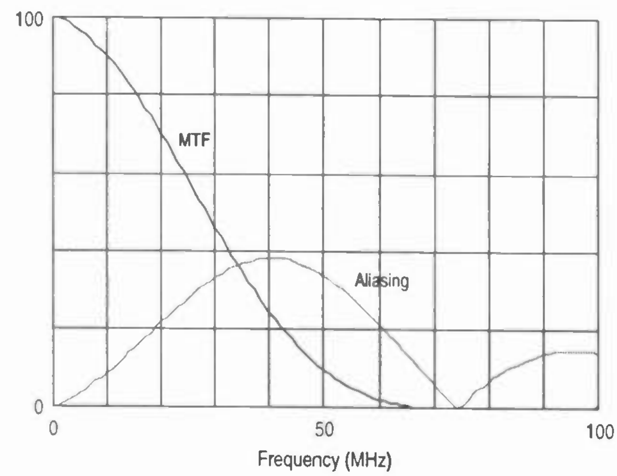

Figure 5. Shown are the MTF of the camera-head and the residual aliasing due to folds at the pixel sample frequency of 74.25 MHz

The frame-transfer image cell has a large aperture and therefore has intrinsically good horizontal and vertical aliasing behavior for higher spatial frequencies (greater than the Nyquist frequency). An optical low-pass filter helps to reduce aliasing further, especially at lower frequencies, by introducing dips (or notches) in the MTF. These dips must be at the vertical sampling frequency (92.6 line-pairs/mm) and at the horizontal sampling frequency (74.25 MHz, or 92.6 l-p/mm) for maximum effect (Figure 5). The need for a vertical anti-alias filter is unique to this progressive scan camera, since interlaced CCD sensors typically have considerable overlap in the even and odd scanning apertures that performs a similar function at the cost of vertical resolution.

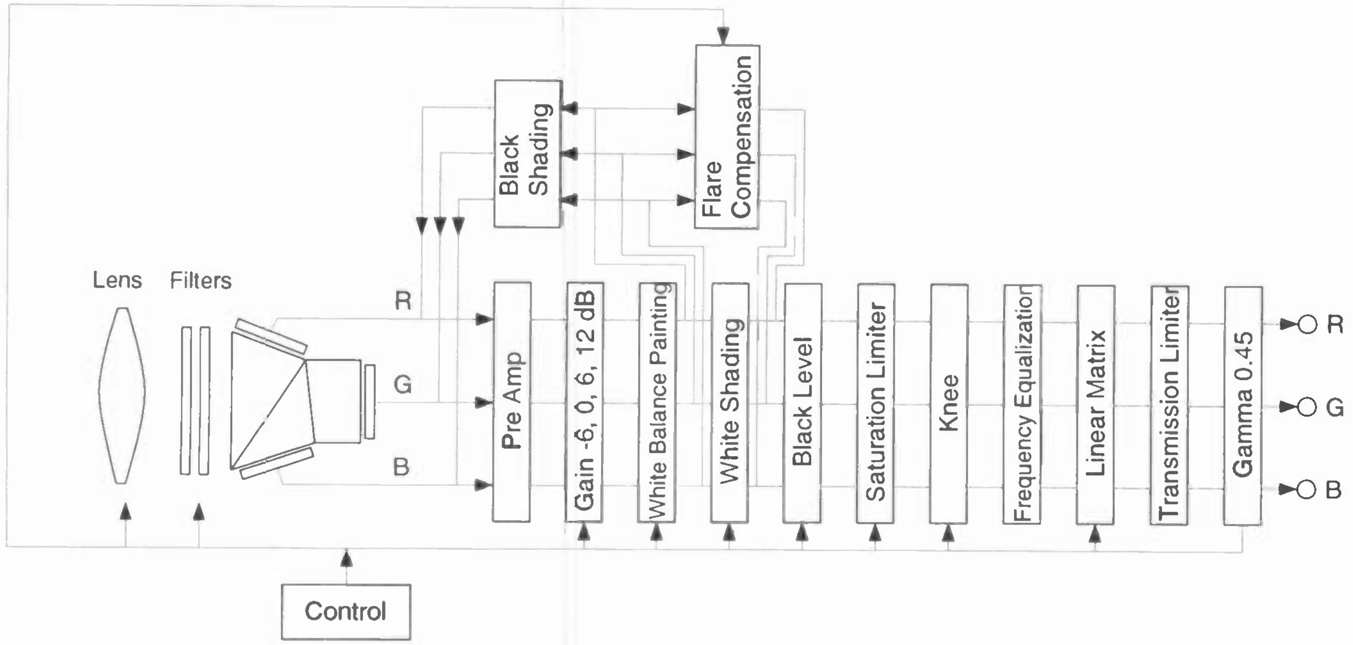

Figure 6. Camera head video processing.

VIDEO PROCESSING INCLUDING CONTOURS

Video processing in a progressive scan camera is not very different from the processing in an interlaced camera. The processing of the LDK 9000 camera has already been described in an earlier paper [1]. It consists of a part in the camera head (Figure 6), and further processing in the camera processing unit (CPU) (Figure 7).

Important design objectives for the LDK 9000 video processing were:

- Gain control over full temperature range.
- High dynamic range.
- Headroom before highlight compression of more than 14 dB.
- Signal/Noise deterioration due to video processing less than 1 dB.
- High quality, reliability and operational flexibility.
- Low power consumption.

Operation following the 1280 x 720 progressive standard calls for some specific adaptations as compared to the 1920 x 1080 interlaced standard:

- Line time is changed from 29.6 usec. to 22.2 usec. This calls for different line delays in the contour delay unit.
- Active line time is changed from 25.8 usec. to 17.2 usec. This calls for a more accurate timing in the video processing as timing errors will be more visible on the display.

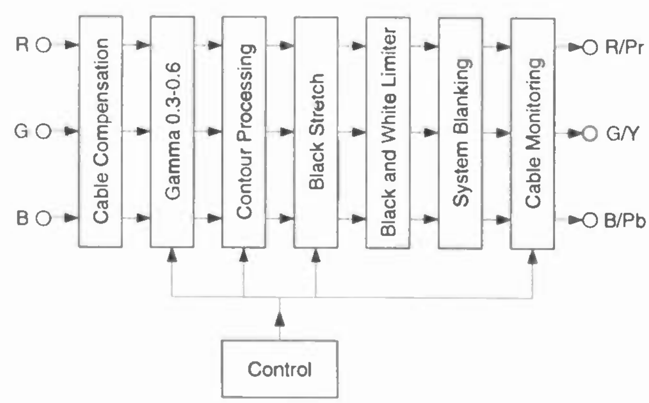

Figure 7. Video Processing CPU

- Vertical contours will look different -- the vertical contour generation in a 1080 line system is field based, with the 0T, 1T and 2T lines 1/540 picture height apart. In a 720 line progressive system vertical contours are generated from lines with a spacing of 1/720 picture height. This results in a higher vertical peaking frequency for vertical contours in a 720 progressive system.
- Horizontal contours will have a lower spatial frequency peak in the 1280 x 720 progressive scan system. This can be changed by shortening the delay lines in the contour processor, but there are practical limitations imposed by the lower Nyquist frequency of the 1280 x 720 system.

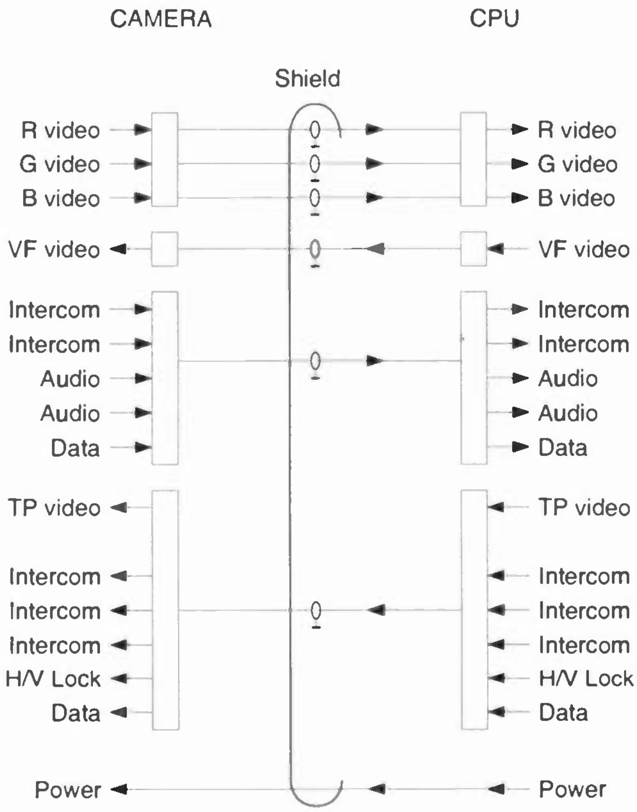

Figure 8. Multicore Transmission system

SIGNAL TRANSMISSION

The connection between camera and processing unit is formed by an interconnection system of cable and electronics specifically designed to maintain signal quality. A multicore cable can be used for short distances up to 300 meters. For longer distances, the multicore cable can be extended with a fiber optic system. Four coaxial cables are required for the R, G, B and view finder video signals. The remaining signals (figure 8) could be multiplexed into a single bidirectional coaxial cable, or into two separate single directional cables. This latter approach is used to achieve a simple interface with an optical fiber. Additionally, power wires are added, yielding a custom-made circular cross-section multicore cable.

The electronics provides automatic compensation for all multicore cable lengths between 0 and 300 meters. This is realized by dividing the total compensation into a fixed part and an adaptive part. The fixed part can compensate any cable length within an increment of 12.5 m. The compensation determined at power-up, by means of a successive approximation measurement. The adaptive part, which is independent in each channel and continuously active, has two functions:

- It has to compensate the last residual cable length within the resolution of the fixed part,
- It has to compensate (the frequency dependant) loss differences as caused by such things as temperature changes of the multicore cable and differences between the individual coaxial lines.

Delay differences between the coaxial lines is kept small by using high quality coax: maximum 1.5 ns between R, G, and B video signals at 300 meter cable length.

7 INCH VIEW FINDER

The main challenge in adapting the view finder to the progressive scan 1280 x 720 format was operating at much higher line frequency (45 KHz) given limitations on power dissipation and demands for high brightness and contrast. A stable high voltage source is required to prevent "breathing" at high beam currents and to secure high resolution performance.

Although spatial frequencies are lower for 1280 x 720 than for the 1920 x 1080 system (27 MHz bandwidth gives 780 TVL for 1920 x 1080 versus 520 TVL for 1280 x 720), focus assist is still a valuable tool for the camera operator. Apart from peaking in the view finder, the HDTV camera system is provided with two focus assisting tools:

- Magnifier: Momentary activation of this function enlarges the center part of the image by approximately 1.6 times, filling the whole screen.
- Crawler: Small details in the picture are converted to a more coarse structure, which gives edges and other fine details a highly visible crawling pattern. Optimum focus is obtained when this crawling serration reaches the maximum intensity. It acts more or less like "peaking" and can be used continuously.

CAMERA SYSTEM

The camera being presented is part of a complete system configured for broadcast applications. The system, as modified for the progressive standard, consists of the following system components:

Camera head
7-inch view finder
Camera Processing Unit (CPU)
Multicore cable
Master Control Panel (MCP)
Operational Control Panel (OCP)
Lens
Accessories

The camera head has been designed as a compact, lightweight, modular unit (Figure 9). The camera features two four-position filter wheels with three neutral density filters and two special effect filters.

The 7-inch view finder can be mounted on a optional specially-designed support above the camera. The support is designed to accept the lightweight camera combination. The camera, with an optional 1.5-inch view finder, can be

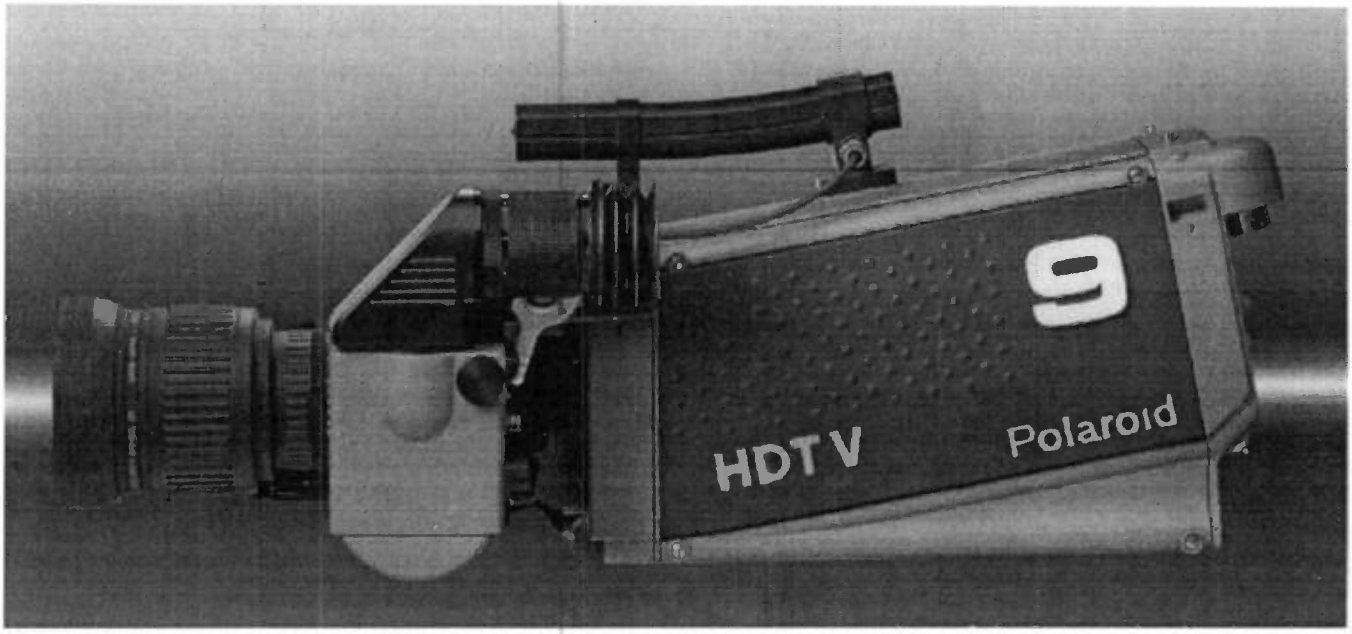

Figure 9 Camera head

easily placed upon or taken out of this support, leaving the support and 7-inch view finder on the tripod.

The CPU is a 19 inch rack mountable component which is 3 standard units high. The device is constructed using standard Eurocard PC boards and a rear connector panel with all signal interconnection options commonly used in broadcast studios.

The control panels follow the Series 9000 control philosophy, as used with all BTS standard TV cameras. The Master Control Panel gives access to most of the set-up controls via menus. The Operational Control Panel (OCP) provides all the operational control functions of the HDTV camera. The OCP is arranged with user-friendly directly accessible controls.

The camera can be used with a wide range of lenses built with internationally standardized interfaces. The camera presented is equipped with an 11 x 11 barrel-type lens from Fujinon. This heavier barrel-type zoom lens is supported by standard film-style accessories: a bridge plate underneath the camera accepts support rods, lens supports, matte boxes, etc.

The main camera system characteristics are summarized in Table 4. Performance specifications are summarized in Table 5.

For picture evaluation during the development period a Barco color monitor, Reference Calibrator model 121, was used. This monitor is capable of displaying 1280 x 720 progressive signals without effecting picture quality. Noise measurements were done on the Rohde & Schwarz VNA (Video Noise Analyzer). The Tektronix 1730HD Waveform monitor was used.

- An electronic white balance range from 2500K to 15000K.
- Highlight compression in automatic and manual mode.
- Black stretch in Y and R,G,B.
- Colorimetry according to EUREKA/EBU standard.
- 2-Dimensional contours.
- Electronic shutter with 5 and 2 msec exposure time. Also 50 Hz and 60 Hz lighting positions are available.
- Camera power consumption approximately 22 W.

Table 4. Main camera system characteristics

- Modulation Transfer Function of over 40% at 520 TVL (27 MHz) without contours.
- Limiting horizontal resolution of 700 TVL.
- Sensitivity of 1200 Lux at F/4.
- S/N ratio of 50 dB at a bandwidth of 30 MHz.
- The max. lens aperture is F/1.2.
- Dimensions approx. 140 x 210 x 350 mm.

Table 5. Camera system specifications

CONCLUSION

We report here on the first CCD HDTV broadcast camera to demonstrate the newly recommended 1280 x 720 progressive standard (750/60/1:1). A frame-transfer CCD was custom-designed, and a broadcast-quality interlaced HDTV camera was modified to meet the progressive standard.

A prototype camera has been built and is demonstrated at the 1996 NAB Exhibition. This camera meets all specifications as presented in this paper, and meets all goals toward proving the feasibility of the 750-line progressive HDTV standard.

The LDK 9000 system, with the 1250/50/2:1 standard, has already been in use for 4 years in Europe. During this time these systems have been used, to complete satisfaction, at a wide variety of events. A modified system for the new American 1920 x 1080 production standard (1125/60/2:1) was used for the production of test material by the Advanced Television Test Center. It is anticipated that this proven record of reliability and success will carry over to the progressive scan camera.

This project represents a successful embodiment of collaboration between industry, university, and research laboratory to accomplish more in a shorter period of time than any one could do alone.

ACKNOWLEDGMENTS

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525-LINE PROGRESSIVE SCAN SIGNAL DIGITAL INTERFACE STANDARD

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Abstract:

The 525-line progressive scan(525P) is mainly used for input signals of EDTV-II in Japan, which is an NTSC-compatible letter box transmission scheme. This signal is also considered to be one of the signal formats for an SDTV(Standard Definition Television) signal format in the US.

Two signal interface formats were proposed through ARIB (Association of Radio Industries and Businesses, previously BTA) in Japan and are being proposed to SMPTE in the US.

1)4:2:2P:

2)4:2:0P:

These interfaces are based on our study for a 525P signal format. This paper describes the reasons for proposing these interfaces as well as their advantages.

1. What is 525-line Progressive Scan?

525P is a type of progressive (noninterlaced) scan format, as used in computer displays. Basically, the scanning speed is exactly double that of the conventional 525-line interlaced scan(525i). Therefore, it has good compatibility with the conventional 525i.

This signal is used for the signal source as follows:

(1)EDTV-II signal input source:

EDTV-II is an NTSC compatible letterbox coding scheme that was started in July 1995 in Japan. EDTV-II is quite similar to PALplus in Europe, but PALplus is based on an interlaced signal source, while EDTV-II is based on a progressive scan signal. 525P is an input signal source to the EDTV-II encoder.

(2) SDTV signal source:

525P is one kind of signal format in SDTV(Standard Definition TV). SDTV is a type of digital TV broadcasting(ATV) in the US proposed by the FCC ACATS(Advisory Committee on Advanced Television Service).

(3) Digital Satellite Broadcasting signal source in Japan:

In Japan, the signal format used for digital satellite broadcasting is 525i or 525P. This is decided by the Ministry of Posts and Telecommunications in Japan.

2. The advantages of 525P:

The advantages of 525P are as follows:

1. Good resolution in moving pictures
2. Ease of scanning format conversion
3. Compatibility with 525 interlaced system
4. Compatibility with computer system
5. Good cost performance
6. High-quality source for the interlace signal

These advantages are explained in the paper in reference 1.

3. 525P Serial Digital Interface Standard

The digital interface standard for 525P has been proposed for SMPTE[2] in the US and ARIB[3] in Japan. Two types of digital interfaces have been proposed for the 525P interface :

1) 4:2:2P
2) 4:2:0P

These interfaces are described as follows:

3.1 4:2:2P:

The color difference data can be reduced to half the data rate of the luminance by reducing the horizontal sampling points of the color difference signal into half those of the luminance signal. This is quite common, the same with 525i or 1125 interlaced. With 4:2:2P, the luminance sampling is 27 MHz and the color difference sampling is 13.5 MHz, which is exactly twice as fast as the sampling rate of the 525 interlaced sampling(4:2:2) described in SMPTE 125M because the scanning speed is exactly double that of 525i.

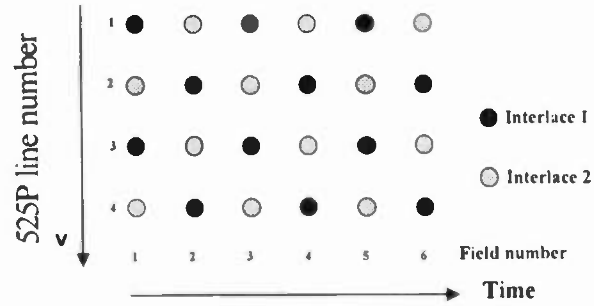


Figure 1 4:2:2P Sampling Structure

The progressive signal can be divided into two interlaced signals, as shown in Figure 1.

Therefore, two links of interlaced 4:2:2 can interface with the progressive signal (270 Mbps x 2). Although it is rather troublesome to connect two cables between equipment, this interface has an advantage in that existing conventional 4:2:2 interlaced equipment or devices can be used to process the progressive signal. This is a great benefit in terms of system construction, especially in the early stage of development, when not every sort of progressive equipment has been developed yet and is therefore currently unavailable on the market.

Another advantage of this dual-link 4:2:2P system is the switchable function. The interlaced and progressive mode switchability is easily performed with this interface. The single use of the 4:2:2 interface can process the interlaced signal, and dual use of the 4:2:2 can process the progressive.

3.2 4:2:0P:

As described in the previous section, 4:2:2P has the disadvantage of two cables being required to connect the equipment.

Therefore, a single-link connection 4:2:0P is also proposed. The 4:2:0P signal is generated as follows:

The color difference signal of the dual-link 4:2:2P is vertically filtered and sub-sampled into half lines as the interlaced structure. Therefore the luminance signal is a 27 MHz sampling, and the color difference signal is a 6.75 MHz sampling in the case of 4:2:0P. The sampling structure of the 4:2:0P is shown in Figure 2.

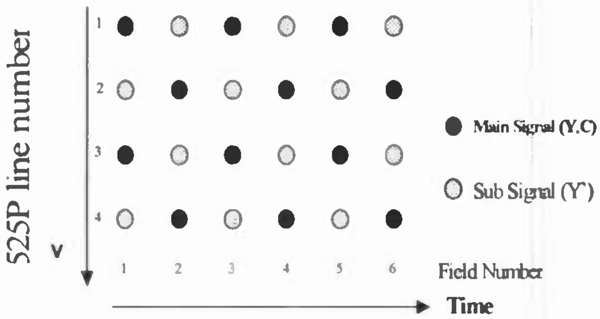


Figure 2 4:2:0P Sampling Structure

The serial data rate of this 4:2:0p was originally supposed to be:

27 MHz x 10 bit + (6.75 MHz x 10 bit) x 2 = 405 Mbps

This data speed is unique, and devices operated at this speed would have to be newly developed. Considering the development cost and time, this data speed is therefore reduced to 360 Mbps without discarding the active image data. This is achieved by shrinking the horizontal blanking area. The 360 Mbps data rate is originated for the 18 MHz sampling version of 4:2:2 525i interface as defined in SMPTE 259M. This data structure difference between the conventional 18 Mbps sampling 4:2:2 serial digital interface for 525i and 4:2:0P digital interface is shown in Figure 3.

The HANC (horizontal ancillary data) area is compressed as shown in the figure.

This reduction of HANC area is an abnormal process, but it has the advantage that the conventional ICs and equipment developed for 360 Mbps can be used. This blanking area is used for ancillary data, such as embedded

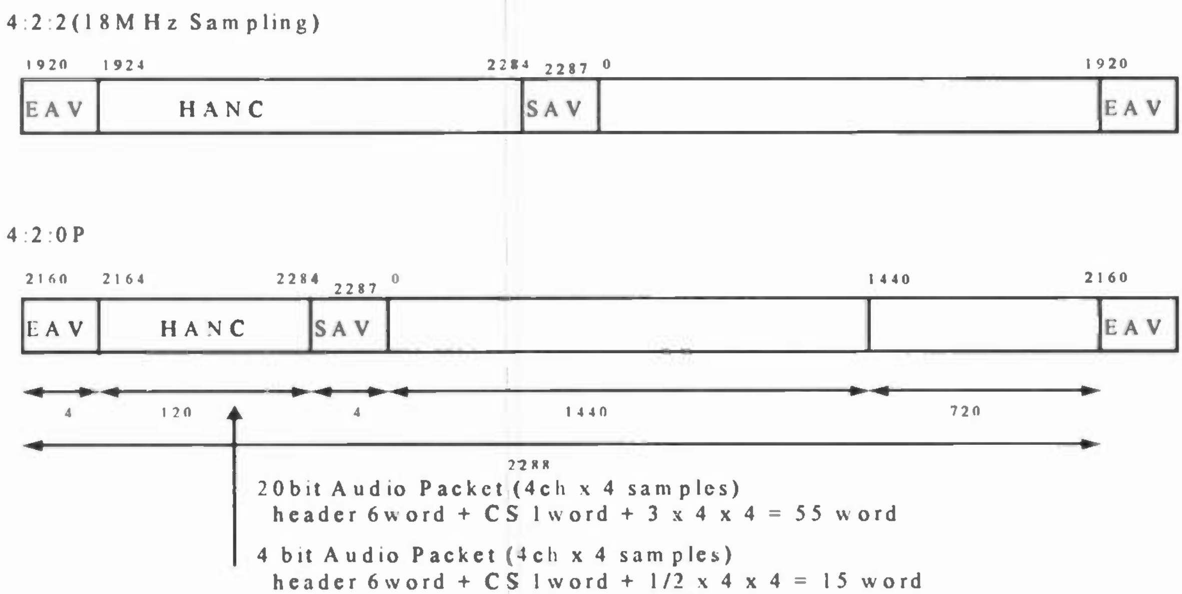


Figure 3 The Data Structure difference between 18 MHz sampling 4:2:2 and 4:2:0P

audio. Even with the 4:2:0P, 8 channel, 20 bit audio data can be installed in the HANC.

The serial data structure of 4:2:0P is shown in Figure 4.

4. The study before finalizing 4:2:0P

Several signal formats were studied to minimize the data rate of the progressive signal without degrading the image quality (based on 4:2:2 interlaced plus progressive helper signal) before the 4:2:0P signal format was finalized. They are:

1) 4:2:2+4

2) 4:2:2+2

3) 4:1:1+2

where 4 means 13.5 MHz sampling information , 2 means 6.75 MHz, 1 means 3.375 MHz sampling information.

4.1 4:2:2+4:

In this case, the helper information is the 13.5 MHz sampling rate. It is natural to assign this helper signal to the luminance signal because it is the simplest data format. Therefore, total luminance information is the 27 MHz sampling rate (13.5 MHz + 13.5 MHz). 13.5MHz is the sampling frequency of 525i(4:2:2). This provides the same horizontal luminance sampling number as 525i(4:2:2), which is 720 samples per active line, because the scanning frequency of 525P is twice that of 525i and 27MHz is twice 13.5MHz.

As for the color difference signal, it provides better image quality to decrease the spatial resolution vertically as well as horizontally rather than decreasing only horizontally. Therefore the sampling rate is horizontally reduced to half of the luminance data, and also vertically reduced into half of the luminance signal.

Several vertical sub-sampling structures were studied, such as an interlaced sampling structure, every two lines (cosited, the same lines in every field), and sampling between lines like MPEG 4:2:0. The interlaced structure is employed because of the following reasons:

(1) Compatibility with interlaced system

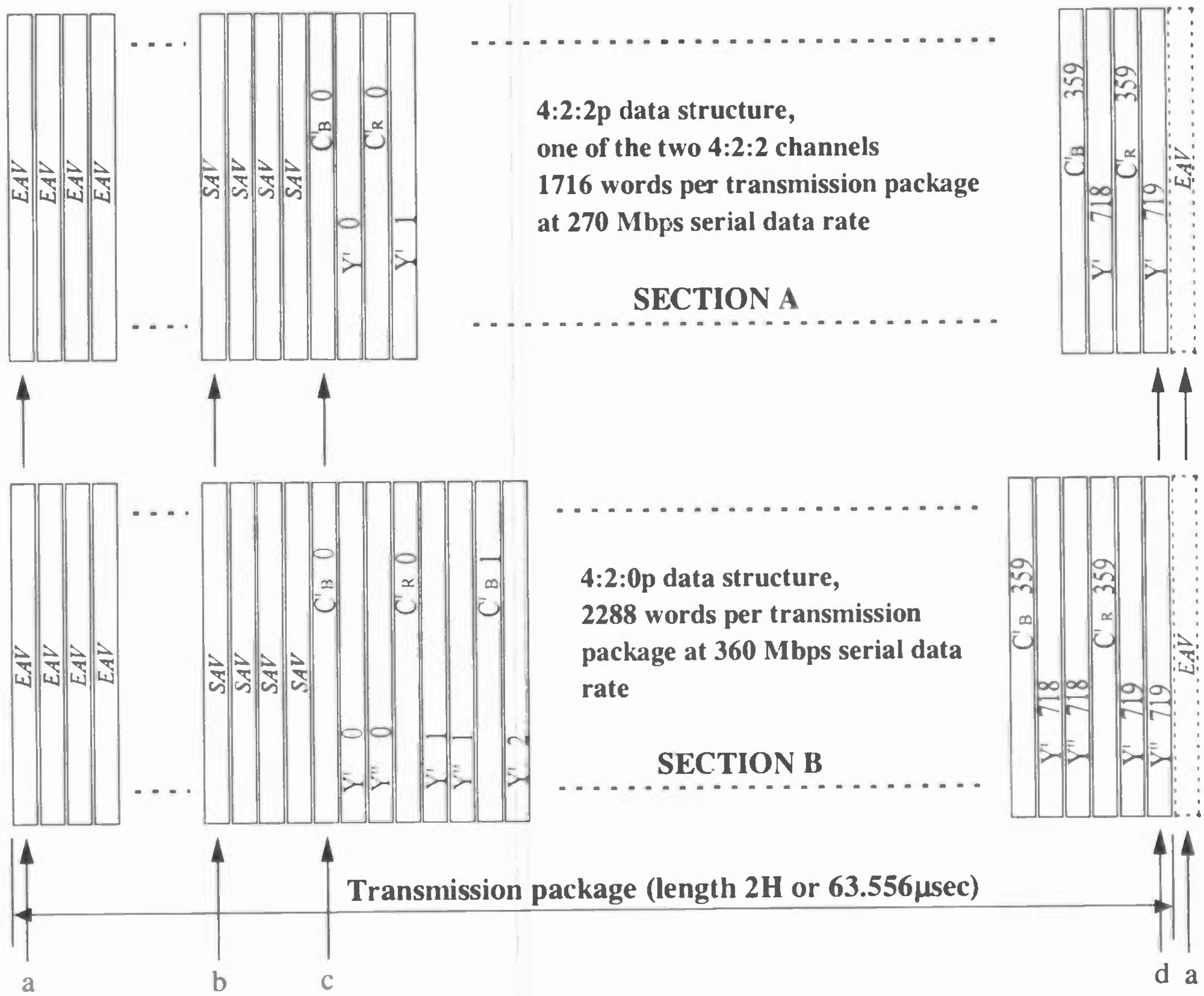
(2) Higher vertical color resolution in still pictures

In summary, in the case of 4:2:2+4, the luminance sampling frequency is 27 MHz and the color difference sampling frequency is 6.75 MHz, which is the same as the sampling structure of 525i. This is the origin of 4:2:0P.

The subjective assessment result for 4:2:0P versus 4:2:2P(mean opinion score of 4:2:0P-4:2:2P) using the 5-level double stimulus continuous quality scale method is shown in Figure 5. This shows 4:2:0P provides no image quality degradation.

4.2 4:2:2+2, 4:1:1+2:

Further study was carried out to reduce the data rate even more without degrading the image quality. Based on the 4:2:2 interlace structure, an additional helper signal for the progressive signal was studied.



			Word number			
			a	b	c	d
System	Total words per transmision package	Total words of active image data per transmission package	Start of EAV	Start of SAV	Start of active image data	End of active image data
360 Mbps 4:2:0p	2288	2160	2160	2284	0	2159
270 Mbps 4:2:2p	1716	1440	1440	1712	0	1439

Figure 4: - Multiplexed horizontal data stream

Note1: The time scales for the 4:2:0p and 4:2:2p systems are different.

Note2: Color difference samples are cosited with even numbered luminance samples and take the half value of the associated luminance sample number.

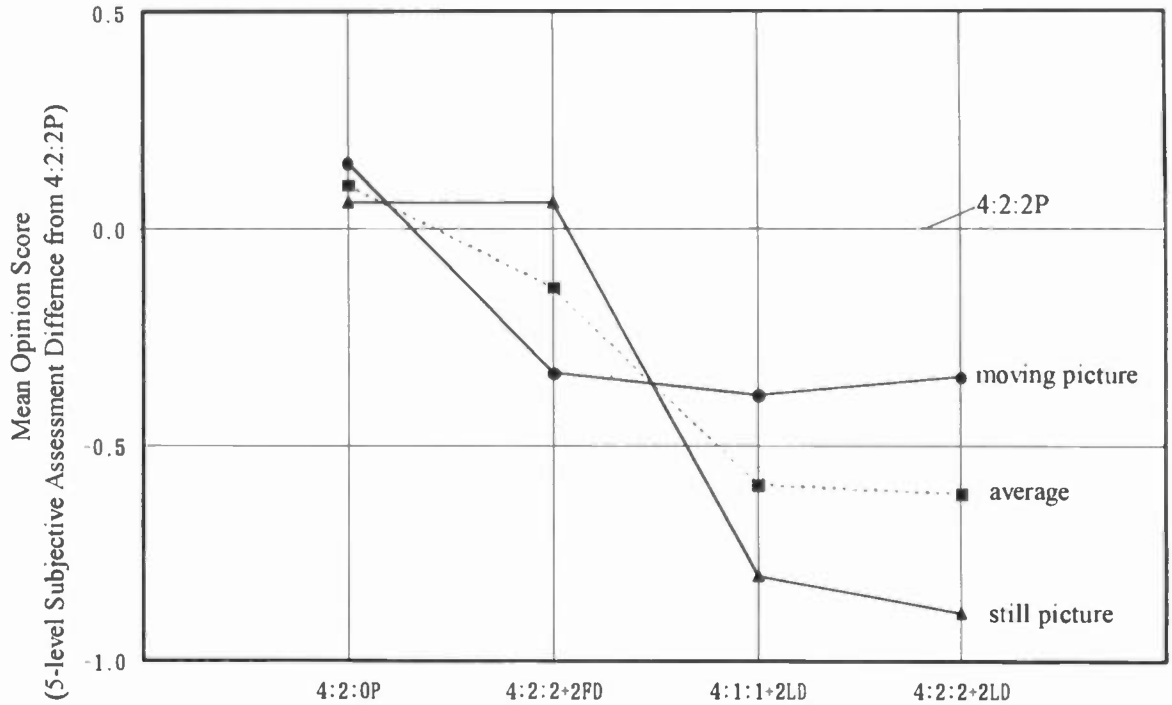


Figure 5 Subjective Assessment Result for 4:2:0P, 4:2:2+2 and 4:1:1+2 versus 4:2:2P

Luminance signal:

Considering correlation in the data from nearby lines, it would provide better image quality to assign this helper signal as the data difference from neighboring lines. The field difference information (FD) from the same line in the previous field (shown in Figure 6), and the line difference information (LD) from the upper line (shown in the Figure 7), are assigned to the helper signal. The sampling frequency of this additional helper signal is 6.75 MHz, which is represented as "2" (2 of 4:2:2+2).

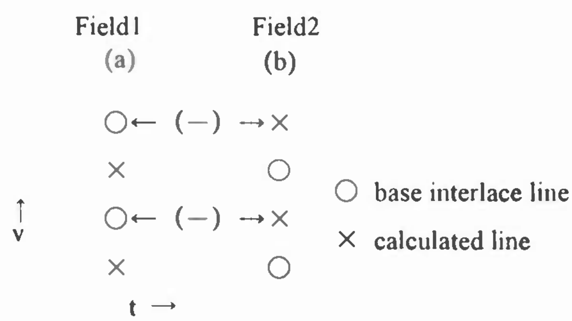


Figure 6 Field difference signal FD(a-b)

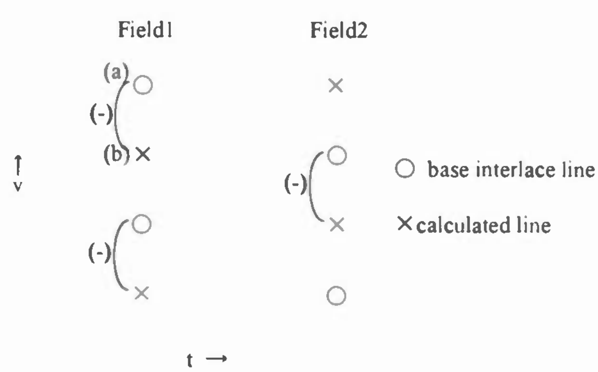


Figure 7 Line difference signal LD (a-b)

The subjective assessment result for 4:2:2+2FD and 4:2:2+2LD, (versus 4:2:2P) using the double stimulus continuous quality scale method is shown in Figure 5 .

Color difference signal:

Furthermore, the color difference signal was reduced to a 3.375 MHz sampling rate, which is represented as "1"(1 of 4:1:1+2).

The subjective assessment result for 4:1:1+2LD versus 4:2:2P using the double stimulus continuous quality scale method is also shown in Figure 5.

The subjective assessment result shows that there was some degradation in the 4:2:2+2 and 4:1:1+2 formats. Therefore, these formats were not used for the studio signal standard.

5. Conversion between 4:2:2P and 4:2:0P

A conversion between 4:2:2P and 4:2:0P may be necessary.

5.1 Conversion from 4:2:2P to 4:2:0P:

In the case of 4:2:0P, the vertical color difference sampling point is half that of the 4:2:2P. Therefore, a vertical low-pass filter is necessary before subsampling the color difference signal of 4:2:2P to avoid alias. The ideal vertical low-pass filter for this is attenuation 0 in the passband(≤ 240 lph) and the attenuation is infinity in the rejection band(>240 lph) without ringing. Actually this is quite difficult to achieve because of the complexity of the hardware, especially in the vertical filter. Therefore, an adaptive vertical filter is proposed to solve this problem. The structure of this filter is shown in Figure 8.

This adaptive filter is based on the idea that if there is a vertical high-frequency color difference component, it must be low-pass filtered. If there is no high-frequency component, it should not be filtered. The upper part of the filter coefficient [-1,2,-1]/4 is a high-pass filter, and the lower part of the filter coefficient [1,2,1]/4 is a low-pass filter. If the output of the high-pass filter is higher than the threshold level, which implies there is a high vertical frequency component, the signal must be filtered and should take the signal of the filtered path. If the output of the high-pass filter is lower than the threshold level, which implies high frequency component is little, the signal should not be filtered and should take the nonfiltered path.

Although this filtered structure is quite simple, the adaptive filter is quite effective, especially in the multiple conversion between 4:2:2P and 4:2:0P.

5.2 Conversion from 4:2:0P to 4:2:2P:

In this case, interpolation of the color difference signal is necessary to obtain 4:2:2P. One example of this filter is shown in Figure 9. This filter is designed to have no effect on the actual line(noninterpolated line). This is a great advantage in multiple conversions between 4:2:2P and 4:2:0P because it minimizes conversion degradation.

6. Conclusion:

This standard has already been finalized in Japan as BTA T-1004. In SMPTE's case, this standard was discussed in Working Group S17.39(Advanced Television Production). It will be published as proposed SMPTE

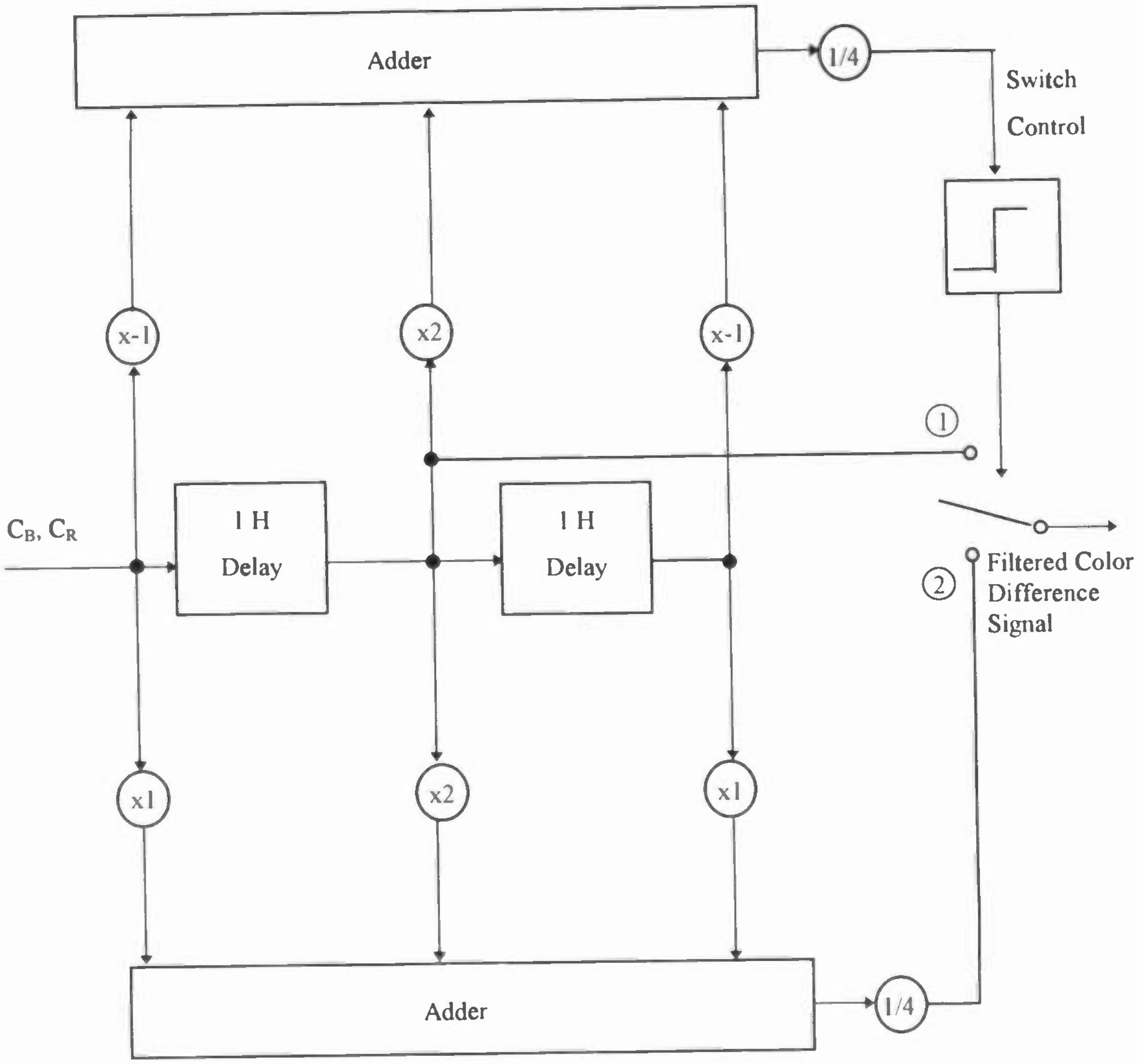


Figure 8 An example demonstrating an adaptive filter used for the color difference components before subsampling the 8:4:4 data into a 4:2:0p quincunx signal.

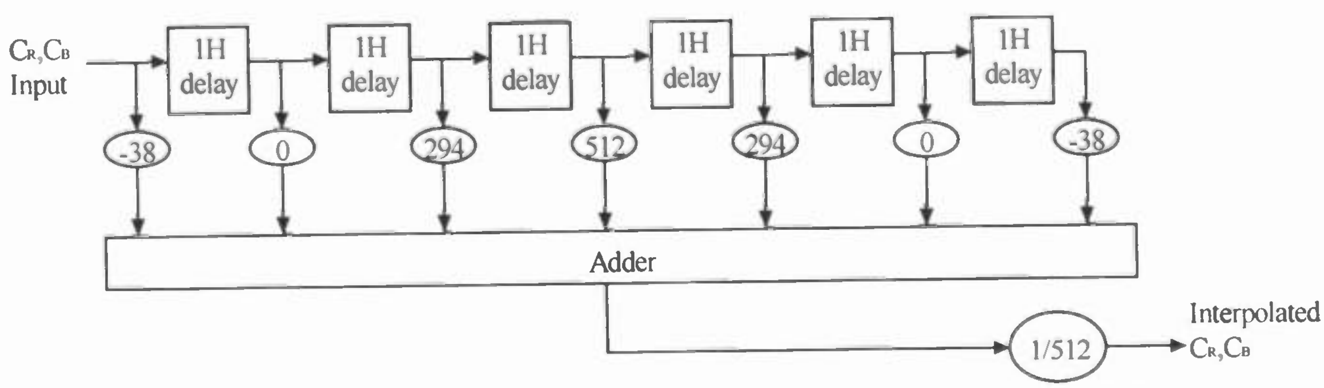


Figure 9 An example of color difference interpolation filter from 4:2:0p to 4:2:2p

standard 293M (Digital Representation) and 294M (Bit Serial Interfaces) in the April issue of SMPTE Journal. Finalization of the standards will be completed in the middle of this year.

I hope these interfaces will be utilized and that many 525P equipment components will be developed in the near future.

References:

1. Akihiro Hori, "525 Component Progressive Scan OB Van System," NAB 1995 Broadcast Engineering Conference Proceedings pp 353-359

2. Proposed SMPTE Standard for Television-

   293M:

   720 x 483 Active Lines at 59.94 Hz Progressive Scan Production - Digital Representation

   294M:

   720 x 483 Active Lines at 59.94 Hz Progressive Scan Production - Bit Serial Interfaces

3. BTA report T-1004: Video Signal Interfaces for EDTV-II Studio Equipment

DIGITAL AUDIO IMPLEMENTATION

Monday, April 15, 1996
10:30 am - 12:00 pm

Session Chairperson:
Dutch Doelitzsch, WDDD, Marion, IL

PICK AND MIX
APT-X AND APT-Q
Fred Wylie
Audio Processing Technology
Belfast, Ireland

AUTOMATION: THE JOCK'S BEST FRIEND
James Thomason
Harris Broadcast Equipment
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PICK AND MIX...APT-X AND APT-Q

Fred Wylie
Audio Processing Technology
Belfast, Ireland

Abstract

apt-X100 introduced in 1989 is a well established and popular algorithm. It has a fixed ratio of compression set at 4:1 and offers a range of audio bandwidths from 7kHz mono at 56kbit/s up to 22kHz stereo at 384kbit/s. An important feature of this algorithm is the extremely short processing delay, 2.5ms at the highest sampling frequency of 48kHz.

apt-Q, being introduced at NAB96 for the first time is a new scaleable algorithm and is offered with two practical levels of compression, nominally 12:1 and 18:1. It will deliver 15kHz stereo at 56kbit/s or 20kHz stereo at 128kbit/s, the processing delay being better than other transform algorithms.

This paper now examines in more detail how the apt-X and apt-Q compression algorithms achieve lower bit rates while sustaining high quality audio, mono or stereo.

Introduction

The benefits of representing an analogue audio signal by its digital equivalent have been recognised for decades. A digital signal is easier to regenerate at points along a telecommunications highway, it is easier to store and it is perfect for protection from transmission errors and the non-linearities associated with traditional analogue equipment.

However the bandwidth requirement of a digital bearer circuit is many magnitudes greater than for an analogue circuit and in storage terms requires many megabytes of disk space. For example a one hour stereo compact disc, a 16 bit linear PCM format would require 1.5MHz of digital circuit bandwidth for distribution of it's 1.411Mbit/s output compared with two analogue circuits at 20kHz each. It would also require something like 630MB of disk storage space. It is therefore easy to see that there are clear economical advantages for any system which would reduce this space hungry signal.

The basic requirement of any real time compression system is therefore to reduce the bit rate of linear PCM with a minimum of delay and perceived loss of quality, either from signal distortion or injected quantisation noise. The perception left with the listener is that it is a real time and transparent process. However, with such a wide range of possible applications, outside broadcasts and remotes, contribution and distribution, transmission and storage/post production, it is almost impossible for a single algorithm to to match every situation. At one end of the scale it may be a once only news grab or sports event where a more aggressive form of coding would suffice, whereas if the audio is to be subjected to further compression then the user needs to be more circumspect in the selection of an algorithm by using a more modest rate of compression.

Due to the repetitive nature of most complex sounds the linear PCM signal contains large amounts of redundancy and irrelevancy. Redundant sounds are quantifiable and can be removed during the coding process and replaced again in the decoder. Irrelevant sounds, those that the human ear cannot hear can also be identified and removed using a perceptual masking process which matches the response of the human ear. However, once removed they cannot be replaced. Another fact is that almost without exception the fundamental frequency content of most complex sounds rarely go above 4kHz. Taking advantage of these phenomena is the basis for compression.

Compression Techniques

Current compression technology requires the PCM signal to be delayed momentarily to enable an analysis to be carried out as to its level, frequency and energy content. Processing delay therefore will play a significant part in the choice of codec. In some telecommunications configurations any long processing delay would create unacceptable line echo effects in two way hook-ups, even on short propagation paths. Whereas it is not so critical for straightforward one way transfers of audio except maybe in some audio + pictures scenarios. Some codecs are therefore more suitable than others for two way or interactive applications.

There are two main techniques currently utilised in digital audio data compression; time domain prediction coding and frequency domain transform coding.

apt-X100

apt-X100 (see Fig) is a prediction or subband ADPCM (**A**daptive **D**ifferential **P**ulse **C**ode **M**odulation) algorithm. Differential coding reduces the bit rate by coding and transmitting only the difference between a predicted level for an audio sample and the absolute level of that sample, thus demonstrating the exploitation of the redundancy contained in the signal. It is claimed that apt-X100 with extremely accurate prediction loses as little as 2% of the original input audio in the process and coupled. With a processing delay of a minimum of 2.5ms at 48kHz sampling through to a maximum of 7.6ms at 16kHz this is for all intents and purposes a lossless and transparent process.

Adaptive differential coding is dependent on the energy of the input signal and alters the step size represented by each quantising interval. The adaption being controlled by an analysis of previous samples. In apt-X100 this equates to the so called backwards adaption process and involves the analysis of 122 previous samples.

Time domain subband algorithms implicitly model the hearing process and indirectly exploit a degree of irrelevancy by accepting that the human ear is less sensitive at higher frequencies. This is achieved in the subband derivative by allocating more bits to the lower frequency bands. This is the only application of psychoacoustics exercised in apt-X, all the information contained in the PCM signal is processed, audible or not, ie no attempt is made to remove irrelevant information. It is the unique allocation of bits to each of the four subbands coupled with the filtering characteristics of each listener's auditory system that achieves the satisfactory audible end result.

The apt-X100, hardware based, compression algorithm has a wide range of applications including storage, ISDN and other digital terrestrial and satellite telecommunications distribution circuits. This algorithm with four subbands and fixed 4:1 compression can deliver audio bandwidths ranging from single or dual/discrete channels at 7.5kHz up to similar configurations at 22.5kHz each. The user defined output bit rates are from 56 to 384kbit/s achieved by using various sampling frequencies from 16kHz to 48kHz. The allocation of bits for each of the four subbands is fixed and this generates a 16 bit word at the coder output which is representative of the content of 4 × 16 PCM bit words at the input, a bit rate reduction of 4:1.

apt-Q

apt-Q (see Fig) is an extremely robust and flexible compression algorithm. With a compression ratio of approx 18:1 a linear PCM signal at 1024kbit/s can be successfully reduced to 56kbit/s and still deliver 15kHz bandwidth stereo audio. Increasing the bit rate to 128 kbit/s with a compression ratio of 12:1 will deliver 20kHz stereo from a PCM signal sampled at 48kHz.

apt-Q employs the other compression technique mentioned, being a subband transform or perceptual algorithm transforming time blocks of the linear PCM signal into the frequency domain to enable the coding to take place. To achieve the necessary reductions in bit rate apt-Q exploits the large amounts of irrelevancy and redundancy contained in the linear PCM signal on two fronts. It employs source coding techniques to remove the redundancy and psychoacoustic masking techniques to identify and remove the irrelevancy content. These techniques, central to this form of transform coding take advantage of the phenomena of the human auditory system and its inability to distinguish between the wanted audio signal and accompanying noise which is in the same spectral and temporal plain.

The processor acts just like the automatic gain control in the ear, continually adjusting in response to the dynamics, at all frequencies, of the incoming audio signal. This phenomenon of the human auditory system is exploited in subband transform coders by using a simultaneous or frequency domain masking process (see Fig) to detect the differences between the perceptually critical audible sounds, the non-perceptually critical sounds and the quantisation noise already present in the signal and then to adjust the masking thresholds, performing to a preset perceptual model, to suit. This defines a operating point below which the quantisation noise unavoidably injected by the compression process cannot be perceived.

To allow for better frequency resolution and efficient redundancy extraction apt-Q uses a Modified Discrete Cosine Transform filter bank. When extracting the redundancy from a normal complex signal the algorithm takes a time block of 2048 samples of the PCM signal and transforms them into 1024 equally spaced frequency subbands. However it can also successfully deal with the onset of more complex audio using a process of non-simultaneous or time domain masking. This takes advantage of another phenomenom of the ear where a strong signal will mask a weaker signal, before or after it in close time proximity (see Fig). For rapid changes or transitions in the audio content the algorithm automatically switches to a much shorter 256 sample block transformed to 128 frequency subbands. This approach deals effectively with the problem of pre-echo which can result from coding inefficiencies, the shortfall of many algorithms with a long and fixed width transform window. It is this temporal masking for which the apt-Q algorithm is particularly noted, dynamically switching between long and short signal blocks depending on the input, thereby dealing effectively with sharp transitional sounds.

The output data from each of the filtered subbands is quantised and coefficients set with just enough bit resolution to maintain adequate headroom between the accumulating quantisation noise and the masking threshold for each critical band. The maintenance of these signal to masking threshold ratios is crucial if further compression is contemplated in any post production or onward transmission process.

The limit for data compression is reached when the injected quantisation noise rises above the masking threshold and is detected as part of the wanted audio, processed as such and is then heard. This is particularly relevant in tandem or multiple coding applications hence, it is advisable to limit the use of more aggressive coding in a complex audio set up.

It has been found that when coding stereo audio and using a particular perceptual model to determine the masking thresholds, a signal may be masked by both the L and R signals individually but may not be when the L and R signals are offered on separate channels. In apt-Q the perceptual model used determines the masking thresholds for not only the L and R signals but also each of the sum M and difference S signals. The threshold values are then computed having consideration for the spread of masking energy across each signal. From this computation a decision is made as to whether coding the M, S or L,R signals would provide a more efficient coding process. This approach leads to better audio quality and permits a higher compression ratio.

A further stage of bit rate reduction takes place in the apt-Q coder. This is known as adaptive entropy encoding or as it is more readily called, noiseless compression and this takes advantage of any redundancy present in the representation of the quantiser coefficients. Huffman codebooks are used to encode the quantised coefficients for each of the frequency subbands and communicating this information to the decoder imposes a significant bit overhead demand. If transmitted in full it would reduce the number of bits in each data frame that would be available for the audio information. apt-Q gets round this problem by grouping similar codebooks into sections and then using only one codebook to represent that section, the rest being redundant.

Conclusion

Digital audio data compression can play a major part in the exchange, production and transmission of high quality audio in stereo or mono. Interfaced with the dial up national and rapidly increasing international ISDN telecommunications networks it can significantly reduces the operational costs for the capture and exchange of audio throughout the world with a quality which matches the CD benchmark. But, with current technology it is still a destructive process and any potential user must understand the application and then by comparing the operational parameters of the range of available compression codecs and algorithms make an informed match.

apt-X100 with its fixed 4:1 compression ratio is a safe and robust, low compexity algorithm particularly suitable for a whole range of applications, ISDN, STL, storage and multichannel/multilingual digital recording and distribution, but particularly suitable for tandem and multiple coding situations.

apt-Q, a higher complexity algorithm with it's more aggressive form of compression is aimed primarily at broadcasting and other institutions such as advertising, PR agencies and the film industry who have a need for the capture of quality audio and see the ISDN network as the infrastructure to enable this to happen, economical in time and cost.

Acknowledgements

The author wishes to thank his colleagues, Paul Browne Manager R&D and John Knapton Sales Director at Audio Processing Technology for their valuable contribution in the preparation of this paper.

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2 Signal Compression Based on Models of Human Perception by Nikil Jayant, James Johnston and Robert Safranek. Proceedings of the IEEE Vol 81 No 10 October 1993.

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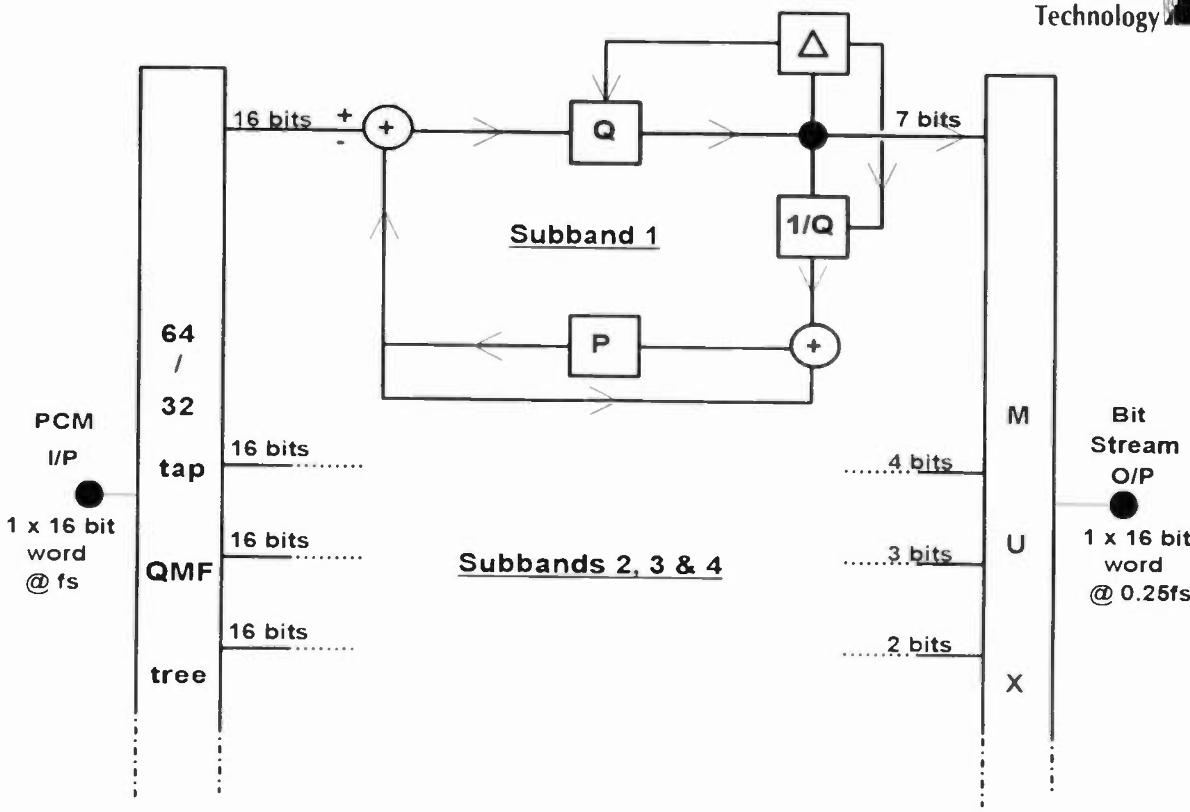


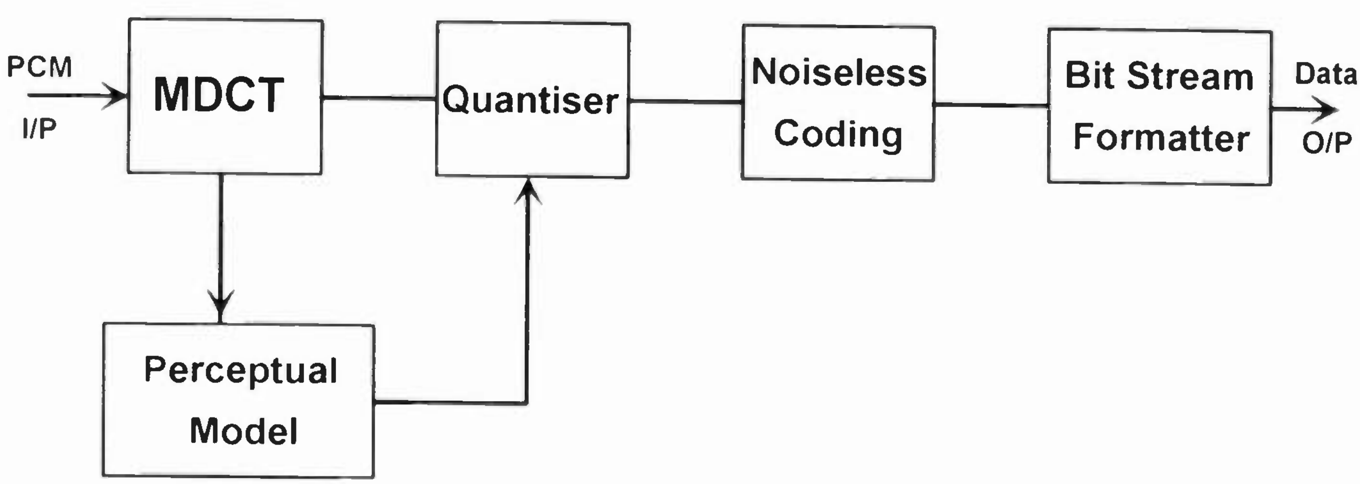
Q Backward Adaptive Quantiser

P Backward adaptive predicter

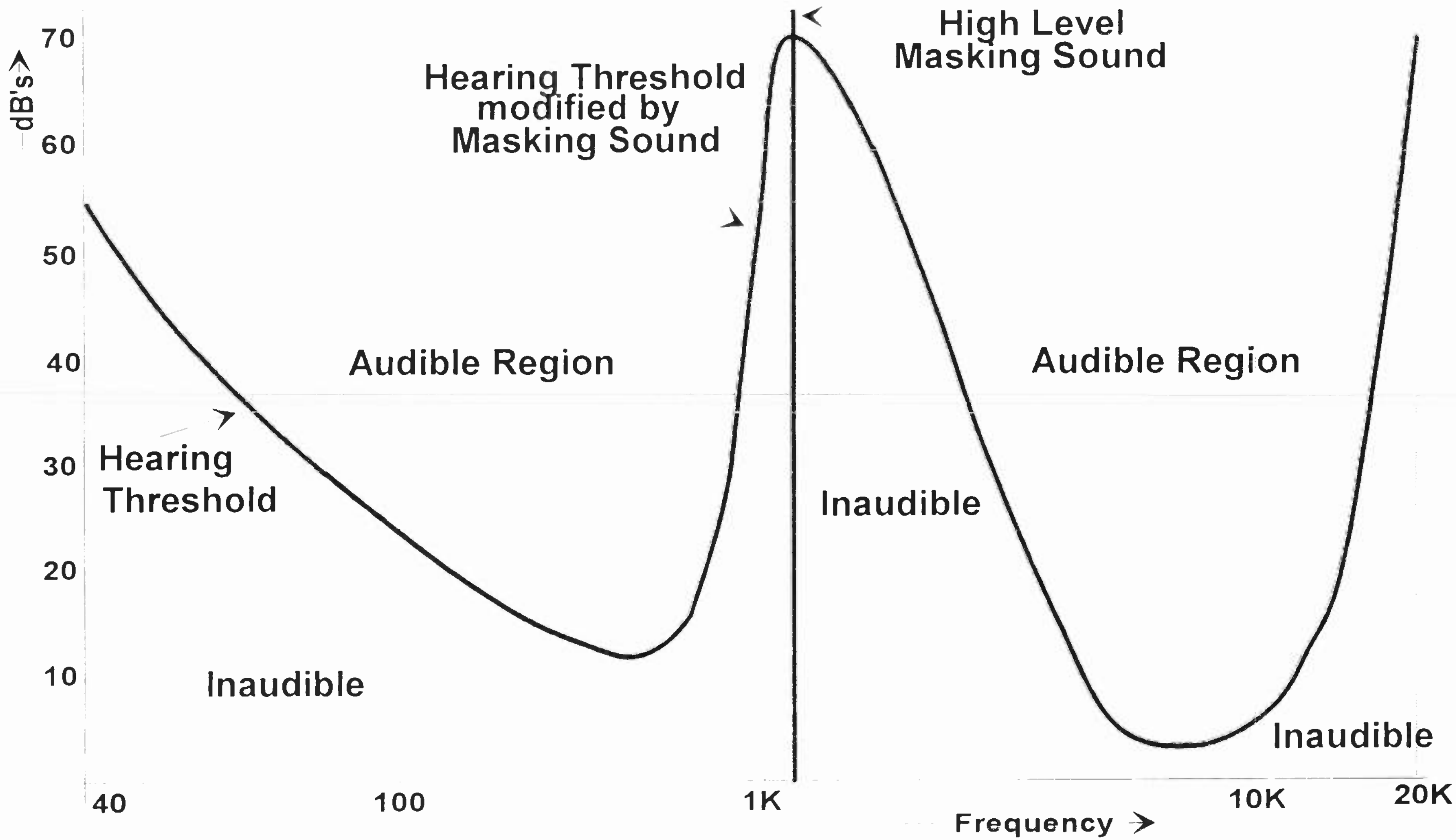
1/Q Inverse backward adaptive quantiser

△ Quantiser step size adapter

Section of apt-X100 coder



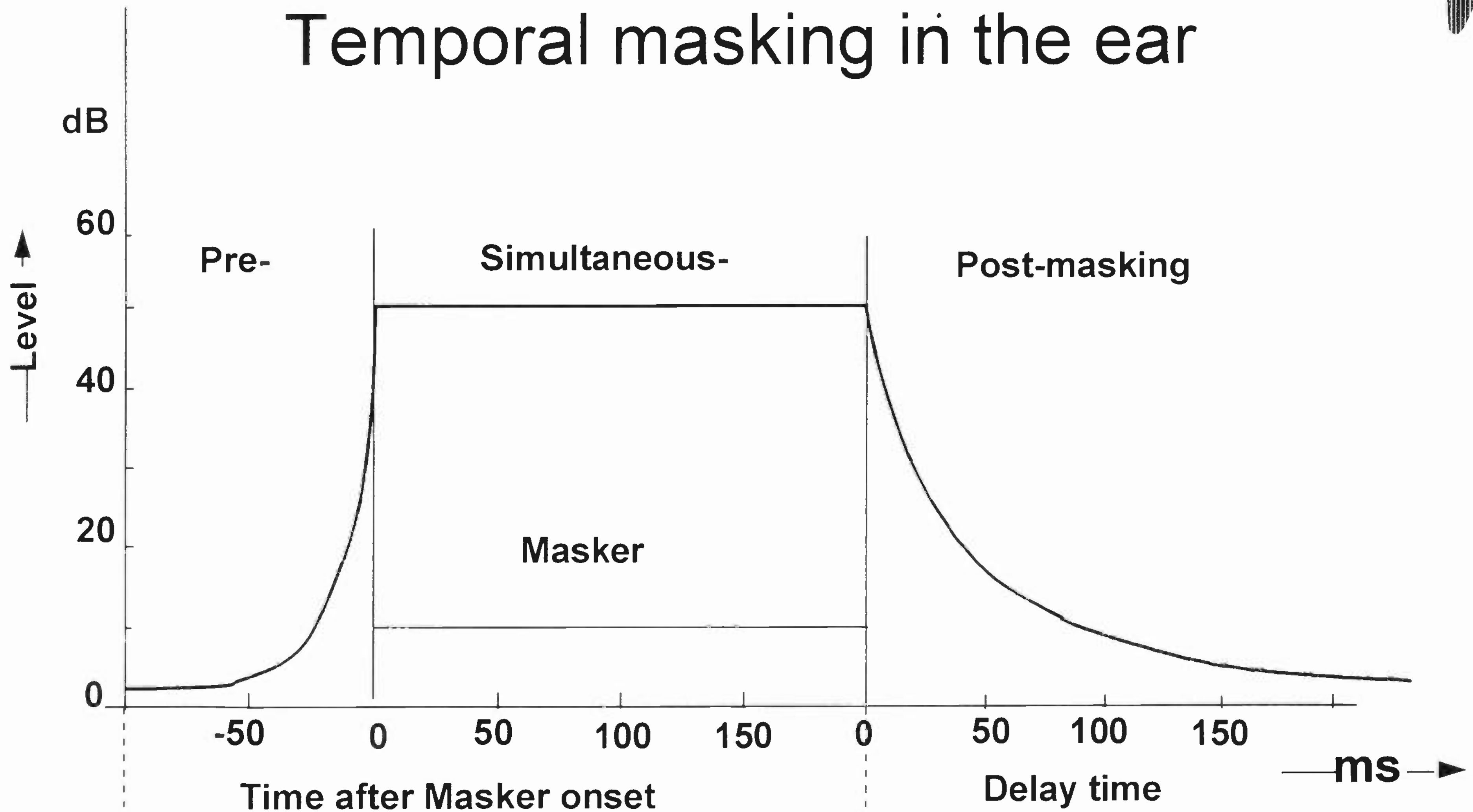
Stereo apt-Q coder



Masking effect of high level sound

Audio Processing Technology

Temporal masking in the ear



AUTOMATION: THE JOCK'S BEST FRIEND

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Digital automation, if used properly, will improve the small market sound. From helping ownership improve the bottom line to competing with the big markets. How do you sell an automation system to your airstaff? What are the real benefits to your airstaff? Automation in the hands of intelligent ownership will reap big dollars and provide a great training ground for future big market jocks. In today's radio markets, competition is like warfare. Don't let an automation system become a loaded gun that isn't used.

There are two areas that need to be looked at when going to a hard disk automation system. The first, things to consider before purchasing. The second, ideas on how to keep and motivate a staff with an automation system.

CHOOSING YOUR SYSTEM

There are many things to remember when choosing your system. Planning for the future of your broadcast facility is the key. Expandability and having the ability to fit your ever changing needs must be the priority. You need to break down each individual department and what they need. From traffic, programming, administration, production and billing, each department needs to write down their needs and the things that will make their jobs run smoother. The following is a brief breakdown of things to consider.

Besides being able to directly interface with your traffic software, how the traffic department interacts with programming is very important. Many automation systems allow the programming department to continue having the freedom to schedule their own jingles and promos. Many program directors enjoy the freedom in doing their own scheduling because it allows them to get the latest promotion on the air quickly without having to go through the traffic department. This will also free up the traffic department to get their job done quickly without having to deal with programming. Having complete and accurate records that can be generated by an automation system makes the end of the month billing more bearable. What used to take three to four days, can know be done in one. How your billing person gets their job done should be reflected in your automation purchase.

The speed of getting production done and implemented is crucial. Having production done on a digital editor means less generation loss. Having the ability to transport those sound files directly to an automation system

can save time and keep the quality constant. If your production people already have a favorite digital editor, find out how it interfaces with the automation system you are looking to purchase. Time spent on commercial transportation can be a concern, because if your production people are able to get their production done more efficiently they might have more time for those all important spec spots. The lack of an interface for your digital editor may not keep you from purchasing the automation system, but it should be put on your pros and cons list.

There are many functions to consider with administration. The ability to run reports showing spot placement and availability is possible with a good traffic and billing system. However, with a good automation system, those reports can be exact. Administration must also be able to integrate security features and preventative maintenance.

Programming is the most important department. Besides having the freedom to schedule promos and jingles, careful format logistics must be considered. An example would be the ability for a top 40 station to have the availability of overlapping the music and being able to have a recorded voice post to the vocals. Different automation systems call this function many things. The term most used would be "Segue Editor". A segue editor can keep your station sounding live. Just because this subject is about automation, it shouldn't change the sound of your station. The automation system can also serve as a record of what music was played and how quarter hours were managed. Another concern might be the availability of a two track editor for phone editing. Many automation systems offer this on their systems. Getting rid of the razor blades in the control room can mean fast, tight edits for all request programs.

The idea is to consider each and every department that will be affected by your purchase of an automation system. Write down all the pros and cons. There are more than one hundred different automation systems on the market, you can find the system that will fit most of your needs for today and the future.

APPLICATION

There are many thoughts and ideas on the best way to use automation. The following is based on the idea that radio can make money and continue to be a training ground for young talent. Many radio stations already run with a reduced staff. Making the move to a digital automation system should not mean cutting personnel, but make the staff you have more effective. The following ideas might allow you to keep your airstaff happy and employed.

Many stations have a morning person who is also the program director. Often times, this person wears even more hats. They are usually the person who has been at the station the longest and is probably the most well known. This person understands the station, community positioning, station imaging, general manager expectations, etc. If an automation system is used properly, it can keep this individual from getting burned out. Delegation is often hard because everyone has many responsibilities. An automation system can spread out more responsibilities and keep the staff from getting burnt. For the sake of examples, let's discuss markets 250 plus.

Let's say we decide to keep mornings all live. Middays are important, but this shift does allow for some flexibility. Let's automate 10 a.m. to noon and then go live for the lunch time. If we consider that many radio stations are going to half hour music sweeps, that means only a couple of recorded jock breaks per hour. By using some automation tricks, like the segue editor, we can keep these breaks sounding great. Those two hours can now be used for producing the station imaging liners, putting together the next promotion or even producing a program for a sister station. In this example, you are getting seven to eight hours of work in a four to five hour time frame. Depending on the system you purchase, production can even be done in the studio so that the disc jockey is still doing a great midday show, but they are also taking some of that pressure off of your morning person. Many disc jockeys take great pride in sharing the duties. Making them in charge of a piece of the pie can keep quality high.

Let's move on to afternoons. Keeping in mind that we are talking about markets 250 plus, this could mean a short drive time. Consider automating 2 p.m. to 4 p.m., this can mean another two hours of creative production or perhaps music scheduling. Again, this creates six to seven hours of productivity in a time span of four to five hours.

Evenings are often used as all request times. Automating the half hour sweeps would allow for more prep time. Having every request finely edited can reap some big rewards. There are many stations that cut off the requests at 10 p.m. and then automate from 10 to midnight. This is the time when the disc jockey works on production or perhaps bits for the next days show.

Overnights are no longer the training grounds they used to be. An automation system can change that. If your overnight jocks is also the traffic director, you can get two jobs accomplished. You are providing an entry level on air job with better pay.

CONCLUSION

When considering the move to a hard disk system, make a chart of all the departments ant their interaction. On this same chart, make a list of equipment that will need to interface. This list will help guide you through this difficult process. One thing to consider on your shopping list is the reputation of the company you are purchasing from. Is this a company that is proven? How many systems do they have on the market? Will they be around five years from now? Remember that your purchase will be with you for many years to come. A good customer service reputation can go a long way.

We need to face the fact that radio is a business. The idea of making a fortune in radio as a disc jockey is for the few. We don't need to completely change the past. Smaller markets can continue to be great training grounds. Being out of work sounds worse that taking on a few more responsibilities. Automation systems can be a disc jockeys best friend.

DATACASTING TECHNOLOGIES FOR TELEVISION

Monday, April 15, 1996
10:30 am - 12:00 pm

Session Chairperson:
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*Paper not available at the time of publication.

WIRELESS INTERACTIVE VIDEO AND DATA SERVICE SYSTEMS AND APPLICATIONS

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ABSTRACT

A number of different system concepts for and applications of Interactive Video and Data Service (IVDS)operating in one of two 500 kHz segments in the 218-219 MHZ band and having radio frequency bandwidths ranging from 5 kHz to 125 kHz are being tested and implemented. Tests to date indicate that system coverage is meeting or exceeding predictions. A number of diverse applications of IVDS in conjunction with other frequencies are being considered and some are being tested. These involve (1) TV related applications such as viewer response to advertising, opinion polls, home shopping, education, etc.; (2) financial and data services; (3) industrial applications; and (4) home information and control systems. Tests of some IVDS applications are in early stages and show promise of being useful and cost effective.

IVDS TRANSMISSION SYSTEMS

A number of different transmission systems concepts for Interactive Video and Data Service (IVDS) operating in one of two 500 kHz segments in the 218-219 MHz band are being tested and implemented. The IVDS is a point-to-multipoint, multipoint-to-point, short distance communications service in which licensees may provide information, products, or services to individual subscribers located at fixed locations in the service area, and subscribers may provide responses. Five different cell transmitter stations (CTS) and four different response transmitter units (RTU) have received type acceptance by the Federal Communications Commission (FCC). These various systems have radio frequency bandwidths ranging from 5 kHz to 125 kHz and employ different types of modulation and channel access techniques. Both one-way and two-way systems operating at IVDS frequencies are represented.

The FCC has adopted power, antenna height and duty cycle limitations for IVDS systems in order to avoid interference with TV Channel 13 operating at 210-216 MHz. Because of these limitations many of the IVDS system designs have been similar to cellular systems.

An IVDS system consists of a central Cell Transmitter Station (CTS) in each cell and Response Transmitter Units (RTU) at each subscriber location as shown in Figure 1. Frequency reuse is a feature of the cellular system because of the power, antenna height and duty cycle limitations. Four manufacturers have FCC type accepted systems that use a number of CTS locations in a given market. In order to reuse frequencies relatively narrow bandwidths are employed ranging from 5 kHz to about 40 kHz.

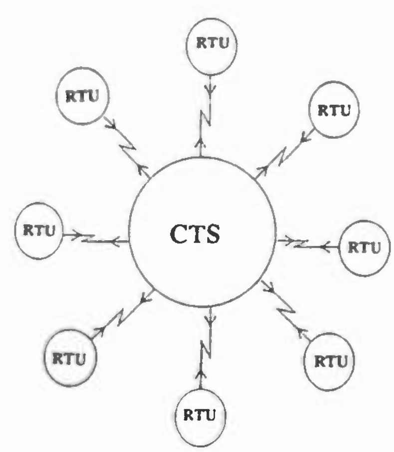

Figure 1. IVDS Cellular System Diagram

The system developed by the *EON Corporation* is a two-way system which divides the band into 15 channels each with a bandwidth of 31.25 kHz (one-half channel at each of the upper and lower ends are set aside for guard bands). Each CTS transmitter uses one of the channels to send information to RTU's. A number of remote receivers are used to receive messages from the RTU's on different channels (see Figure 2). All transmitters, remote receivers and RTU communications as synchronized by a global positioning satellite (GPS) system. Remote receivers are used because EON has limited the RTU power to 100 miliwatts. Each remote receiver handles on the average of three IVDS subscriber messages per second. A cell consists of one CTS (includes one receiver) and up to 14 additional remote receivers with a total coverage area of about 25 square miles (64 km^2). A system using this technology is operating in part of the Washington, D.C. area with reported good coverage results.

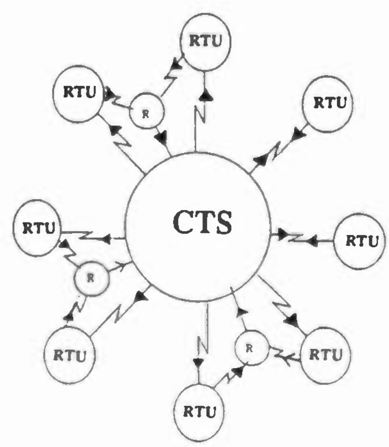

Figure 2 IVDS Cellular System
with remote receivers

A system with similar bandwidth has been developed for *IVDS Affiliates, LLC*. This is a one-way system in the IVDS band, with only data transmission broadcast from the CTS to the RTU's. Coverage area for a single cell of this system can be 70 square miles (180 km^2) if the transmitter is in the Grade B contour of TV channel 13 to 340 square miles (700 km^2) if the transmitter is in the City Grade contour. If the CTS is 10 miles beyond the Grade B contour a coverage area of about 1300 square miles (3500 km^2) is possible. A system of this type is currently being tested in the San Francisco, CA area. Coverage distances in certain directions appear to be exceeding expectations because of the unique topography in the area. *IVDS Affiliates* has recently announced that, in conjunction with the *ARDIS Company*, it will use a mobile data system in the 800 MHz band as a return path.

Two other suppliers have type-accepted two-way narrowband systems. They provide a greater number of channels with reduced data rates per channel. For example, *SEA* has 5 kHz channels; while *GLB Electronics, Inc.* (*GLB*) can go up to 20 kHz channels. The *SEA* system is undergoing tests for data communications in the Philadelphia, PA area with a duty cycle waiver granted by the FCC in areas outside of the Grade B contours of New York City and Baltimore, MD. It is understood that the *GLB* system is being tested in Minneapolis, MN. No data on these tests has been released. Coverage areas for a single channel would be similar to that of *IVDS Affiliates*.

Where multiple transmitters are utilized at the same CTS site, which might be the case for *SEA*, *GLB* and *IVDS Affiliates* systems, the FCC power limitations (being for total power within the band) would require reduction of the power per channel proportional to the number of channels. This would reduce the coverage area for each channel.

Another system, developed by *Radio Telecom & Technology, Inc. (RTT)*, takes advantage of the horizontal and vertical blanking intervals of Channel 13 to avoid interference. The CTS is allowed to transmit only during the vertical and horizontal blanking intervals which reduces interference to TV. An RTU can transmit only during the horizontal blanking interval to avoid interference in an area surrounding the RTU. At greater distances from the RTU propagation loss is sufficient to eliminate interference.

The *RTT* system (also known as the *T-Net System*) in an omnidirectional configuration, has cells which are concentric circles which are formed by the time of travel of the radio waves. Angular sectors are used to subdivide the cells as shown in Figure 3. The RTU's are interrogated in sequence for each range cell. This produce a form of a packet switched network. The omnidirectional system can handle up to 30,000 subscribers using 125 kHz bandwidth channels for subscriber RTU-to-CTS communications, and one 125 kHz CTS-to-subscriber RTU channel. Each RTU is interrogated once a second. When the omnidirectional system reaches subscriber saturation, sector antennas can be used to increase capacity. A four sector system can handle up to 120,000 subscribers with a one second per subscriber polling time.

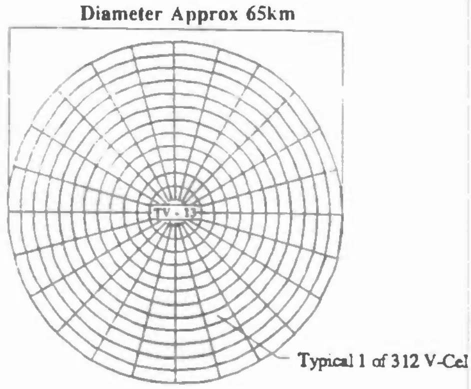

Figure 3. Virtual Cell Arrangement

The *FCC Report and Order of January 16, 1992*, which established the IVDS allocation and rules for its use, indicated, in a footnote, that they would consider, on a case-by-case basis, waivers of the power, antenna height limitation for the *RTT* suggested system or any other measures which are taken to avoid interference. Thus the *RTT* system without waivers will produce coverage areas similar to the *IVDS Affiliates* coverage areas. With a waiver, coverage areas will be much greater; in some cases exceeding 2000 square miles (5100 km^2).

An *RTT* system is operating under an experimental license in the Los Angeles, CA area. An initial licensed system has been constructed in the Boston, MA area with reduced power because of antenna height. Tests of coverage have been made with coverage meeting or exceeding predictions. A waiver request has been submitted to the FCC to allow full 20 watt CTS power at the higher antenna height for Boston. A grant of the waiver is pending.

APPLICATIONS

A number of diverse applications of IVDS frequencies and of IVDS in conjunction with other frequencies are being considered and some are being tested. These involve:

- TV related applications such as viewer response to advertising, opinion polls, home shopping, education, etc.;
- financial and data service such as stock market quotes, credit card verification, automatic teller machine, data communications, etc.;
- industrial applications such as energy management and control, security, vending machine inventory, etc.; and
- home systems such as utility meter reading, remote ticket ordering, central station alarm monitoring, etc.

The business and industrial applications of IVDS include automatic teller machine (ATM) and point-of-sale credit card verifications communications, data communications (WAN or LAN), industrial security, vending machine inventory, industrial utility monitoring and control and state lottery communications between sales agents and the central computer. Nearly all of these applications are currently using telephone communications. In some cases, such as industrial security, utility monitoring and control and vending machine monitoring wireless communication has been used at various frequency bands. In Massachusetts the state lottery uses multiple address systems at 900 MHz for sales agent to computer communication.

The IVDS licensee (Kingdon R. Hughes) in Philadelphia, PA which has TV Channel 13 Grade B coverage only in small portions of the northern and southern parts of the area, determined that, with the narrowband *SEA* system he chose to use, the IVDS duty cycle limitations would not support ATM applications. He requested and was granted an FCC waiver of these limitations in areas not covered by Channel 13. It may be possible to use a wider band system for this application without a waiver of the duty cycle limits. Point-of-sale credit card verification applications have similar duty cycle requirements to those of ATM and might also be accommodated by IVDS.

For other applications in this category IVDS may be able to provide a common network that will consolidate communications in a cost effective manner.

A number of consumer related applications require no home display, these include reading of utility meters, managing utility loads, remote control of heating, air conditioning and appliances, wireless home central station security, etc. Currently some of these applications are being done by other means. For example utility meter reading has been done manually, via telephone lines and using low power wireless communications with a vehicle interrogating meters as it drives by. Utility load control by turning off central air conditioners and water heaters for a part of an hour during peak load condition has been done at several different radio frequency bands. Many of these are one-way systems with no direct reply indicating that turn-off has occurred. Home central station security systems primarily use telephone lines and in some cases multiple address systems in the 900 MHz band.

IVDS offers the opportunity of combining these relatively low duty cycle applications in one transmitter/receiver RTU in the home.

Certain other applications require some type of simple display in the home. For these applications a small printer or LCD display could be used or alternatively a display on a home TV could also be employed. These applications include purchase of tickets to entertainment events (concerts, stage plays, sporting events, etc.), credit card purchases or payment of bills, messaging (simplified E mail), stock quotes, state lotteries or other forms of gambling (if permitted), consumer polls, coupons for discount on merchandise, and home banking (transfer between accounts, account balances, bill payments, etc). All of the above applications with the exception of state lotteries are currently being done via telephone lines.

EON has developed a hand held wireless messaging device. A proposed rule making to allow mobile use of IVDS with messaging as an ancillary service is under consideration by the FCC. *IVDS Affiliates* working with *Data Broadcasting Service* is testing IVDS to transmit stocks quotes to subscribers on portable or fixed receivers in the San Francisco, CA area. Presently *Data Broadcasting* provides this service using FM subcarriers or TV vertical blanking intervals (VBI) but they want to increase the current data rates by using IVDS.

Again IVDS provides an opportunity to consolidate these applications along with those requiring no display into one receiver/transmitter RTU in a home.

The main purpose of the FCC in allocating frequencies for and establishing the IVDS as a radio service was to provide a means of allowing viewers to interact with TV programs and advertising. Some of the applications which can be utilized with IVDS are instant response to talk shows, infomercials, local and national polls, advertising, (both national and local) and educational programs. In addition IVDS can provide for program ratings, home shopping, ordering of pay-per-view movies or events, playing simple games and, when used in conjunction with a CDi player or a server, can provide a means of catalog ordering. It may also be possible to use a TV set as a simplified home computer with access to various data bases.

Many of these applications are presently handled by telephone or mail or not at all. Where responses are given there are time delays or if there are many responders, such as national polls, only a small fraction of the responses can be handled because of capacity limitations of telephone networks.

Several systems have been developed to allow for down stream communications to only the subscribers of the system. *Interactive Systems Inc.* (IS) modulates the intensity of the TV lines to provide information which can be displayed on screen to subscribers. The intensity variation is not perceptible to a viewer. It uses a telephone return path to download responses on an after-the-fact basis. This system has been in use in Spain and is starting to be used in Japan. It is understood that they are considering IVDS as a return path for use in the United States.

Welcome To The Future (*WT²F*) uses a vertical blanking line of a host TV station as a downstream path to the subscriber who would receive information in a picture-in-picture display. The return path is IVDS or telephone. *WT²F* has conducted tests using a special temporary authority for IVDS in conjunction with Channel 13 in the Baltimore, MD area. They are also conducting tests in Columbus, OH in conjunction with Dispatch Communications, a subsidiary of WBNS-TV Inc. who holds the IVDS license in Columbus, OH and Indianapolis, IN as well as TV and radio licenses in both cities. *WT²F* does not presently have an FCC type accepted RTU but had earlier announced their intent of developing a radio frequency modem or using modems of other IVDS system suppliers. They use a server to access various data bases.

EON had originally developed a color overlay on-screen display but because of cost have been concentrating on a black and white display. The display utilizes menus and icons.

RTT response function in its basic interactive system (BiTV) relies on having no on-screen display. Rather they compare the time of response to the programing on the Channel to which their set-top is tuned. This allows for interactive responses such as requesting information on a product, like or dislike of a program, poll response, etc. A more advanced on-screen display system in conjunction with a CDi player is being developed which will also allow for shopping from CD catalogs that now are getting more and more popular.

In addition, other groups are working with the possibility of using Personal Computers (PC) applications. *Ferderer Communications* is working with *IVDS Interactive Acquisition Partners* (*IIAP*) the licensee for Minneapolis-St. Paul, MN-WI in testing access to the Internet via IVDS. *Wave Interactive* is considering PC applications and is reportedly exploring such uses.

CONCLUSION

During the past few years there has been considerable activity concerning interactive television. The cable companies and the telephone companies have conducted pilot tests and have announced plans for implementation of such services in conjunction with expanding the number and types of video services available to their customers. These planned implementations have been postponed primarily because of cost.

Interactive television can be a valuable asset which can improve TV programing, enhance ratings, provide valuable services to the public and increase revenues to the television community. IVDS allows for rapid introduction of interactive responses to standard broadcast TV, LPTV, cable, DBS and broadcast radio as well as for the provision of non TV related services in a very cost effective manner. Further, if program content providers and advertisers produce their material with on-screen prompts for interactive TV greater public acceptance can result and better delivery of services can be achieved.

Most important is the ability of **IVDS** to provide the **only wireless return path** that is readily available and inexpensive to implement. This allows the viewer to be part of a new two-way entertainment and information age in the very near time frame.

D-CHANNEL
A TRANSITION INTO DATACASTING

Brit Conner
Digideck
Menlo Park, CA

INTRODUCTION

Digideck's new D-Channel technology is one of the most exciting and important enhancements a station can add to its signal in the next three to five years. Developed originally by staff of the David Sarnoff Research Center, and extended by joint efforts between Digideck and Applied Signal Technology, Inc., this signal becomes the vital transitioning tool for terrestrial broadcasters between NTSC and ATV. In the paper that follows, we explain what D-Channel is, why it's so important, and how it can be used.

WHAT IS D-CHANNEL TECHNOLOGY?

The term "D-Channel" was coined to refer to Digideck's high speed *data channel* which is opened within the existing 6 MHz NTSC signal spectrum. The data is carried on a new, low-level subcarrier inserted at the transmitter in the lower vestigial sideband region of the modulated video IF signal. The new subcarrier does not impact the audio, video or VBI portions of the existing television signal, yet it can be reliably received by an inexpensive card specially designed to process the signal.

The signal, depicted in Figure 1, uses DQPSK modulation at 700 kbps (350 kbaud). With 20 percent excess bandwidth filtering, it occupies a 420 kHz bandwidth centered 1 MHz below the picture carrier. Twenty-five percent of the bitstream is used for $t = 8$ Reed-Solomon forward error correction (FEC), along with a 340 byte interleaver, netting 525 kbps of data throughput, and over 500 kbps after overhead for packetizing, privacy and other protocol tasks.[1] This is more than enough for interactive television applications, even enough for a complete full-motion video channel of teleconferencing quality.

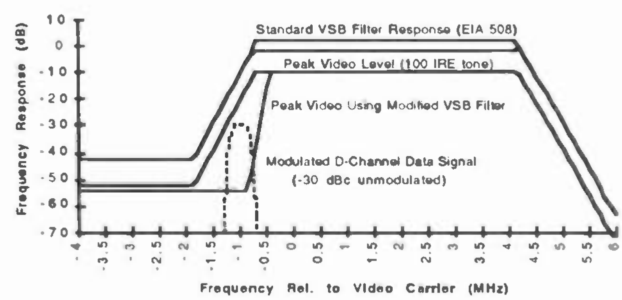

Fig. 1 -- Standard VSB Filter Response, Modified Response and Data Signal

Table 1 summarizes some results from tests conducted by the National Data Broadcasting Committee. Even though the signal is inserted some 30 dB below the peak of video sync, test data indicates a 16 dB noise margin at the Grade B contour, with more resistance to adjacent and co-channel interference than its host video, yet no more impact to adjacent and co-channel signals than current NTSC. Most importantly, expert observers give the D-Channel signal extremely high marks for picture quality, even on images designed to heighten visibility of the data within the picture. D-Channel test scores reached 4.83 out of 5.0 as an average over 16 sets, 3 images, 2 reception levels and 2 observers, well above the 4.0 threshold of "perceptible but not annoying", even to an expert.

[1] The FEC may change in the final configuration, but throughput should remain at or near 525 kbps.

In part, the excellent picture quality scores arise from D-Channel's careful placement of the data carrier to take advantage of the Nyquist filter found in all television sets. As shown in Figure 2 the standard Nyquist filter attenuates the D-Channel data signal approximately 20-25 dB before demodulation, rendering it essentially invisible to a television receiver.

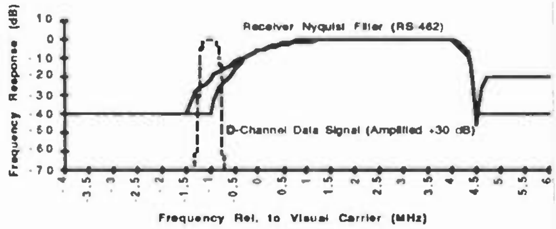

Fig. 2 -- Receiver Response

Also included in Table 1 are comparative results for WavePhore's original and modified TVT systems, originally running at 384 kbps throughput, and now at 300 kbps. The transmission advantages of the D-Channel are clearly evident. Not shown are other competing systems, such as Intercast's new system, which uses traditional VBI data transport, and Matsushita's 253/205 kbps system, which uses quadrature modulation of the picture carrier. All of these competing systems lack D-Channel's high speed and excellent performance.

STATION INSTALLATION

A part of the studio equipment will consist of appropriate devices for receiving, storing and manipulating the data feed(s). Sources may be internal, local, or distant. Data files will generally be stored on a new digital file server which will incorporate control software for additions, deletions, scheduling and reporting, and control hardware to allow interface to the station's master control. The latter is critical, to allow synchronization between programming and data files.

A second part of the equipment will take the scheduled files, and translate them for delivery over a new STL link to the transmitter site. This segment will allow for FCC override control as well. At the transmitter, the data stream -- already packetized for application use, is additionally formatted for broadcast delivery. The STL receiver is connected to an external D-Channel inserter unit which formats the data, adds error correction, modulates it and sets the appropriate insertion level. In a second path it takes the modulated video from the existing exciter, filters it in a custom SAW to remove a further portion of the VSB region, and combines the video and data, and feeds the resulting signal to the existing transmitter's linearization circuitry. Figure 3 depicts the D-Channel Inserter unit.

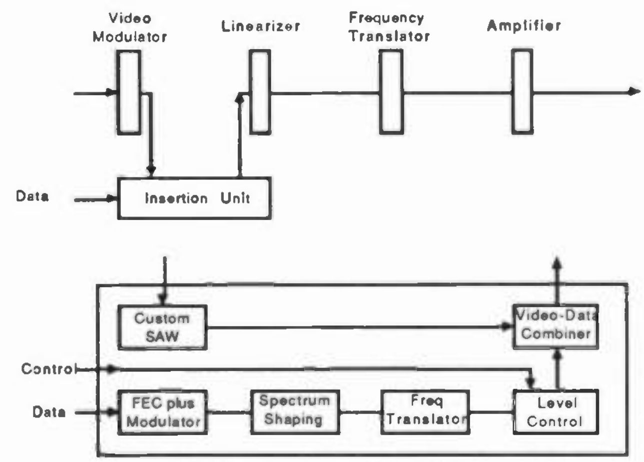

Figure 3 -- Insertion Unit

WHY IT'S IMPORTANT

In one aspect adding the D-Channel signal is similar to adding the additional signals in MTS stereo -- existing equipment is not disturbed, but new equipment can receive and utilize this signal in some very exciting ways, which we will explore below. But in another aspect, the D-Channel is dramatically different from our industry's experience with the addition of stereo -- this time the station owners can get paid for installing it.

The nature of the D-Channel insertion creates an important strategic advantage for the local broadcaster. Because the data signal must be inserted at the transmitter, it will not be incorporated in the existing distribution methods. It won't be present on network feeds, and it won't be incorporated on normal program tapes. It may be provided by a small distribution infrastructure owned by a network or a consortium of cooperating stations and content providers, but it can only be added by the individual station.

This fact gives the station owner a strong hand in choosing how to make the most of this new opportunity. Not only does the D-Channel signal open up new vistas for delivery of both television and non-television related data, but the station owners will now find themselves with a new "freedom of choice" -- to cut new deals for content, to work within or outside their existing video affiliations, and at the same time to reach out to entirely new categories of listeners and viewers, sponsors and subscribers.

APPLICATIONS

The most important question people ask today is, "What are the uses for this new signal?" Let's look at some examples:

Interactive Television -- In this application, the broadcast data is used as a "teaser" for initiating a true two-way interactive event. Suppose the program consists of a musical group. Clicking the remote brings the lyrics on-screen. Clicking again brings up jacket liner notes or other interesting information. Clicking a third time brings up the group's tour schedule. And, noting that the group is coming to the viewer's area, clicking a fourth time tells an embedded cable or telephone modem to dial up a pre-arranged ticket office. The ticket office server contains the same software to drive the TV screen as was used in the original programming, so the ticket purchase event continues without skipping a beat on the user's TV set.

PC Data -- In this application, news, information, software packages and myriad other files are transmitted to thousands of simultaneous users without tying up the Internet. Unlike the expensive one-to-one client/server model of Internet, the power of broadcasting is used to inexpensively reach vast audiences who share, if even only for a few moments, a common interest in the data at hand. For example, the data might be further details on an event, or the text of a speech being given at an event currently taking place somewhere in the nation. Or the files might be updates to software of special interest to a particular set of users. In any case, transmission using television tower based delivery eliminates all the infrastructure cost which, when combined with the multiplicative power of broadcast reception, makes for a system which will cost far less than cable, telco and satellite systems.

Distance Learning -- In 1995 Digideck had over two dozen requests and proposals to use the D-Channel equipment for distance learning applications. The 525 kbps data rate comfortably exceeds several of the standard rates for teleconference quality video transmission, making it well suitable for delivery of classroom programming or local interest events. While this may not represent a big financial opportunity, the ability to deliver 24-hours per day of educational or local programming on a "second channel" is of keen interest in many rural and remote areas.

Kiosk Update -- In another rather interesting application, a communications system house explained a developing modern transit system which would provide realtime information on bus locations to displays at bus stops (for example, estimated time to next bus), traffic congestion information, and other information. All of this could be accomplished with traditional low data rate communications links, but the system only made economic sense if they installed advertising based video kiosks at the bus stops. And the kiosks could be economically updated using the D-Channel to reach bus stops in a 40-50 mile radius.

CONCLUSIONS

In the end, one is only limited by his imagination for uses of a high speed data carrier. And all of them give the station owner a great new way to profit as the basic insertion equipment is relatively inexpensive -- not much more than the cost of replacing the video modulator in your existing exciter.

Table 1 -- Results from NDBC Laboratory Tests --
Digideck D-Channel vs. Original and Modified WavePhore TVT-1/4 Systems

General Information	**Units**	**Digideck**	**Orig WP**	**Modified WP**	**Winner**	**Delta**
Transmitted Bit Rate	kbps	700	599 burst			
Net Bit Rate	kbps	525	*	300	Digideck	75%
Video Impairments	**Units**	**Digideck**	**Orig WP**	**Modified WP**	**Winner**	**Delta**
Degradation to Host NTSC (Video)						
First Laboratory Tests	2 Observers	(each score is observer's avg. of 2 msmts. on each of 16 sets)				
Weak Desired Signal	1-5 absolute quality rating					
Data ON						
Image 1		5.0, 4.9	3.6, 4.2		Digideck	
Image 2		5.0, 4.9	4.5, 4.3		Digideck	
Image 3		4.9, 4.9	4.6, 4.5		Digideck	
Data OFF						
Image 1		5.0, 5.0	3.4, 4.3		Digideck	
Image 2		5.0, 5.0	4.4, 4.1		Digideck	
Image 3		5.0, 5.0	4.5, 4.5		Digideck	
Strong Desired Signal	1-5 absolute quality rating					
Data ON						
Image 1		5.0, 4.4	3.5, 3.9		Digideck	
Image 2		5.0, 4.5	4.4, 4.2		Digideck	
Image 3		5.0, 4.5	4.5, 4.3		Digideck	
Data OFF						
Image 1		5.0, 5.0	3.5, 4.3		Digideck	
Image 2		5.0, 5.0	4.4, 4.3		Digideck	
Image 3		5.0, 5.0	4.5, 4.7		Digideck	
Second Laboratory Tests	4 Observers	(each score is observer's avg. of 4 msmts. on each of 8 sets)				
Strong Desired Signal	±3 comparative quality rating					
Small Sets	(0 = WP II is same as NTSC, -1 = slightly worse, -2 = worse, -3 = much worse)					
Image 1				-1.00, -1.55, -2.38, -1.25		median -1.5
Image 2				-0.59, -0.83, -0.72, -0.39		median -1.0
Image 3				-0.47, -0.19, -0.88, -0.28		median -0.5

Video Impairments (cont.)	Units	Digideck	Orig WP	Modified WP	Winner	Delta
Large Sets						
Image 1				-1.58, -2.13, -1.47, -1.75		median -2.0
Image 2				-1.02, -1.77, -1.52, -1.47		median -1.5
Image 3				-1.06, -1.22, -0.98, -0.86		median -1.0
Effects on Ghost Canceller		None	Believed OK		—	
Unweighted SNR (@-26 dBm)	dB					
Data OFF		52.7	53.7		—	
Data ON		46.2	Msmt. invalid		—	
VM-700 Measurements					Digideck	
Refer to detailed results showing Digideck OK, some color effects w/ WavePhore and WavePhore II						
Recordability of Host NTSC		Some audio noise	Some color smearing		—	
NTSC Degradation (Audio)	1-5 rating					
Weak Desired Signal	2 observers	(each score is observer's avg. of 4 msmts. on each of 6 sets)				
Sound 1		5.0, 5.0	5.0, 5.0		—	
Sound 2		4.5, 4.8	5.0, 4.9		WavePhore	
Sound 3		5.0, 5.0	4.9, 5.0		—	
Strong Desired Signal	2 observers	(each score is observer's avg. of 4 msmts. on each of 6 sets)				
Sound 1		4.9, 5.0	4.9, 5.0		—	
Sound 2		4.6, 4.9	5.0, 5.0		WavePhore	
Sound 3		5.0, 5.0	4.9, 5.0		—	
Interference to Closed Captioning		None	None		—	
Interference to Host VBI Signal		None	None		—	
Co-Channel Interference	D/U dB					
NTSC+ into NTSC		41.84	40.22		—	
NTSC into NTSC		41.58	40.28		—	
Audio	Neither system exhibited more interference to audio than NTSC					

Video Impairments (cont.)	Units	Digideck	Orig WP	Modified WP	Winner	Delta
Adjacent Channel Interference	D/U dB					
Upper NTSC+ into NTSC		-7.24	-8.02		—	
Upper NTSC into NTSC		-7.29	-8.26		—	
Lower NTSC+ into NTSC		6.01	5.01		—	
Lower NTSC into NTSC		8.29	5.96		—	
Audio	Neither system exhibited more interference to audio than NTSC					

Data Robustness	Units	Digideck	Orig WP	Modified WP	Winner	Delta
Random Noise into NTSC+	Req'd CNR dB					
Video CNR in 6 MHz req'd for 10^{-5} BER		28.08	44.23	39.97	Digideck	11.89 dB
Note that WP II transmitter injection level measured 4 dB stronger than WP I						
Impulse Noise into NTSC+	dB above TOV					
Survivability		14.5		*	—	
* This test could not be run as test bed background noise floor exceeded WP II receiver's random noise threshold						
Immunity to Co-Chan. Interfence	D/U dB					
NTSC into NTSC+		11.87		26.05	Digideck	14.18 dB
NTSC+into NTSC+		14.97		25.00	Digideck	10.03 dB
Immunity to Adj. Chan. Int.	D/U dB					
Upper NTSC into NTSC+		-35.14		-17.89	Digideck	17.25 dB
Upper NTSC+ into NTSC+		-34.90		-17.08	Digideck	17.82 dB
Lower NTSC into NTSC+		-21.98		-22.03	—	
Lower NTSC+ into NTSC+		-22.09		-21.07	Digideck	1.02 dB
Immunity to Static Multipath	BER					
Ensemble 1		OK		OK	—	
Ensemble 2		OK		OK	—	
Ensemble 4		OK		OK	—	
Immunity to Dynamic Multipath	BER					
Ensemble 3		OK		OK	—	

Other Parameters	**Units**	**Digideck**	**Orig WP**	**Modified WP**	**Winner**	**Delta**
Data Recordability at Studio (D2)		No		Yes	WavePhore	
Out -of-Channel Emissions	spectrum	OK		OK	---	
Realtime Delay	millisec	12		4060	Digideck	337x
Acquisition Time	sec	1.25		5.70	Digideck	78%

* Not all parameters were remeasured in the laboratory test of WavePhore's modified system. Results from the modified system replace those from the original system where NDBC data was taken, with the exception of the video quality tests where both sets are included for comparison.

DATACASTING TECHNOLOGIES FOR RADIO

Monday, April 15, 1996
1:00 - 5:00 pm

Session Chairperson:
Dutch Doelitzsch, WDDD, Marion, IL

FMeX - A SYSTEM FOR HIGH-SPEED DATA SUBCARRIER TRANSMISSION IN THE FM-SCA BAND
Dr. Tim Dyson
Mikros Systems Corporation
Princeton, NJ

***A REPORT FROM THE NRSC HIGH-SPEED FM SUBCARRIER SUBCOMMITTEE**
Michael Rau
EZ Communications, Inc.
Fairfax, VA

***FM DATACASTING: THE NEXT 20 YEARS**
Gordon Kaiser
CUE Network Corporation
Irvine, CA

***A TUTORIAL ON FM DATA BROADCASTING**
Eric Small
Modulation Sciences, Inc.,
Somerset, NJ

***FM DATACASTING**
Mo Gardner
Mainstream Data, Inc.
Salt Lake City, UT

***A SYSTEM FOR DATA TRANSMISSION ON THE AM BAND**
Henry Silcock
Mikros Systems Corporation
Princeton, NJ

***A STATUS REPORT ON RDS**
Scott Wright
Delco Electronics
Kokomo, IN

***DATA FOR THE LISTENER: THE BENEFITS OF RDS**
John Casey
Denon
Parsippany, NJ

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*Paper not available at the time of publication.

FMeX - A SYSTEM FOR HIGH-SPEED DATA SUBCARRIER TRANSMISSION IN THE FM-SCA BAND

Tim Dyson and J. Burns
Mikros Systems Corporation
Princeton, NJ

19 December 1995

ABSTRACT

A system for the transmission, reception and decoding of digital data, using high-density hybrid modulation over the FM-SCA band is described. The system is designed to achieve high capacity data communications rates over the FM-SCA band, as defined in 47 CFR sect. 73.295. The system is dependent upon several core technologies which are combined uniquely for this application. Among these are the use of Hexagonal Close-Packed (Optimum) Modulation, Trellis Coding, adaptive channel equalization in the FM baseband, synchronous clock recovery, and fully automated message presence and frame boundary detection. Performance demonstrations have indicated operating levels of up to 3 bps/Hz.

I. Introduction

High-speed data transmission over the existing commercial radio band is an area of fierce activity. Currently, typical SCA data suppliers operate at from 1200 to less than 14000 bps for applications such as paging, news and sports reporting, and financial data transmission using commercial FM broadcast media.

Recent advances have produced systems producing up to 19,200 bps using Duobinary, BPSK and QPSK modulation. Further, AT&T, Amati and USADR have fielded prototype test systems using IBOC and IBAC methodology to support upwards of 256 kbps within the FM channel broadcast mask. These systems, however, utilize the entire available FM bandwidth (about 400 kHz), and although they conform to the FCC's regulations, the question of interference with established SCA Subcarriers (e.g., RBDS and numerous pagers) is still unclear. Additionally, it has been found that the reduced subcarrier levels required in these techniques to operate within the FCC FM Broadcast RF transmission mask greatly limit their area of coverage.

The FMeX system developed by Mikros Systems Corporation provides up to 76 kbps (38 kbps to 57 kbps after FEC, depending on modulation) in a 21 kHz bandwidth within the FCC's defined Subsidiary Communications Authorization. One of the key factors making the approach attractive is that it requires no modification to the existing FM broadcast methodology or equipment. Additionally, since a limited portion of the allotted FM-SCA bandwidth is utilized, mutual interference with other subcarriers (e.g., RBDS) is controlled at the source allowing competing subcarriers.

Section II discusses the transmitter and receiver components of our technology and provides a theory of operation. Section III presents some of the typical laboratory and field testing results for the Mikros Systems HSDS-SCA System. Section IV discusses some of the potential customers for this technology and evaluates several potential trade-offs that can be made to customize the technology to a customer's needs.

II. System Structure

The receiver/decoder system component consists of a standard high-sensitivity FM receiver/demodulator, a clock recovery system, digital message presence and frame boundary detector, an adaptive channel equalizer, Viterbi Decoder/Forward Error Correction. Figure 1 presents a general diagram of the overall HSDS Receiver/Modem. The basic system supports three different data rates designed around an optimum modulation constellation. Typical data rates available are 28.5 kbps, 38 kbps and 57 kbps, with expansion to higher rates straight forward, depending on bandwidth available and channel characteristics. Trellis Coded QAM and optimum modulation is used: Hexagonal Close Packed (8-ary, 16-ary or 64-ary HCP). Figure 2 illustrates the symbol constellation geometry of HCP which maximizes the intersymbol distance in 2-space at a given subcarrier power [1]. Trellis-coded modulation (QAM or quasi-optimal HCP) will be used in production units.

(a) Receiver/Modem

The baseband output of the FM Demodulator is fed to both the clocking recovery circuit and to the A/D input section of the all-digital modem. A PLL receives the output voltage from a tuned circuit cascade centered at the FM pilot frequency (19 kHz). The PLL VCO output is set to produce a clocking signal suitable for Nyquist sampling of the FM baseband signal in the A/D circuits. This process then serves to deliver near-synchronous sampling of the transmitted HSDS digital data to the equalization and decoding processes. The modem input data is derived from the externally clocked A/D. Data arrives at the modem input and is streamed into a complex demodulator (digital heterodyning) tuned to the HSDS-SCA subcarrier frequency. The resulting complex signal is digitally lowpass filtered to the HSDS data bandwidth and fed to the data presence/frame synchronization processor where data frames are aligned with the symbol decoding/equalization processing.

Each encoded frame consists of a known symbol header and channel encoded data. Frame length is approximately 0.3 seconds ending with a 26 symbol EOF mark. Each frame is further subdivided into sub-frames representing 30.3 msec of data and is available to the data broadcaster for independent data feed segments. (see Figure 3)

Binary feed data is processed to add parity to each byte. Each subframe consists of 36 blocks of 36 bits each. Each block is 3/4-rate Reed-Solomon encoded and the output is subjected to inter-block interleaving within a data feed sub-frame. This data is then subjected to frame level symbol interleaving over a 242 msec. maximum latency. Trellis coding is then performed. On the receiver side, Maximum Likelihood Sequence Detection is applied in the receiver to decode the received frame level symbols . The data is de-interleaved before Viterbi decoding. The binary stream is then block de-interleaved and fed to the binary error correction processing. Scrambling is employed for both error protection and security of data access in the receiver, thus allowing complete control of user access to the transmitted data stream. Scrambling code keys can be downline changed from the transmitter.

The equalizer used is an adaptively adjusted Decision Feedback Equalizer utilizing fractional symbol feed-forward tap spacing ([2], [3]). (Refer to Figure 4) Symbol quantization is achieved by Mikros' patented Constant Cost HCP Decoding Algorithm for feedback. Trellis decoding is based on soft symbol Viterbi Decoding.

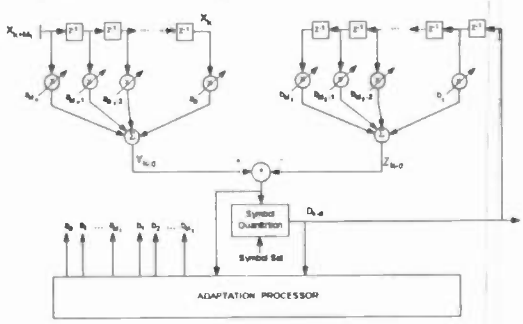

Figure 4 HSDS-SCA Equalizer

(b) *SCA Transmitter/Injector*

At the radio transmission site, Mikros provides a data feed insertion device which accepts one or more independent binary data feeds, at or below the user data rate for the network, and converts this data to a HSDS-SCA signal. The Mikros HSDS-SCA Generator provides the following functions:

1. Binary data feed bit scrambling.
2. Insertion of independent data feed subframes (144 bytes per subframe) into a continuous data stream for transmission.
3. Interleaving
4. Reed-Solomon and Trellis encoding
5. Binary-to-Symbol mapping.
6. Header/EOF field insertion and frame formation.
7. SCA waveform generation.
8. Local pilot extraction and sampling clock generation for DAC.
9. Injection level control for SCA insertion at the FM transmitter.

Additional features of the HSDS-SCA Generator include remote dial-up for status/problem reporting, multiple SCA subcarrier capability, data rate control for the widest possible applicability, station unique Generator ID, variable injection level control and continuous self-monitoring of on-air broadcast quality.

III. Performance

(a) *Mutual Interference*

The HSDS-SCA System offered by Mikros has been specifically designed with the rampant competition for SCA real estate in mind. Aside from the system's high bandwidth efficiency, the placement of subcarrier bands, as well as bandwidths (and therefore data rate), is quite flexible. The table below shows two subcarrier placement examples designed to control mutual interference with the common SCA subcarriers at 57 kHz (RBDS), 67 kHz and 92 kHz . The data shown illustrates the level of HSDS energy in the FM baseband at key frequencies. As can be seen in the table, the roll-off of the HSDS bands is more than adequate to prevent cross-talk interference into any adjacent subcarriers.

Table 1
FM-HSDS Band Level Measurements

Low Band (59.5 - 83 kHz)			
Injection Level	Frequency (kHz)	Level (dBm)	Rel. to Carrier
10%	53.0	-59.2	-73.8
	59.5	-57.5	-72.1
	83.0	-58.5	-73.1
	92.0	-58.9	-73.5
20%	53.0	-58.2	-72.8
	59.5	-49.9	-64.5
	83.0	-57.5	-72.1
	92.0	-58.5	-73.1

Table 1 (cont'd)

High Band (83 - 97 kHz)			
Injection Level	Frequency (kHz)	Level (dBm)	Rel. to Carrier
10%	67.0	-59.2	-73.8
	83.0	-57.9	-72.5
	99.0	-59.2	-73.8
20%	67.0	-59.5	-74.1
	83.0	-57.2	-71.8
	99.0	-59.2	-73.8

Undeviated Carrier Level: +14.6dBm
Low Band: 19K baud, 8 HCP, Subcarrier @ 71.25 kHz
High Band: 9.5K baud, 8 HCP, Subcarrier @ 91.0 kHz

(b) Reception Quality

Numerous experiments have been undertaken to assess the quality of the data received by the HSDS-SCA system. Figure 5 shows the behavior of a 76 kbps system in the presence of severe multipath in the commercial FM band. The chart shows the absolute best performance that can be achieved in the curve labeled "Upper Bound". The horizontal dashed line in the figure indicates the 2% bit error rate threshold, a practical performance limit for a usable transmission. We also show the behavior of our system to the GSM Urban multipath model (scaled to FM Broadcast frequencies) for comparison.

Figure 6 shows the comparative performance of our HSDS-SCA system, again at 76 kbps versus an existing 9600 bps CPFSK system under identical testing conditions. This data clearly shows the (E_b/N_o) superiority of Mikros' HSDS-SCA system to the low data rate FSK.

(c) Coverage Area

As of this writing, an HSDS-SCA system experimental prototype has been field tested over a large area of central and southern New Jersey and eastern Pennsylvania north of Philadelphia. The transmitter was located at the WPRB (103.3 MHz FM) site at the Princeton antenna site of NJ Public Broadcasting. WPRB radiates at 14.5 kWatt with an essentially omnidirectional radiation pattern. The antenna is at approximately

700' elevation with respect to the surrounding area. Hence line-of-sight is about 25 miles. The system was tested over this 2000 sq.mi. area with acceptable BER out as far as 22 miles. Performance was not uniform over the entire area. Poorest reception occurred to the North and East (toward New York and Atlantic City), with acceptable results mainly in the South and Southeast, as well as due West. Good reception to the Southwest (towards Philadelphia) occurred out to about 15 miles. Some of the typical results, expressed in terms of uncoded BER are shown in Figure 7.

IV. Application Areas

Although FMeX targets the FM SCA band and the radio data broadcasting market, the technology employed is widely applicable to wireless communications. In particular, the combination of high density digital modulation, adaptive equalization and DSP-based receiver miniaturization are directly applicable to most forms of Personal Communication Services (PCS), wireless networking, direct and two-way paging systems, and especially in any application where a low cost "last-mile" wireless connection is required.

Clearly, the most fully developed application area is still FM-SCA data broadcasting. However, Mikros' technology, as well as our on-going R&D into both AM piggy-back methods and general linear modulation applications is well suited to our goal of bringing technology to all segments of the wireless communications market place which optimizes spectrum efficiency for wireless communications.

References:

[1] Forney, G.D.; Gallager, G.; Lang, G. R.; Longstaff, F. M.; and Qureshi, S. U. "Efficient Modulation for Band-Limited Channels." *IEEE Trans. Sel. Areas of Comm.*, Vol. 2, No. 5, (Sept. 1984): pp 632-647.

[2] Monsen, P. "Theoretical and Measured Performance of a DFE Modem on a Fading Multipath Channel." *IEEE Trans. on Comm.*, Vol. 25, No. 10, (Oct. 1977): pp 1144-1153.

[3] Kobayashi, H. "Simultaneous Adaptive Estimation and Decision Algorithm for Carrier Modulated Data Transmission Systems." *IEEE Trans. on Comm.* Tech., Vol. 19, No. 3, (June 1971): pp 268-279.

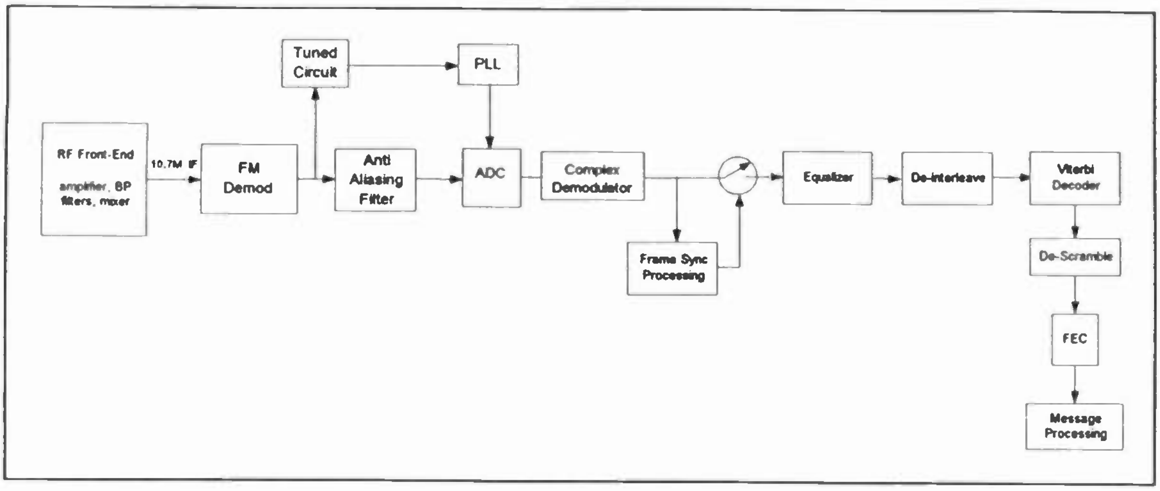

Figure 1 HSDS-SCA Receiver

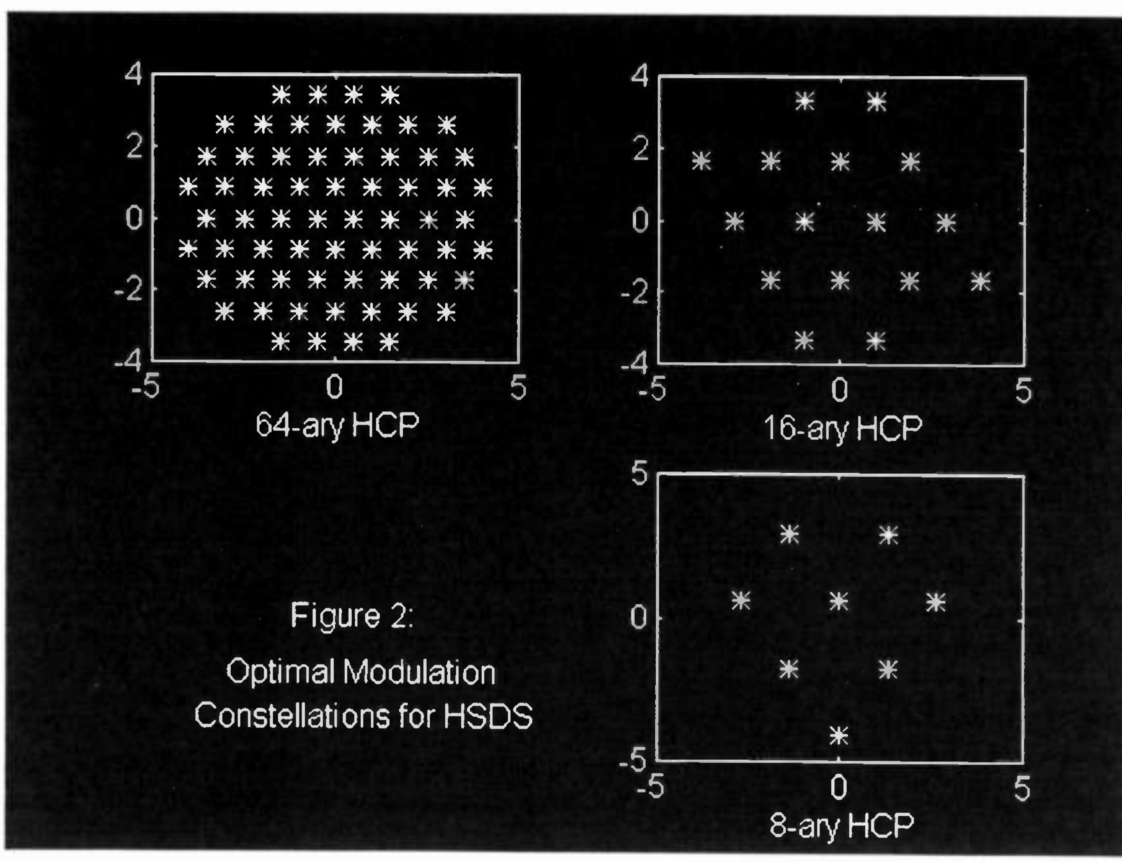

Figure 2 Modulation Formats

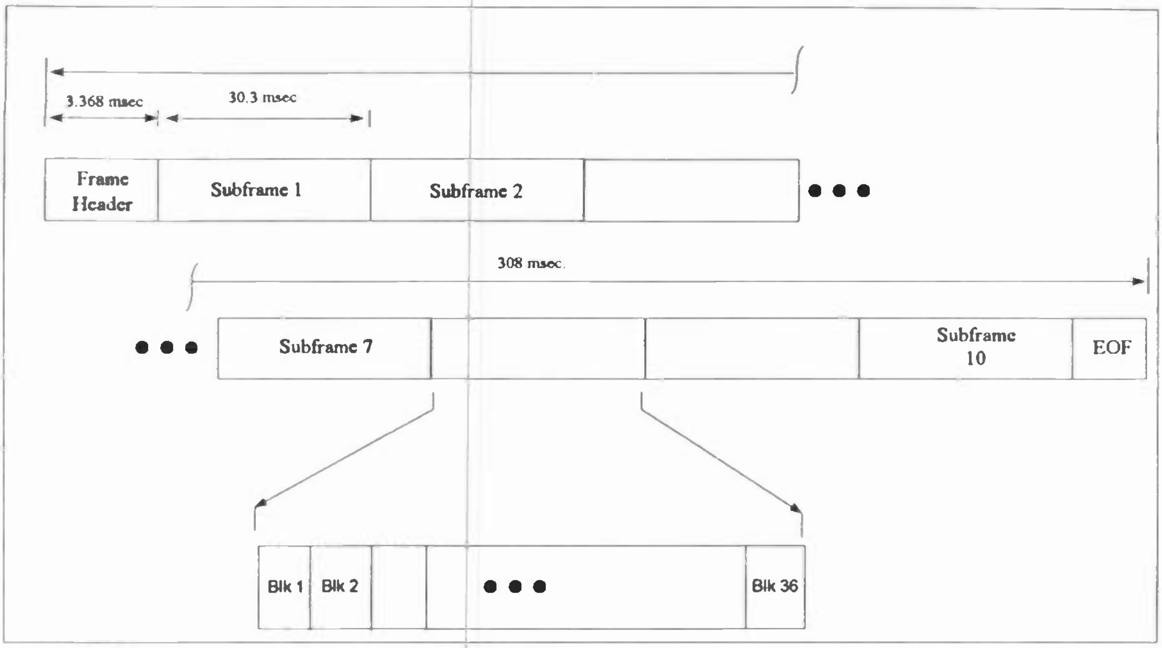

Figure 3 Frame Structure for FMeX HSDS System

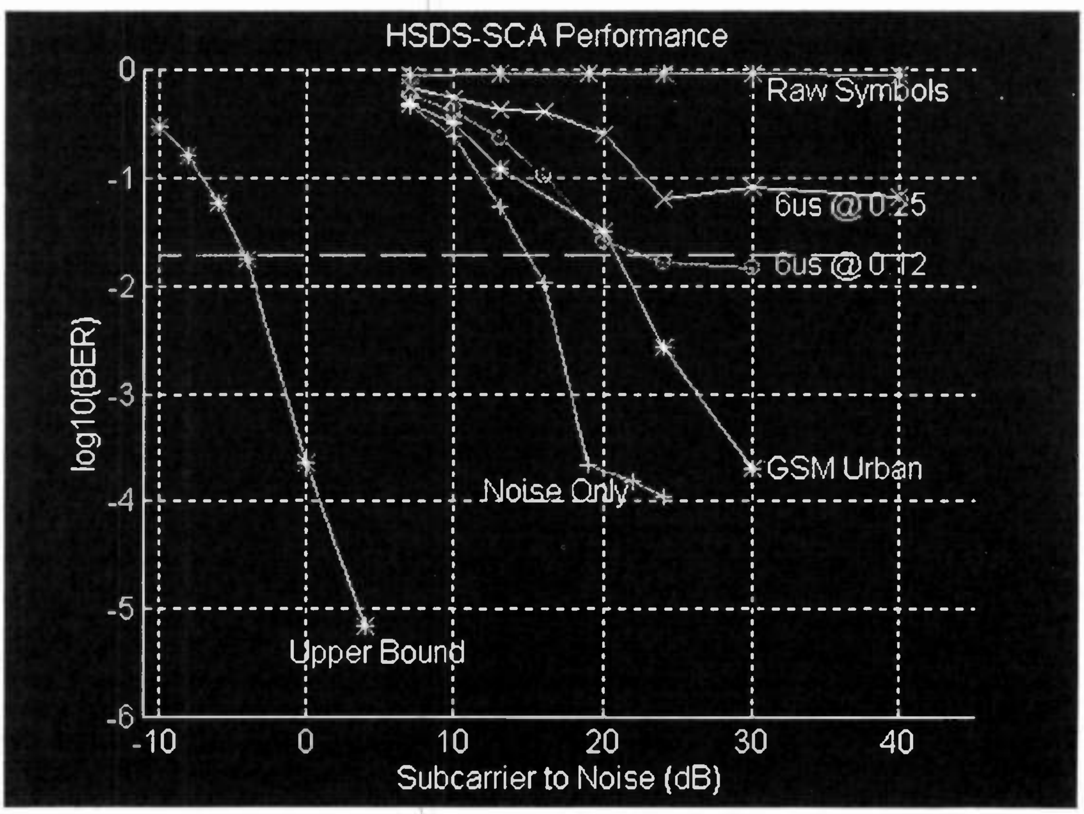

Figure 5 HSDS System Multipath Performance

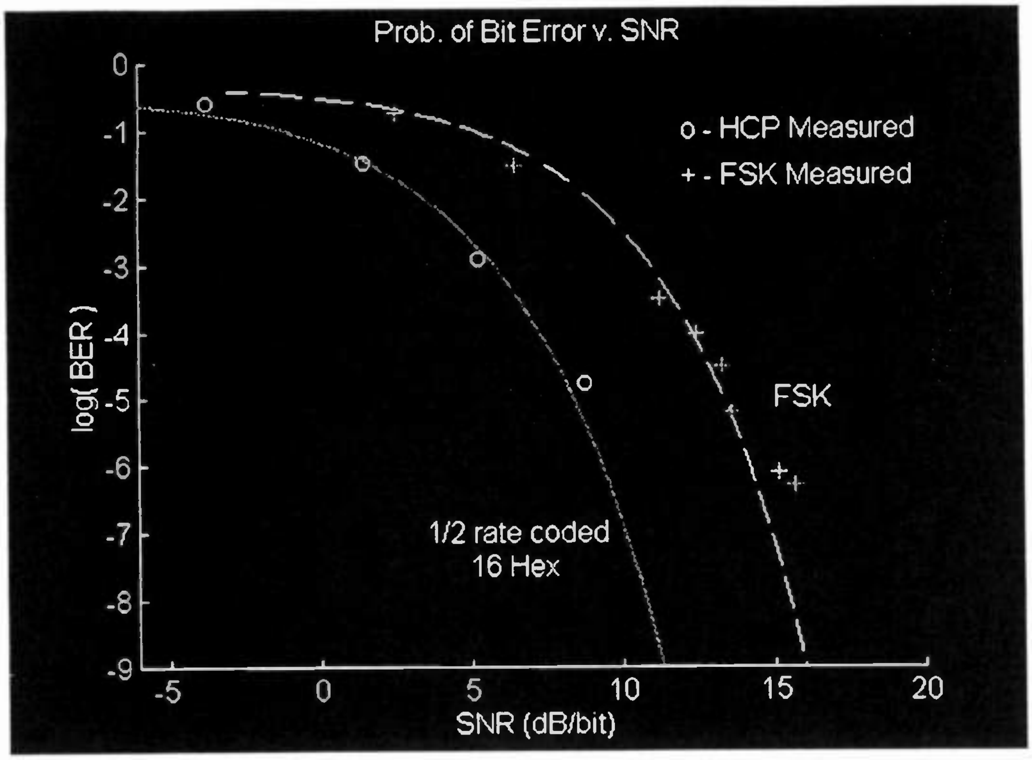

Figure 6 HSDS System Energy/bit Comparison to FSK

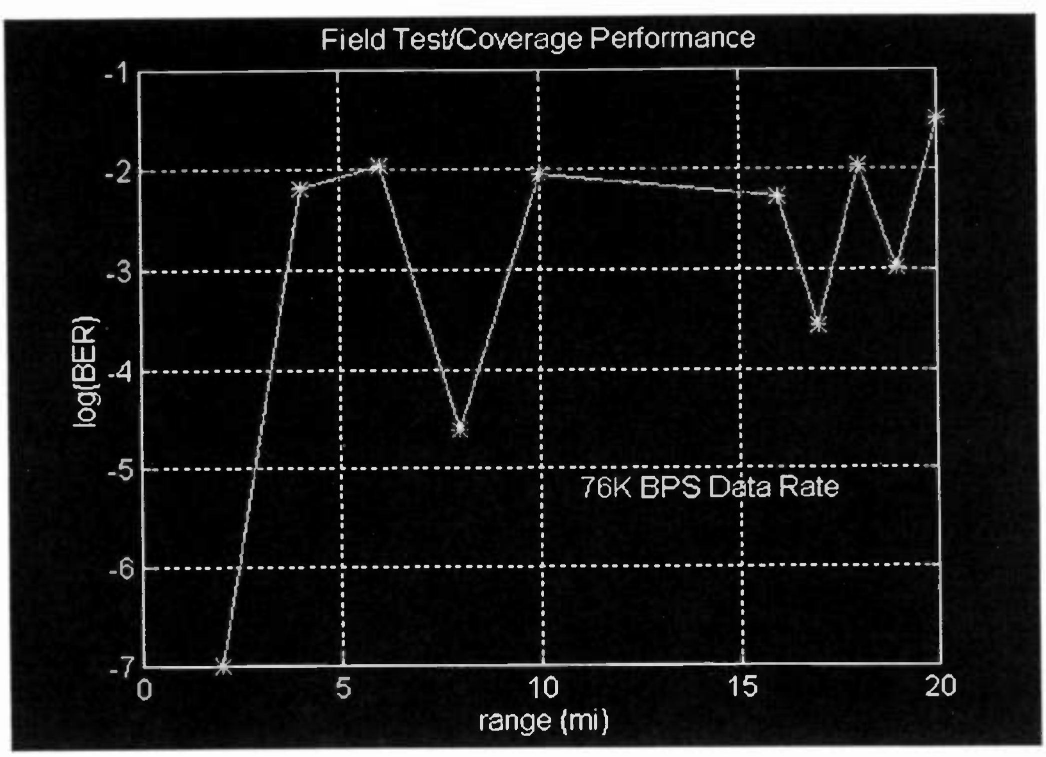

Figure 7 Field Test and Coverage Performance Results

VIDEOSERVERS: TECHNOLOGIES AND IMPLEMENTATION

Monday, April 15, 1996
1:00 - 5:00 pm

Session Chairperson:
Andy Butler, PBS, Inc., Alexandria, VA

STORAGE AS THE BACKBONE OF VIDEO SERVER PERFORMANCE
Andy D. Hospodor
Quantum Corporation
Milpitas, CA

***VIDEO SERVER SYSTEM INTERFACE AND CUTOVER**
Donald A. Messina
CBS-TV
New York, NY

VIDEO SERVERS: TECHNOLOGIES AND IMPLEMENTATION
David C. Guo
Micropolis Corporation
Chatsworth, CA

***MULTIMEDIA ASSET MANAGEMENT COMES OF AGE**
Roger LeMay
Illustra Information Technologies, Inc.
Oakland, CA

SELECTING THE RIGHT RAID DISK ARRAY FOR VIDEO
Bill Moren
Ciprico, Inc.
Plymouth, MN

***IMPLEMENTATION OF SERVER-CENTERED & NETWORKED BROADCAST TELEVISION VIDEO PRODUCTION SYSTEMS**
Craig Jones
Avid Technology, Inc.
Tewksbury, MA

IMPLEMENTATION OF MULTI-CHANNEL AUTOMATION
Barry Goldsmith
Drake Automation Ltd.
Welwyn Garden City, United Kingdom

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*Paper not available at the time of publication.

STORAGE AS THE BACKBONE OF VIDEO SERVER PERFORMANCE

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Quantum Corporation
Milpitas, CA

ABSTRACT

A world with 10,000 channels and movies on demand whenever and wherever imaginable appears to be more of a reality with each passing day. Yet what most people don't consider is the massive amount of storage needed to beam videos and television shows to viewers all over the world. Companies like Quantum are already manufacturing and further enhancing storage technology to serve the needs of the video server masses.

Along with exceptionally high storage capacities, video server technology requires disk drives with consistently high sustained throughput and data transfer rates. Quantum's AV-ready drives meet the demands of multiple simultaneous MPEG-2 video streams, a common video server standard. These enhanced storage devices ensure that the maximum amount of data fits into the smallest amount of space; enabling more information to reach viewers round the clock.

INTRODUCTION

Video servers provide multiple simultaneous streams of video to users. Movies are encoded using MPEG, and lately MPEG-II formats. The movies are commonly striped across several drives to distribute access load and permit RAID strategies. Each MPEG-II video stream requires approximately 420KB/second, and a feature length film requires 3 to 4GB of disk storage.

VIDEO SERVER ENVIRONMENT

The server environment typically performs large requests across multiple drives, with the majority being reads. The simulation presented here assumes a 99% Read ratio, and 128KB requests randomly made to disks. The simulated requests have exponentially distributed seek distances, with an average of 0.33 stroke. A further assumption is that each video stream has 1MB of buffer in the set top box. The maximum allowable response time is then:

$$\frac{1\ MB}{420\ KB/s} = 238.1\ ms \qquad (1)$$

Each set top box requires another 128KB of video data every 238 ms, otherwise frames are dropped and degrade the quality of the movie. The response time requirement is stated simply as 200 ms in the remainder of the paper to allow for delays associated with the disk array and network.

$$\frac{420\ KB/s}{128\ MB/IO} = 3.3\ IOS/sec \qquad (2)$$

Each video stream is broken into 128KB chunks and transferred to the set top box. The MPEG-II rate translates into a system load of 3.3 IOS/sec per video stream. The single video stream is simply stated as 4 IOS/second in the remainder of the paper to permit encoded data rates up to 512 KB/s.

$$1\ Video\ Stream \cong 4\frac{IOS}{sec} \quad (3)$$

BASELINE DISK PERFORMANCE

Atlas is a 4.3 GB disk drive rotating at 7200 RPM. Data is stored on 10 disks, and accessed by 20 Thin Film recording heads. Data is buffered using a 1MB DRAM, located between the recording channel electronics and the 16 bit SCSI bus. The recording channel employs the common Peak Detect method of retrieving user data.

Recall that a single video stream requires four IOS/second. System cost, measured in $/stream, is reduced whenever the same set of disks can service more users. Figure 1. shows that the Atlas drive supports up to 25 IOS/second, or **6.25 simultaneous MPEG-II video streams**.

EFFECT OF ADVANCED RECORDING CHANNEL

Atlas II is a 9.1GB disk drive rotating at 7200 RPM. Data is stored on 10 disks, and accessed by 20 Magneto-Resistive recording heads. The data buffer and interface are similar to Atlas, however, Atlas II employs a faster, more advanced recording channel.

The difference in performance between Atlas and Atlas II is primarily due to differences in the recording channel. The faster recording channel of the Atlas II supports up to **8.1 simultaneous MPEG-II video streams,** a 30% improvement over Atlas.

EFFECT OF ZERO LATENCY READS

Both Atlas and Atlas II contain special hardware to allow Zero Latency Reads (ZLR). ZLR allows the drive to read blocks out of order, eliminating rotational latency on long transfers. The on-board disk drive buffer re-orders the blocks so the user see the proper order.

Performance increases from 6.25 to 6.5 simultaneous video streams on Atlas. Atlas II performance increases from 8.1 to **8.3 simultaneous MPEG-II video streams**.

CONCLUSIONS

An advanced recording channel and ZLR enhance the performance of disk drives operating in MPEG-2 Video Server environments. The addition of the advanced recording channel and ZLR results in a performance improvement of 33% over the previous generation disk drive. These features of the Atlas II disk drive could potentially service 33% more users than existing technology. This translates into a reduced cost per video stream.

	Streams	Streams wZLR	%Imp.
Atlas	6.25	6.5	4%
Atlas II	8.1	8.3	3%
% Improvement	30%	28%	

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Fouad A. Tobagi, et. al., "Streaming RAID - A Disk Array Management System for Video Files," ACM Conference on Multimedia, Anaheim CA, August 1993.

Leana Golubchik, John C.S. Liu, Richard Muntz, "Reducing I/O Demand in Video-On-Demand Storage Servers," pp 25-36, ACM Sigmetrics 1995.

Atlas Disk Drive Product Manual, Quantum Corp, Milpitas, CA, June 1995.

KB - Kilobyte = 1024 Bytes
MB - Megabyte = 1 048 576 Bytes
ms - millisecond = 0.001 second
GB - Gigabyte = 1024 MB
RAID - Redundant Array of Independent disks
SCSI - Small Computer System Interface
ZLR - Zero Latency Read

Dr. Hospodor manages the Advanced Storage Applications Group at Quantum Corp. Prior to joining Quantum, he was a research fellow with the Institute for Information Storage Technology (IIST) at Santa Clara University. He has also worked with National Semiconductor, Scientific Micro Systems, and IBM, designing and developing storage controllers and devices. Dr. Hospodor received his B.S. degree in Computer Engineering from Lehigh University in 1981, and M.S. and Ph.D. degrees in Computer Engineering from Santa Clara University in 1986 and 1994. He is currently senior member of the IEEE, active in the Magnetics Society and Computer Society. Dr. Hospodor may be reached via email at ahospodo@qntm.com .

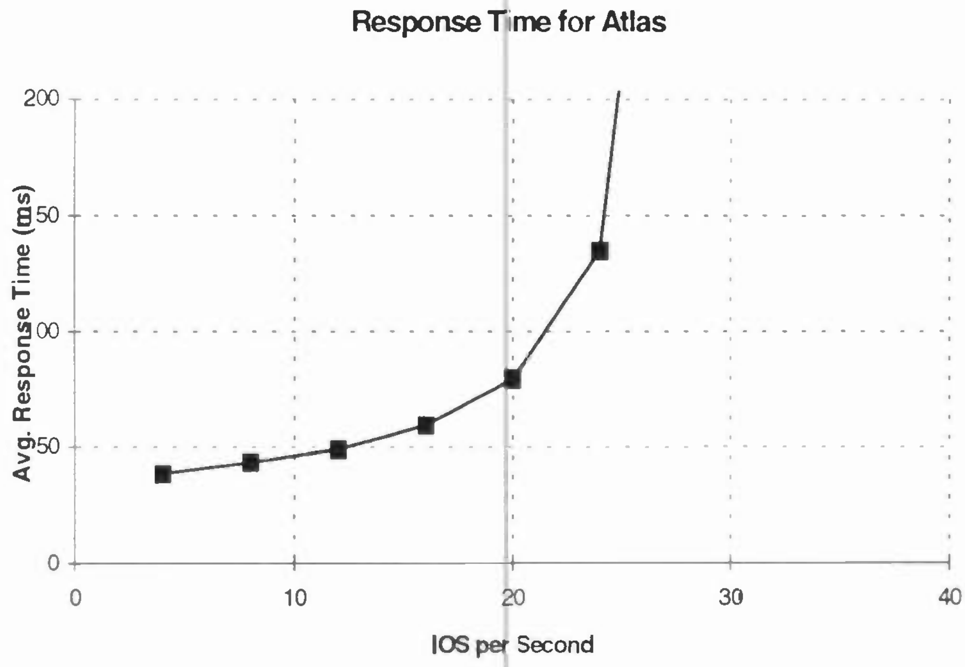

Figure 1. Atlas-II Disk Drive Performance

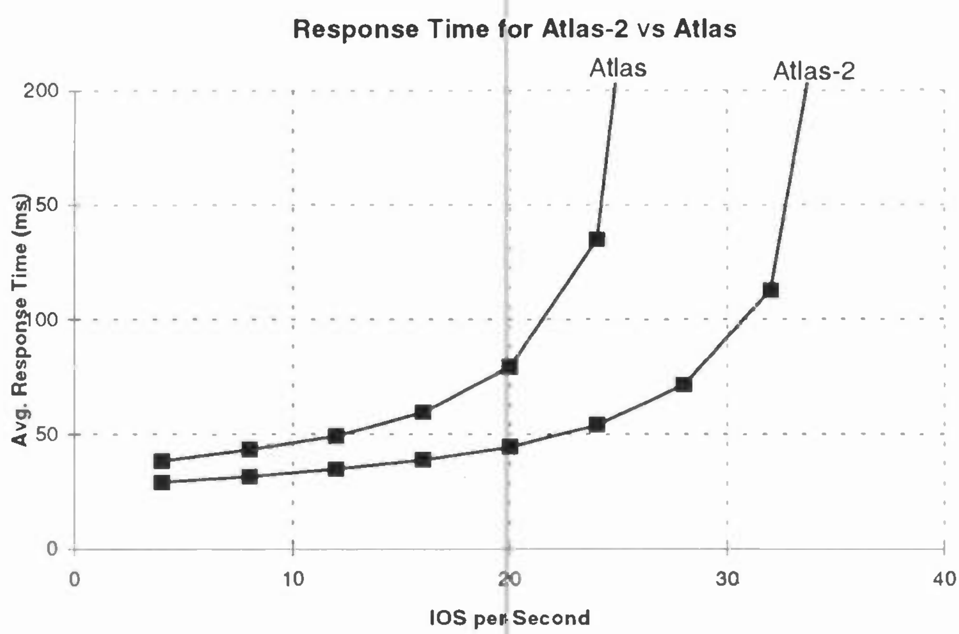

Figure 2. Atlas-II Disk Drive Performance

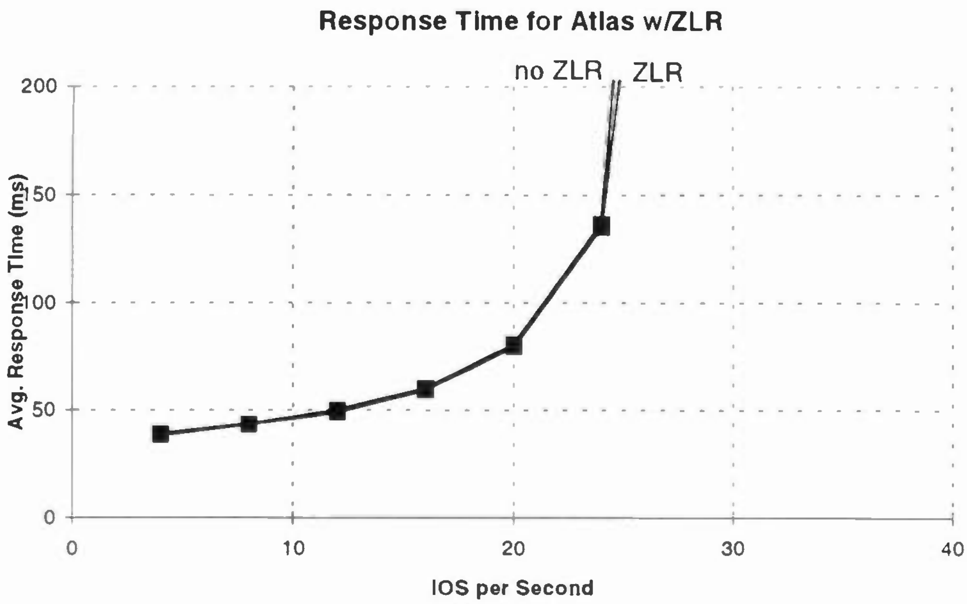

Figure 3. Atlas Performance with ZLR

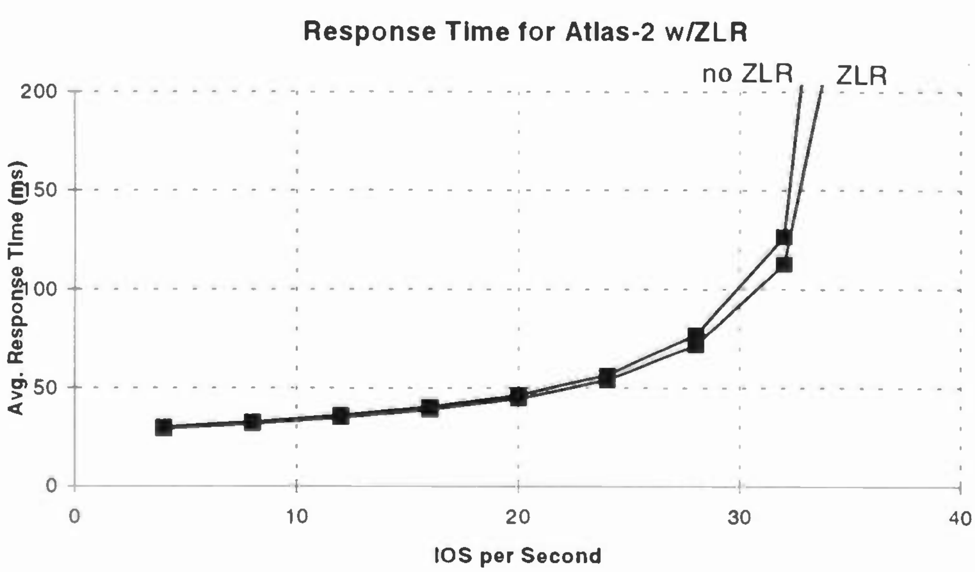

Figure 4. Atlas II Performance with ZLR

VIDEO SERVERS: TECHNOLOGIES AND IMPLEMENTATION

David C. Guo
Micropolis Corporation
Chatsworth, CA

ABSTRACT

In 1993, we saw wide acceptance of digital video. In 1994, we saw the first video servers emerge. In 1995, we saw convergence of technologies to make digital video delivery and broadcasting possible. In 1996, the focus is video server applications.

As competition looms, the focus in 1996 is specific applications. By now, the market and product requirements are well defined. The winners and losers will be determined by how well each product meets the needs of the customer -- specifically, cost performance.

An examination of each market segment from broadcast to video-on-demand brings us to one conclusion -- it is a supply side problem, not demand. Everything ultimately comes down to cost-per-stream and affordability. If traced from one end of the system to the other, it is evident that access to storage, not capacity, is the problem; it is the ability of the disk drives to meet the I/O requests. Here lies the challenge for video server manufacturers, system designers, and commercial deployments in 1996.

THE GREAT ARCHITECTURE DEBATE

In 1994, with the emergence of the first video servers, several industry icons proffered their designs to the eager public. First came SGI with mega-Gigahertz bus Challenge, then came HP with the loosely coupled computing architecture of the Media Streamer; then it was Oracle/nCube with a hypercube, massively parallel processor architecture, followed by DEC with their multi-Alpha processor-based Alpha Server. As no great surprise, Micropolis also introduced a video server based on their disk centric storage architecture.

Looking back, it shouldn't take a crystal ball to see that we were being sold "recycled" products. Everyone seems to have provided a new slant on their existing core technology and retooled an existing product to be able to "pump" video. Granted, it wasn't easy; no one really knew what the market wanted, except that the product had to do everything, and it had to be inexpensive. And then of course, there was Microsoft Tiger which claimed to utilize existing, standard, off-the-self hardware to thoroughly confuse everyone.

By mid-1994, the video server architecture debate focused on cost per stream -- the magic number was sub-$500. In addition to that mind boggling number, no one seemed to agree on what a stream was. So like all pioneers, we marched forward, admittedly somewhat blind. Then in late 1994 and early 1995, came the true test -- who could deliver and how much. By now Full Service Network had already begun the installation and trial of their SGI Challenge Servers, but to everyone's amazement, a technology trial suddenly turned into a marketing trial. Early numbers started to trickle in. HP initially said under $3,000 per stream for 500 streams. nCube said $500 per stream for 5,000 streams. But, in reality everyone was all over the map because none of the products had the same configuration of I/O ports, storage, and/or system management capabilities.

VIDEO SERVER ARCHITECTURE 101

Believe it or not, it all starts with disk drives. This is only logical because the disk is where the data is stored. How quickly you can access the data and how fast you can read or write that data determines how many "streams" or people you can service.

The typical 5400 RPM drive is capable of servicing an average of 4MBytes per second or 8 streams at 4Mbits per second. Of course, these numbers change if there is a lot of random access or a requirement for different bit rates, but for typical movie watching, 8 streams at 4Mbits per second is a safe bet.

The next equation is to determine capacity. This is an easy computation based on the data rate of the video material multiplied by how many seconds, minutes or hours you want to store. Additional drives can be added for redundancy.

The next part gets complicated. To get the data from the drives to the video network (may it be analog or digital), requires some sort of routing device. This is where most video server manufacturers differ on their design. Some computer manufacturers, like SGI, are used to putting data on a super fast CPU bus and "pushing" the data out. Others like HP and DEC choose to use multiple CPUs and buses to do the same thing. Still others like nCube choose a switch type technique using their MPP architecture. But the net result always has to be the same -- bits of data from all the different disk drives must travel through the system to be arranged and ordered into a video stream and then piped out the right port for the right user.

The major cost discrepancies exist in the routing devices since every video server, large or small, digital or analog, must use the same number of video network ports and the same number of disks to support a fixed requirement, i.e. X number of streams by Y number of video clips of Z length. The only real difference is in how the data is routed. Of course, the elegance of any design is 100% scalability, and that is just as elusive as the Holy Grail. But at least with a simple exercise of counting the number of drives and number of ports a bench mark can be established for almost 75% of the cost of the competitive range of servers, if not more.

The last 25% of the server leaves you to debate on how much you want it to do. If you want it to generate customer invoices, you might be asking too much. My simple philosophy is that the less a server does the better, which is to say that it should be efficient and cost effective, and thus more or less modular.

EARLY ADOPTERS

The standard answer from a video server salesman when asked how much his equipment costs was typically, "it depends". Even if you knew how many streams, what type of interface, and how much storage, no one really knew how much it was going to cost to get the system up and running at a specific site because every site was different. So where does this lead us? Quickly to nowhere. Then in mid-1995, a few potential customers on the industry fringe showed up and said, "we're not interested in interactive television for consumers, but we do have a need for a video server that does exactly *this* and costs no more than *that*.." That's all it took for the engineers and finance people to understand that specific applications were the driving force for video servers. With this realization, the industry took a giant step forward.

APPLICATIONS

At first there were the hotel pay-per-view operators who wanted to find a more efficient way to update movies. The by-product of having movies-on-demand was just a bonus. Their incentive was to save the labor cost at each site, thus the requirements were simple: replace the 32 video cassette players at each hotel.

What they needed was a server with 32 channels that holds 25 movies at 3Mbits per second, an MPEG-2 analog video and audio output, a satellite data port, and a simple command protocol to play and stop movies. All of this equipment at no more than $40,000. They demanded just one must-have feature, it must be cheap, even at the expense of redundancy.

The second application to materialize into a market was cable ad-insertion. The estimated number of cable headends engaged in ad-insertion was 2,000 - 3,500.

The requirement was for a 16 channel video server with MPEG-2 output capable of sustaining at least an 8Mbits per second data rate. The system, however, had to have more stringent performance tolerances since miss-delivering one ad was at least 1,000 times more expensive than a movie in a hotel room. The most important feature on an ad-insertion server is video quality; thus the requirement for full CCIR-601 compliance on video output. Second, full chroma genlock input was required to maintain network video synchronization. And third, the system must be accurate to at least 100 milliseconds in execution and logging of each command; the entire video network requires proper synchronization and verification of each play out.

Aside from these 3 must-haves, the only other requirement is price, which is pegged at $40,000 for a 16 channel system with 24 hours of storage using 8 Mbits per second MPEG-2 video. A Satellite link option is also desirable, along with capabilities to be high speed networked with an MPEG-2 encoding system.

Another application which has re-emerged as a real market is Near-Video-On-Demand (NVOD) or Pay-Per-View (PPV) for cable operators. The system is essentially the same as the hotel system, however, a cross functionality is required for systems that also deliver ad-insertion. The most common scenario is to build an ad-insertion system with additional capacity for PPV.

Interactive television (ITV) is taking a back seat to the Internet. The servers once touted to serve 3,000 streams of video are now being retooled to do the same over the Internet. Currently, the Internet is just not quite ready to run video, but with the advent of cable modems things could change quickly. The truth of the matter is that whether it's the Internet or ITV, the problem with servers is almost identical: how to get massive amounts of data on the network.

Post production servers for editing will remain computer-based systems like SGI's because of the processing power required to generate and edit images, video and audio. The system is likely to consist of powerful graphics engines to manipulate images and video. A large storage system is used to sustain very few streams at very high data rates and to hold uncompressed video and graphics data. The whole system will likely be networked with high power video processing systems using Asynchronous Transfer Mode (ATM) technology at 155Mbits per second. The resulting network will also consist of MPEG-2 compression systems that produce an MPEG program stream ready to be aired.

Digital Satellite Systems (DSS) and Digital Video Broadcast (DVB) systems are very similar to ad-insertion systems due to the stringent demands of broadcasting. Whether DVB is received via satellite or delivered through cable, the system architecture on the back end is likely to contain a digital RS-422 interface.

Multiple MPEG streams are fed through a multiplexer to create the DVB programming. The DVB system is essentially a real-time digital video encoding system since most programming aired is provided in analog format. The video server is mainly used for ad-insertion or for programs which are repeated through multiple channels, thus the output and storage requirement is relatively low.

THE VIDEO SERVER EVOLUTION

As is the case for all industries and products, there will be an evolutionary path which all will take that will make things more homogeneous. This is the result of standardization of technology and integration of products with other systems. Video servers take no exception.

The eventuality of the video evolutionary process is -- I hate to admit this -- the vision which Bill Gates had all along. The only way to make things cheap is to use existing, high volume, standard hardware, and of course Bill's software, (because once it comes out it will be so widely accepted that no one will use anything else, so you shouldn't either). Which means that every video server will look like every other server today. So what does this mean? Eventually, my guess, is that most video servers will be more or less identical to data servers today. They will be mostly based on Intel processors on a PCI bus architecture. They will have a fast network adapter such as ATM or Fast Ethernet, and will have some sort of RAID storage subsystem. And of course, they will be running Bill's new operating system like NT, Bill's new server software such as Microsoft Back Office and Bill's new Internet server software, Gibraltar WebServer.

So, as they say, the network does the computing. But in reality, you still need to solve the storage problem because one disk drive by this time, I presume it will be 1997, will still only be able to deliver no more than 10-15 streams. And the problems of accessibility and throughput we face today will no doubt remain the same.

MATRIX STORAGE

A fresh new view of storage is taking shape as network computing architectures are being rapidly adapted. Much like what networking brings to computing, i.e. the accessibility of data and sharing of information among peers; intelligent storage systems are quickly changing the paradigm of monopolistic service. The slave storage device to one computer is now able to service multiple "hosts", but a bus-based architecture such as SCSI faces the same arbitration limitation as a computer bus.

The break-through is in switch-based storage, where an array of computers can access a array (or array of an array) of disks through a switch matrix. Using techniques such as Synchronous Time-Division-Multiplexing (STDM), each computer server is given access to the entire array all the time. The problem of accessibility to data is thus removed. No longer is data duplicated because of bandwidth limitation of one server or one storage device. With data stripped across the entire matrix, accessibility is guaranteed, with improved data throughput using parallel transfers. The underlying technology which Micropolis developed is a Serial Storage Architecture (SSA) switch. The SSA matrix couples an array of disk drives to an array of computers running cluster type operating system technology. The clustering network is used for read/write coordination. Effectively, this means that each host or computer server sees the storage array through its switch ports as its own and can address each file as such. A host of other applications can now be built on top of this operating system with seemingly unlimited storage resources. (In fact many already exist today.)

The side benefits to Switch Storage Technology (SST) is the reduction of routers and bridges in a network. If storage is centralized, with each server given 100% access and availability to data, the network can be simplified by reducing the number of interconnects between work groups. In fact with Matrix Storage, no interconnects are necessary, because one server no longer needs to supply data to another over the network. The net result is reduced network traffic, and thus more efficient network utilization.

CONCLUSION

The debate is not over, or rather, the debate is not about who has the best server design architecture, (because it will all become generic), but who has the best storage system. Ultimately, it is the storage system that determines system efficiency and effectively the price.

SELECTING THE RIGHT RAID DISK ARRAY FOR VIDEO

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INTRODUCTION

The convergence of video production, broadcast, communications, and digital computing has spawned a new class of computing system, the video server. A video server is related to its cousin, the file server, because both 'serve' data to fulfill requests from clients. The term 'serve' refers to the action of retrieving data from digital storage and forwarding it to the user, via a communication network. Video servers, notwithstanding their digital similarities to file servers, have performance demands placed on them which make the architectures employed quite unique. A key component of these architectures is high performance redundant disk storage.

THE VIDEO SERVER PARADIGM

Video servers are used to deliver digitized video data to a communication outlet for distribution of the video, in real-time. Upon request or on a pre-programmed schedule, the video server must retrieve video (which has been digitized and compressed) from on-line storage, forward it through its system buffers to a communication outlet for transmission to the viewer. This process must be performed in real-time to ensure the video is delivered at a continuous 30 frame per second (fps) rate. A typical video server environment is shown in *Figure 1*.

The video server itself is typically a powerful workstation with hardware and software designed to deliver video in real-time. The hardware architecture of video servers must allow for many streams of video to move from the on-line disk storage through internal system buffers and out through communications connections. While any of these components could limit the server's performance, it is the disk subsystems which, if improperly designed, has the potential to degrade performance by fifty percent or more over optimal configurations. Further, as the disks in a video server account for about fifty percent of the server's cost, anything less than optimal configurations can dramatically impact the server's cost effectiveness.

Unlike file servers, which manipulate data of any type, the video server's sole task is to retrieve video data. The video data stored on a video server consists of programming which has been digitized and compressed. Typically there will be a large number of different titles stored in the video server, of arbitrary length (depending on the type of video - movies, news, commercials, etc.). Each title may be played at differing times. Further, multiple copies of the same title may also be active, simultaneously. Advanced server software, typically found on systems serving metropolitan area systems, may schedule multiple requests for the same title into slots to eliminate redundant data requests. For each slot, regardless if a single or multiple video requests are being serviced, the video server must supply a single video 'stream'.

The sequence of digitized data that comprises a

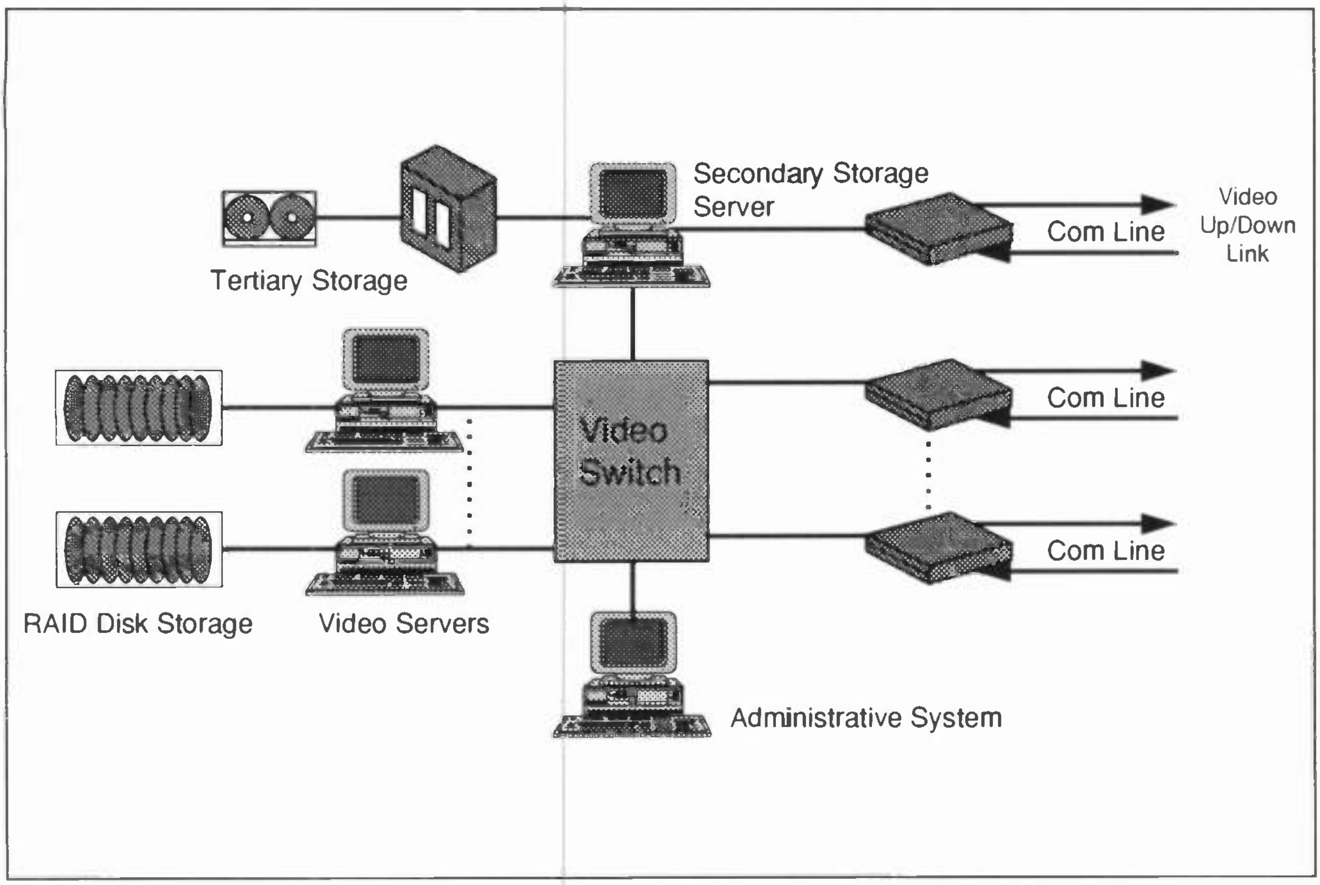

Figure 1 - Video servers are used in conjunction with video switches, communication links, administrative systems, and various methods of downloading content.

piece of video, when transferred in such a fashion as to satisfy a viewer's request, is said to be a stream. A stream is sequential chronologically, though it isn't required to be physically stored sequentially. In practice, however, video streams are typically stored sequentially for performance reasons. If a viewer pauses or rewinds, a new stream is generated when the playback resumes, at least from the video server's perspective. Video servers typically support numerous simultaneous streams. For performance modeling, a random distribution between streams is often assumed. Video steams also have isochronal characteristics. Specifically, each frame in a video stream must be delivered *every* thirty three milliseconds (time per frame at 30 fps). An *average* of thirty-three milliseconds is unacceptable for video playback because some video might fall below the minimum rate. Within the server, the method used to provide isochronous performance may accommodate some components inability to operate truly in an isochronous mode. For instance, system buffering may allow disk subsystems to load a sequence of video into the server at a rate much faster than 30 fps. The server then transfers out of the buffer at precisely a 30 fps rate. Fault-tolerance is another key video server trait. It is not unusual for video server applications to operate twenty four hours per day, seven days per week.

VIDEO SERVER DISK STORAGE

The fundamental objectives for video storage are straight forward. First, the storage should be as cost effective as possible. This implies that the storage subsystem employs an architecture that delivers the highest stream to spindle ratio possible. This ensures maximum performance for the lowest cost. Second, there must be adequate capacity for the total content to be available online. Finally, the storage must be fault tolerant, enabling continuous real-time operations.

Individual disk drives alone do not satisfy the objectives set for a video server's storage. A single drive delivers only a few streams of compressed video. To determine the number of streams a disk can deliver, there are several variables to consider: the request size of the stream, the drive's sustainable bandwidth, the drive's access latencies, and system overhead.

The total number of streams (S) a drive can support is the ratio of the time per request at the compressed video rate (Tv) and the time per request at the sustainable disk rate (Td) $\{S=Tv/Td\}$. A request is the transferring of an arbitrary amount of data in a unitary disk operation, and typically encompasses many frames of video.

Tv (time at video rates) is simply the amount of data requested (L) divided by the video stream rate (Rv), after compression (typically 1 - 15 Mb/s). Td (time at disk rates), on the other hand, consists of the sum of the drive's latencies and the time for the actual data transfer (Tx). A drive's latencies consist of the a seek (Ts), a rotational period (Tr), and overhead (To). Data transfer time (Tx) is request length (L) divided by the drive's data rate (Rd).

Expressed algebraically, the total number of streams a single drive can support is:

$$S^1 = L / ((L/Rd + Tx + Tr + Ts + To) * Rv)$$

For example, a 7200 RPM drive, with average access times of around 13 milliseconds and transfer rates of around 6 MB/sec will support approximately 15 streams of video compressed to a rate of 3 Mb/sec. This same drive, if worst case performance is evaluated (full-stroke seeks, full revolution rotational latencies), only supports about 10 streams of video at the same compression rate. For better quality video streams (e.g. lower compression), which require higher data rates, the number of streams supported by this class of drive drops. *Figure 2* depicts graphically the performance of this type of drive for varying data request lengths (L).

While the performance of a single drive has at best modest stream performance, the capacity is also rather small in the context of video. Video compressed to 3 Mb/sec requires over 22 MB of storage for one minute playback. On a 4 GB disk, approximately 180 minutes of capacity are available, enough for only about 1.5 movies. For a server with any reasonable total capacity, many spindles will be required.

Regardless of the total number of drives used in a video server, media redundancy is required to ensure on-air real-time playback. Even though mainstream drives have high Mean Time Between Failures (MTBF) ratings, they do fail and at a rate that may be surprising. For instance, for a family of drives with an MTBF of 800,000 hours, the expected failure rate over the service life of the drives is over 5%[2]. This analysis assumes no drive design failures in either the hardware or microcode, nor any manufacturing process problems endemic to a particular lot or facility.

RAID FOR VIDEO SERVERS

Fortunately there is a storage technology referred to as Redundant Arrays of Independent Disks (RAID)[3] which address the performance, capacity, and redundancy needs of video servers. RAID was conceptually presented in a paper published by the University of California at Berkeley in the mid-1980s. The paper offered a series of data storage architectures which provided media redundancy, large capacity, and high performance. The architectures are colloquially referred to as *RAID*

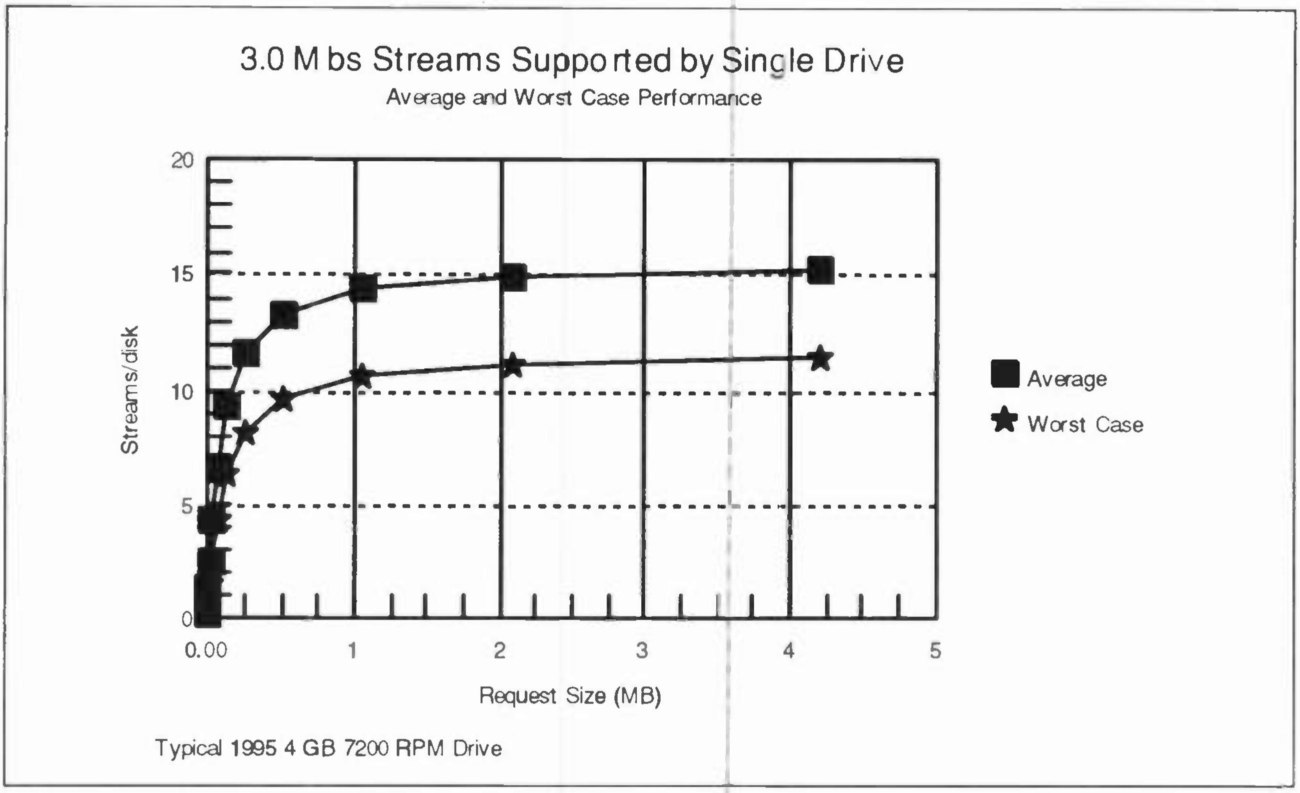

Figure 2 - 3.0 Mbs video streams supported by a 7200 RPM disk.

levels and were arbitrarily numbered one through five to identify each level[4].

RAID level 1 (RAID-1) is disk mirroring. Mirroring is a technique common to mainframe storage architectures and pre-dates the RAID paper. It was used as a frame of reference in the paper for the other RAID levels presented. Mirroring provides redundancy by simply duplicating every disk in the storage system. The remaining RAID levels incorporate a data striping technique in which data is evenly divided across a group of data drives. Error correction information, which can be used to regenerate the data on a failed drive is stored on a redundant drive. RAID levels 2 and 3 stripe the most elemental unit of disk data, the block, across all data drives. The difference between these two levels is the redundancy technique, in which RAID-2 uses multiple redundant drives while RAID-3 uses a single drive. Because RAID-2 offers no significant benefit over RAID-3 and has higher costs, it is not considered a commercially viable alternative, and won't be considered further. RAID levels 4 and 5 stripe blocks, with a single, or group of blocks, entirely contained on a single drive. Like RAID-2 and RAID-3, the difference between RAID-4 and RAID-5 is in the method of redundancy. RAID-4 stores its error correction data on a dedicated drive while RAID-5 distributes this information across all drives. While RAID-4 may be easier to implement, it offers lower performance and no cost savings as compared to RAID-5, and it too will not be considered further.

RAID-1, RAID-3, and RAID-5 share a common trait; any single drive in a RAID configuration may fail and all the data stored in the RAID will remain accessible. The similarities end there. Each of the RAID levels have differing levels of normal performance, performance after a failed drive, and media costs. These differences define the suitability of these RAID levels for video server applications.

SELECTING THE RIGHT RAID FOR VIDEO SERVERS

While the old adage, "your mileage may vary", is appropriate when considering different vendors RAID implementations, a good understanding of the underlying RAID principles will remove any doubt as to the upper limits of each RAID level. To compare objectively the RAID levels 'n' drive configurations (e.g. an arbitrary though equal number of drives) of each will be considered. *Table 1* compares four important metrics for each RAID level.

RAID Level	Performance - Normal	Performance - Failed Drive	Usable Capacity	Cost for n Usable Drives
1	n	n-1	n/2	2n
3	n-1	n-1	n-1	(n+1)/n
5	n	n/2	n-1	(n+1)/n

Table 1 - Performance and cost characteristics of RAID levels for video server applications

The first metric to compare is the media cost. It is in this comparison that the primary detriment to RAID-1 becomes apparent. RAID-1, because it must duplicate every disk, requires twice the total number of drives needed for any arbitrary capacity objective. As a result, the total cost for a usable capacity equal to n drives is $2n$ (100% premium). By comparison, RAID-3 and RAID-5 (RAID-3/5) only use a total of one drive's capacity for redundancy.

As a result, the premium for redundancy is only a fraction of the total capacity purchased. A five drive RAID-3/5 only requires one for redundancy, 20% of the total installed. As the width of the array increases, the premium decreases even further. A nine drive array, eight of which are usable, has a redundancy premium of only slightly more than 10%. In general, the cost for n usable drives in a RAID-3/5 configuration is $(n+1)/n$.

Figure 3 graphically illustrates the difference in redundancy premiums for RAID-1 and RAID-3/5 as compared to just a bunch of disks (JBOD), with no redundancy whatsoever. In addition to RAID-1's much higher cost for redundancy, the large number of drives associated with RAID-1 also increase the packaging and cooling complexities of a system, while lowering the overall reliability due to the larger total number of components. As a result, RAID-1 will not be considered further in this paper.

THE STRIPING RAIDS - RAID-3 AND RAID-5

The popular convention today is to consider the striping RAIDs for use in video servers. Since the media costs are identical for both RAID-3 and RAID-5 the primary consideration in selecting one or the other is the comparative performance capabilities. Video servers operate in real-time environments. As a result, the performance of a RAID under *all* operating conditions is the crucial consideration. While it is important to consider the performance of a RAID during normal operations, when all drives are functional, it is equally important to consider the performance of a RAID after a single drive has failed[5], also a normal operating condition. This consideration is required because a video server is typically guaranteed to deliver a minimum level of performance under all operating conditions. Therefore the lowest performance capability of the RAID under any operating situation it may encounter is the specification that dictates the server's specified performance.

RAID PERFORMANCE - ALL DRIVES OPERATING

Because one drive in a RAID-3 is dedicated to redundancy, it cannot contribute to data transfer performance. However, the remaining drives are used for data operations. Because of RAID-3's parallel striping technique, the sustained transfer rate approaches the media limits of the data drives. For example, a RAID-3 in a 4+1 configuration

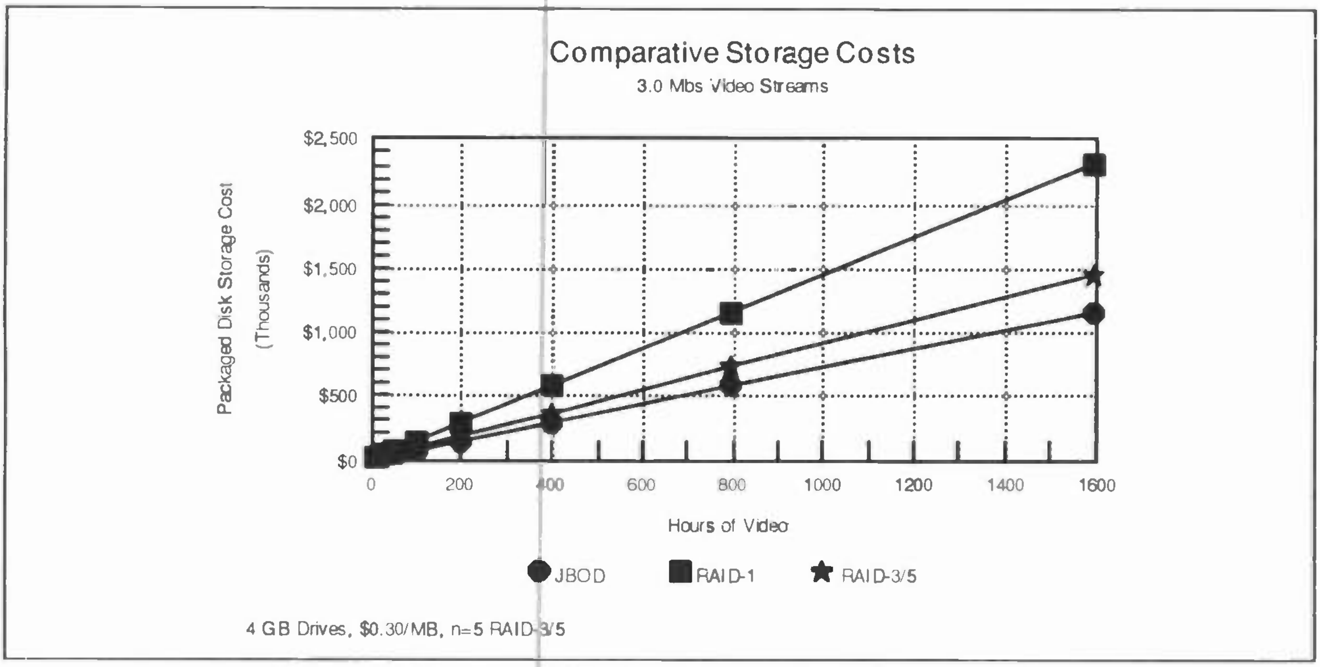

Figure 3 - Media costs for RAID-1 is significantly more than for RAID-3/5.

(4 data, 1 redundant drive) will have a sustained transfer rate approximately equal to four times the sustained transfer rate of an individual drive. In general, for a *n* drive RAID-3, the total number of streams supported is equivalent to *n-1* drives, with adequately large data requests.

The analysis for a RAID-5 is a bit different. RAID-5 distributes redundancy information across all *n* drives. Further, each drive is accessed individually. This enables each of a RAID-5's drives to service a data request simultaneously (providing the data requests are evenly distributed and there are no hot spots, a significant presumption). As a result, a RAID-5 theoretically can support *n* drives worth of streams. *Figure 4* compares the theoretical performance capabilities of RAID-3/5.

In practice, RAID-3s track theory much better than RAID-5. This is due to the unpredictability of the request distribution. With a RAID-3, all requests access all drives, in parallel. This results in very predictable performance, albeit at an efficiency of *(n-1)/n*. With a RAID-5, however, any request distribution pattern which doesn't keep all the drives busy results in performance degradation. A RAID-5 performance is much more difficult, if not impossible, to model accurately for a real-time environment. Many RAID-5 users find performance to actually benchmark at or below the *n-1* level.

PERFORMANCE AFTER A DRIVE FAILS

The strength of RAID-3 is its performance after a drive fails. In fact, there is no performance degradation after a drive has been removed from a RAID-3 array. These arrays perform data striping on-the-fly, using special hardware. The information stored on the redundant disk is also generated on-the-fly, using hardware. Because every piece of user data is striped on all drives, all operations occur in parallel. If a drive fails, a RAID-3 controller turns on hardware which regenerates the missing drive's data by combining the data from the remaining data drives and the redundancy drive. The hardware which performs this function typically is in the controller's data

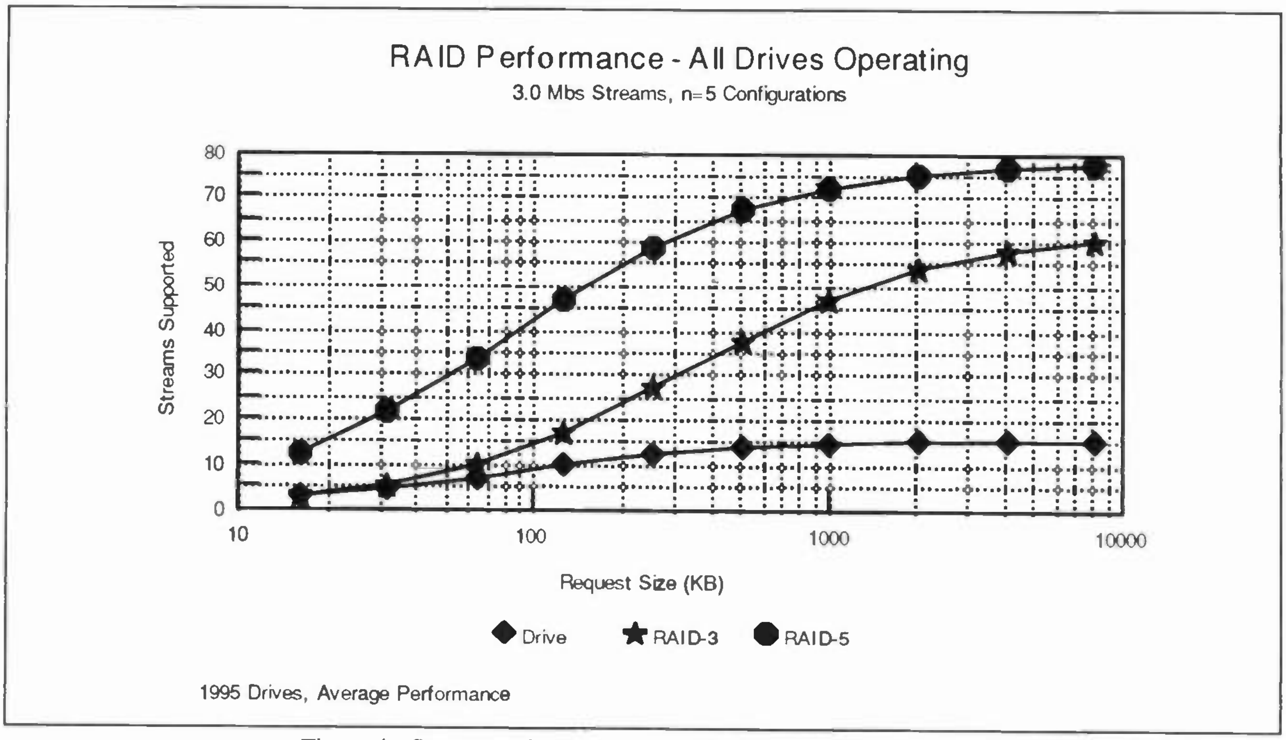

Figure 4 - Stream performance comparison with no drive failures.

path. These circuits are designed to operate at rates equal to or greater than the media data rate, with no additional latency.

A RAID-5, on the other hand, has a significant performance degradation after a drive fails. It is RAID-5's strength, independent access to each drive, which causes the severe performance loss after drive failures. When a RAID-5's drive fails, the remaining drives must be accessed for every request of the failed drive. This, in turn, prevents the functional drives from servicing their own requests. To quantify the performance loss, something called the Array To Drive Request Ratio (A/D) must be examined.

The A/D is simply the ratio of the total number of array requests to the total number of disk requests required to complete the array request. In other words, an array request is simply a data request from the host. The drive requests are the actual disk operations inside the RAID-5 array to carry out a host request.

When all drives are operating, the A/D is one, because every host request corresponds to exactly one disk request. However, after a drive has failed, there are far more disk requests (accesses) than host requests. Assume an even distribution of requests to all n drives, from the host's perspective. After a drive fails, $n-1$ of the n requests will be for the remaining $n-1$ functional drives. One of the n requests will be for the failed drive. The total array requests equal $n-1$ *(for the good drives)* $+1$ *(for the failed drive)* $=n$.

For the drive requests, the $n-1$ requests for the functional drives will correspond to $n-1$ drive requests, because each functional drive can perform one request. For a request to the failed drive, another $n-1$ requests are generated, because all of the functional drives are used to regenerate the failed drive's data. In total, after a drive has failed, a RAID-5 will generate $n-1$ *(for the good drives)* + $n-1$ *(for the failed drive)* drive requests. Or, more simply, $2*(n-1)$ drive requests. Hence, the ratio of host requests to drive requests (A/D)

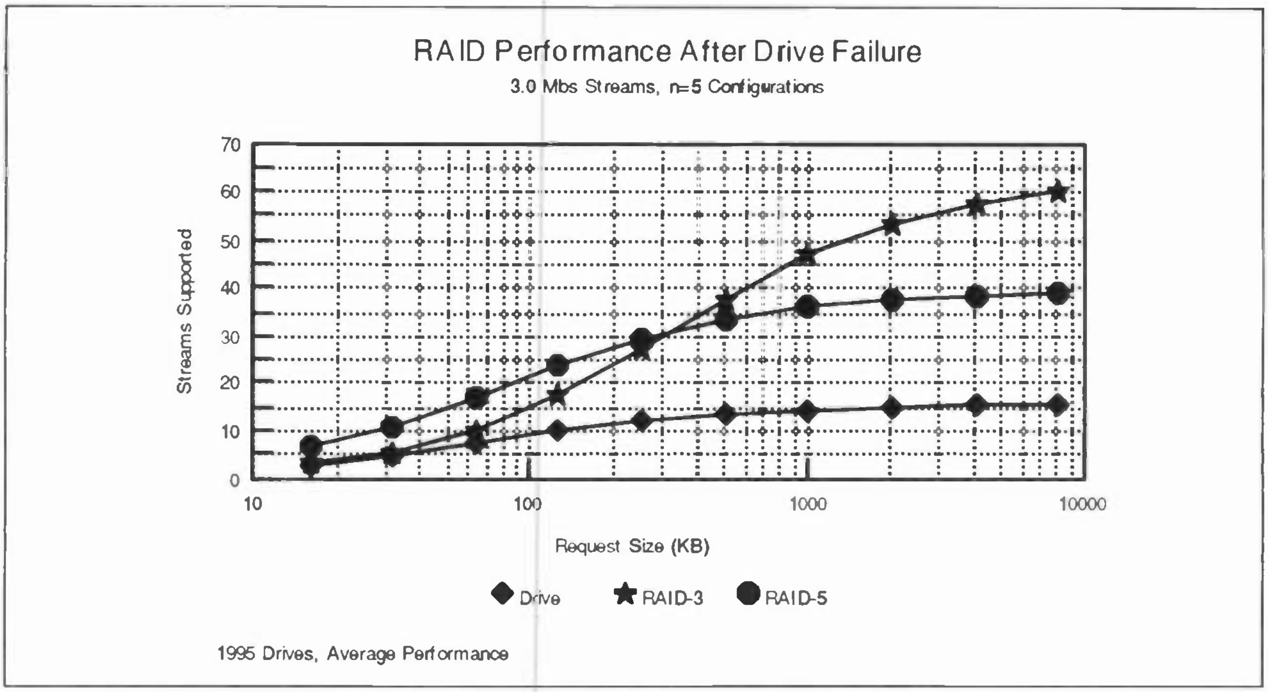

Figure 5 - RAID-5 performance falls by 50% after drive failure. No degradation in RAID-3 performance.

becomes $n/(2*(n-1))$.

To complete the analysis, the *A/D* is factored with the number of drives in a RAID-5 after a drive failure and the number of streams an individual drive can support (*S*). In general, the number of streams a RAID-5 will support is the product of the array to disk ratio (*A/D*), the number of usable drives in the array, and the number of streams supported per drive. Algebraically, total streams = *A/D* * number of drives * *S*. Substituting the expression for *A/D* and *n-1* for the total number of drives in a RAID-5 array after a drive failure, the total streams supported = $n/(2*(n-1)) * (n-1) * S$. Simplifying, the total streams supported = $n/2 * S$. Hence, the performance of a RAID-5 after a drive failure is only 50% of its theoretical maximum when all drives are operating. *Figure 5* shows the performance of each RAID level after a single drive failures.

SUMMARY

The converging worlds of video production, broadcast, communications, and computing are placing an ever increasing burden on the architects of video servers. These architects must design systems that deliver many streams of video, in real-time, and for a price that is palatable. The storage component of the video server accounts for about 50% of the total system cost and is a key performance component. It is for these reasons that RAID disk arrays have become an integral part of video server architectures.

RAID-1, or mirroring, duplicates every drive. Compared to alternative RAID architectures, this approach is not cost effective. Further, the decreased reliability of a large number of spindles and the complexities of packaging, powering, and cooling a RAID-1 implementation is prohibitive.

RAID-3/5 stripe user data and only require a single drive's worth of capacity for redundancy. They differentiate themselves in performance. While, in theory, RAID-5 may have a slight performance advantage when all drives are operating, in practice this is not often realized.

But the main difference in these two RAID architectures is the performance after a drive fails, a normal operating condition for RAID. While RAID-3 doesn't experience any performance degradation, the RAID-5's performance will drop by 50%. For the real-time environments of video servers, it is RAID-3's robust performance, over all operating conditions, that make it an ideal choice.

REFERENCES

[1]Analysis courtesy of Tom Ruwart, University of Minnesota, 1995.

[2]For more on this analysis, refer to Ciprico's white paper *RAID Disk Arrays and the 100 Year Disk Drive.*

[3] D. Patterson, G. Gibson, and R. Katz, *A Case for Redundant Arrays of Inexpensive Disks (RAID)*, ACM SIGMOD conference proceedings, Chicago, June 1-3, 1988

[4]Because the RAID levels were numbered, a popular misconception persists that the higher the RAID number, the better the level. In truth, each RAID level has its own unique properties, positive or negative, which makes each RAID level suitable for different applications.

[5]More than a single drive failure simultaneously causes the RAID to fail. However, this condition is so remote it does not require serious consideration.

IMPLEMENTATION OF MULTI-CHANNEL AUTOMATION

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ABSTRACT

This paper presents significant developments in multi-channel automation that have taken place in Europe since the last NAB. Experiences gained during implementation of a 64 channel Near Video on Demand (NVOD) service in The Netherlands and a multi-channel General Entertainment service operating out of the UK are outlined. Both systems utilise the latest developments in automation. It is believed that these two fundamental operations form the basis upon which future programme service packages will be developed. How such demands have been anticipated and implemented is discussed in the context that the demand to economically originate in excess of 50 channels has only become possible by the availability of true multi-channel automation control of transmission equipment.

Video disk servers operating under automation control have enabled the broadcaster to maximise the opportunity presented for multi-channel operation. For twenty four hour multi-channel operation to be economically managed and serviced demand that operational staff are fully supported by highly interactive and reliable control systems. The acceptance and integration of full automation control requires a fundamental operational change within the broadcast station this, is discussed with examples of how such changes have been adopted and managed.

INTRODUCTION

Since the inception of television the scheduler chose what the viewer watched. Viewer choice is restricted by the limited number of network channels generally available. More recently that choice has become greatly extended by the advent of thematic channels, and those viewers on cable have an even broader selection than those watching via the ether, whether terrestrial or satellite.

By the introduction of digital transmission technology the number of channels available to the viewer is about to exponentially increase, the viewer will be empowered as never before. Broadcasters are now faced with the adoption of significant changes and choices, non acceptance will not be permitted as these changes will be forced through on the grounds of viability.
It is an inevitable fact that as the number of channels increase the number of viewers per channel will reduce. Further, indications are that the revenue coming into the industry as a whole will increase only slightly above inflation. In order for viable operation or Revenue equals Costs, the cost of producing and delivering each programme stream must of necessity be reduced.

We have considered three principle elements to these costs:

1. **Programme production and origination costs.**

It is expected that these will generally remain the same or reduce irrespective of assembly or delivery mechanism. Savings will be expected to be made in post production costs.

2. **Programme assembly costs.**

Significant saving can be made in this area, more of this later.

3. **Programme delivery costs.**

These costs will be reduced significantly with the advent of digital transmission, further savings will be made as channel bandwidth selection is adjusted to suit the programme content. However it is considered that such savings by themselves are not sufficiently necessary for multi-channel viability.

It is against this background that the **"Implementation of Multi-Channel Automation"** must be considered. This automation is the control of equipment merging programmes and interstitial material into multiple contiguous programme streams.

BACKGROUND

The history of automating the delivery of programme networks has had varied results. These have ranged from the "not delivered system", to the "delivered but unreliable system" to the "delivered but unusable system". In a small number of cases the operational staff looked after their own short term job prospects by ensuring that the automation system is never used. With today's explosion in channels it is paramount to the survival of the industry that multi-channel automation system(s) are reliable, deliverable, useable and real.

Many of the automation systems that are available today stretch and extend a single channel control system concept to manage a greater number of channels. It is our contention that this approach is flawed.

From experience it is recommended that at the planning and design stage the maximum number of possible channels should be considered, and then the system scaled down to meet the foreseeable future. This allows a system modular structure to be adopted that meet the demands of eventual growth. Only in this way will a suitable overall systems emerge that will evolve to meet the needs of the 21^{st} century; less than 4 short years away.

It is understandable that many broadcasters will find it difficult to imagine the number of channels that they will be originating in the next millennium. Predicting the future has at best been a difficult occupation, some consider it better to leave this to the Astrologers. However to get the future wrong by omission can be very costly. In Europe the number of satellite channels that will be available at the start of the next century will be in excess of 3,000, some consider this to be a conservative estimate.

NEW GENERATION SYSTEMS

In order to meet the new demands of multi-channel broadcasting it was found necessary to combine, multi-channel playout system concept from the North American market, together with the required European features. Having merged these two it was necessary to interface with the new types of systems, such as disk stores, Encryption, Compression and Conditional Access Centres. As would be expected in such an evolving and demanding environment there is no standard approach. However by working closely with other pioneers in the field the approach detailed in this paper forms what we believe is the basis of a New Generation of Automation system.

The new generation of automation starts by the ability to accept multiple playlists which carry far more comprehensive information than previously generated. Examples of this include for each event, the programme indent, start time, duration, instructions for when the logo is to be shown when to come off, when the voice-over should be automatically switched on/off and where an alternative source for the programme, silence detection, DSK control and reserve control. These are only some of the requirements per event, per channel.

KEY ISSUES

1. **Traffic Systems:** in this new multi-channel world it will be necessary to put a much greater reliance on downloading accurate and complete schedules. In most broadcast stations considerable reliance is placed

on the transmission controller fixing the schedule, this is practicable with one channel to control, it is not practical with more than ten channels to control. With more accurate schedules being downloaded there will be less for the operator to do, this may lead to:

2. **Operator boredom:** this will be a challenge for the station manager in tomorrow's multi-channel world. In order that the channel controllers can react quickly they need to be provided with effective:

3. **Monitoring of many channels:** the Transmission Controller needs to be provided with effective displays in the monitor stack, these displays should give him/her all necessary information about the status of each of the channels without the necessity to interrogate the control terminal. The monitoring needs to be largely visual as audio monitoring of multi-channels is not practicable. Audible alarms are necessary to warn the operator of potential errors at which point the control terminal is used. In this way the Transmission Controller will see failures at the earliest possible moment and can react quickly. With so many revenue dependant channels:

4. **Reliability is essential:** the three most important factors in the system are, Reliability, Reliability and Reliability. This is paramount as so many channels are on the air 24 hours a day 365 days a year. The system design should be such that maintenance is possible. This can be achieved with a modular approach to the design, it also means that it must be possible to:

5 **Upgrade,** whilst on air. Live upgrading of the software and hardware must be achievable.

To explore the overall aspects of two distinct multi-channel environments we will examine the demands of a multi-channel NVOD operation and a multi-channel General Entertainment operation. Both have one important factor at their conception; anticipation of factors with respect to the potential total channel numbers, then develop a scaling down process that allows a definable and achievable channel module size. This approach which is basically a top down strategy ensures that irrespective of time-scale a modular building approach of say 12 channels at a time allows for growth over a long period.

NVOD IMPLEMENTATION

The overall concept employed for a potential 64 channel service is schematically illustrated by Figure 1. The initial phase incorporated 16 channels, this has since been expanded to 24 channels this extension was contracted prior to completion of installation of the initial 16 channels.

A key requirement of any automation system is reliability. If there are dedicated Transmission Controller personnel for each programme stream they can probably devise a short term work around if a failure condition exists. This situation is not possible in a multi-channel environment the speed of decision is critical. Any failure condition has to have a one button correction if the disruption to the output is to be minimised. K.I.S. (Keep it Simple).

In this case the system comprises a pair of computers, specially engineered to ensure high mechanical and electrical reliability, operating in a master slave relationship. Detection of a failure either automatically or manually results in the off line computer coming on line and taking over control, this changeover must be seamless.

Anyone entering the NVOD field should seriously consider duplicated paths, to minimise the possibilities of failure and ensure continuity of revenue. As can be observed from figure 1 a completely dedicated reserve path is available and would be constantly running identical material to that in the main path.

In addition to the constant playout of films, consideration was given to how promotions could be introduced cost effectively for each channel. This was resolved by the implementation of a Barker channel. A Barker channel can be considered as a virtual channel, the output of which is from a video server providing free running promotional material. Under automation control any channel can be switched to the Barker channel, at the end of a film. This switch must be done in a smooth way. Prior to broadcasting the next film the channel is switched away from the Barker channel hence to the film in a seamless fashion.

Operational staff with only a small number of channels have time to fix the schedules, this has allowed scheduling errors to be tolerated. In the multi-channel world much greater demand will be placed on accurate Traffic Systems, their data-bases and staff. The Transmission Controllers will not have the time to fix any errors or omissions. Few broadcasting operations have considered this aspect. Further, imagine the operator having to watch eight copies of a film being played out many times during his/her shift. How is the operator to be alert in the case of failure, so that he/she will react quickly. Boredom will be a great danger. Broadcasters need to pay considerable attention to this area. Remember the operator is unlikely to be alert and may not react quickly in the event of a failure if bored.
The management of multiple channels requires that the operator is not presented with any operational surprise, hence the automation system must be constantly looking ahead to ensure, on a per channel basis, that no source, material or timing conflicts exist. If they do, then controlled advance warning must be given to allow resolution. The process of such management also demands uncluttered and clear operational screens. Dual serial control lines between the two automation computers and, where possible to equipment under control, may in this day of Ethernet networking solutions appear inelegant, however for real time, reliable and dedicated control it is still the best solution. Very high speed serial lines are becoming available.

The actual monitoring of multi-channel output can be very confusing unless the operator is provided with a range of basic tools. In these installations an automation control panel presents the operator with a high degree of comfort in that schedule, material or plant errors are quickly highlighted. Integrated with the automation system are Under Monitor Displays providing immediate indication of Material identification, Next Source to be used, Time of Next Event and Count Down to the event. In addition there are panels which show the currently aired source. Such visual information is vital in an environment where many channels are playing identical time displaced material.
With multi-channel and multi-lingual operation the effective monitoring of audio becomes vital. In the system illustrated, each channel has a range of silence detectors interconnected with an event driven automation alarm system. Remember some programmes have periods of silence.

GENERAL ENTERTAINMENT IMPLEMENTATION

For the purpose of definition we have classified non repetitive channel programme material with repetitive interstitial material as a General Entertainment Channel.
The implementation of a 12 channel General Entertainment multi-channel systems is illustrated by figure 2. This systems demonstrates the application of having a large, in this case 40 hour video server, feeding under automation control a series of duplicated live video servers. The system is again based on a highly fault resilient basis with a main and reserve path allocated for each channel. The automation system maintains a watching brief over both paths. As the overall system is based on the use of Serial Digital Video with embedded audio, SDI. It was necessary to use embedders and de-embedders for voice over operation. This was accomplished with all four functions carried out in the same unit, DSK (Down Stream Keying), logo, voice over and still store. In this case FlexiCarts with Drake machine controllers were used for the playout of programme material.
The demand for general entertainment also included the need to enable break away operation for live events. This was accomplished by the provision of 3 co-located Control Rooms which dynamically interact with the General Control Room. Handshaking routines between the main Control Room and sub control room enable a smooth handover to be achieved. During live events the operator is assisted by the automation system for interstitial playout. An important element of the automation system in the context of live events, is the ability to allow ripple down time and

adjustments of succeeding events if the live event finishes early or late.
A typical floor layout for 12 channels of General Entertainments, 16 channels of NVOD and 3 single channel control rooms is illustrated by figure 3. From this diagram the full effect of real estate savings acheived by use of video server technology coupled with a new generation of automation system can be fully appreciated.

WHERE NEXT

Once Automation is commonplace the broadcaster may be able to accept an unmanned transmission control room. This will become a necessity because of the "bored operator syndrome". Enabling this to become a reality will be the challenge to broadcast equipment suppliers. Effective monitoring and alarm systems will need to be developed. Once these have become reliable and accepted the next stage will be to use them to control the change-over to back-up systems.

SUMMARY

This paper has addressed the experience gained by Drake in installing its automation system D-MAS in multi-channel environments. A 12 channel General Entertainment System to BSkyB in the UK and the first NVOD system in Europe with NetHold in the Netherlands.

Key issues considered at the formative stage of system design were :

1. To try and define the total number of channels to be transmitted long term.

2. The maximum number of channels to be under the control of one person.

3. Labour savings by integrating manually operated channels with automated playout at selected times during the day.

4. Definition of the material to be transmitted.

4.1. Duration of segments.

4.2. How repetitious?

4.2.1. If interstitial material, what is the total duration of the instantly available material?

4.2.2. If there is repetitious programme material, how often is it repeated?

4.2.3. In the case of NVOD consider that a 2 hour movie with a 15 minute cycle for start times, if run for 24 hours a day, and aired for 4 weeks will need to be repeated 2,688 times. Is this practical or cost effective with tape?

Other programme material may be repetitious, is it to be repeated on the same channel? Are the start times such that only one copy is needed? etc.

5. What degree of failure can be tolerated?

5.1. In the PPV NVOD case it is unlikely that any but the smallest degree of failure can be tolerated.

Consider the Subscriber Management Centre besieged by angry customers wanting a refund of their money for a failed performance blocking subscriber sales of future performances.

5.2. If advertising is the source of revenue then advertising revenue lost can never be retrieved. The advent of Disk technology has made duplication of resources cost effective and practical. Cacheing allows for preview and eradicates head clog problems in an on-line application.

6. Style of presentation.

6.1. Major networks require sophisticated presentation with mixes and effects at the programme transitions. Is this affordable in today's multi-channel environment?
Is it necessary to provide a sophisticated presentation
control desk?
Very few broadcasters can afford to put every channel through a Presentation Mixer in a Multi-channel world. This is particularly true with a totally digital transmission solution.
Can equipment providers come up with a cost effective solution?

6.2. In the cable and satellite world cuts between programmes and interstitial are acceptable.
This is more cost effective than using presentation mixers. Further, significant savings could be made with a totally digital transmission path.
Will the viewer notice or care?

6.3. Is "Hotel" style of presentation acceptable?

If yes to this question then the signal can be fed from MPEG II Consumer servers.

7. Real Estate
The system must be able to evolve without major physical movements.

8.Capital Costs Per Channel

8.1. Target costs per channel in the NVOD case were $200k. This includes a fully duplicated transmission chain.

8.2. Target costs per channel in the General Entertainment case were $300k.

8.3. These figures are set against network control equipment costs in the of $1.5m - $9m.

9. Running Costs

9.1. The degree of operational running costs could be significantly reduced by one person controlling a large number of channels.

9.2. The degree of playout equipment must be totally under automation control.

9.3. With only a limited number of operators on duty at any one time manual back-up in the event of automation failure is not feasable. Total automation failure is not acceptable.

CONCLUSION

The two major systems described in this paper are in the forefront of NVOD and General Entertainment broadcasting in Europe. There is one non technical aspect that becomes highly relevant when implementing multi-channel operation and that is associated with the operational culture. Such operations now require a frame of mind which is targeted towards instant recovery leaving others to analyse and resolve technical problems and issues. Further, constant last minute changes to schedules will not be viable in a 100 channel system. Technically they can be achieved, mentally or physically we doubt it.
The advent of digital transmission technology has placed before us the opportunity of mass channel broadcasting, it is predicted in Europe a market size in excess of 3,000 channels within the next few years. Whether you believe such numbers or not, the general increase in channel availability will provide the viewer with a degree of power not yet fully appreciated or understood. The introduction of thematic channels and NVOD distribution will mean focused but restricted number of viewers. Hence the viewer per channel ratio will reduce and bring on a demand for lower programming and transmission costs. Without the implementation of a new generation of automation system the costs associated with multiple channel transmission playout will become a constraining element to growth.

ACKNOWLEDGEMENTS

The authors wish to express their appreciation to both BSkyB and NetHold for allowing the system implemented to be presented.

NVOD - NEAR VIDEO ON DEMAND

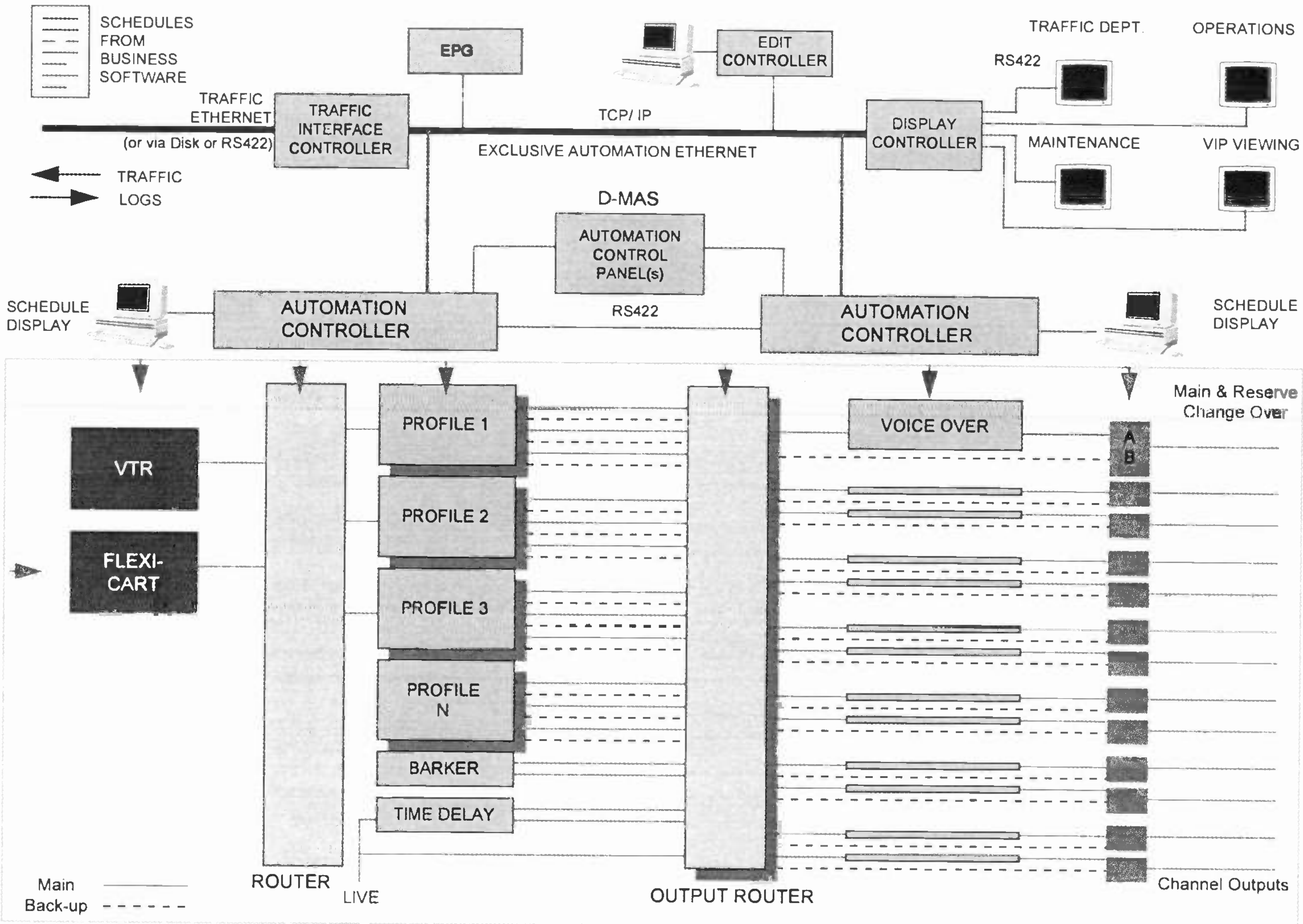
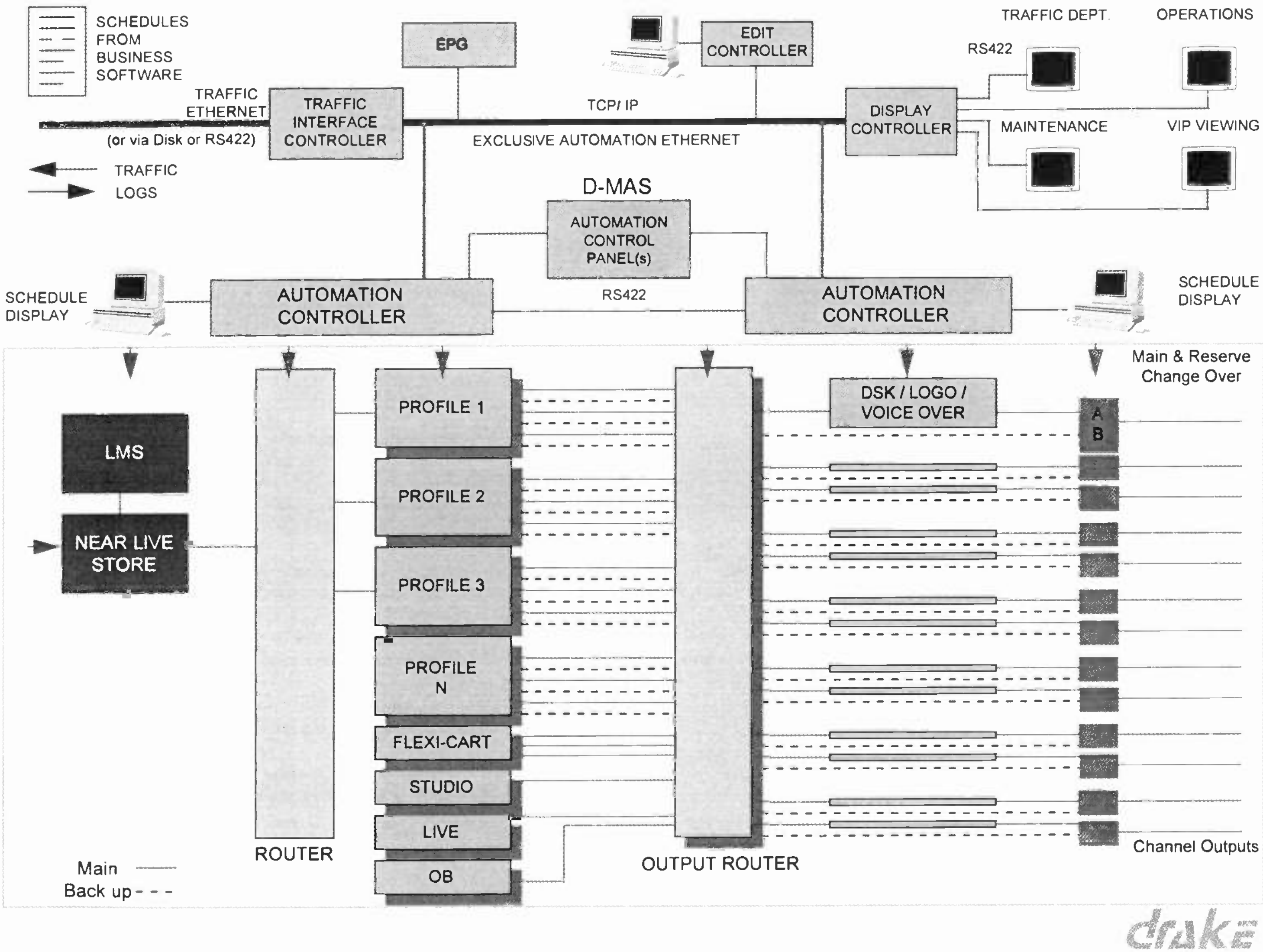


Figure 1



SCHEDULES FROM BUSINESS SOFTWARE
TRAFFIC ETHERNET
(or via Disk or RS422)
TRAFFIC
LOGS
TRAFFIC INTERFACE CONTROLLER
EPG
EDIT CONTROLLER
TCP/ IP
EXCLUSIVE AUTOMATION ETHERNET
DISPLAY CONTROLLER
TRAFFIC DEPT.
OPERATIONS
RS422
MAINTENANCE
VIP VIEWING
D-MAS
AUTOMATION CONTROL PANEL(s)
RS422
SCHEDULE DISPLAY
AUTOMATION CONTROLLER
AUTOMATION CONTROLLER
SCHEDULE DISPLAY
Main & Reserve Change Over
LMS
NEAR LIVE STORE
PROFILE 1
PROFILE 2
PROFILE 3
PROFILE N
FLEXI-CART
STUDIO
LIVE
OB
DSK / LOGO / VOICE OVER
A
B
ROUTER
OUTPUT ROUTER
Channel Outputs
Main
Back up
drake

Figure 2

drake Automation Ltd

Typical area required for

16 Channel NVOD
12 Channel General
3 Single Channels
Equipment Room

FLOOR PLAN

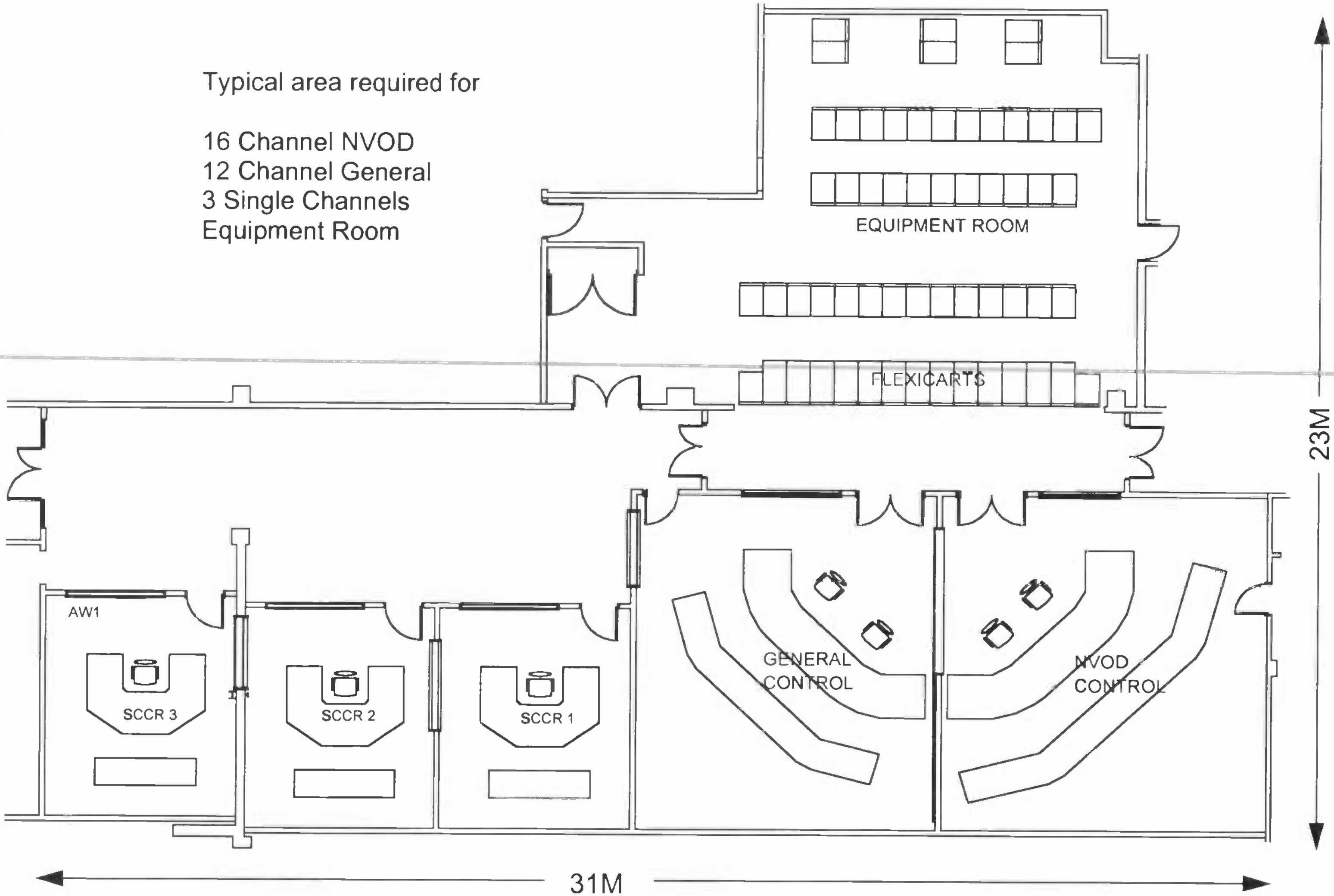


Figure 3

HIGH-QUALITY RF FOR RADIO

Tuesday, April 16, 1996
9:00 am - 12:00 pm

Session Chairperson:
Bill Ruck, KNBR-AM/KFOG-FM, San Francisco, CA

COMPUTER CONTROL AND MONITORING OF A COMPLEX MF DIRECTIONAL ARRAY
Grant Bingeman
Continental Electronics Corporation
Dallas, TX

TRANSMITTING SITE EVALUATION USING A MOBILE SPECTRUM MEASUREMENT SYSTEM
Stanley Salek
Hammett & Edison, Inc.
San Francisco, CA

AMERICA'S FIRST EXPANDED BAND STATION WJDM ELIZABETH, NEW JERSEY
Charles A. Hecht
Charles A. Hecht & Associates, Inc.
Pittstown, NJ

FM MEASUREMENT LIMITATIONS AND TECHNIQUES
Edwin R. Twitchell
Harris Corporation, Broadcast Division
Quincy, IL

LOW PROFILE ANTI-SKYWAVE ANTENNA (LPASA)
Marshall W. Cross
MegaWave Corporation
Boylston, MA

DIRECTIONAL ANTENNA BASICS
George Whitaker
Practical Radio Communications
Arlington, TX

***AN EFFICIENCY COMPARISON: AM/MEDIUM WAVE SERIES-FED VERSUS SKIRT-FED RADIATORS**
Tom F. King
Kintronic Labs, Inc.
Bristol, TN

*Paper not available at the time of publication.

COMPUTER CONTROL AND MONITORING OF A COMPLEX MF DIRECTIONAL ARRAY

Grant Bingeman
Continental Electronics Corporation
Dallas, TX

ABSTRACT

A real-time computer in a 600 kW, seven-tower, five-pattern directional antenna system monitors RF impedances, currents, phases, voltages, powers, VSWR, tower-lighting, arcs, interlocks, air pressure, variable capacitor positions, RF switch positions, RF sample selection, antenna monitor mode, etc.

An operator can select and adjust (in real time and at full power) radiation patterns from local, extended, or remote control locations. A color-graphics user interface permits both mouse and keyboard entries. Screens include a Smith chart, bar graphs, dial-type meter faces, antenna monitor readings and deviations, an RF path schematic, an interlock block diagram, real-time pattern plots, faults, etc.

The computer system consists of a local and an extended industrial 486 PC connected to a web of small, inexpensive slave computers distributed among the antenna huts. Fiberoptic and RS422 serial communication operate well in the hostile RF and transient environment.

The computer control and monitoring concept reduces adjustment and maintenance time in a complicated directional array of tall towers and parasitic elements. Simultaneous real-time impedance measurement at the common-point, transmission line inputs, and tower feed-points eliminates a common source of error during the initial adjustment process. Also, error-sum displays of overall system mismatch and pattern deviation reduce adjustment time tremendously.

INTRODUCTION

This paper deals with the control and monitoring of the antenna equipment used in a new high-power medium-wave radio station located in Kuwait. There are seven self-supporting towers which produce four directional radiation patterns and one-omni-directional pattern (see Figure 1, site plat). This paper will emphasize the automatic control and adjustment of the networks that produce the desired tower currents, and avoid antenna theory.

The primary mission of the antenna control system is to produce a particular set of readings on the antenna monitor for each of the five patterns. The antenna monitor-samples the RF current in each tower, and displays a magnitude and phase for each current. The computer reads this analog information from the antenna monitor, and converts it to a digital form, which is used in various calculations within the software, and is presented to the control room operator in a color-graphics format.

The secondary mission of the antenna control system is to increase the efficiency of the antenna adjustment team, by providing a lot of instantaneous, real-time information during the adjustment process. For example, since an adjustment to one tower's current influences the currents in all the other towers, it is sometimes not evident in which direction to turn a crank. In other words, is it a counter-clockwise or clockwise adjustment of a particular capacitor which has the better overall effect on reducing the error between the desired and actual antenna monitor readings? And later during the adjustment process, does that optimal direction change? If the computer can provide an instantaneous error sum for the operator while he is adjusting a coil or capacitor, the tuning process almost becomes enjoyable, even with a myriad of interacting factors.

The tertiary mission of the antenna control system is to provide quicker failure diagnosis during and after initial array adjustment. To this end, a number of fault screens and testing functions are built into the computer software.

SYSTEM DESCRIPTION

The computer control and monitoring system consists of a rack mounted industrial 486 computer in antenna house number five. Since this is a 600 kilowatt station and the RF components are quite large, the word "house" is more appropriate that the conventional word "hut." The centrally located (local) computer is connected to an RS422 network of eight smaller computers, which are dispersed among all seven antenna houses (Figure 2). Eight separate full-duplex RS422 channels are used instead of a two-wire RS-485 buss, in order to increase reliability by redundancy, and partly because of the trench layout at the site. Each of the eight slave computers collects "parallel" data from equipment located in its house, and controls various motor drives, contactors, etc. The parallel data are formatted into a serial data stream which is sent to the central computer on command.

The peripheral equipment connected to the computers include interlocks, pressure and temperature sensors, relay and contactor position microswitches and control coils, capacitor position potentiometers, stepper motors, fire and intrusion alarms, an antenna monitor, RF ammeters, RF impedance meters, user-defined devices, tower lighting current and voltage sensors, arc detectors, reflectometers, etc.

The extended control computer is a sister 486 that is located in the transmitter building about a mile away, and connected via fiberoptic lines. Remote control is available at an RS422 port in the extended computer.

LOCAL 486 COMPUTER

This machine is a typical industrial PC with several special I/O cards plugged into a passive backplane. There is an eight-channel RS232/422 card, two 16-channel 12-bit A/D cards with a DAC output, one 32 channel opto-isolated digital input card, one 32 channel opto-isolated digital output card, an external fiber-optic modem, a 15 inch color monitor, a color printer, a standard keyboard and buss mouse, and an UPS to run the computer and monitor during power failures and protect it from transients. In addition, a magnetic-resonance voltage regulator is installed between the UPS and the AC mains.

The local computer polls all the slave computers and various parallel status and analog lines, then formats the information into the color-graphics screens that an operator can readily understand. A lot of calculations are performed in the local computer, such as operating impedances, transmission line VSWR, Smith chart and pattern plots, and individual house temperatures.

EXTENDED 486 COMPUTER

This machine is very similar to the local computer, except it does not have most of the specialized I/O cards that are used in the local computer. However, it does have an internal telephone modem and tape drive, which the local computer does not. Many of the parts are exchangeable between the local and extended 486 computers (for example, the CPU card, video card, floppy drive, etc.). Since the local and extended 486 computers do not have what is conventionally called a mother board (they use a 12-slot passive backplane), it is easy to upgrade the CPU, as it only involves switching plug-in cards.

The extended computer runs on a large UPS that powers much of the control room, so an individual UPS is not provided for it.

TYPICAL SLAVE COMPUTER

Each Z180 slave computer is mounted on a panel with a power supply and two I/O cards. There are 40 ten-bit A/D inputs, ten digital inputs, 22 open-collector control outputs, two form-C outputs, one ten-bit DAC output, separate RS422 send and receive pairs, and a reset line. This computer is housed in a metal box, and has a 2 by 20 LCD plus a 12-button keypad. A site operator would not normally use this keypad or LCD, as they are intended primarily for initial installation and check-out. Typical connections to antenna house equipment are shown in Figure 3.

Since the hostile RF and transient environment of the high-power antenna site can wreak havoc with low-level control and monitoring signals, the usual special techniques are employed to filter out RF, modulation, motor noise, etc. These techniques include passive and active filters, software averaging, zero-crossing sampling, high-quality balanced and shielded lines, double-shielding, termination of lines in the correct surge impedance, balanced sources, optical isolation, careful cable routing, isolated power supplies, etc.

SOFTWARE

The software for the 486 computers is written in C, and uses many modules that have been developed over the years at Continental Electronics. Various third-party libraries are also used. The programs in the local and extended computers are identical, but the extended computer program is run with a /E option to tell it where to get its data (COM1). These color-graphics programs run in DOS, and are about two megabytes in size, but use EMM and overlays in order to function within the 640K DOS limit.

The software for the slave computers is also written in C, and runs from a ROM in its final form. During the development phase, this program runs in RAM so a debugger can be used. The same program runs in all of the eight slave computers.

Continental Electronics uses a large number of common software modules in its various transmitter and site control applications. This permits quicker development of control and monitoring software for a new hardware application, and it provides the end-user with a software product containing many more useful features, since development cost is amortized over a large number of projects.

SPECIAL HARDWARE

Continental Electronics uses a directional coupler designed for high voltage and high current applications that feeds RF samples to a proprietary box which produces the low-level DC signals for a computer to calculate operating impedance.

Several signals are affected by modulation, and special buffer amplifiers remove the modulation where necessary. For example, the output from the antenna monitor when switched to the amplitude mode (as opposed to the ratio mode), contains an undesired modulation component. The buffers remove the modulation component but leave the DC component. The bandwidth of these buffers is about four Hertz, similar to that of an ordinary analog meter. But modulation components above 30 Hz are reduced to an insignificant level and so will not interfere with the digital computations. By the way, the computer can select any of the three antenna monitor modes: ratio, amplitude or test.

The various tuning capacitors are driven by stepper motors. The controllers for these motors use various techniques for controlling the motor speed, including analysis of position error. They allow operation of the motors in open or closed-loop modes.

RF NETWORKS

A schematic of the antenna couplers, power dividers, common-point matchers and phase adjusting networks (Figure 4) looks a bit complicated in this array because of the fact that there are five patterns and seven towers. All seven of the towers are either parasitic, driven or de-tuned depending on the chosen pattern. Because of the high RF power output from the transmitter, most of the antenna RF components are quite large physically, and have high voltage and current ratings. This is convenient during the initial low-power adjustment process, as there are not likely to be any component failures caused by the wide impedance swings possible at the start of array tuning.

Since the assigned frequency is at the high end of the broadcast band, shunt stray capacitance and series inductance are important factors in the operation of the system. Real-time display of all transmission line, coupler load, and common-point impedances is a definite advantage to an engineer assigned the task of adjusting such a complicated array.

The actual tower mutual impedances can be determined from the measured self impedances, and any set of initial operating impedances and antenna monitor readings, when a simultaneous set of equations is solved. This information can then be used to determine what the actual operating impedances will be for a particular set of antenna monitor values, which aids in the initial setting of the network components.

GENERAL PROGRAM OPERATION

Many of the control commands entered at the keyboard require arguments. If an illegal argument is entered, a warning message is displayed and the command will be ignored. The on-line help text can be accessed to determine the valid range of argument values for a given command.

Most of the control commands will normally be issued from the main screen using the mouse. However, the 48 function keys can be programmed by the user, or the standard command mnemonics and applicable arguments can be entered at the keyboard from just about any screen.

When in doubt, hit the Esc key; this clears the keyboard and other buffers, and will eventually get an operator back to the last real-time screen. The Home key (or simultaneous left and right mouse click) will always return to the main real-time summary screen.

A string search function called FIND will display and highlight a desired string wherever it appears in the help text. This is a very powerful tool for those who are still learning to operate the control system.

COLOR-GRAPHICS SCREENS

The graphics mode is 16-color 640 by 480 (VGA mode 12H). Mouse or keyboard inputs select and modify the information presented on the screens. Both the left and right mouse buttons are used in single-, double- and dual-click protocols. On-line help text is available for all command functions, and a list of available commandsis automatically presented after an unrecognized command is entered (Figure 5).

Real-time screens can be recognized by the clock running in the upper right corner. Set-up screens (for example, analog warning limits) are not real-time, hence do not have a time and date stamp. However, when a graphics screen is printed, a time and date stamp is placed on the print-out. When reviewing logged data, all the real-time screens will display a fixed date and time and a message stating the log sequence number. When simulated data are displayed on a real-time screen, a message next to the time and date stamp will remind the operator that the information is not real-time.

The main screen (Figure 6) shows a summary of the overall antenna system status, plus some of the more important meter readings. The meters can be scrolled by clicking on their part of the screen. The radiation pattern can be changed simply by clicking the mouse cursor on the desired push-button symbol. More detailed status or control screens can be invoked by clicking on the appropriate summary box. In general, if a particular status box is green, all is okay; if yellow, then something is in transition or needs attention; if red, than a serious fault requires immediate attention.

If the FAIL box on the main screen were yellow, an operator could click on that box and the Failures and Warnings screen would appear. Figure 7 shows all boxes green except "pattern fail" and "logs full." The latter would alert an operator that he should transfer the logs to a floppy diskette for the archives, and reset the log counter sometime in the near future. More importantly, however, he would note that the desired radiation pattern had not been achieved. In order to narrow down the cause of this failure, he could start with the Network screen (Figure 8) to determine if a particular RF contactor is out of position. Or he could refer to the Radiation Pattern screen (Figure 9), and compare the actual pattern shape with the desired pattern. If more detail is desired, the Antenna Monitor screen can be invoked (Figure 10). Or perhaps too many arcs or reflectometer trips have occurred, and the transmitter has simply reduced power, in which case the Faults or Interlocks & Arcs screens (Figures 11 and 12) would point the operator in the right direction.

Also, if an antenna coupler component failure has occurred which is not severe enough to knock the transmitter off the air, but does have a significant effect on the antenna monitor readings, the Smith Chart screen (Figure 13) may show one transmission line with a higher than normal VSWR. This information could save a lot of leg work, since there are seven antenna houses at this particular site.

MAINTENANCE

A number of software commands are available for individual equipment tests. For example, LINK allows an operator to test RS422 communication with an individual slave computer over a long period of time to find out if its error rate is degenerating compared to previous tests, or compared to the other slave computers.

A tape backup should probably be run once a week at the extended computer, and the logs may be archived on floppy diskettes more frequently if necessary or convenient. There are a number of user- and adjustment-defined files which should also be archived in a safe place, to use as a starting point in case of a massive failure at some future date.

A command called ENDS allows a capacitor drive to be run from one end-stop to another. If the position indicator is different from the previous test, then the drive should be checked for mechanical slippage or other problems. Normally this command is used during the initial system setup, or during fault diagnosis. In general, a command such as this one should not be used on a daily maintenance basis; once a month may be more appropriate.

SUMMARY

A network of inexpensive, but hardened slave computers can be used to multiplex a large number of control and monitoring signals in a widely dispersed antenna farm so that a central computer can provide real-time information to an operator or engineer to aid in the diagnosis of problems, or to speed up the adjustment process.

In a complicated array, such a control and monitoring system may pay for itself in the amount of professional labor saved during the initial tuning and commissioning of a site. Of course, this assumes that the control and monitoring system is designed and installed efficiently.

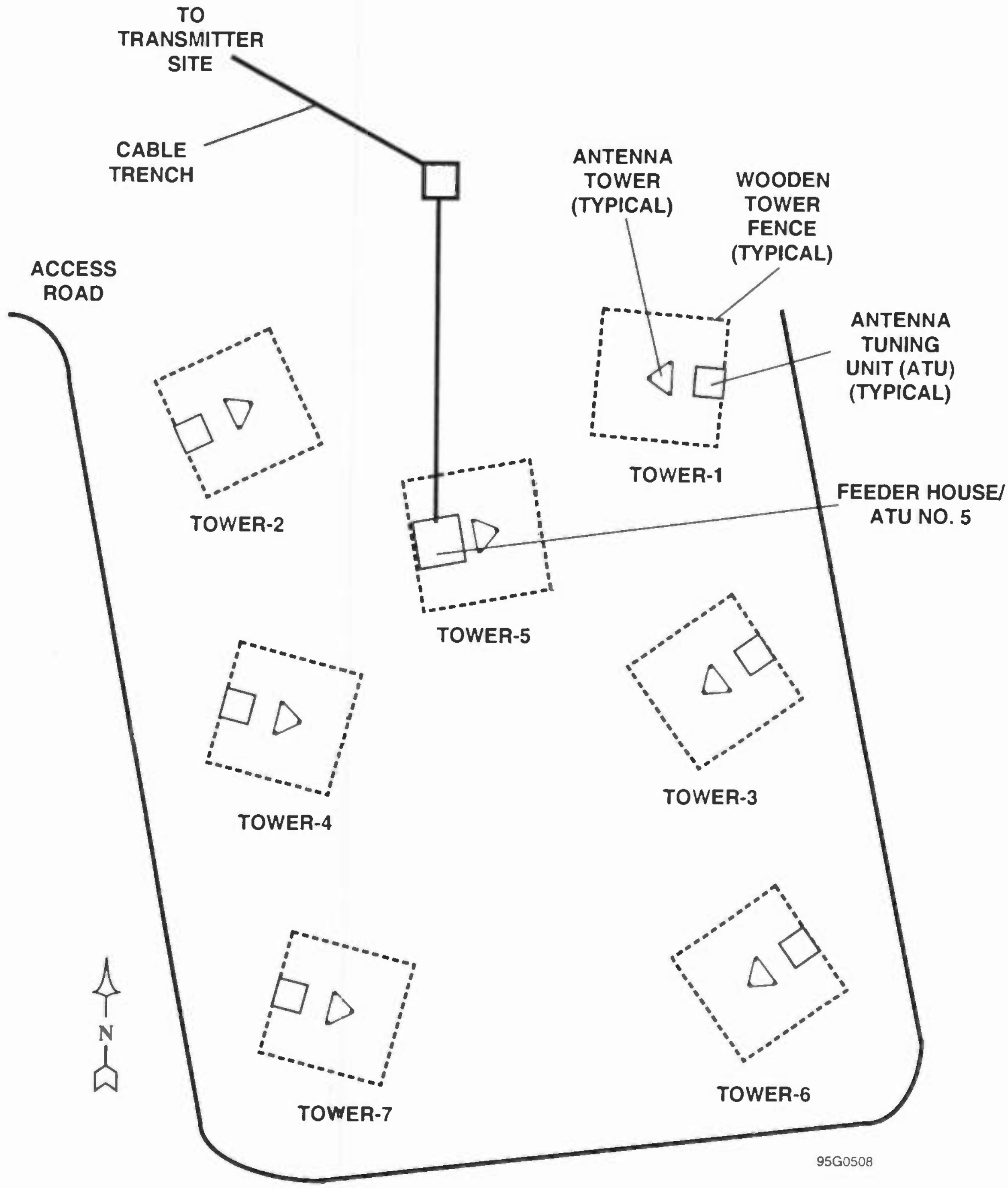

TO
TRANSMITTER
SITE
CABLE
TRENCH
ANTENNA
TOWER
(TYPICAL)
WOODEN
TOWER
FENCE
(TYPICAL)
ACCESS
ROAD
ANTENNA
TUNING
UNIT (ATU)
(TYPICAL)
TOWER-1
TOWER-2
FEEDER HOUSE/
ATU NO. 5
TOWER-5
TOWER-4
TOWER-3
N
TOWER-7
TOWER-6
95G0508

Figure 1

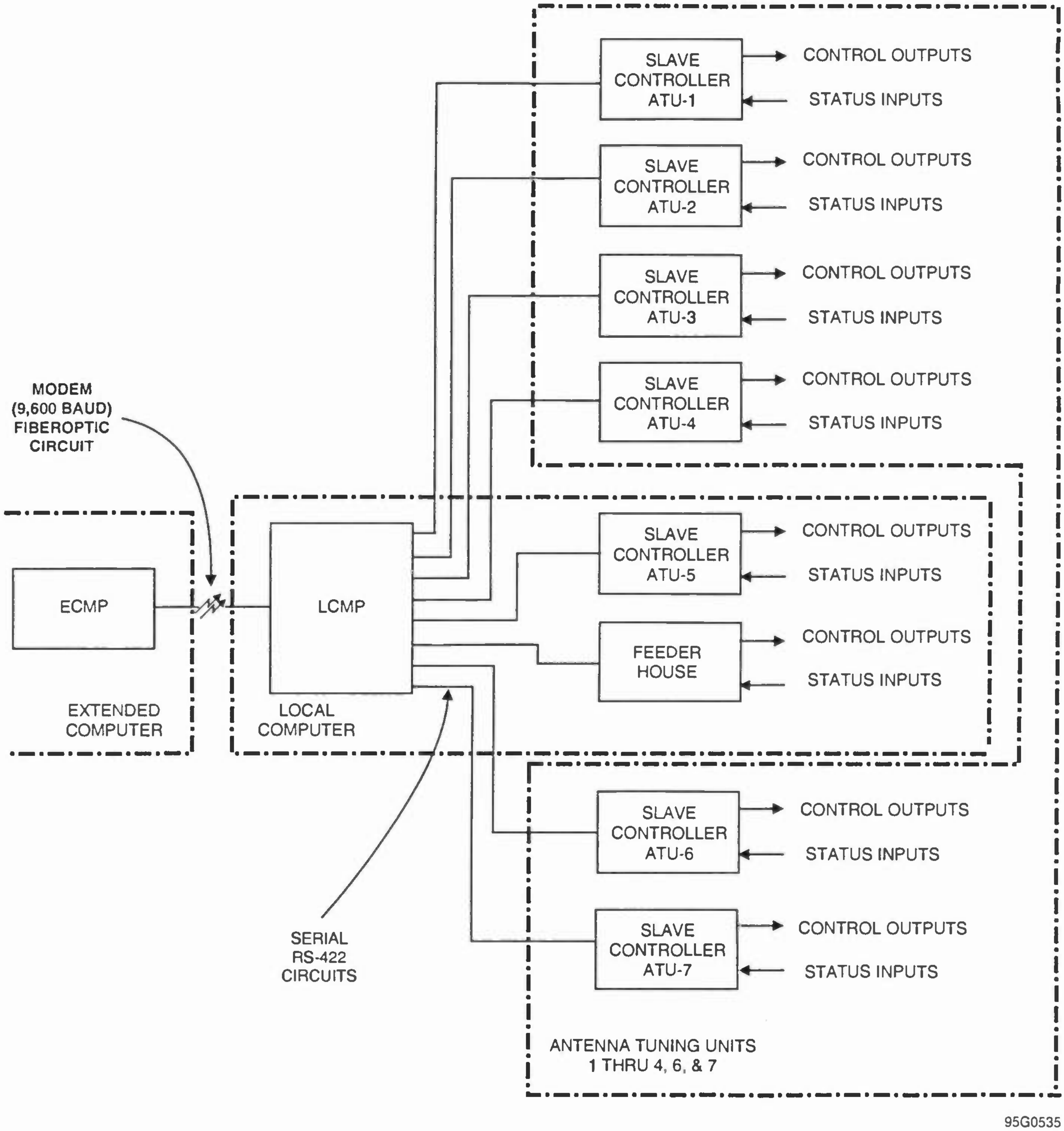
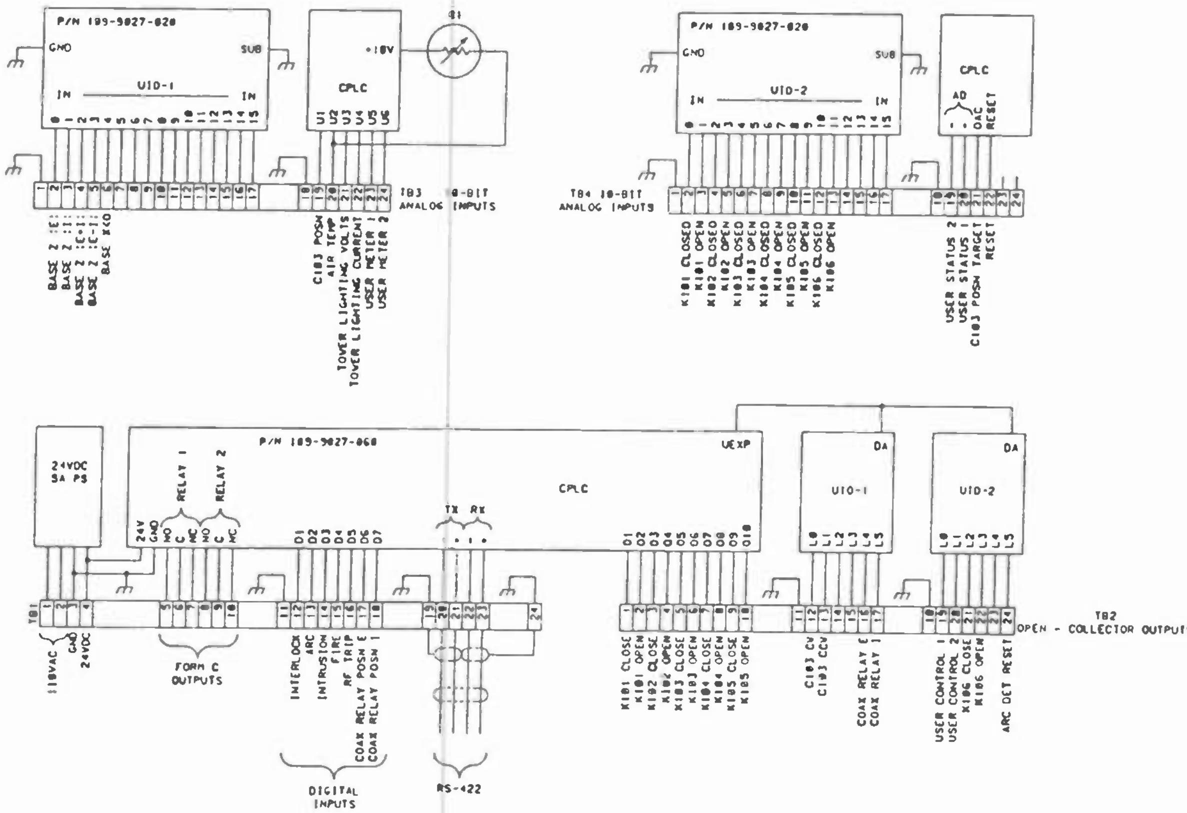

SLAVE CONTROLLER ATU-1
CONTROL OUTPUTS
STATUS INPUTS
SLAVE CONTROLLER ATU-2
CONTROL OUTPUTS
STATUS INPUTS
SLAVE CONTROLLER ATU-3
CONTROL OUTPUTS
STATUS INPUTS
SLAVE CONTROLLER ATU-4
CONTROL OUTPUTS
STATUS INPUTS
MODEM (9,600 BAUD) FIBEROPTIC CIRCUIT
ECMP
LCMP
EXTENDED COMPUTER
LOCAL COMPUTER
SLAVE CONTROLLER ATU-5
CONTROL OUTPUTS
STATUS INPUTS
FEEDER HOUSE
CONTROL OUTPUTS
STATUS INPUTS
SLAVE CONTROLLER ATU-6
CONTROL OUTPUTS
STATUS INPUTS
SLAVE CONTROLLER ATU-7
CONTROL OUTPUTS
STATUS INPUTS
SERIAL RS-422 CIRCUITS
ANTENNA TUNING UNITS 1 THRU 4, 6, & 7
95G0535

Figure 2



NOTES: UNLESS OTHERWISE SPECIFIED

1. PARTIAL REFERENCE DESIGNATIONS ARE SHOWN. PREFIX WITH UNIT NUMBER OR SUBASSEMBLY DESIGNATION OR BOTH FOR COMPLETE DESIGNATION.
2. ALL TERMINALS EXCEPT GND, RS422, 110 VAC ARE BYPASSED TO GND WITH .01 UF, 50V.

FIGURE 3

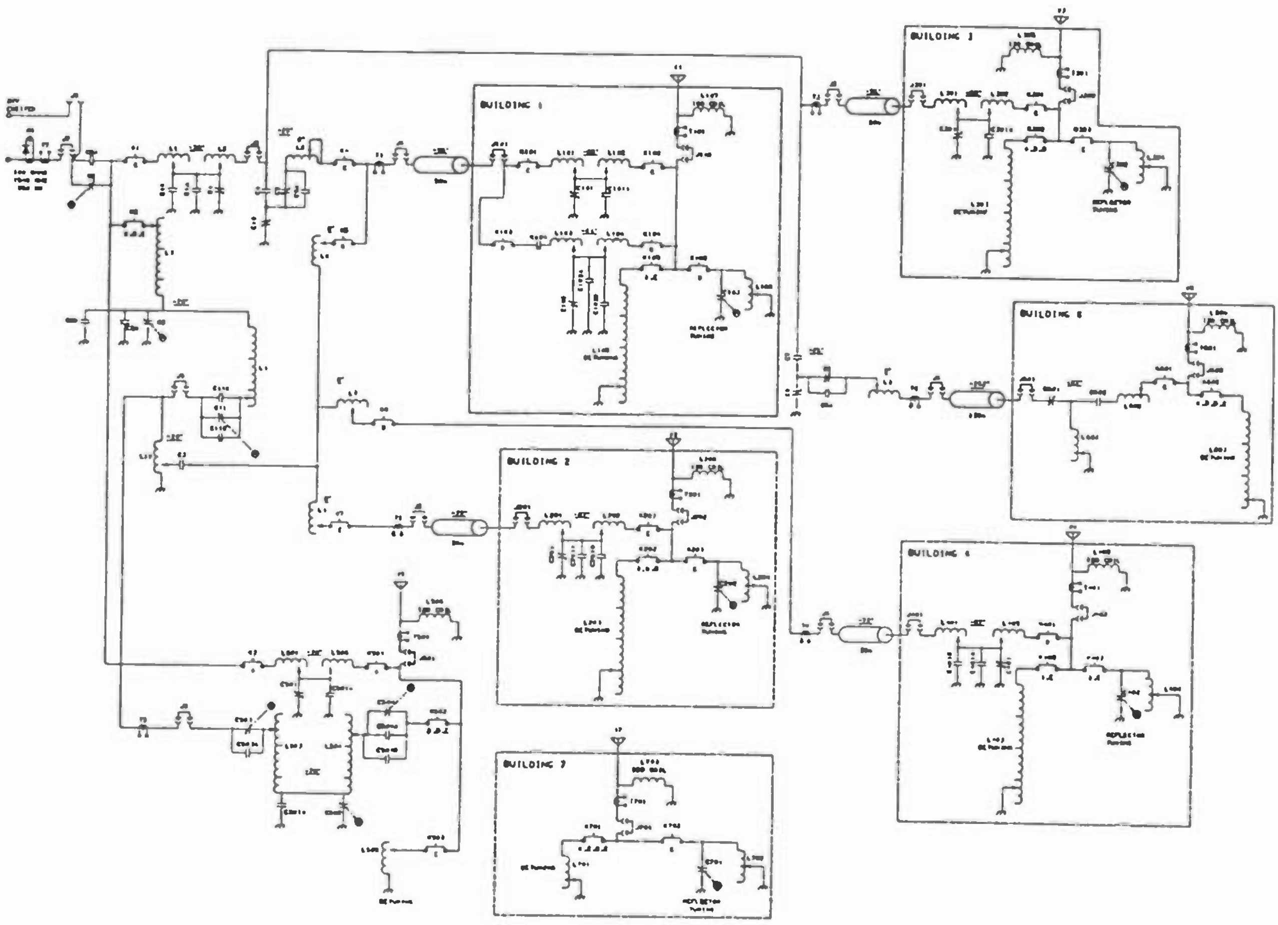


FIGURE 4

ACTS - LOGGING ACTIVITY
BARS - BAR GRAPHS
BASE - DATABASE NAMES
CALC - CALCULATIONS
CAPS - CAP POSITIONS
DATA - DATABASE IN HEX
FAIL - DETAILED FAILURES
FALT - DETAILED FAULTS
FIND - SEARCH FOR STRING
HELP - PROGRAM PROTOCOL
INTL - INTERLOCKS AND ARCS
LOGS - READ A LOG
MAIN - MAIN PANEL
MAPS - POSITION STORAGE GRID
MATH - PROCESSED METERING
MISC - USER-DEFINED FLAGS, ETC
MON - ANTENNA MONITOR
NTWK - RF PATH
PLOT - PATTERN PLOT
SMIT - SMITH CHART
SIMU - SIMULATOR
TABL - FIELD RATIOS
TITL - TITLE PAGE & REV LEVEL
USER - USER-DEFINED SCREEN
ZDAT - SLAVE DATABASE
ZMET - SLAVE METER READING
ZSTA - SLAVE STATUS BYTE
+ - /* - CALCULATOR OPERATIONS

MUTE(F1) - DISABLE RF
ENAB(F2) - ENABLE RF
DACn, m - SET LOCAL DACMV
DACZz, M - SET ZWORLD DAC MV
DIGLz, N - SET ZWORLD OUTPUT LO
DIGHz, N - SET ZWORLD OUTPUT HI
ENDSN [9] - FIND MOTOR END STOPS
PATN - PATTERN SELECT
LOWRC - LOWER CAP POSITION
RAISC - RAISE CAP POSITION
POSNC, P - SET CAP POSITION
RSET (F11) - RESET INDICATORS
SAMP (F4) - ANT MON SAMPLE
REMT - LOCAL/REMOTE
LOOPZ - ZWORLD LINK TEST
+16 USER - DEFINED COMMANDS

alt-R - REVERSE VIDEO
alt-P - PRINT SCREEN
COM - COM PORT TEST
COMZ - Zworld COM PORT TEST
CMND - DEFINE SPARE COMMANDS
DATE - CHANGE CALENDAR
DFLT - INSTALL DEFAULT VALUES
DOS - ISSUE DOS COMMANDS
DUMP - TRANSMIT DATABASE
EVNT - CLOCKED CONTROL EVENTS
EXIT - LOGOUT REMOTE TERMINAL
FUNC - FUNCTION KEYS
HIST - FREEZE-FRAME SEQUENCE
ID - DEFINE SECURITY IDs
LIMT - METER BOUNDARIES
METR - DEFINE SPARE ANALOG IN
PAL - COLOR PALETTE
PASS - SET PASSWORD
PROG - CONTROLLER
QUIT - QUIT PROGRAM
REP - REPEAT LAST COMMAND
SPAR - DEFINE SPARE DIGITAL IN
SCAL - METER SCALING FACTORS
STLG - STORE A LOG
STOR - STORE CAP POSITIONS
TIME - CHANGE REAL-TIME CLOCK
TRND - TREND ANALYSIS

USE CURSOR OR CLICK LEFT TO SELECT

FIGURE 5

95G0013

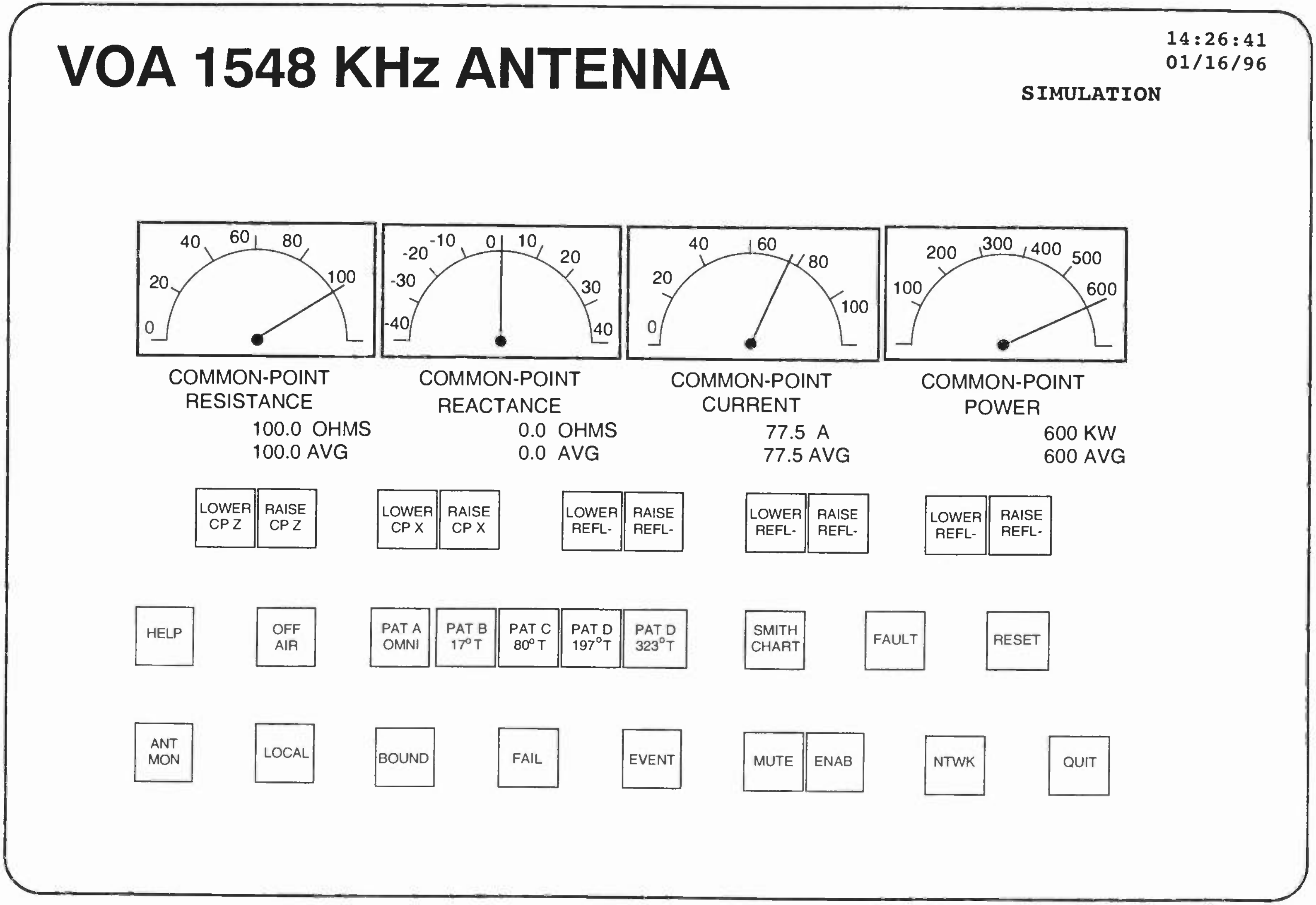


FIGURE 6

96G008

FAILURES and WARNINGS

COMMAND DID NOT EXECUTE	Z1 DATALOAD FAIL
COM 1 ERROR	Z2 DATALOAD FAIL
COM 2 ERROR	Z3 DATALOAD FAIL
COM 3 ERROR	Z4 DATALOAD FAIL
PATTERN FAIL	Z5 DATALOAD FAIL
LOGS FULL	Z6 DATALOAD FAIL
PASSWORD FAIL	Z7 DATALOAD FAIL
KEYBOARD TIME-OUT	Z8 DATALOAD FAIL
LOCAL A/D FAIL	Z1 LOCAL
MUTE FAIL	Z2 LOCAL
REMOTE DATALOAD FAIL	Z3 LOCAL
	Z4 LOCAL
	Z5 LOCAL
	Z6 LOCAL
	Z7 LOCAL
	Z8 LOCAL

FIGURE 7

96G002

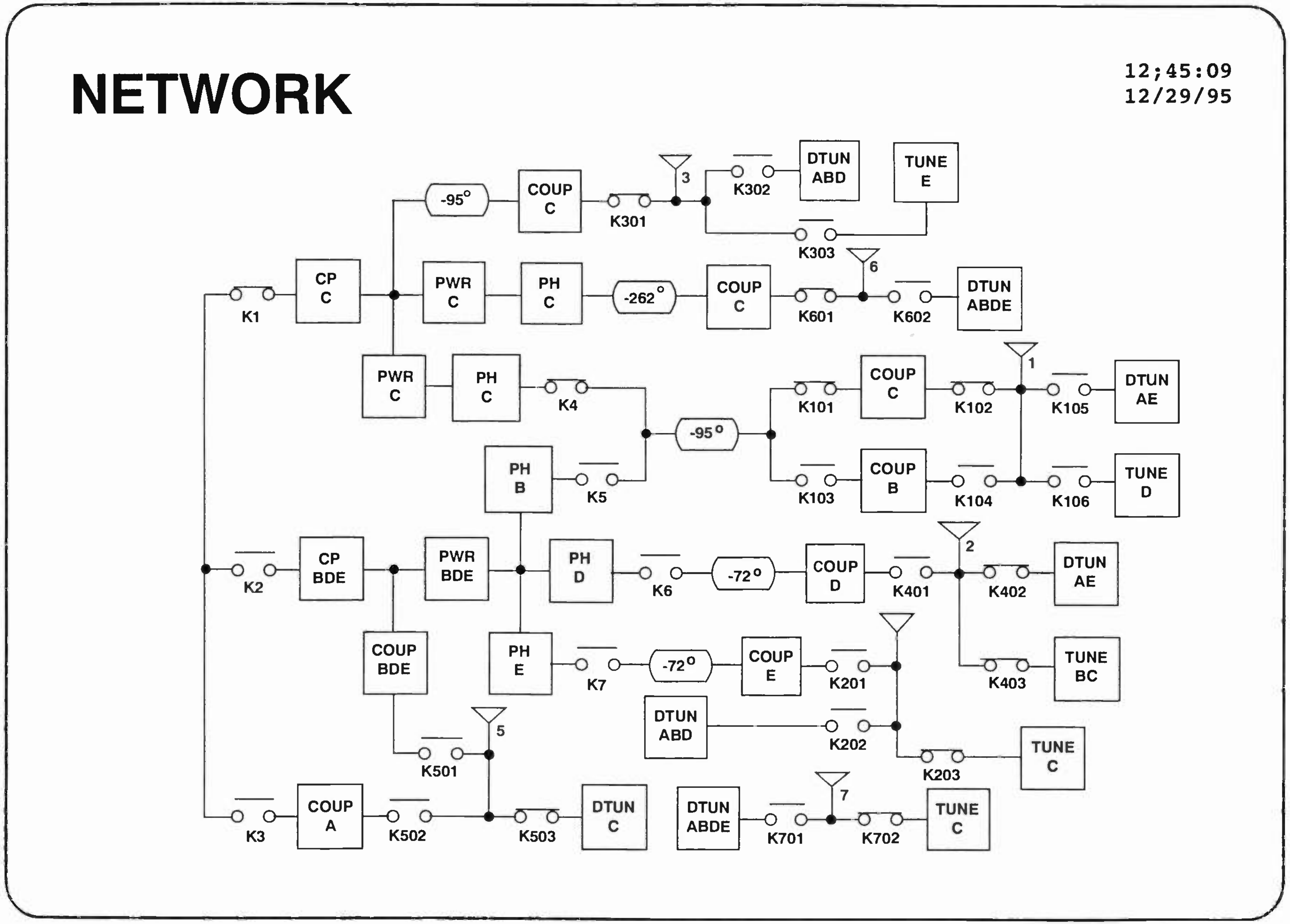

NETWORK
12;45:09
12/29/95
-95°
COUP C
K301
3
K302
DTUN ABD
TUNE E
K303
CP C
PWR C
PH C
-262°
COUP C
K601
6
K602
DTUN ABDE
K1
PWR C
PH C
K4
-95°
K101
COUP C
K102
1
K105
DTUN AE
PH B
K5
K103
COUP B
K104
K106
TUNE D
K2
CP BDE
PWR BDE
PH D
K6
-72°
COUP D
K401
2
K402
DTUN AE
COUP BDE
PH E
K7
-72°
COUP E
K201
K403
TUNE BC
DTUN ABD
K202
5
K501
K203
TUNE C
K3
COUP A
K502
K503
DTUN C
DTUN ABDE
7
K701
K702
TUNE C

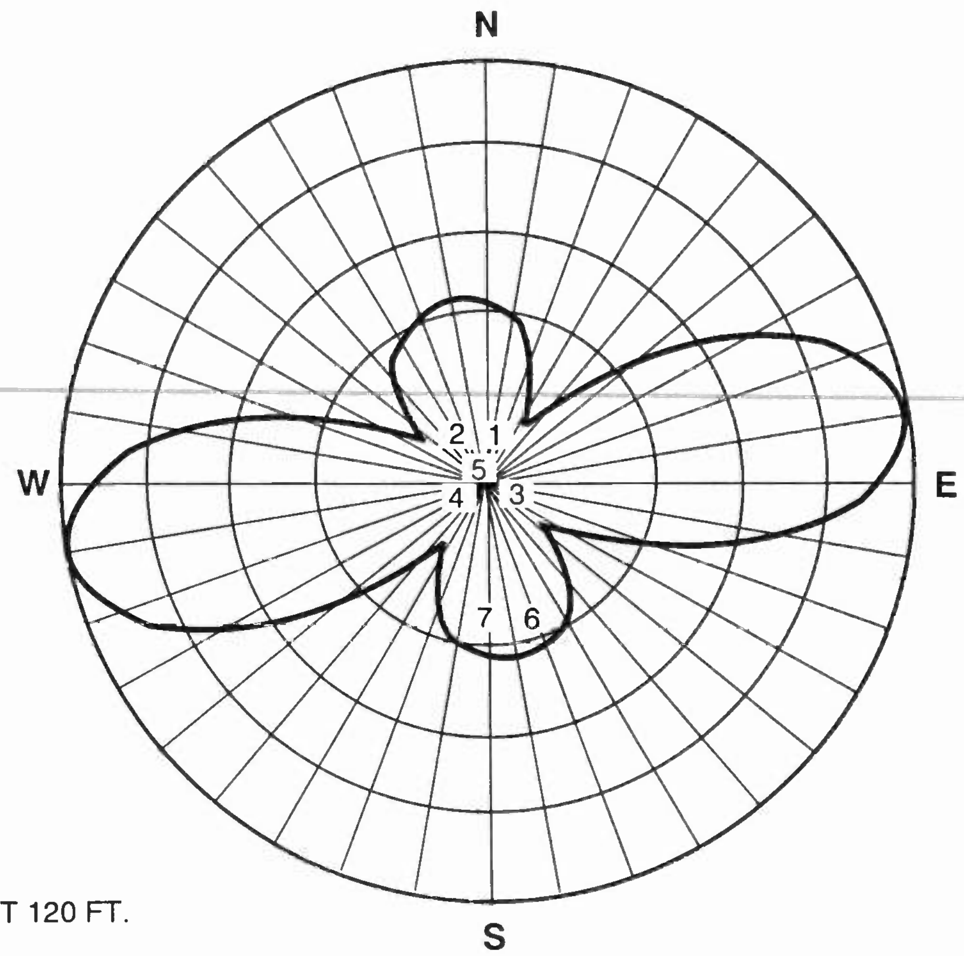
FIGURE 8

RADIATION PATTERN

14:30:52
01/16/96
SIMULATION

EAST (80° T)

SAMPLE		SPACING	BEAR	MAG	DEG
1	LOOP	100.6	16.5	1.00	0
2	LOOP	100.6	323.4	1.00	0
3	LOOP	100.6	143.4	1.00	0
4	LOOP	100.6	196.5	1.00	0
5	LOOP	0.0	0.0	1.00	0
6	LOOP	273.7	160.5	1.00	0
7	LOOP	273.7	179.5	1.00	0



COMMON- POINT POWER 360.kW
ANTENNA MONITOR MODE: RATIO
ONE KM TOWER FIELD= 135mV/m PER AMP AT 120 FT.

FIGURE 9

93G004

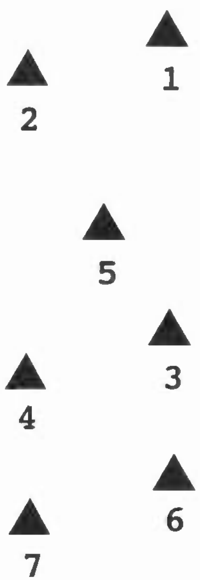
ANTENNA MONITOR

EAST (80° T)

14:28:46
01/16/96

SIMULATION

		DESIRED		RATIO MODE					LINE	
TOWER	kW	MAG	DEG	MAG	ERR	DEG	ERR	SAMPLE	VSWR	ERR
1	0.	.500	.0	.999	.499	.0	.0	LEG 1	1.00	.00
2	0.	.570	95.0	.999	.429	.0	95.0	LEG 1	1.00	.00
3	0.	1.000	.0	.999	.001	.0	.0	LEG 1	1.00	.00
4	0.	1.050	95.0	.999	.051	.0	95.0	LEG 1	1.00	.00
5	0.	.000	.0	.999	.999	.0	.0	LEG 1	1.00	.00
6	0.	.500	.0	.999	.499	.0	.0	LEG 1	1.00	.00
7	0.	.570	95.0	.999	.429	.0	95.0	LEG 1	none	
					2.907		285.0			.00



TOTAL ERROR 287.907

FIGURE 10

96G0009

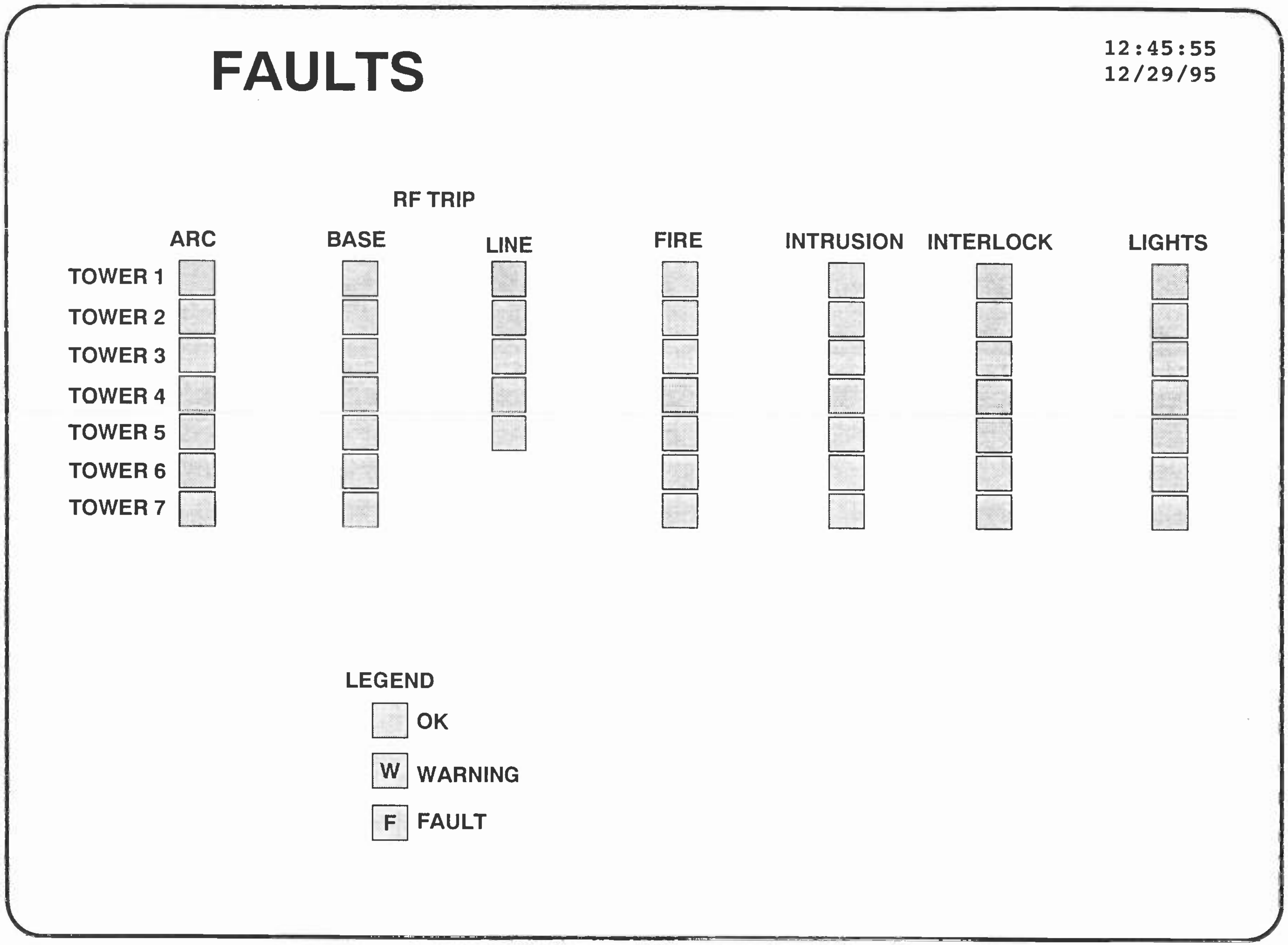
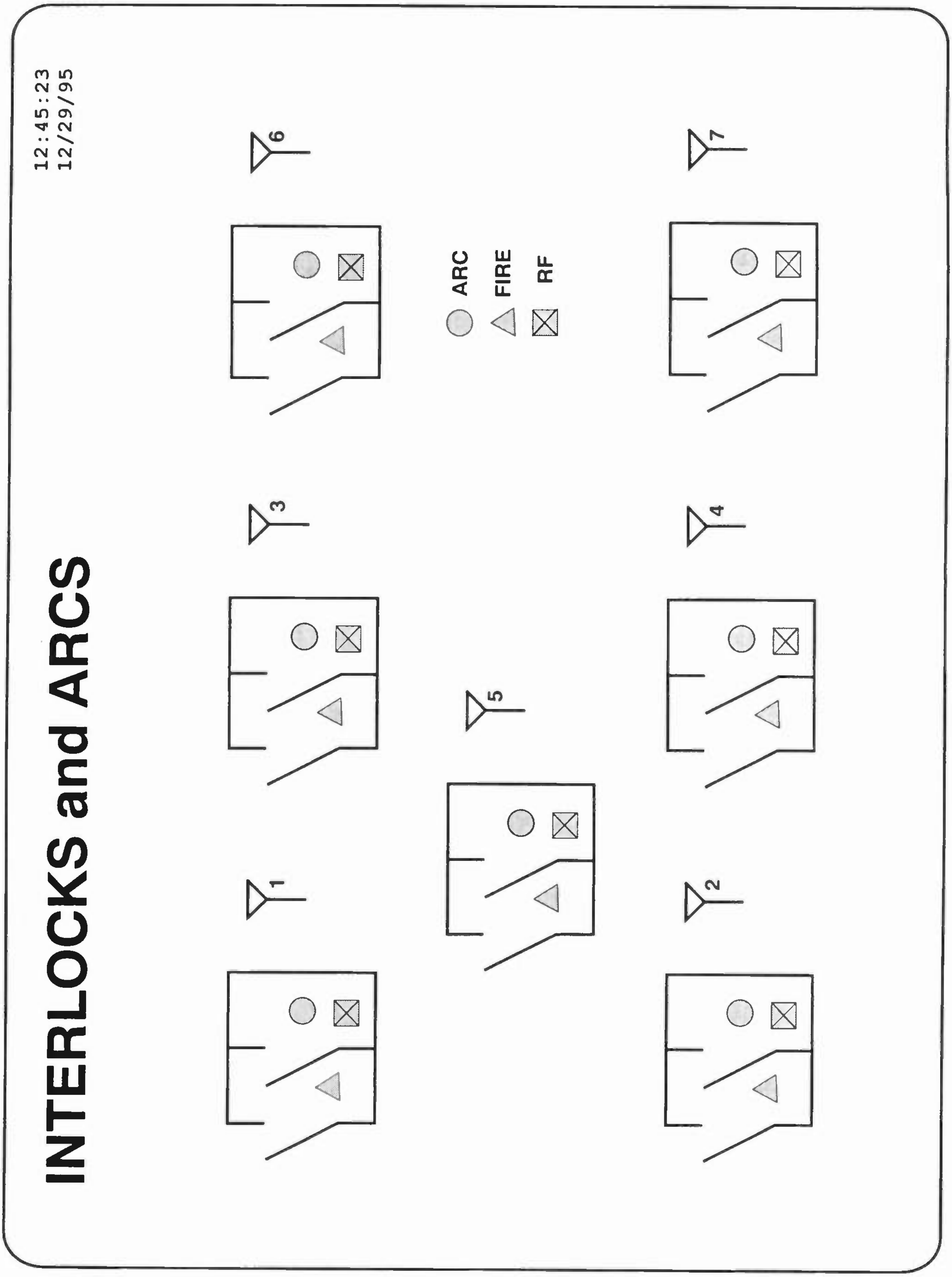


FIGURE 11

96G005



96G006

FIGURE 12

SMITH CHART

TRANSMISSION LINE

	R	X	(RHO)	VSWR
C	100.0	0.0	0.00	1.00
1	70.0	50.0	0.33	1.98
2	100.0	0.0	0.00	1.00
3	100.0	0.0	0.00	1.00
4	100.0	0.0	0.00	1.00
5	100.0	0.0	0.00	1.00
6	100.0	0.0	0.00	1.00

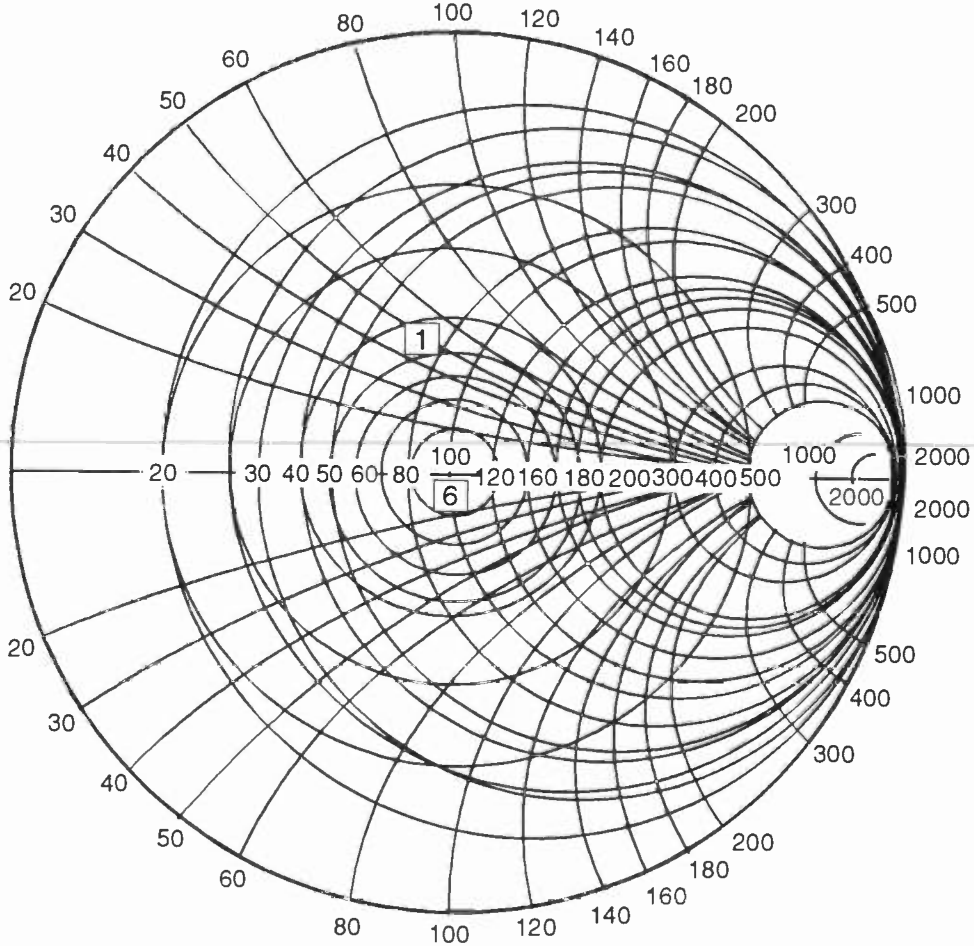


FIGURE 13

96G003

TRANSMITTING SITE EVALUATION USING A MOBILE SPECTRUM MEASUREMENT SYSTEM

Stanley Salek, P.E. and Robert P. Smith, Jr.
Hammett & Edison, Inc.
San Francisco, CA

ABSTRACT

This technical paper describes general development and application of a mobile coverage data collection system, employing an RF spectrum analyzer, a Global Positioning System (GPS) receiver, and a custom digital signal processor interface. The paper also describes how the system was used to gather coverage data on an existing lower power FM facility already operating from the potential relocation site of a client FM station. That data was then appropriately scaled and studied using powerful coverage analysis and mapping tools, resulting in the development of a relocation profile for the client station.

Other system applications include comparative analysis of station coverages in a particular home market, an adjacent market, or in weak signal areas, and evaluation of FM booster transmitter performance. Gathered field strength data can be linked with similar data predicted by popular terrain-sensitive computer algorithms, providing a "predicted-to-actual" performance comparison. This ability additionally allows easier troubleshooting of transmitting installations that provide coverage perceived to be poorer than expected.

INTRODUCTION

The need to better understand the physical nature of signal propagation has driven development efforts to produce improved computer modeling and measurement verification tools. These tools have been used to predict and verify the coverage of broadcast facilities, especially that of TV and FM stations.

With respect to field verification, mobile field strength data gathering systems for coverage analysis are becoming increasingly common. Typically, these systems are based on single-channel or scanning receivers. These receivers obtain signal level information for only one channel at time and, unless multiple receivers are employed in parallel, some settling time must be allowed after a frequency change. When multiple channels are being measured, this requirement often restricts the data sampling interval on each channel. Typical receivers also have difficulty in providing accurate signal level information in the presence of strong adjacent-channel signals, and they are also subject to receiver-induced intermodulation effects; both of these factors can contaminate measurement data. Also, some receivers have a limited dynamic range for use in conducting mobile field strength measurements, and the inclusion or removal of external filters, attenuators and/or preamplifiers must be constantly addressed by a trained operator or an automated process to avoid the possibility of signal loss or receiver saturation.

A broadband spectrum analyzer is a signal measurement instrument that overcomes many of these limitations. With proper selection of the frequency span, multiple signals can be measured nearly simultaneously, and proper choice of resolution bandwidth can yield accurate results, even in a crowded signal environment. Further, the considerable viewable dynamic range available on most spectrum analyzers allows simultaneous measurement of weak signals in the presence of strong signals. The spectrum analyzer is still a swept measurement instrument, so data for multiple signals is not truly available simultaneously, but no settling time is required to yield usable data. However, a spectrum analyzer is traditionally a "bench" instrument. The goal of the present effort was to develop a recording system that could utilize the advantageous measurement capabilities of a spectrum analyzer in a mobile data collection system.

SYSTEM DESIGN

The system is designed around a low cost Tektronix Model 2710 spectrum analyzer. The key component in the system is an interface module between the spectrum analyzer and a portable computer. The module uses a digital signal processor (DSP) circuit to digitize the analog sweep output signal from the analyzer, perform signal processing, and convert each display sweep into a

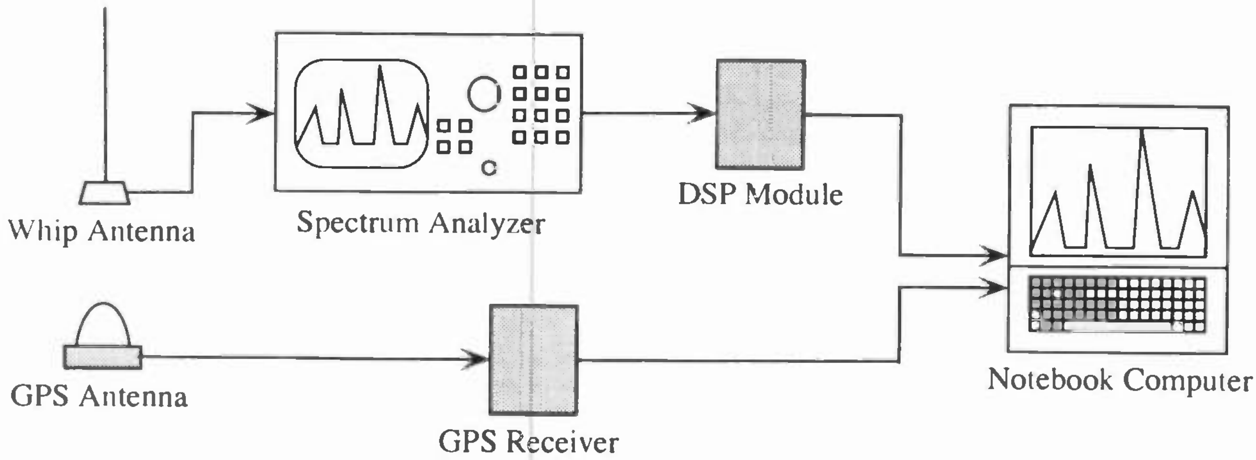


Figure 1. Block diagram of spectrum analyzer recording system.

packet of data that is then transmitted to the computer. The system also includes a GPS satellite receiver to provide position data. Custom software running on the computer controls the system and records the data. Figure 1 shows a block diagram of the data gathering portion of the system.

Receiving antenna. A quarter-wave vertical whip antenna was selected for use with the system, because it is most representative of antennas commonly installed on automobiles for FM reception. While other types of nondirectional mobile antennas could be employed, the vertical whip antenna is easy to use and to mathematically characterize. It is adjusted by setting its length to correspond with the wavelength of a desired signal being measured or to the center frequency of the selected spectrum analyzer span. The metal roof or body of the vehicle on which it is mounted provides the required counterpoise. Magnetic mount versions of whip antennas are easily transported, for use on rented vehicles.

Spectrum analyzer. The selected spectrum analyzer covers a frequency range of 100 kHz to 1.8 GHz with 70 to 80 decibels (dB) of usable dynamic range [1]. Through careful selection of sweep span and resolution bandwidth parameters, the sweep time can be relatively short, in the 0.25 to 0.5 second range for a full sweep, making it well suited for mobile measurement applications where signals are changing rapidly. The spectrum analyzer provides output signals suitable for processing and recording, including an analog sweep trace signal and a digital sweep gate signal.

DSP module. The DSP module contains a Texas Instruments TMS320C50 DSP chip and a TLC32040 analog interface circuit (AIC). The DSP chip [2] is an advanced 32-bit processor running at a clock rate of 20 MHz. The AIC [3] contains an analog-to-digital converter with 14-bit resolution capable of sampling at rates of up to 20,000 samples per second. The output signals from the spectrum analyzer are fed to the DSP module, with the sweep trace signal connected to the AIC and the sweep gate signal connected to processor interrupt lines. The DSP code receives interrupts at both the start and end of the sweep gate. The DSP module digitizes the sweep trace signal only during the active portion of the sweep. Digitized data for each sweep is delivered to the control computer in a single data packet through a full-duplex asynchronous serial interface. The DSP has sufficient processing power to perform serial transmit-and-receive signal processing as well as other functions; no separate serial interface hardware is needed.

The sweep trace signal is digitized at a sampling rate of 19,200 samples per second. This rate was chosen because it allows serial data transmission to be driven by the analog sampling interval, simplifying the code design. Depending on the spectrum analyzer sweep parameter settings, the duty cycle of the sweep gate can be as high as 90%. Consequently, the DSP module cannot store the sweep data for transmission during the inactive portion of the sweep gate, so serial transmission to the computer is synchronized with data sampling. The DSP performs peak-hold processing on the digitized signal to reduce the sampling rate for transmission. Serial transmission limits the data size to 8 bits per sample, so the signal is clipped and scaled from the 14-bit signed output of the AIC to an 8-bit unsigned signal for transmission to the computer.

The operational code for the DSP is downloaded from the computer when the system is initialized. This feature simplified and accelerated the development process, allowing the code to be revised quickly and easily and greatly shortening the debugging cycle.

GPS receiver. The GPS receiver employed in the measurement system is a Magellan five-channel module, which requires an external controlling device communicating through an asynchronous serial interface. Software on the control computer provides a status display and user interface for the GPS receiver. The receiver provides position updates to the computer at one-second intervals.

Computer and recording software. The computer must initialize and control the DSP module and the GPS receiver. A multi-tasking operating system is used on the computer so the software applications controlling each device can be entirely separate. This scheme greatly simplified and accelerated the development process. The operating system used is Linux, a popular public-domain operating system developed by a collaboration of programmers on the global Internet. While there was some trepidation at using non-commercial software, Linux met or exceeded every expectation, and proved to be extremely reliable.

The spectrum analyzer software application initializes and controls the DSP module and provides a real-time graphical display of the digitized sweep data from the analyzer. The system operator can visually compare the corresponding screen displays on the analyzer and computer to confirm that the system is functioning properly. The software records each analyzer sweep, along with a time stamp and control information, to a file on the computer hard disk drive. The operator may manually enter reference markers in the data stream to note significant events or locations during recording. The GPS software application controls the GPS receiver and records position reports to a disk file. Time stamps are included, and since they are referenced to the same clock being used by the spectrum analyzer recording software, the separate data streams are correlated. The operator may switch between the two applications on the computer screen whenever desired without interrupting the execution of either.

The first use of the system was with a notebook computer containing a 386SX processor running at 16 MHz, with 4 megabytes of memory and a 60 megabyte hard disk drive. This modest platform was perfectly adequate to the task, although the small disk drive limited the total recording time; the spectrum analyzer recording software generates data at the rate of 3 to 4 megabytes per hour. The computer has since been upgraded to another notebook employing a 486DX2 processor running at 66 MHz, with 8 megabytes of memory and a 500 megabyte disk drive. The more powerful computer increases total recording time, and it also facilitates data post-processing and analysis in the field. Separate post-processing and analysis software, discussed later, requires a considerably more powerful computer platform than does the recording software.

SYSTEM IMPLEMENTATION

The system is designed to be installed in a vehicle; it can be set up in less than 30 minutes. The spectrum analyzer requires 120 VAC power, so the remainder of the system was designed to operate from a 120 VAC source. The system is typically installed in a modified GMC "Safari" field measurement van, which is equipped with interior bench space and a high capacity AC power system. Antennas for GPS and RF reception are permanently installed on this vehicle. The total power requirement of the system is a relatively modest 175 W, so if the system must be used in a different vehicle, power can be supplied by a portable DC inverter. Portable magnetic-base GPS and vertical whip antennas would be used in the latter configuration. The only inconvenience encountered with the portable DC inverter is the need to run power cables directly to the vehicle battery to accommodate a current drain greater than the capability of common cigarette lighter jacks.

The most significant installation problem encountered was radio-frequency interference (RFI) from the recording system hardware. The spectrum analyzer itself is very well shielded and not a direct source of RFI. The DSP module and GPS receiver are both constructed in shielded enclosures and were found to be minimal contributors. It was expected that the largest RFI source would be the computer itself, but surprisingly, it too was found to be only a minimal source of interference. Conducted energy on the power and control cables, however, was significant. Extensive use of shielded cabling and ferrite cores reduced the problem to a tolerable level, sacrificing less than 5 dB of the measurable dynamic range. Depending on the frequency band being measured, an additional significant source of RFI can be vehicle engine ignition systems. This interference can originate not only from the measurement vehicle itself, but from nearby and passing vehicles on the road. Thus, shielding the measurement system from such interference or eliminating it at the source is impossible. Fortunately, the appearance of this interference in the recorded spectrum data has a unique signature that allows it to be eliminated through virtual filtering in the post-processing software, as discussed later. Figure 2 is a photo of the system installed in the Hammett & Edison measurement van.

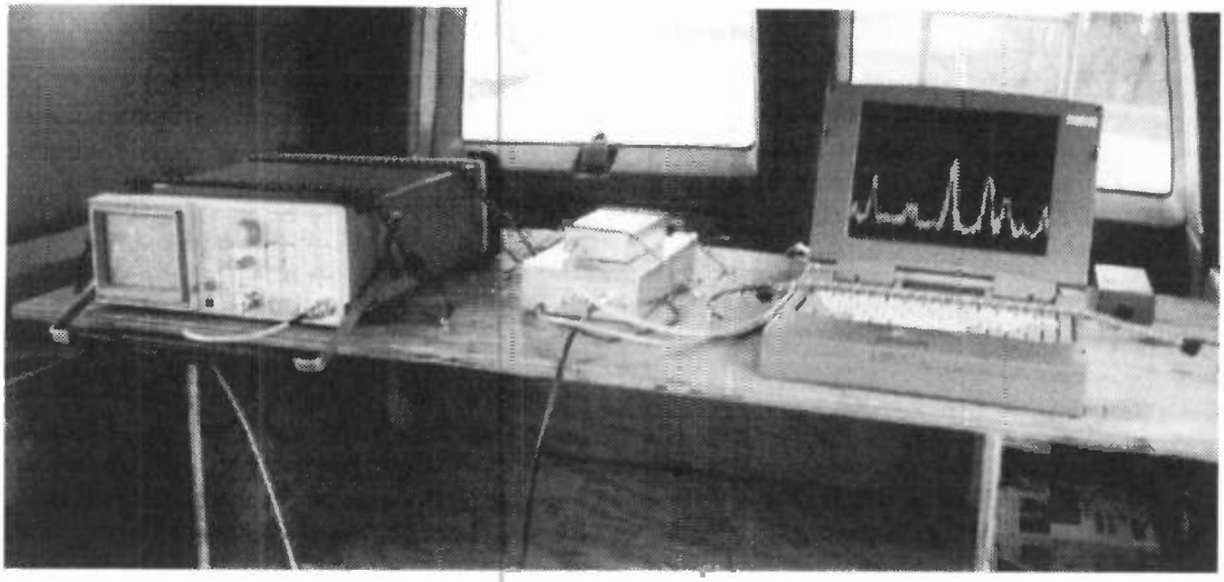


Figure 2. System installed inside measurement vehicle.

SYSTEM OPERATION

During actual data gathering, the system requires relatively little operator intervention. It can be used in an unattended manner, with a single individual used both to operate the system and drive the vehicle. The operator starts the system and initiates data recording, then proceeds to drive the measurement route without the need for any further attention to the recording system. When used in a moving vehicle, the GPS system is prone to brief outages, during which position reports are either inaccurate or not available. They occur due to localized obstructions interfering with one or more of the GPS satellite signals, or due to the need for the receiver to periodically change the specific satellites being tracked. Outages are usually insignificant, and the analysis software can interpolate between periods of valid position data to reconstruct the route. GPS data may not even be required if an operator is available to record accurate position markers.

DATA ANALYSIS

The recording system gathers volumes of raw data, which must be processed into usable signal level versus location information. The analysis process occurs in two steps. First, the raw sweep-trace data from the spectrum analyzer is processed and converted to produce signal level versus time data. Then, separate applications correlate this data with the GPS position data or manual position markers to produce the final results.

RF system calibration. Before the raw spectrum analyzer sweep data can be processed, a correlation of the recorded digital values to the screen graticule of the analyzer is needed. This produces a data mapping of the raw digital values from the recording system to equivalent relative positions on the analyzer screen. This data was obtained several times over a period of months, and correlation was found to be highly stable; the correlation data remains constant as long as the analyzer itself does not undergo a factory recalibration. Overall accuracy ultimately depends on the calibration of the analyzer instrument itself, so frequent factory recalibration of the analyzer is important.

With correlation data available, the operator of the analysis software need only provide the range and sweep parameters used during a particular recording session. These parameters include the amplitude reference level, the sweep center frequency, and the sweep range per division. The analysis software can then translate any point in the recorded data to a signal level at a known frequency.

The signal level value obtained by this process represents receiver terminal voltage at the analyzer input. To equate it to actual field strength, the receiving antenna and interconnecting cables used also must be considered. For the vertical whip antenna, the following formula, derived from antenna theory, can be used to convert the voltage at the antenna terminals to a corresponding electric field value:

$$E = V + 14.65 - 20 \log(\lambda) + L,$$

where E is the electric field in dBuV/m (or dBu), V is the voltage measured at the antenna terminals in dBuV, λ is the wavelength in meters, and L is the loss of the antenna lead in dB for a given wavelength. The equation assumes an antenna gain of 5.15 dBi [4] and a system load impedance of 50 ohms. Practical use of this equation assumes that the antenna/vehicle combination performs as a theoretical quarter-wave antenna over a perfect ground plane.

When highly accurate field strength data is desired, supplementary measurements are taken using a calibrated reference receiver and antenna to determine field strength at specific locations, and then they are compared to the indicated results at the same locations with the measurement antenna. In general, though, most applications of the system are not intended to determine absolute field strengths at specific locations. Rather, they are intended to produce relative levels and field strength trends that can be correlated to results predicted by terrain-weighted computer propagation analysis, such as that produced by the Terrain-Integrated Rough Earth Model (TIREM) algorithms [5]. Experience with the system has shown that measurements in the broadcast FM band, using a measured-length vertical whip antenna on the roof of a metal-bodied vehicle, can yield field strength figures that correlate well with critical measurements and with the results of computer propagation studies.

Spectrum data analysis. Once adequate calibration data is obtained, the operator uses an additional software application to study the recorded spectrum analyzer data and produce signal level versus time data. The analysis software provides the operator with a graphical display that mimics the screen of the spectrum analyzer. The software can replay the recorded sweep data in real or accelerated time, or allow the operator to single-step through the recorded sweeps. Many sweep and display features available at the spectrum analyzer front panel can be emulated by the software, such as a peak-hold display. The major focus of the software, though, is to produce amplitude data for the signals being studied. To accomplish this task, the operator can define one or more channel windows, entered as a center frequency and frequency span. When displaying each recorded sweep, the software will determine the peak signal observed in each of these channel windows. The software can convert the measured signal level values to various units, including a field strength based on calculated or measured antenna parameters. The values are displayed on the computer screen, and on user command they can be recorded to a disk file for use in a second stage of analysis. The output data indicates the time at which a particular sweep was recorded along with the signal values for all defined channels.

Several problems that can exist in the recorded sweep data may require additional data processing features in the analysis software. As mentioned earlier, the recording system is inherently and unavoidably prone to RFI from vehicle engine ignition systems. The interference appears in the sweep data as a series of extremely narrow signal spikes as the periodic short-interval broad-band noise bursts from ignition firing occur. The data analysis software eliminates much of this ignition-spike noise by passing the recorded sweep data through a simple magnitude-comparison routine, discarding small groups of data samples that greatly exceed their neighbors in magnitude. This approach is effective as long as the spectrum analyzer sweep parameters are chosen so the resolution bandwidth is much smaller than the expected occupied bandwidth of the signals being measured, resulting in short-interval spikes in the data that are never desired information.

Even after the ignition-spike noise is reduced or eliminated, recorded data still can be very rough, due to the effects caused by signal modulation. Using a single peak recorded value in a channel window to determine signal strength produced data with a high level of random variation. A method was needed to eliminate as much of this spurious information as possible. An approach was chosen to treat the recorded sweep data as a simple time-varying signal and apply low-pass filtering to eliminate noise. After filtering of the ignition spikes as described above, the analysis software uses digital finite impulse response (FIR) filtering, with variable parameters, to smooth the data for a recorded sweep. The software can then take the single peak value from the filtered data in each channel window and convert it to a signal level having a much lower degree of random variation due to modulation effects and measurement artifacts, with any variation remaining likely to be significant data. Figures 3 and 4 show a single recorded sweep before and after these filtering steps are applied.

Signal analysis. The final stage of data analysis is to convert the signal level versus time data, from the spectrum analysis, to signal level versus location data. Additional software routines correlate the recorded GPS position data to the signal level data based on time. The GPS data often requires some additional processing to eliminate spurious position reports due to localized disruptions in GPS signal reception. The analysis software filters the recorded position data by applying limits on the apparent velocity of the vehicle. When two position reports are farther apart than could be possible given the time interval between them, both are discarded.

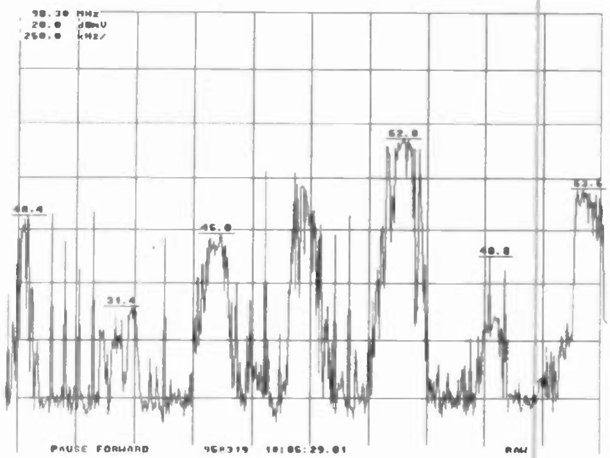


Figure 3. Raw recorded spectrum sweep data.

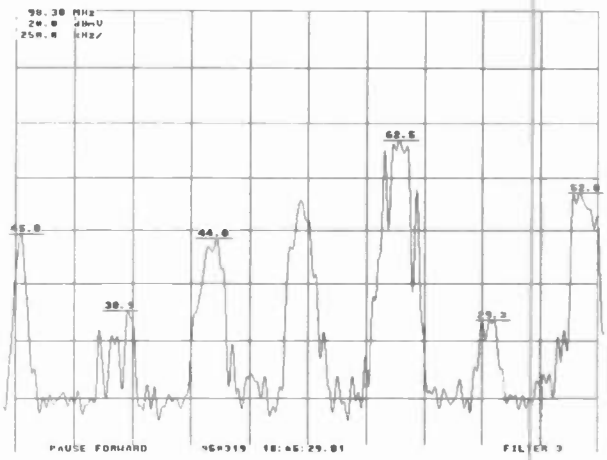


Figure 4. Spectrum sweep data after filtering.

Once the position data has been filtered, an interpolated position for each recorded signal level value is determined based on the time stamps in the data. The software does have the ability to supplement the GPS position data with location markers recorded by the system operator during the measurement process, so signal data can be used even when GPS is not available, as long as accurate and sufficiently frequent markers are recorded.

The final result of the analysis process is a set of geographical locations and signal level readings at those locations. These data can be presented on a map, or further processed to produce signal level contours or to determine information such as desired-to-undesired (D/U) signal ratios.

SYSTEM APPLICATION EXAMPLES

Having presented information on how the system was designed and how it operates, the real proof of its value lies in the results that it can provide with respect to actual project applications. To date, some of the more interesting system applications have involved evaluation of FM transmission systems with regard to transmitting site selection, booster performance verification, and comparative signal level determination.

Transmitting site evaluation. The developed system actually has its roots in a project to evaluate a prospective transmitter site for a client FM broadcast station, and to determine if operation from that site would provide improved coverage in desired market areas over the station's existing transmitting facilities. While the prospective transmitter site provided greatly improved height, its location was considerably father from the desired coverage areas than the existing site. Therefore, a proper selection was not intuitive.

Traditional analysis involving FCC methods was not useful to determine the better site, because terrain obstructions in the signal propagation paths from each of the sites could not be adequately modeled using "average terrain" data. Terrain-weighted computer propagation studies were conducted, but the collected data yielded inconclusive results when a possible statistical error of several dB was considered. It was determined that objective signal level measurements, taken simultaneously from both transmitting sites, would provide the most useful comparative information. The proposed site already supported another FM facility, albeit operating at much lower power than could the client station at that site. However, signal level measurements taken on the lower power facility could be appropriately scaled to provide valid data for the proposed facility.

Field measurement studies were conducted in several populated areas within the desired station listening area. The spectrum analyzer was set up to display both the station's existing signal and the signal of the proxy station. To be sure that the collected data would be valid at the center of the measurement area, each area was traversed in different intersecting perpendicular directions (north-to-south, then east-to-west, for example), in paths that exceeded at least several city blocks, so that collected data could be correlated.

In the analysis phase, a mean signal level was calculated for each station using a one-minute data sample centered at the intersection location, using approximately 240 total measurement points. It was found that data processed in this manner for the intersecting paths yielded excellent correlation, with only a 1 to 2 dB difference at most measurement locations. With regard to the determination of the better site, the analysis showed that the proposed transmitter site would provide greater signal level in 72 percent of measurement locations, the existing site would provide greater signal level in 14 percent of the locations, and that the signals would be essentially equal (± 3 dB) at the remaining locations. Thus, the proposed site was clearly shown as superior, provided that signal losses in the 14 percent of areas served better by the existing facility would be acceptable.

FM booster performance verification. Another project involved field verification of synchronous FM booster transmitter performance, and determination whether the booster would cause significant interference to the signal of the station's main transmitter in populated locations.

Previous laboratory and field work has yielded data on the key performance traits of FM booster transmitters, and how they interact with their host FM stations [6]. In summary, interference between main and booster facilities, which operate on the same FM channel, was noted when the associated signal levels were found to be within about 15 dB of each other. As expected, terrain shielding between the transmitting sites and coverage areas of the main and booster stations provided the least interference between the facilities and, hence, the best performance. Though optimum separation of transmitting sites is not often possible, carrier and modulation synchronization of the facilities, utilizing an adjustable time delay [7], may be used to "steer" predicted interference zones into unpopulated areas, while maintaining a good transition between main and booster facility signals in desired coverage areas.

The measured client station operates a main transmitting facility that is situated on the north-facing slope of a mountain. While most of the desired population covered by the station also lies to the north, recent population growth in the area south of the mountain yielded a significant number of persons without adequate service from the station, even though that population was inside the "protected" FCC contour of the station. Thus, construction of a booster transmitter was initiated. The actual booster facility was constructed on the southern slope of the same mountain, employing a directional antenna oriented to the south. Calculations were made to set adjustable system delays, such that the main and booster signals would be time synchronized at a populated location along a major highway where both signals were predicted, using the TIREM terrain-weighted computer propagation analysis algorithm, to be equal in strength.

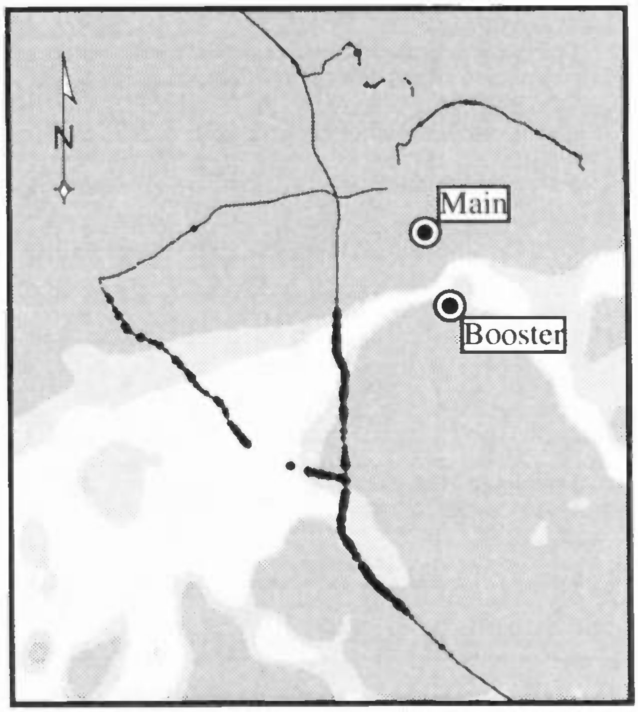


Figure 5. Calculated and measured D/U ratios for example FM booster installation.

Field verification of the booster system design involved separate measurement of the main and booster transmitting facilities, one at a time, along identical measurement routes. (Only the facility being measured was activated during measurement.) The collected data was then synchronized for position so the facilities could be compared and D/U ratios could be determined. Figure 5 shows the results of the study. The thin lines show the routes traveled, as derived directly from GPS data; discontinuities in the lines are due to interruptions in GPS data caused by localized terrain features or changes in GPS satellite geometry. A thickening of the lines indicates a reduction in D/U ratio for measured signals within 15 dB of each other, with the lowest relative D/U ratios at the thickest parts of the lines. The map is overlaid onto a calculated TIREM plot of the areas predicted to have reduced D/U ratios; the darkly shaded areas are predicted to have D/U ratios greater than 15 dB, while the next two lighter shaded areas depict D/U ratios of 10 to 15 dB and 5 to 10 dB, respectively. White areas are predicted to have D/U ratios of 5 dB or less. Thus, for the routes measured, good correlation was found between TIREM predictions and actual D/U level measurement.

Figure 6 is a graph of overlaid relative main/booster signal levels versus distance, showing the proper location for transmitter time synchronization. It was found that the measured synchronization point precisely matched the point that had been previously calculated using TIREM data. Thus, the booster system was already properly adjusted for best performance in the transition area.

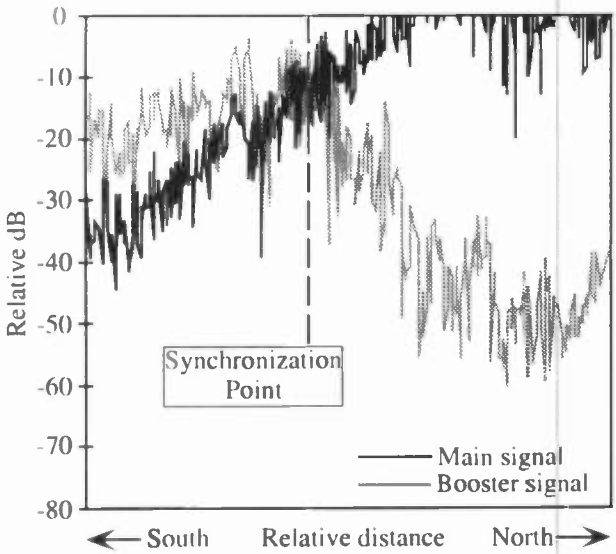


Figure 6. Measured signal levels of main and booster stations near area of lowest D/U ratio.

Other applications. In probably the simplest and most straightforward application, the system has been used to measure the relative signal levels of multiple FM stations in a market, allowing direct comparison of the strengths and weaknesses of various facilities. Other specific applications include interference analysis between FM stations in distant markets and analysis of cellular telephone base station performance. In general, the RF signal levels of almost any communications system used in the mobile environment can be characterized and, as required, compared to similar facilities.

CONCLUSION

Valuable information can be obtained by using a low cost spectrum analyzer for RF signal level measurements in a mobile environment, especially when it is tied to appropriate data sampling and logging apparatus. The developed system overcomes many of the drawbacks associated with the use of fixed-tuned receivers in similar field applications, and it allows critical data to be post-processed in a form that preserves multiple variables, allowing thorough and concise post-measurement analysis.

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AMERICA'S FIRST EXPANDED BAND STATION WJDM ELIZABETH, NEW JERSEY

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ABSTRACT

Since the FCC announced the expansion of the AM band several years ago, the potential for high quality operation offered by the Expanded Band has intrigued broadcasters. This paper explores the design, construction and technical operation of America's first Expanded Band Station: WJDM Elizabeth, New Jersey. The technical highlights of the WJDM Expanded Band operation include diplexing with only 130 kilohertz frequency separation, the use of a folded unipole antenna and wide band stereo audio performance. In addition, project management aspects that should be common to constructing similar Expanded Band facilities will be discussed.

BACKGROUND

The Expanded Band is an extension of the AM broadcast band from 1600 to 1700 kilohertz. As in the existing AM band, the channels are spaced at 10 kilohertz intervals, thereby creating ten new channels. The purpose of the Expanded Band is twofold. The first is to diminish interference in the existing AM band by moving the stations that currently cause the most interference to the Expanded Band. The second is to foster superior signal quality for the stations that will operate in the Expanded Band through the creation of what the FCC designates as a Model 1 facility. In Section 73.14 of the Commissions Rules and Regulations, the FCC defines a Model 1 facility as "a station operating in the 1605-1705 kHz band featuring fulltime operation with stereo, competitive technical quality, 10 kW daytime power, 1 kW nighttime power, nondirectional antenna (or simple directional antenna system), and separated by 400-800 kM from other co-channel stations". After a station is assigned to the Expanded Band, it is permitted to operate on its original frequency for a period of five years, after which operation on the original frequency must cease. Consequently, interference to the stations remaining on the original AM broadcast band is reduced.

When the FCC formulated the Expanded Band allocation process, a three tier priority system was created. The goal of this system is to relocate the stations that cause the most interference to other stations. Under this plan, which is contained in Section 73.30 of the Commission's Rules and Regulations, all stations within their respective tiers were ranked by the FCC according to the "improvement factor" that relocating a station to the Expanded Band would provide to the remaining stations in the AM broadcast band.

The first priority was for daytime only stations licensed to cities with populations in excess of 100,000 which had no local fulltime aural service. The second priority was for fulltime stations with the highest improvement factors.

WJDM is licensed to Elizabeth, New Jersey. Based on the 1990 U.S. Census, Elizabeth's population is 110,002. Therefore, WJDM qualified for first priority in the Commission's allocation plan and was ranked number one overall of all the stations selected to receive an Expanded Band allocation. Radio Elizabeth, Inc., the licensee of WJDM, applied to the FCC for a Special Temporary Authority (STA) to construct their Expanded Band allocation immediately so that a local fulltime service could be established for Elizabeth, New Jersey at the earliest possible time. The FCC granted Radio Elizabeth's request.

TECHNICAL DESCRIPTION

Prior to the establishment of its Expanded Band operation, WJDM's operation was limited to daytime hours only on 1530 kilohertz with power of 1000 watts except during the critical hours when power is reduced to 670 watts. The antenna system is nondirectional and employs a folded unipole which consists of a three wire skirt which is mounted on a grounded self supporting tower of uniform cross section. The electrical length of this antenna is 57.7 degrees. The Expanded band operation transmits on 1660 kilohertz with power of 10 kilowatts daytime and 1 kilowatt nighttime and is diplexed into the WJDM 1530 antenna system. The electrical length of the unipole at 1660 kilohertz is 62.6 degrees. Both stations broadcast in stereo and utilize solid state transmitters.

The WJDM Expanded Band transmission system consists of a Harris DX-10 transmitter, a Harris AMS-G1 stereo generator, and a custom designed diplexing system manufactured by Phasetek. Audio is fed to the transmitter site via a pair of 15 kilohertz equalized telco circuits which feature identical cable routing in order to maximize stereo performance. Audio processing is provided by an Optimod 9100B2.

DIPLEXER DESIGN

Due to the close frequency separation between the two WJDM operations, 130 kilohertz, careful design choices for the matching and filter circuits in the diplexer were made to provide adequate frequency isolation and optimum system bandwidth. As previously described, the WJDM antenna is a folded unipole of short electrical length. It was necessary to short (stub) the unipole to provide reasonable impedance values and slopes for the desired frequency ranges. The final setting yielded a feed point resistance of 32.5 ohms for 1530 kilohertz and 155 ohms for 1660 kilohertz. Both impedances had a high inductive reactance.

FIGURE 1 is a schematic diagram of the diplexer system. The circuitry for each frequency includes a full "T"

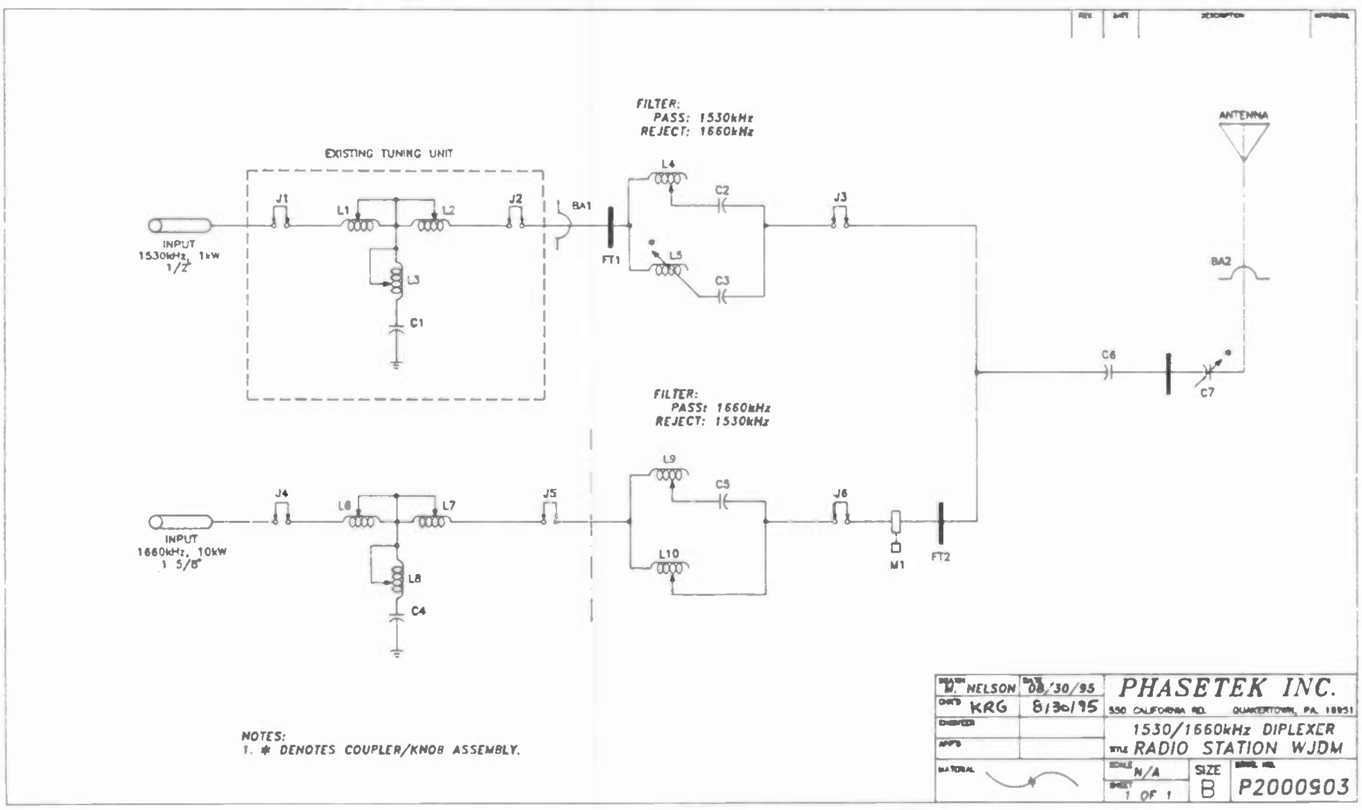

FIGURE 1
Schematic diagram of the WJDM diplexer

matching network and series reject filter. The filters were designed to provide 40-50 db of rejection while minimizing bandwidth limitation. The matching networks for both frequencies were optimized for the final measured impedances yielded from the unipole setting and filter response. In addition, the shunt leg of the 1530 kilohertz matching network (L3/C1) was configured to shunt 1660 kilohertz signal. This design consideration provided an additional 10 db of isolation with minimal bandwidth change. Fixed capacitor C6 and variable capacitor C7 are in series with the feedpoint to slope and reduce reactance at this point. Variable capacitor C7 was adjusted for 1660 kilohertz near resonance to lower filter voltage and deliver a considerable bandwidth improvement. While tuning the system, it was observed that at frequencies above 1600 kilohertz, stray capacitance and mutual coupling were much more predominant. At these frequencies, it would be desirable to increase inductor spacing 25-30% to minimize these effects. Severe space limitations at the WJDM transmitter site precluded increased inductor spacing in the diplexer. As with any diplexer, careful adjustment was required to obtain the desired results.

SYSTEM PERFORMANCE

Based on the use of a diplexer with close frequency separation, there were four design goals considered to be paramount for the WJDM Expanded Band facility. The design goals included adequate RF isolation in the diplexer between the two WJDM carrier frequencies, attenuation of spurious emissions in accordance with Section 73.44 of the Rules. RF efficiency that meets or exceeds the standards of Section 73.189 of the Rules and audio performance that is significantly in excess of FCC minimum standards.

As previously discussed in the DIPLEXER DESIGN section of this paper, RF isolation approaching 60 db was obtained. This figure proved to be adequate to eliminate undesirable effects between the two WJDM operations. Field strength measurements of spurious emissions, which were taken every 90 degrees at a distance of one kilometer from the WJDM transmitter site, were in compliance with FCC requirements. Additional field strength measurements were conducted to ascertain inverse distance field radiation values in order to verify proper radiation efficiency. The results of the measurements indicated good radiation efficiency for both frequencies. Skywave performance has exceeded expectations, with documented DXer reports and audio tapes received not only from throughout the United States, but from European countries as well, including Italy, Finland, Norway and England.

An audio proof of performance could not be completed in time to meet the publication deadline for this paper. However, listening tests and preliminary measurement data, such as frequency response within plus or minus 0.75 db between 50 and 9000 kilohertz, point to strong performance. Final results of the audio proof of performance will be discussed at the oral presentation of this paper at NAB96, and documentation will be provided.

PROJECT MANAGEMENT

The construction of a new transmission facility, such as the WJDM Expanded Band project, is a major undertaking. Since there will be approximately eighty similar Expanded Band facilities constructed in coming years, universal techniques useful in creating a successful project will be discussed. Successful project management requires planning, organizational and supervisory skills. These skills are applicable to any type of major project. The author assumes the reader is skilled in basic broadcast engineering techniques.

Prudent planning is the cornerstone of a successful project. Poor or inadequate planning will delay a project's completion and increase its cost. Before construction begins, planning must be completed. Planning begins with a site inspection.

Although photographs can be helpful, there really is no substitute for a site inspection. The goal of a site inspection is to accumulate information of all kinds that will enable the project to be properly organized. Information that should be collected includes determinations of the availability of electrical service, telco service, water and sewer (if necessary) and site access. Photographs of the site should be taken for future reference.

After the site inspection is completed, it will possible to ascertain what utility or outside services (subcontractors) will be required. It is imperative to remember that the utility companies or the subcontractors often need a long lead time to get their part of the project finished and generally do not understand the unique specifications that the construction of broadcast facilities presents. Therefore, it is a good idea to meet these people at the site, explain the special needs for the project, if any, and answer any questions they might pose. A typical example is that many electric companies are particularly loath to provide three phase power with a closed delta configuration that the manufacturers of most high powered transmitters require or recommend. A site meeting and explanation will ensure the desired service is provided. Inquire if various types of building permits (construction, electrical, etc.) are needed and whose responsibility it is to obtain them. When explaining the project, be specific as to your specifications and project timetable. Obtain phone and pager numbers for key personnel. File all contact information and project correspondence in the same place. Provide copies of all correspondence, as well as frequent project updates, to the station owner or general manager.

If the site is going to be constructed on a virgin piece of property, investigate whether advance municipal, county, state or federal approvals, such as zoning permits, use variances or wetlands permits must be procured. Although it is not the scope of this paper to explore zoning issues, zoning requests can take months, sometimes years to process, with attendant high expense and no guarantee of a positive outcome.

It is especially important after a site meeting with any subcontractor to obtain a proposal from the subcontractor providing specific details for the price, payment schedule, specifications and timetable for the work proposed. If the proposal is acceptable, acknowledge it in writing and reiterate key aspects. Treat equipment vendors in a similar manner and request a firm delivery date.

Next, continue the organization process by putting together a project schedule. When formulating the schedule, do not forget often overlooked but important details such as allowing time for mandatory municipal inspections. Often, municipal inspections are performed only once a week and the inspector may not be immediately available after the job is done. Reconfirm the schedule and stay in regular contact with everyone involved. If one vendor, subcontractor or utility fails to meet their project commitment, it can cause significant delay or temporarily stop the project. One of the prime duties of good project management is to attempt to foresee problems so that delays can be prevented or limited to the maximum extent possible. When dealing with large companies, such as utilities, do not rely or assume that your contact from one division of the company has contacted another division regarding a critical component of the project. Independently verify that everything is progressing the way it was planned.

When the on site work begins, the project manager should make daily visits to the site to supervise, inspect and answer any questions which may arise. This procedure will expedite the project's completion and prevent problems from developing.

Despite the most meticulous planning, a few problems will probably develop anyway. Weather related delays are common and should be provided for in a project's schedule. The delay factor should be based on a project's geographic location and season that it is to be constructed. The only thing that is predictable about the weather is that it is unpredictable.

Other problems that may develop include personality conflicts between the various personnel at the job site. A professional demeanor and a judicious application of tact and diplomacy will go a long way toward a prompt resolution of any problem. Sometimes problems appear during construction that could not have been foreseen previously. When an unforeseen problem is present, think it over, then take a logical course of action to correct or alleviate the problem.

As each stage of the project is completed, it should be carefully inspected to ensure that the correct materials were used and that the work performed was in accordance with its proposal. If something is not satisfactory, it is the project manager's responsibility to have it corrected. Do not sign any paperwork regarding job completion or authorization for payment until the corrections have been made. Also, for jobs that require municipal inspections (electrical, concrete work, etc.) do not sign anything until the job is successfully inspected by the appropriate official.

ACKNOWLEDGEMENTS

The author wishes to thank Don Neumuller, Chief Engineer WJDM, Bud Williamson of Digital Radio Engineering, Inc. and Kurt Gorman of Phasetek Inc. for their capable assistance and long hours of hard work on this project.

FM MEASUREMENT LIMITATIONS AND TECHNIQUES

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Abstract

Recent advances in FM exciter technology have pushed the performance envelope to new limits. In the past, standard off-the-shelf test equipment could be used to adequately test exciter FM signal quality. Current FM exciter performance has now exceeded the capability of industry standard measurement practice. For an FM station to achieve the best RF signal possible, the broadcaster must be knowledgeable of FM test equipment limitations. Assuming that the FM demodulator and/or stereo decoder are perfect can lead to a degraded FM signal. This paper will show why the FM demodulator and stereo decoder specifications are critical in optimizing system performance. In addition, measurements that are susceptible to inaccurate readings are identified along with techniques on how to reduce measurement errors.

Advances in FM Exciter Technology

The Direct Digital Synthesizer (DDS) or Numerically Controlled Oscillator (NCO) provides a mechanism for FM to be generated digitally [1]. The Harris DIGIT™ is the first commercially available digital exciter to take advantage of this technology. Early on in the development of the DIGIT™ digital exciter, it was noticed that the measured performance levels were not equal to that of its counterpart the analog exciter. Great lengths were taken to ensure transparent signal conversion from the composite input to the RF output. It seemed that a "brick wall" had been hit which defied any attempts made in performance improvements. In the process of isolating the problem, the FM demodulator became suspect. What was discovered was that the FM demodulator was in fact limiting the measurement levels. In addition, it was found that analog exciters were being aligned to compensate for non-linearities in the FM demodulator resulting in artificially low measurements. The digital modulation process of the DIGIT™ offers no means for non-linear compensate and consequent measurements were being limited by the non-linear demodulator. This discovery led to considerably different measurement approaches and techniques than previously used. Many of the results of these discoveries and test techniques have been incorporated into this paper.

History of Distortion Measurements

Distortion measurements have evolved in the Broadcast industry. The first THD measurements were actually THD plus Noise or THD+N measurements. When FM exciter performance exceeded 65dB linearity levels, THD+N became essentially a S/N measurement because the harmonic energy was less than the noise energy. To overcome this limitation, 75µs de-emphasis was used to reduce the noise energy by reducing the noise bandwidth to about 2.1kHz. Assuming white noise, this provided a theoretical 33dB S/N improvement over a measurement with a 100kHz noise bandwidth. With 75µs de-emphasis, linearity could be measured beyond the 90dB levels. Also, there was a perceived performance benefit in addition to noise reduction caused by the 6dB per octave attenuation of the distortion products. When the Audio Precision Model System One became the industry accepted test equipment, the perceived performance levels improved even more, especially at high frequencies. The reason is that the Audio Precision applies de-emphasis only to the distortion and noise, but measures the fundamental separately before filtering. This technique provides a 27dB of improvement for a 50kHz fundamental frequency. De-emphasis and the Audio Precision measurement technique do not represent a real performance improvement. They offer only a perceived performance improvement at high frequencies where distortion is typically worst. Yet they have become the defacto standard for THD measurements.

A far more informative THD measurement is spectral analysis of the recovered composite baseband using either a spectrum analyzer or an FFT processor. Using spectral analysis, the measurement becomes exclusively a distortion measurement. Noise has essentially no affect on the reading for small resolution bandwidths, or fine frequency resolution in the case of FFT analysis. With no de-emphasis to attenuate the distortion harmonics, a "real" picture of distortion is available. High frequency distortion is no longer masked and better insight into the type of distortion is inherent. Spectral analysis for THD measurement is the approach recommended by Harris Broadcast.

Practical Distortion Limitations

FM systems, in general have two principal mechanisms for signal distortion. Distortion can be caused by non-linear group delay in the FM channel, and by a non-linear modulation process. Only an analog FM exciter suffers the latter problem since digital modulators are intrinsically perfectly linear. In the FM modulation process, the important signal characteristic is the zero crossings of the FM carrier. In an ideal system, amplitude variations have no importance for linearity. The only important characteristic is frequency or phase. Admittedly, there are problems associated with amplitude variations (AM modulation) of an FM carrier due to a phenomenon known as AM to PM conversion. But in general AM plays a rather minor roll in distortion measurements due to hard limiting of the FM signal. Most distortion in an FM system outside of the analog modulator is caused by non-linear group delay. There is however an additional indirect source of distortion which is universally overlooked. The FM demodulator can be the origin of FM distortion without having any direct control of the broadcast signal.

Traditional analog FM exciter technology utilizes a modulated oscillator approach to FM generation. The FM signal is obtained by forcing an oscillator to change frequency proportional to the modulation. This is usually accomplished by applying the modulation signal to a varactor diode which is part of the oscillator's tank circuit. All varactor-tuned oscillators have an inherently nonlinear modulating characteristic because the varactor diodes do not exhibit the desired square law voltage versus capacitance relationship [2]. This is especially true for a wide-band FM exciter. Pre-distortion of the modulation signal is typically used to linearize analog exciters. The problem is that there are no perfectly linear demodulators that can be used to align the pre-correction. Inevitably, the pre-distortion will be adjusted to compensate not only for VCO non-linearity but also for demodulator non-linearity as well. This condition is shown in Figure 1.

Figure 1 shows that any non-linearities in the demodulator may in fact improve the "perceived" exciter performance. However, if the spectrum of the exciter were inverted in frequency before being applied to the demodulator, then the distortion measurement would actually be worse than the true performance of the exciter (see Figure 2).

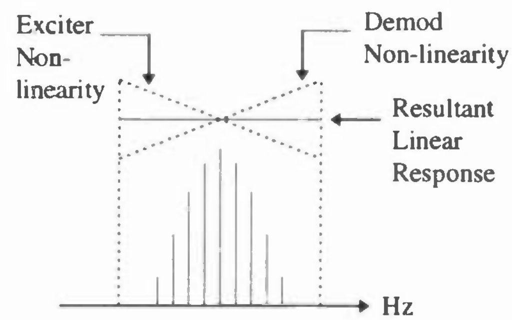


Fig. 1 Non-linear FM Demodulator and Compensation by Non-linear Exciter

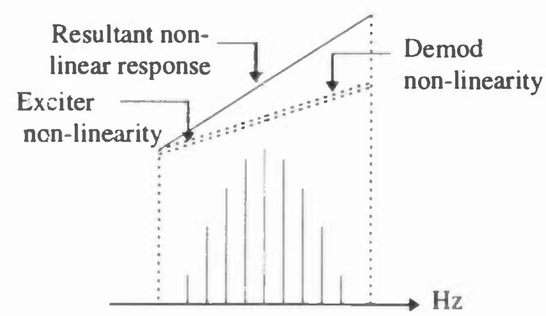


Fig. 2 Non-linear FM Exciter in Non-compensating Condition with Demodulator

Inversion of the FM spectrum is a rather simple test to accomplish, therefore it provides an easy determination of over-compensation in FM exciters. Most demodulators provide an easy means for inverting the FM spectrum. Demodulators that have no means of spectral inversion can still be used if they have an IF input. In these cases, an external mixer can be used to down-convert the FM signal to the IF of the demodulator. The IF spectrum is inverted by selecting an L.O. frequency which provides either high side or low side mixing.

With a numerically generated FM signal from a digital exciter, a nearly perfect FM waveform can be applied to the Demodulator as shown in Figure 3. If the FM source is perfectly linear, then no change in the THD reading will occur when the FM spectrum is inverted, regardless of how non-linear the demodulator is. By the same argument, it can be shown that a perfect demodulator will provide identical THD measurements given an inverted spectrum, no matter how distorted the exciter is.

Finally, if the harmonic content changes when the FM spectrum is inverted, then both the exciter and demodulator have non-linearities which are non-symmetric.

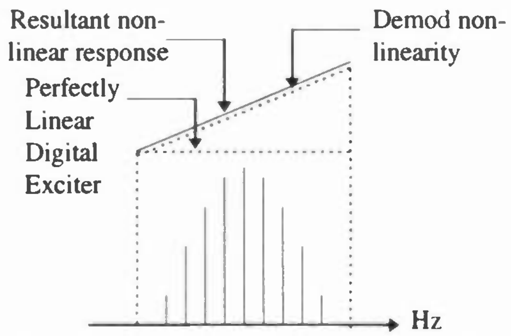


Fig. 3 Perfect Digital FM Exciter: Linearity Measurement Independent of Spectral Inversion

Non-symmetric non-linearity as shown in Figures 1-3 is inherent in all varactor tuned analog exciters and also in FM demodulators. This type of non-linearity is so-called because of the non-symmetric group delay response relative to the FM center frequency. There are also symmetric non-linearities which complicate the characterization of demodulator non-linearity. Symmetric non-linearity, as the name implies, is caused by amplitude and group delay variations which are symmetric about the FM center frequency (see Figure 4). Symmetric non-linearity is predominant when Band Pass Filters are used for example.

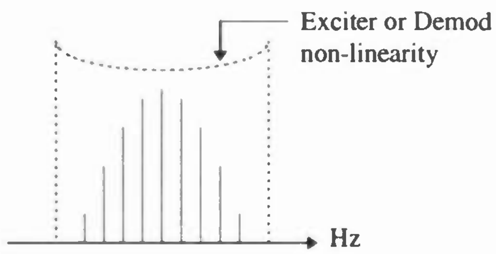


Fig. 4 Example of Symmetric Non-linearity

Inverting the FM spectrum is not useful in detecting symmetric non-linearity. To detect the presence of symmetric non-linearity, a different test method must be used. Symmetric non-linearity is usually more pronounced for frequencies far away from the carrier center frequency. As the deviation and/or modulation frequency increases, more of the Bessel sidebands will occupy frequencies further from the carrier center frequency. As more of the spectral energy moves to the less linear regions of the FM channel, the distortion will increase. Plotting THD vs. Modulation frequency is an easy test to detect symmetric non-linearity. Since the use of de-emphasis in THD measurements masks high frequency distortion, the spectral analysis or FFT type of measurement is most useful.

Comparison of Exciters and Demodulators

A rule of thumb for test equipment is that it should be 10dB better than the measurement it is trying to make. A THD measurement of 0.003% would require the demodulator to have greater than 0.001% (100dB) linearity to ensure a reliable reading. A demodulator with 0.003% linearity could provide reliable distortion measurements no better than 0.01%. This places a significant burden on demodulators that are used to measure THD at these levels. The fact is that there are no demodulators which publish specifications at these levels. The inevitable outcome is over-compensation of the analog FM modulator due to demodulator non-linearity.

To validate the preceding argument of exciter over-compensation, the following test was performed. Three different exciters were tested back to back. A digital exciter (The Harris DIGIT™) and two analog exciters were used. One of the analog exciters tested used pre-distortion. The other analog exciter used no pre-distortion. Three FM demodulators were used in the test, namely the Rohde & Schwarz FMA, the HP8901A, and the Belar FMM-2. The Rohde & Schwarz FMA and the HP8901A offered selectable spectral inversion. The FMM-2 required an external L.O. to invert the 650kHz IF spectrum. An Audio Precision System One was used as the modulation source and an HP3589A spectrum analyzer was used to measure the spectral distortion. The test set-up was as shown in Figure 5.

Each exciter was tested with every demodulator, providing 9 sets of measurement data. The goal of this test was to determine the levels of both symmetric and non-symmetric distortion in the FM demodulators and exciters and to verify that a perfect digital modulator is unaffected by spectral inversion.

The first test performed was a distortion test applying the pre-distortion theory presented above. The recovered 2nd harmonic energy was measured relative to the fundamental modulating frequency. The energy in the 2nd harmonic was converted from dB (relative to the fundamental frequency) to absolute FM deviation in Hertz. The FM spectrum was then inverted and the measurement repeated. The two 2nd harmonic levels representing the normal and inverted spectrums were then compared and the difference between the two was calculated. The difference was calculated in absolute FM deviation rather than dB so that the direction of variation (positive or negative) could be seen. This became important to show that each demodulator had a unique distortion characteristic.

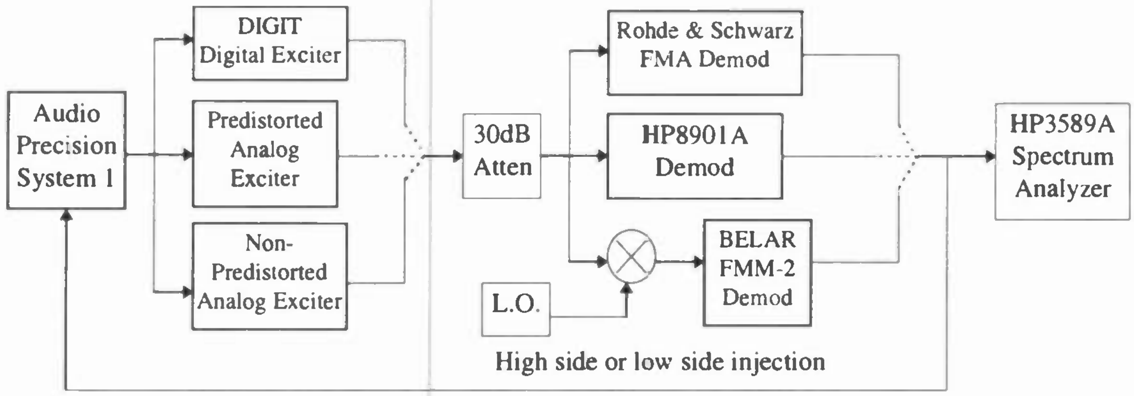


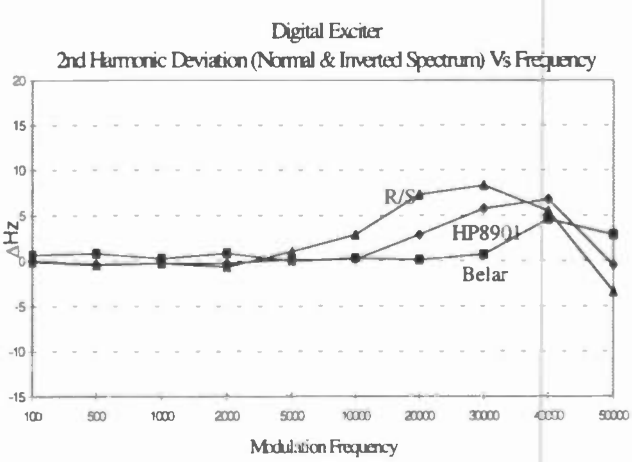
Fig. 5 Test Setup for Exciter and Demodulator Comparisons.

The 2nd harmonic distortion difference was calculated for various modulation frequencies between 100Hz and 50kHz so that effects of both symmetric and non-symmetric non-linearities could be detected. Figures 6a, 7a, and 8a show the 2nd harmonic deviation variations caused by spectral inversion for each exciter. Each of the plots have three different curves corresponding to the three different FM demodulators.

Although the 2nd harmonic difference test is useful for quantifying pre-distortion, it does not indicate absolute distortion. A good distortion measurement is one which demonstrates a low absolute distortion reading as well as a low distortion difference when the spectrum is inverted. To obtain an absolute distortion number the inverted and non-inverted distortion readings were averaged. This average is shown in Figures 6b, 7b, and 8b. Again, there are 3 plots representing each exciter, each of which contain 3 curves relating to each demodulator.

Given the fact that a perfect demodulator would show no change in measured distortion regardless of spectral inversion, it can be seen that none of the demodulators exhibit this ideal characteristic. All demodulators appear to have measurable non-linearities. The opposite assumption being that a perfect exciter would demonstrate no change in measured linearity after spectral inversion suggests that the digital exciter is the most linear of the exciters. But even the digital exciter seems to exhibit some non-linearity at higher modulation frequencies above 10kHz (Fig. 6a). Although the digital modulation process is perfectly linear, there may be some degradation due to the upconversion process required in a digital exciter [1].

A second observation that can be made is that the digital exciter (Fig. 6b) demonstrates a lower average THD than both of the other exciters. The exception to this is when the digital exciter is used with the Rohde & Schwarz FMA demodulator above 10kHz. This demodulator however, is equally poor for all exciters which suggests that the demodulator is the source of the non-linearity.



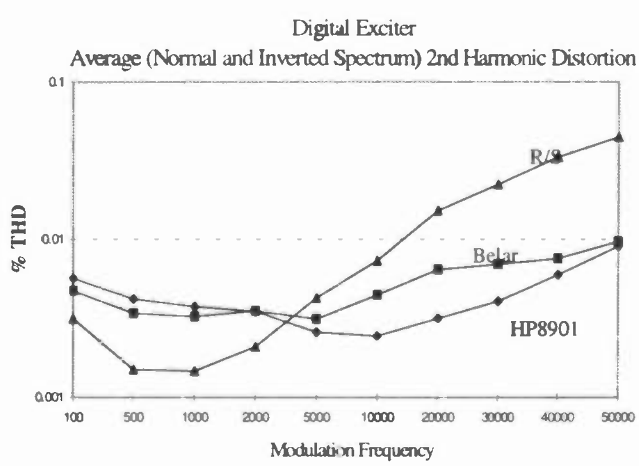


Fig. 6 Digital Exciter a) Distortion Difference Between Normal and Inverted Spectrum Versus Modulation Frequency
b) Distortion Average of Normal and Inverted Spectrum Versus Modulation Frequency.

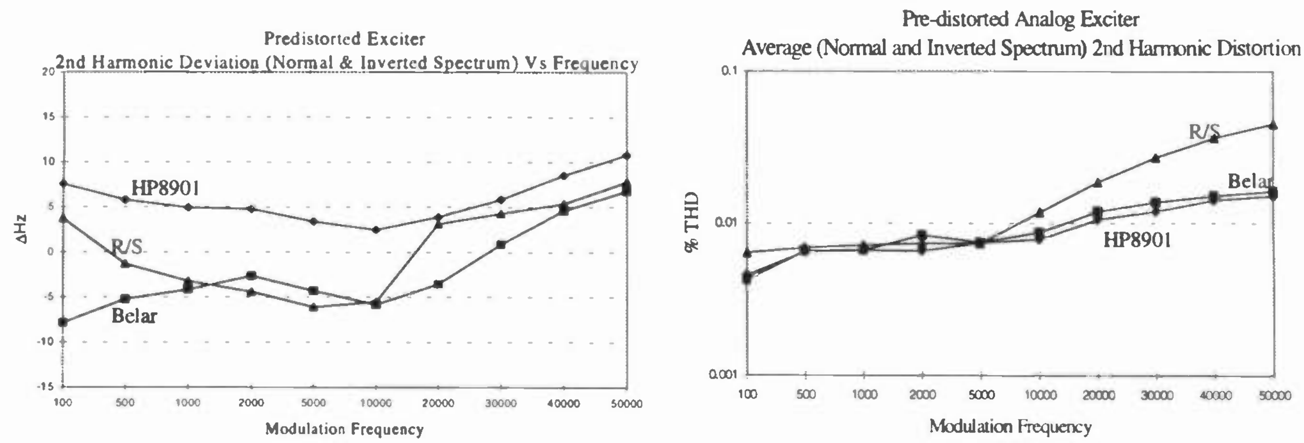


Fig. 7 Pre-distorted Analog Exciter a) Distortion Difference Between Normal & Inverted Spectrum Vs. Modulation Freq.
b) Distortion Average of Normal & Inverted Spectrum Versus Modulation Frequency

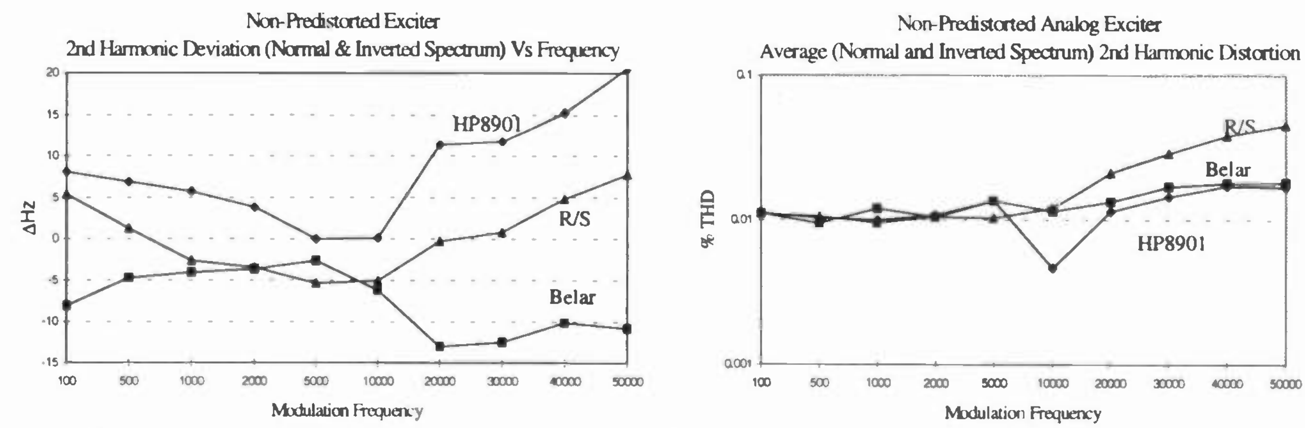


Fig. 8 Non-Predistorted Analog Exciter a) Distortion Difference Between Normal & Inverted Spectrum Vs. Modululation Freq
b) Distortion Average of Normal and Inverted Spectrum Versus Modulation Frequency

From these plots it can be concluded that the digital exciter exhibits the least amount of both symmetric and non-symmetric non-linearity. It can also be concluded that no demodulator exhibits a high degree of linearity (relatively speaking) and none could be used to provide accurate THD readings better than 0.006% (assumes 75µs de-emphasis). It can also be seen that the pre-distorted analog exciter offers essentially no non-symmetric distortion advantage over the non-predistorted analog exciter except at high frequencies where the average distortion levels are high.

It must be pointed out, however, that the distortion measurements of Figures 6, 7 and 8 were made without 75µs de-emphasis. When 75µs de-emphasis is added to the measurement of Figure 6b, the curves of Figure 9 are obtained (assumes that de-emphasis is only applied to the 2nd harmonic and not the fundamental per the Audio Precision System One measurement technique). Figure 9 clearly shows that distortion problems at high frequencies are attenuated significantly when 75µs de-emphasis is used. In fact, after de-emphasis, the low frequency distortion becomes the worst case distortion in the measurement. In Figures 6-8, the Rohde & Schwarz FMA demodulator may not have been considered because of its high frequency distortion. However, Figure 9 suggests that the Rohde & Schwarz FMA demodulator is the best performer (2nd harmonic distortion) when 75µs de-emphasis is used.

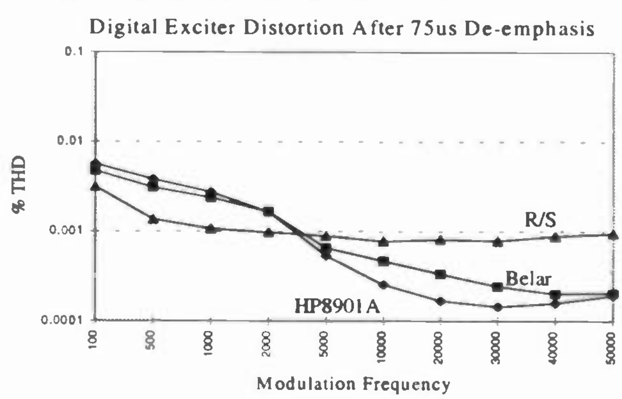


Fig. 9 Digital Exciter Distortion for All Demodulators After 75µs De-emphasis is Applied.

In summary, the above test indicates a need for careful consideration of the FM demodulator when any distortion measurement is being made. Each demodulator tested exhibited a significant and unique non-linear characteristic. Because of this, the broadcast signal quality will be indirectly impacted by the demodulator if an analog exciter is used. The perfectly linear modulation process in a digital exciter exceeds the current FM demodulator measurement capability and has superior distortion characteristics over it's analog counterpart. Finally, spectral analysis is by far the most sensitive and informative technique available for measuring distortion, and the distortion is not masked by de-emphasis.

FM Signal to Noise Ratio

There are two important elements in any signal to noise measurement which can have dramatic effects on the reading. The first is measurement bandwidth and the other is noise sampling and measurement settling. As discussed above, de-emphasis essentially limits the bandwidth to 2,122Hz so the important noise is below this frequency. Because a modulated-oscillator requires very narrow phase-lock-loop (PLL) bandwidths it has no ability to track out the phase noise of the VCO. VCO noise is generated because an oscillator "bootstraps" the noise of the amplifying devices internal to the oscillator. Additional noise contribution can occur from noise in the resonator itself. The frequency-response shape of the noise (or power spectral density) was originally described by Leeson [3], neglecting the additional effect of active device 1/f noise contamination. The bootstrapping occurs due to positive-feedback, which grows exceedingly larger as the observed frequency gets closer to the carrier (more centered in the resonant circuit).

The expression for the power spectral density (relative to frequency-displacement from the oscillator carrier) including the extended accounting for 1/f noise proposed by Scherer is described mathematically as [4]:

$$L(f)=(FkT/2/P)[1+(f_o/2/Q/f_{sb})^2][1+f_c/f_{sb}] \qquad (1)$$

$$L(ssb)_{dB}=NF_{dB}+20\log[(f_o/f_{sb})/2/Q]+10\log[1+(f_c/f_{sb})] \qquad (2)$$

Where:

- $L(f)$ is the noise power per Hertz versus frequency
- F is the noise figure of the active device.
- T is the absolute temperature in degrees Kelvin.
- k is Boltzman's constant
- P is the output power.
- f_o is the center frequency of the oscillator
- Q is the loaded Q of the tank circuit (resonator).
- f_{sb} is the sideband frequency offset from carrier.
- f_c is the 1/f corner frequency.
- NF_{dB} is the noise floor in decibels per Hertz below carrier (dBc/Hz).
- $L(ssb)_{dB}$ is the single sideband phase noise in dBc at offset frequency f_{sb}.

Without the last expression in [ ] brackets (Equation 1), Scherer's equation is the same as Leeson's.

The noise of a typical oscillator increases 6 dB per octave the closer it is to the carrier until it reaches the 1/f noise corner at f_c, where it climbs an additional rate of 3 dB/octave. Thus, very close to carrier, the noise power increases 9 dB per octave. Figure 10 is a sketch of the single-sided power spectral density described in the equations above.

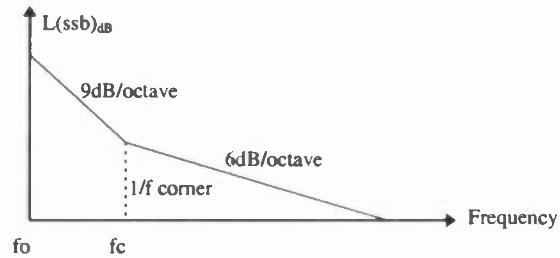


Fig. 10 VCO Phase Noise Based on Leeson Model

An oscillator used as the primary source of frequency-modulation in a broadcast FM exciter has two short-comings:

(1) 1/f noise and ordinary $1/f^2$ noise in the VCO being frequency-modulated harms the signal to noise ratio, and (2) The need to maintain the average center frequency of the VCO via a phase-locked-loop (PLL) demands that the loop ignore the desired modulation. Therefore, the PLL loop bandwidth must be very low.

As the PLL bandwidth is lowered, more and more of the 1/f and $1/f^2$ phase noise of the VCO remains un-attenuated by the loop and the FM noise is degraded. Below the loop corner frequency where the frequency-control-servo begins to operate, the digital logic, phase detector, and frequency-divider noise will all be amplified. The amount of amplification is easy to calculate. To obtain 75kHz peak FM deviation at frequencies down to 30 Hz requires a phase deviation of up to 2500 radians. For a phase detector limited to 1 radian of linear phase excursion, a digital frequency (hence phase) division of 2500 is required. This is equivalent to a noise gain of 68dB. Thus, if the phase detector noise floor were -145 dBc/Hz, it would appear to generate -77 dBc/Hz phase noise at the multiplied-by-2500 RF output for sub-sonic frequencies.

A solution to the problem of high sub-sonic noise in ordinary FM exciters can be found in Direct Digital Synthesis (DDS). The DDS FM generator requires no PLL loop, hence no frequency-division back to a reference. Thus, frequency-multiplication of the phase-comparator noise (due to jitter) is not a problem.

Since the untracked VCO 1/f and $1/f^2$ phase noise and the multiplied divider and jitter noise dominates the analog exciter S/N, it cannot achieve the performance of a digital exciter unless the noise is masked somehow. One way the noise is masked is by specifying a 22Hz lower bound on the S/N bandwidth. A 22Hz high-pass filter attenuates the majority of the phase noise of the Modulated Oscillator.

Another noise masking technique is accomplished by means of a settling filter in the measurement equipment. In an Audio Precision System One the settling filter is important for swept measurements where the signal changes rapidly and erroneous results may be obtained unless the signal settles before the reading is made. But in a static measurement environment where there is no changing stimulus the settling filter can be turned off. In fact the Audio Precision System One recommends no settling when measuring phase noise. The settling filter essentially compares the previous data samples with the current data sample. If the current sample is outside of a specified threshold relative to the previous samples, then the current sample is ignored. When measuring the dominant noise and jitter in an analog FM exciter, the settling filter must be turned off otherwise the majority of samples will be ignored. This is because the noise of an analog exciter easily exceeds the common threshold for settling in most measurements.

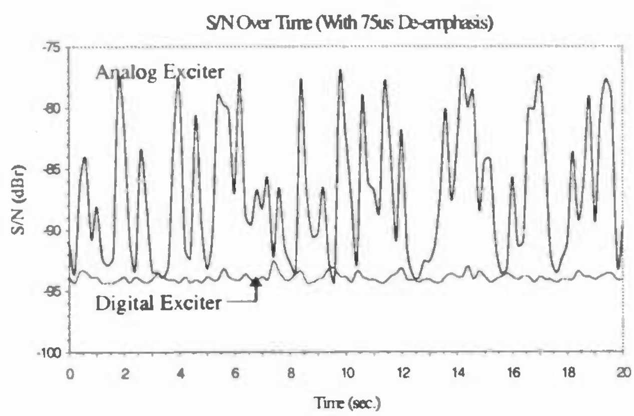


Fig.11 S/N Comparison Between Digital Exciter and Analog Exciter (No Settling or 22Hz HPF)

A test which demonstrates the large S/N difference between a digital FM exciter and an analog FM exciter is one which does not use a high pass filter and one which uses no settling filter. The S/N measurement is taken over a fixed time interval so that bursty phase noise and jitter can be detected. This test is affectionately called the Signal to Noise over Time test or SNOT test. Figure 11 plots the S/N of the DIGIT™ digital exciter and a premium analog FM exciter. It can be seen that the digital exciter does not exhibit the wild S/N variations which are inherent in the analog exciter. This test shows that the analog exciter S/N is over 15dB worse than a digital exciter when no high-pass or settling filters are used.

Stereo Separation

In the past when 40dB stereo separation performance was considered acceptable, the best way to predict stereo separation performance was by measuring the amplitude and phase response of the FM exciter. With the advent of the digital exciter, stereo separation performance is theoretically limited only by the resolution of the digital modulation. For a 16 bit system, this means stereo separation levels greater than 90dB. In practice, however, there is some loss due to upconversion, but the major limitation has become the FM demodulator.

It has also become impractical to predict exciter stereo separation by means of amplitude and phase measurements because of test equipment limitations. Both the audio test equipment and the FM demodulator have amplitude and phase variations which exceed the levels necessary to predict stereo separation performance above 60dB. The accepted method for calculating stereo separation requires the ability to measure the amplitude and phase variation between the main channel (dc - 15kHz) and sub-channel (23kHz - 53kHz) [6]. This equation is given below:

$$Sep = 20\cdot\log\left[\sqrt{\frac{\left(\cos(\theta\Delta)+\frac{Asub}{Am}\right)^2+\sin(\theta\Delta)^2}{\left(\cos(\theta\Delta)-\frac{Asub}{Am}\right)^2+\sin(\theta\Delta)^2}}\right] \quad (3)$$

Where

Asub is the combined amplitude of the two sub-channel sidebands.

Am is the main channel amplitude.

$\theta\Delta$ is the phase difference between the sub-channel and the main channel.

This equation, however, assumes a flat response across the sub-channel from 23kHz to 53kHz. Typically, the upper sideband of the sub-channel suffers more response problems than the lower sideband because it is closer to the composite cutoff frequency. An equation is given below which can be substituted into Equation 3 above to account for gain and phase variations between the sub-channel lower and upper sidebands. Equation 4 is the amplitude offset and Equation 5 is the phase offset for the sub-channel.

$$\mathrm{Asub} := \frac{\sqrt{(\mathrm{Asl}\cdot\cos(\theta\mathrm{sl}) + \mathrm{Asu}\cdot\cos(\theta\mathrm{su}))^2 + (\mathrm{Asl}\cdot\sin(\theta\mathrm{sl}) + \mathrm{Asu}\cdot\sin(\theta\mathrm{su}))^2}}{2} \quad (4)$$

$$\theta\mathrm{s} := \mathrm{atan}\left(\frac{\mathrm{Asl}\cdot\sin(\theta\mathrm{sl}) + \mathrm{Asu}\cdot\sin(\theta\mathrm{su})}{\mathrm{Asl}\cdot\cos(\theta\mathrm{sl}) + \mathrm{Asu}\cdot\cos(\theta\mathrm{su})}\right) \quad (5)$$

Where:

- θsl and θsu are the lower and upper (respectively) sideband phase relative to the main channel (degrees).
- Asl and Asu are the lower and upper (respectively) sideband amplitude values (volts).

Asub and θs can then be substituted into the equation 3 above. Δθ is identical to θs if the main channel phase is referenced to 0 degrees.

These equations indicate that for a non-symmetric sub-channel response variation, the worst case amplitude variation to maintain 70dB of channel separation is 0.01dB. This is based on the assumption that the group delay response is perfectly flat and that only one of the sub-band channel sidebands has imperfect amplitude. Assuming a perfect amplitude response, the worst case non-linear phase variation tolerable is 0.07 degrees. Again this assumes that only one of the sub-channel sidebands is affected. This means that if a 70dB stereo separation is to be predicted based on amplitude and phase, the measurement equipment can have no worse than 0.01dB of amplitude variation and 0.07 degrees of phase non-linearity across the dc-53kHz stereo frequency band. This presents a problem, because demodulators and audio test gear are not specified to these levels.

Figure 12 is a plot of the amplitude response for the three demodulators used in the distortion tests above. The problem with making amplitude comparisons is that like distortion, there is no standard demodulator which can be used as a reference to calibrate the system. In this case, the Rohde & Schwarz FMA demodulator was chosen as the reference and the system was equalized based on it. Figure 12 shows that there is 0.3dB of amplitude variation at 100kHz between the three demodulators. Over the stereo channel from dc to 53kHz, there is 0.06dB of amplitude variation. This level of amplitude variation allows for at best 55dB of stereo separation. Since demodulator performance falls short of the levels necessary to measure 70dB stereo separation, the issue becomes which demodulator does one trust.

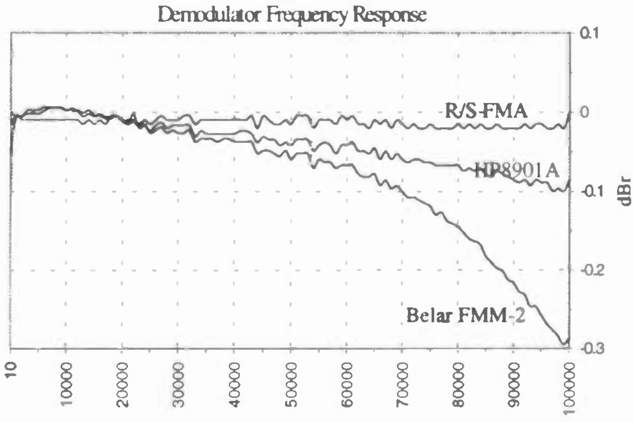


Fig. 12 Amplitude Response for 3 Test FM Demodulators

One obvious solution to this dilemma is through the use of a perfect FM modulator. A perfect FM modulator offers a transparent conversion between the composite domain and the FM domain allowing system variations to be calibrated out except for the FM demodulator. This allows the impairments of the FM demodulator to be isolated. Fortunately the numerical modulation technique of a digital exciter offers a perfect FM source. To avoid the possible system variations caused by frequency upconversion and down-conversion, a special digital FM modulator was designed to provide an FM source centered at 650kHz. This frequency was chosen so that it could be applied to the IF input of the Belar FMM-2 demodulator. The test was set up as shown in Figure 13.

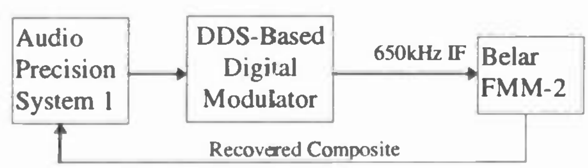


Fig. 13. Perfect Source Test Set-up to Calibrate Demodulator Amplitude Flatness.

The amplitude variations of the input cables and Audio Precision System One were calibrated out so that a flat response was guaranteed through the system except for the demodulator. The frequency response of the Belar FMM-2 demodulator was then measured. This calibrated frequency response of the Belar FMM-2 could then be used as a standard for measuring the response of the other demodulators.

To accomplish this, the DIGIT™ digital exciter was chosen as the reference exciter. Its output was connected to the Belar FMM-2 demodulator and the system frequency response was measured. The resulting measurement was then compared to the calibrated IF frequency response of the Belar FMM-2 from the test shown in Figure 13. A minor adjustment had to be made in the amplitude response of the system because it was found that the Belar FMM-2 down-converter attenuated the amplitude response by about 0.05dB at 100kHz. After calibrating out cables and the Audio Precision System One amplitude variations, the amplitude response was measured to be nearly identical to the calibrated Belar FMM-2 response. Having calibrated the entire modulator and measurement system, the amplitude response of the other demodulators was then measured. Figure 14 shows the calibrated amplitude response of all three demodulators tested.

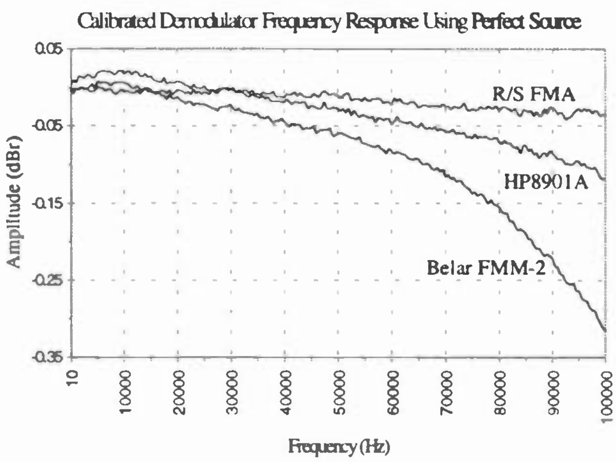


Fig. 14 Calibrated Amplitude Response of FM Demodulators.

Based on the curves of Figure 14, and using Equations 3, 4, and 5, the following chart was generated which predicts the stereo separation for the three demodulators at 400Hz modulation frequency and 15kHz modulation frequency.

Demodulator	Separation @ 400Hz Modulation	Separation @ 15kHz Modulation
Rohde & Schwarz	72.3dB	73.7dB
HP8901A	55.6dB	54.5dB
Belar FMM-2	53.0dB	52.3dB

An actual stereo separation measurement was made using the test setup to verify the calculated results. An HP8904A synthesizer (capable of 80dB stereo separation) was used as the stereo source and a Belar FMSA-1 digital stereo decoder was used in the test. Since the stereo separation calculation assumed perfect phase linearity, the results were not as good as predicted (down about 10dB), but the Rohde & Schwarz FMA demodulator was clearly the best performer. A simple phase corrector was used to linearize the phase response of the system. The stereo separation performance of the three demodulators then closely tracked the predicted levels. All that was left was to compare the stereo separation of the three exciters. Since the Rohde & Schwarz FMA demodulator offered the best stereo performance, it was chosen as the reference demodulator. Figure 15 shows how the three FM exciters compared in stereo separation.

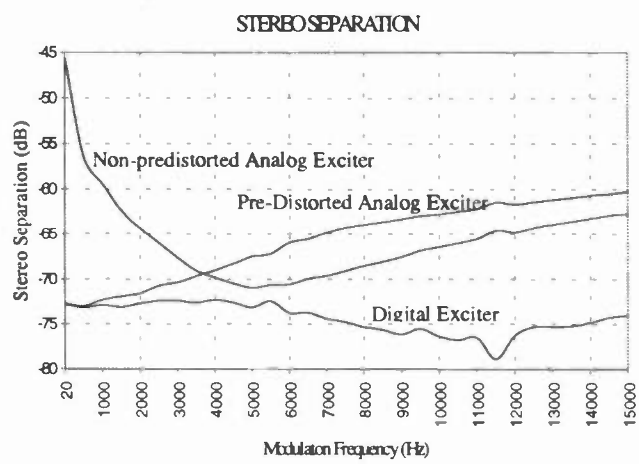


Fig. 15 Stereo Separation Versus Frequency using the Rohde & Schwarz FMA Demodulator & Belar FMSA-1

The results of Figures 14 and 15 required hours of testing and meticulous attention to detail. In general, the broadcaster will not be able to easily duplicate these measurements outside of a laboratory environment. It is the case that system variations of less than one hundredth of a dB have significant impact on stereo separation at these levels. This discussion does, however, suggest that circuits in an analog exciter affecting stereo separation were designed to compensate for an imperfect demodulator. In addition, stereo separation cannot be predicted to levels beyond about 50dB using amplitude and phase response measurements, due to demodulator limitations. Finally, the digital modulator offers a way to accurately measure the FM demodulator response and predict demodulator impact on system performance.

An additional consideration that must be taken into account is the quality of stereo encoding and decoding. The FM exciter and demodulator may degrade stereo separation but they cannot affect the matrix composition of the signal. Any imperfections in the composite domain will impact both Left and Right channel separation equally. If the Left channel into Right channel separation is different than the Right channel into Left channel separation, then one can be assured that the stereo encoder or decoder is the source of the problem. In this case, stereo separation measurements are not reliable beyond the worst case stereo separation reading.

Other Stereo Measurements

Because the noise bandwidth of a stereo signal is three times greater than a monaural signal, the signal to noise ratio for stereo signals is about 10dB worse (assumes 10% pilot injection). Note that this is only additive noise in a contained test environment, and not noise introduced at the receiver antenna which is about 23dB worse than monaural. Given a wideband de-emphasized signal to noise performance of 92dB, the 75µs de-emphasized stereo performance will be about 82dB. A 16dB penalty is suffered if 75µs de-emphasis in not used (assuming white noise). But most of the noise of an analog exciter is at low frequencies, so the reduction in performance is not as severe. Typical non-de-emphasized stereo S/N levels are around 75dB.

This 75dB stereo S/N sets the best case measurement floor for any stereo distortion or separation measurements. For this reason, the best stereo THD specification (really it is a THD+N specification) will be no better than about 0.05%. To measure stereo THD without noise, a spectral-type measurement must be made as discussed earlier. With DSP based digital stereo generators, stereo THD can approach wideband THD performance levels exceeding 90dB. If an FFT or spectrum analyzer is not used for stereo THD, the measurement becomes essentially a stereo S/N measurement.

Disregarding the sub-sonic problems of an analog exciter, most of the exciter distortion is found at high frequencies. Because of this, the worst case stereo THD scenario is when a Left channel = inverted Right channel stimulus is used. This places the stereo energy totally in the sub-band (23kHz - 53kHz) where the linearity is most affected. The most optimistic scenario is when Right channel = Left channel. This places all stereo energy in the main channel where the distortion is less. Left channel only or right channel only represents a compromise between the two extremes above.

Summary

This paper has introduced the idea that digital techniques for generating FM such as DDS produce perfectly linear modulation. This advanced process has allowed a major performance improvement in FM exciters. No longer can FM demodulators be regarded as having a negligible affect when measuring system performance. The DIGIT™ digital exciter has necessitated a change in this paradigm. FM measurement techniques have also needed to change to accommodate the new technology.

This paper proposed the spectral analysis method of measurement as the preferred technique for measuring distortion and separation. Another technique called spectral inversion was provided which identifies excess exciter pre-distortion. This test is applicable for all exciters and serves as a method for estimating demodulator linearity.
Comparisons of two types of analog exciters were made relative to a digital exciter, concluding that the digital modulation process is measurably more linear. In addition, it was shown that the noise of an analog exciter can easily be masked by traditional measurement techniques. A new S/N test was proposed which provides a more accurate way of measuring noise in analog exciters.

Finally, a comparison of FM demodulators was made which indicated a rather wide range of linearity and frequency response variations. A calibrated response curve was generated for three industry standard demodulators so that stereo separation performance could be predicted.

Acknowledgments

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LOW PROFILE ANTI-SKYWAVE ANTENNA (LPASA)

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ABSTRACT

The LPASA is an attempt to address two concerns of AM broadcasters: skywave interference which reduces nighttime signal quality and reduction in the visual profile of AM antennas. The LPASA is a near-earth, horizontal antenna that is a fairly poor exciter of high and medium angle skywaves but a fairly good exciter of surfacewaves. A preliminary design for 1660 kHz indicates, when fed with 2kW, a 3x3 array of horizontal wires, 3 meters above average earth will produce 3.1mV/m at 10 km via the surfacewave, 52 µV/m via skywave at this same distance but only 17 µV/m at 1000km. Several suggested configurations are included as land-area reduction techniques.

BACKGROUND

Efforts to suppress nighttime skywave and thereby increase quality coverage areas date back to the 1930's. Past efforts have included: collinear arrays $2\lambda/3$ to λ in height (1), orthogonal systems (2) one of which primarily excites the more heavily attenuated extraordinary skywave mode (3) and a screened array consisting of a single conventional monopole and several short, ancillary monopoles (4). None of these addressed the need for a low profile structure which today has become important due to public resistance to towers and lack of suitable sites.

Near-earth, horizontal antennas have been studied for many years, beginning with J.H. Rogers (5) in 1919 who discovered that near-earth and buried horizontal dipoles shown in Figure 1 radiated useful energy along the earth. J.R. Carson (6) formulated the properties of horizontal antennas near the earth in 1929. During the Cold War significant research in both the United States (7, 8, 9) and the Soviet Union (10) was conducted due to the military advantages of buried and near-earth antennas: their inherent physical survivability, covertness and ability to be used at extremely low frequencies (ELF) where vertical antennas are impractical.

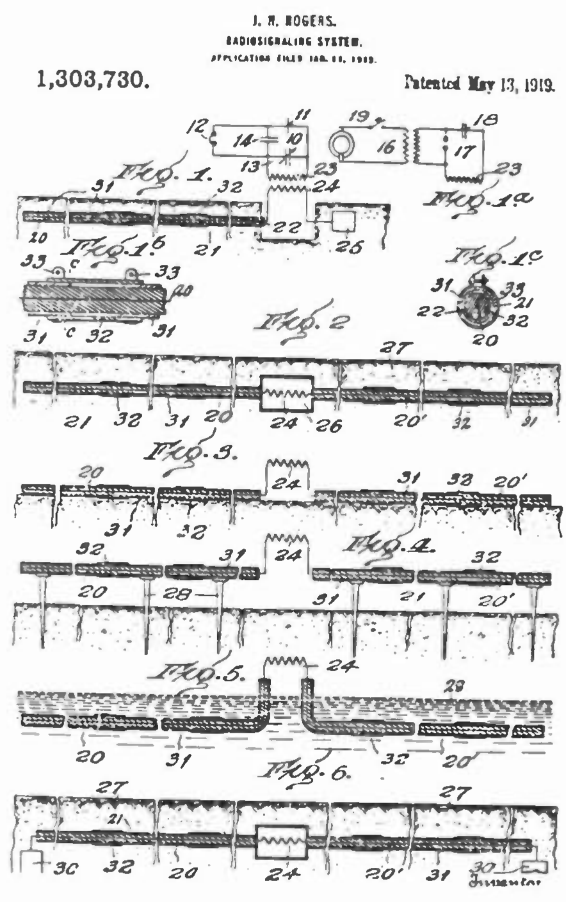

Figure 1. Rogers' 1919 patent

During the 1960's, the U.S. Navy carried out an extensive experimental and theoretical research program aimed at the development of a high power transmitting antenna for use in the very low frequency (VLF) band (11, 12, 13, 14, 15, 16). A series of measurements were made of the performance of horizontal dipoles deployed just above Hawaiian lava beds. During the course of this program it was discovered that a worthwhile increase in efficiency could be obtained by inserting series capacitors along the antennas which modified the wave velocity ratio c/v (free-space wave velocity divided by the wave velocity along the antenna). It was also found that the elevation pattern of the dipole could be varied considerably without loss of efficiency by changing this ratio as shown in Figure 2, taken from Seeley (17). When the ratio is made to be approximately 1.05, the groundwave is maximized and the skywave suppressed. Seeley also studied the possibility of increasing radiation efficiency by paralleling dipoles within an area $\leq \lambda_o/2$ in width and end-loading the dipoles to conserve real estate. Neglecting mutual coupling, the former results in an N-fold increase in radiation efficiency (N=number of dipoles) while retaining the azimuthal pattern of a single dipole. The latter causes a 4-fold increase in efficiency over a short ($l \lambda_o/2$) dipole.

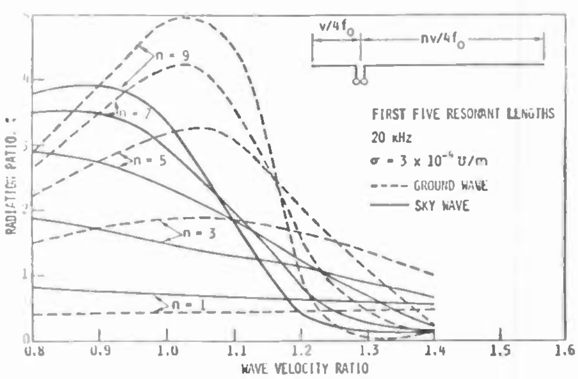

Figure 2. Radiation Ratio VS Wave Velocity Ratio

During the 25 years since Seeley published his final paper dealing with fast-wave horizontal dipoles, a family of moment method computer codes, the latest designated NEC-4.1D (18) have been developed which permit a convenient tool to estimate the electrical performance of wire antennas near a complex boundary. The resulting ground and spacewave fields can in turn be "propagated" to a distant receiver using the WAGSLAB (19) integral equation code for computing MF field strength over inhomogenous, irregular paths covered with forests, buildings or snow. To compute the ionospherically propagated component a ray tracing code which fully accounts for magneto-ionic effects such as Jones-Stephenson (20)can be used.

The purpose of this paper is to, using the above cited research and computer codes, evaluate the expected performance in terms of groundwave coverage and skywave suppression of a preliminary design for a full-size array operating in the expanded AM band and to suggest methods for reducing land requirements.

PRELIMINARY LPASA DESIGN

The physical characteristics of a preliminary design for 1660kHz are shown in Figure 3. The orthogonal conductors and balanced feed lines are dashed for clarity. Short wooden or composite poles could be used as supports and no further insulation would be required between the poles and PVC covered cable. For average earth at 1660 kHz (0.004/10) the NEC-4.1D computed electrical characteristics are:

- VSWR = 1.16 @ 1660 kHz
- ADVERSE PHASE ANGLE $\pm$7.5 kHz = <18°
- POWER PER N-S/E-W SUBARRAY = 1 kW
- TOTAL POWER = 2 kW
- GROUNDWAVE RADIATION PATTERN = ESSENTIALLY OMNIAZIMUTHAL
- GROUNDWAVE RADIATION POLARIZATION = VERTICAL
- SKYWAVE POLARIZATION = CIRCULAR, SENSE CHOSEN TO MINIMIZE 0-MODE EXCITATION
- LOW LEVEL PHASE SHIFT BETWEEN 1 kW TRANSMITTERS = $\pi/2$
- RMS FIELD STRENGTH @1000m, 2m AGL = 0.133V/m

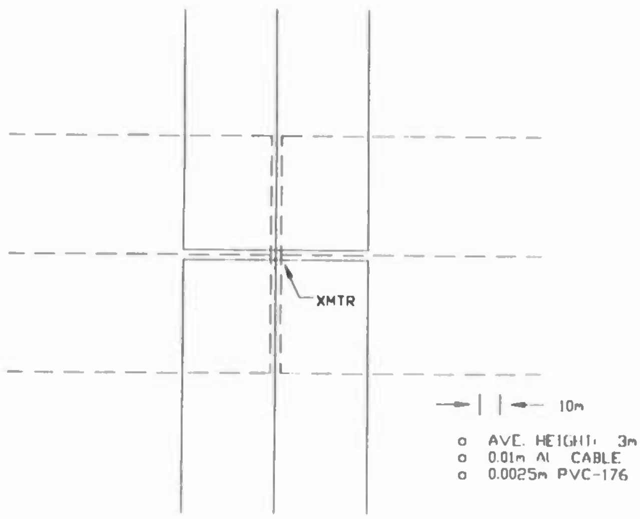

Figure 3. Full Size LPASA Array

Figure 4 indicates the computed elevation pattern for the array shown in Figure 3, where c/v was adjusted to be 1.043. The WAGSLAB code was used to compute the received groundwave field strength at 10 km from the array assuming average earth. For a receive antenna height of 2m, the predicted field strength is 3.1mV/m. At this distance the worst case yearly median value of skywave signal strength was predicted to be 52 µV/m. At 1000 km the worst case yearly median value of skywave decreased to 17 µV/m. Thus, even though the efficiency of the array is increasing at angles typical of 1-hop E and F, propagation (7.5-25°), this is more than compensated for by the increased path loss, resulting in a skywave signal that would protect local groundwave coverage somewhat beyond the rural coverage goal of 0.5mV/m.

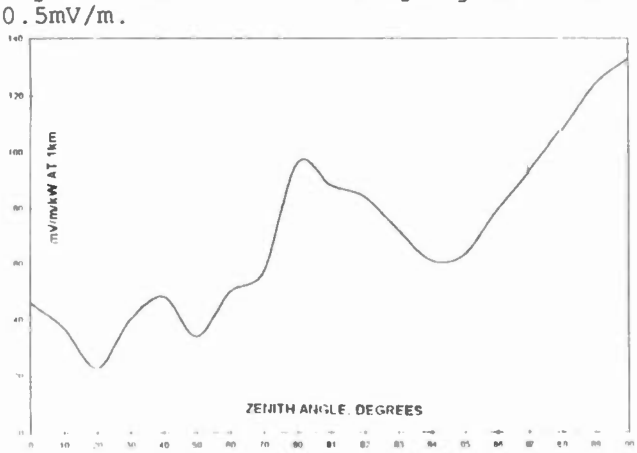

Figure 4. Field Strength vs. Zenith Angle

EXCURSIONS TO THE PRELIMINARY LPASA DESIGN

The preliminary design occupies approximately 9.6 acres of real estate. Two schemes are included to reduce this amount of land: folding the elements and end loading a shortened version of the array shown in Figure 3. The plan views of these configurations are shown in Figure 5. For the folded elements, less than 1.75 acres would be required with a loss in efficiency of about 7dB meaning that the total transmitter power would have to be increased from 2 to 10 kW for average soils. The end loaded 3x3 array is formed by attaching insulated "crow's-foot" conductors slightly buried as shown in Figure 5. In this case 3 acres would be required with a 5dB loss in efficiency, requiring 6 vs. 2 kW of total transmitter power, as compared with the full size LPASA.

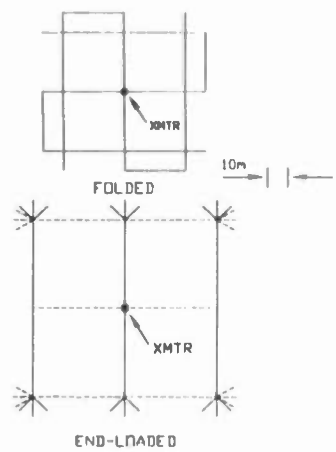

Figure 5. Reduced Size Configuration

CONCLUSIONS

The LPASA described represents an initial attempt to assess the suitability of various configurations of near-earth horizontal dipole arrays for useful coverage in the expanded AM broadcast band. Groundwave enhancement and skywave suppression can be accomplished at MF just as it was shown by Seeley at VLF using low profile, fast-wave structures. Several techniques for reducing the required area have been shown to require an increase in total transmitter power from 2 to 10 and 6kW for folding the antenna elements and end loading respectively. The arrays described are suitable for installation on forested or unimproved lots.

ACKNOWLEDGEMENT

The author thanks Mr. C. T. "Tom" Jones, Jr., President of C. T. Jones Corporation, Springfield, VA for his assistance in the areas of broadcast engineering and regulatory matters concerning this paper.

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DIRECTIONAL ANTENNA BASICS

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ABSTRACT

Quite a bit of material is available on directional arrays. However, almost all of it assumes you are going to be designing them. This paper is for the engineer who will be maintaining an array and contains only enough theory to gain a basic understanding.

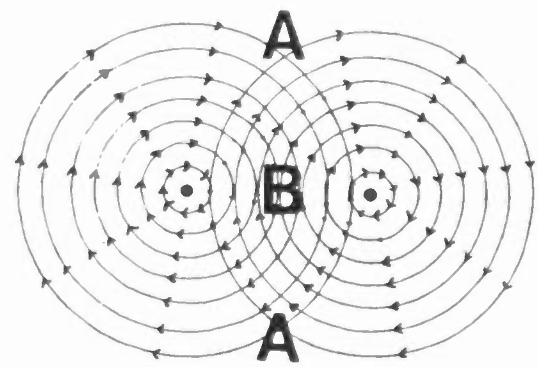
Directional antenna systems are, at the design and set-up level, extremely intricate. However, some of the basic principles are relatively easy to grasp and use. The operator of a station need only know what information is required to be checked on a regular basis and how to interpret that information in terms of good or bad.

With this as our goal, let's examine a directional array.

In order for an AM station to be directional you must have at least two antennas. These are then adjusted in phase and power to develop the desired pattern. To understand this relationship, consider what happens with two streams of water coming out of hoses. If you aim one into the side of the other, the stream changes direction. If you change the angle at which they intersect, or change the flow volume in either one, the stream alters its direction.

The same thing happens with directional arrays, with the angle of attack being the phase and the volume being the ratio of each tower to the reference tower.

If you look at an antenna from the top, the signal comes off the tower in a circular manner. Looking at our two tower example, you will see that, at the A points, the waves are traveling in the same direction, while at point B they are traveling in opposite directions. This will result in combining at point A for more signal in that direction, or, what is referred to as a lobe. At point B we will have two equal, but opposite signals, therefore cancellation and what is referred to as a null.



If we set our towers 90 degrees (1/4) wavelength apart and feed them equal amounts of power in phase, again looking from the top, the pattern produced by our example would look something like figure 1. By simply changing the phasing to 135 degrees, we get a pattern something like figure 2.

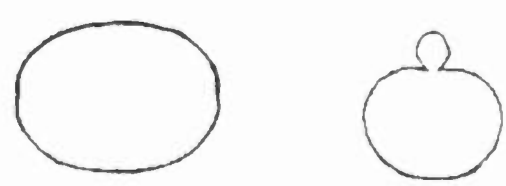


FIG. 1 FIG. 2

You can readily see that, by changing their power relationship, or causing one of the signals to be ahead of, or behind, the other we can alter the shape of the produced pattern.

With this understood, the information derived from the antenna monitor begins to make sense. On an antenna monitor, we derive two basic readings, loop current, or the amount of power in each tower, and phase, or how much does that tower lead, or lag, the reference tower. The reference tower is always the tower with the most antenna current (power) and the other tower's loop current is an expression of it's power as a percentage of the power in the reference tower.

In other words, if tower one is reference and tower two shows a loop current of .815, tower two has 81.5% of the power in tower one. Likewise, phase is expressed as that tower in relation to the reference tower. If you look at your license, you will see what the loop currents and phase angles are supposed to be for your station. You will also find base currents and ratios for each of the towers showing their relationship to the reference tower.

In order to check the health of a directional array, there are three things to check: 1. antenna monitor ratio and phase readings, 2. base current ratios, 3. monitor points. If all of these are in order you can be confident of your array.

In order to interpret the information you derive, you need to know the tolerances. In the case of loop currents and base currents, the commission says you must be within 5% of licensed value. So, if your license specifies tower two to have a loop current of .815, and you read .835, you are still ok since, on this particular tower, you could read from .775 to .855. Your phase readings must be within plus or minus three degrees.

If your antenna monitor and base current readings are within the tolerances of your licensed values, then, the last check is the monitor points. Monitor points are specified for directional arrays to prove that you are not sending more signal in a given direction than you are supposed to. These are read with a field intensity meter.

For monitor points, the upper limit is stated in your license and there is no tolerance. Whatever the license says is the absolute maximum you should read at that point.

However, there are no minimums and, in many cases, the actual reading may be only a small percentage of the allowable. I point this out so that you will understand that it is quite possible to have a monitor point with a limit of 82 millivolts and actually read only 15. This may be normal. So, if you find everything else in order, but you have a reading such as this, it is probably ok.

To be sure, you would want to look it up in the original Directional Antenna Proof of Performance to see what was read at that point. See if what you get is near the reading shown on the original proof. From this you can determine if it is normal. This proof of performance would have been an exhibit filed with the form 302 (*Application for Station License*) filed after the array was constructed. These are required to be kept on file at the station.

As for the requirement for reading monitor points now; there is none. However, the Commission requires that they be within the allowable readings. So, if you don't read them, you don't know whether they are in or not.

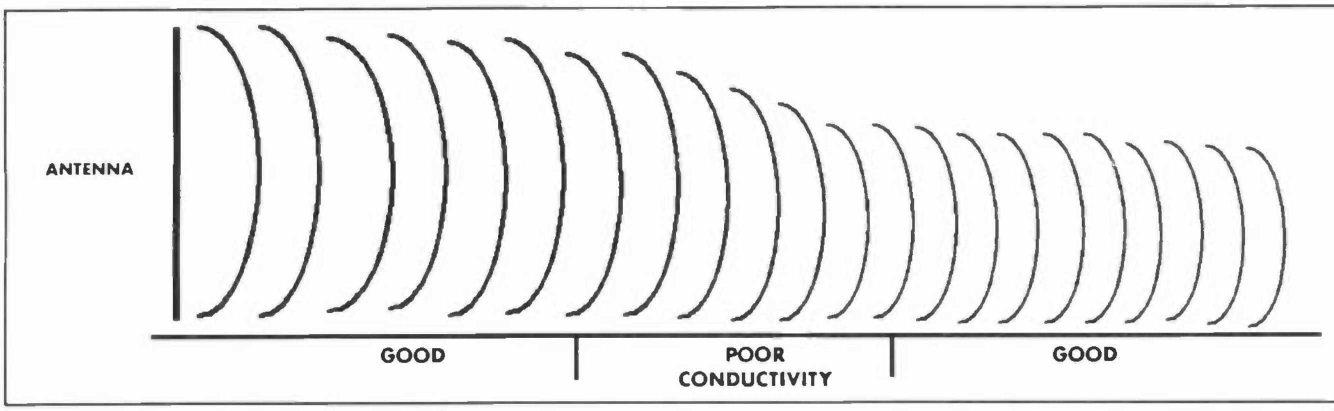
Now let's take a brief look at ground conductivity.

An AM station's coverage area is calculated using ground conductivity. This is measured in Mhos, which is Ohms backwards. For once, something in electronic nomenclature makes sense to me. If resistance is measured in Ohms, then conductivity (the opposite) should be measured in Mhos. For non-directionals it is only considered in a theoretical manner when applying for a license or a modification. In the case of directional stations, we actually make the measurements to prove the conductivity in the area covered by the pattern.

First, lets take a look at a model that makes this a little easier to visualize.

We know that a capacitor will pass AC (in this case RF), and that the signal will pass from one side to the other, even though there is no physical connection between the two sides.

When we build a tower and ground system, we are actually creating a giant capacitor; the tower being one side and the ground system being the other. As the radio waves come off the tower, they are seeking to return to ground. The



better the ground conductivity, the farther the wave will travel before it dissipates.

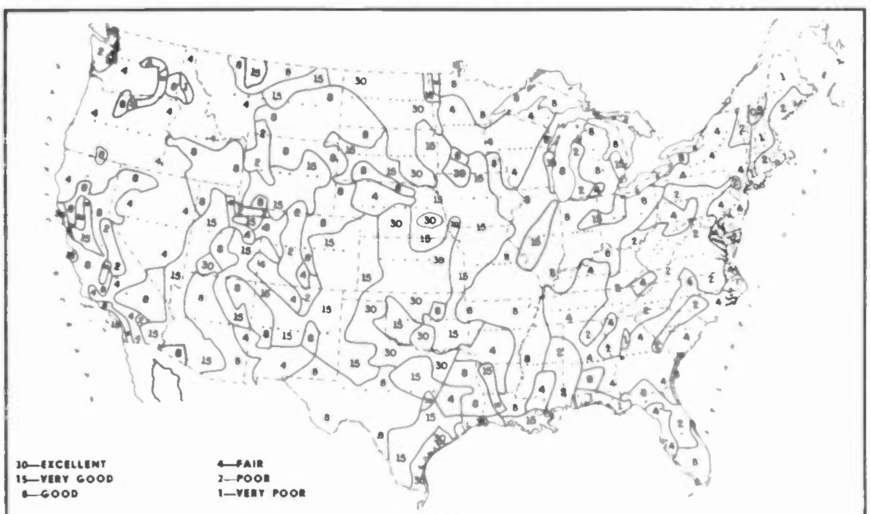
Looking at the figure above, we see that the wave decreases in strength as it moves away from the tower. As it travels over the high conductivity area, it decreases slightly. As it travels over the low conductivity area, it decreases more rapidly.

In setting up a directional array, you first establish where the signal will go in a non-directional mode. This is done by taking a series of measurements in a given direction and then plotting them on a special graph paper.

These measurements are generally made at 0.2 kilometer intervals for the first three kilometers. One kilometer intervals for the next 7 kilometers. Three kilometer intervals out to 25 or 34 kilometers. Certain cases may vary this measurement procedure slightly. For the procedure you can see 73.186 of the FCC rules. This is then used with tables supplied by the Commission to establish what the ground conductivity is in that given direction. The array is then adjusted by referencing it to the original, non-directional measurements.

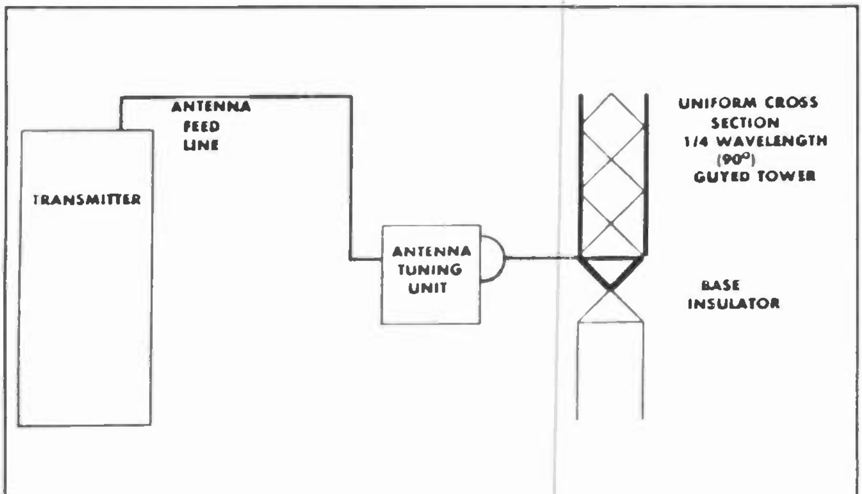
In establishing the allowable coverage contours for a non-directional station, a map issued by the Commission is used to calculate the approximate signal strength at a given distance in a given direction. For directional stations, it is used to produce the orignal predicted contours used in applying for the construction permit. The measured contours are then used when applying for the license.

This map, the M-3, gives the estimated conductivity for all of the United States. If you are in an area that shows estimated ground conductivity of 30, then you calculate the allowable power in that direction based on the idea that the signal will propagate very well. If you are in a area that shows a conductivity of 2, the assumption is that the signal will not propagate well and, therefore, you are allowed more power in order to cover the same distance.



MAP M-3

Before we proceed, let me point out that the small scale reproduction of the M-3 here should not be used in deriving any conclusions. The actual map is extremely large, and in sections. Just a few counties will cover a conference table. I just wanted you to know what it looked like so you could understand how it is used.



SERIES FED ANTENNA

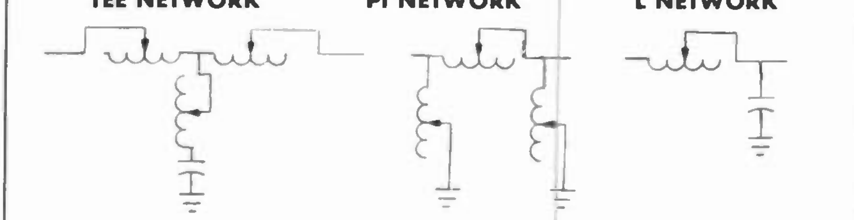


FIGURE 2

Now, in the case of directional arrays, you must measure the signal and calculate the conductivity. Therefore, if the map proves to be wrong in its estimate, it doesn't matter since the licensed pattern will be based on the measurements rather than the estimate shown on the map.

However, in the case of non-directional stations, we generally accept the estimate and no measurements are ever made. What I have found in my experience is that the map estimates are sometimes very generous and the actual ground conductivity is somewhat lower. In a couple of cases I have found it to be considerably lower.

When we set up the KRVA Dallas array, we discovered that, although the M-3 map showed a conductivity of 15 and 30, we actually measured 5 and 6. In our case we simply compensated for this difference in the alignment of the array.

Now, let's take a look at the basic non-directional series fed antenna. Before we get into the array, we need to get a little understanding of each individual tower. However, before going too much farther in antennas, I want to point out that I am just trying to get you to be able to understand the basics of how these things work. Before changing anything on an antenna, you need to really understand what you are doing, and, you must have the proper equipment for measuring and setting everything to its right value. A person can get in a lot of trouble if they try to proceed too fast in this area. This is an area where you learn the basics, then look over the expert's shoulder and ask a lot of procedural questions before trying it on your own.

Impedance is made up of pure resistance and also reactance. Reactance can be described as resistance that varies with frequency. Reactance is known as the J factor, with inductive reactance being labeled +j and capacitive reactance being labeled -j.

Looking at the drawing above we know that the transmitter wants to see 50 ohms j -0- as does the feed line into the ATU (antenna matching unit). The antenna itself exhibits 30 ohms +j 20. Therefore, the ATU will be set to effectively add 20 ohms to the resistance and produce -j 20 to offset the j of the tower.

There are several different networks that can be used to accomplish this task. The Tee is the most common, although you will also find Pi and sometimes L networks. The names describe the shape in which the components are laid out. Figure 2 above shows the schematic of the three types. Depending on whether the tower is + or -J, the components may be somewhat different from the schematics shown. Network design requires consideration of many different factors and there is an almost infinite variety of component arrangements even within the basic

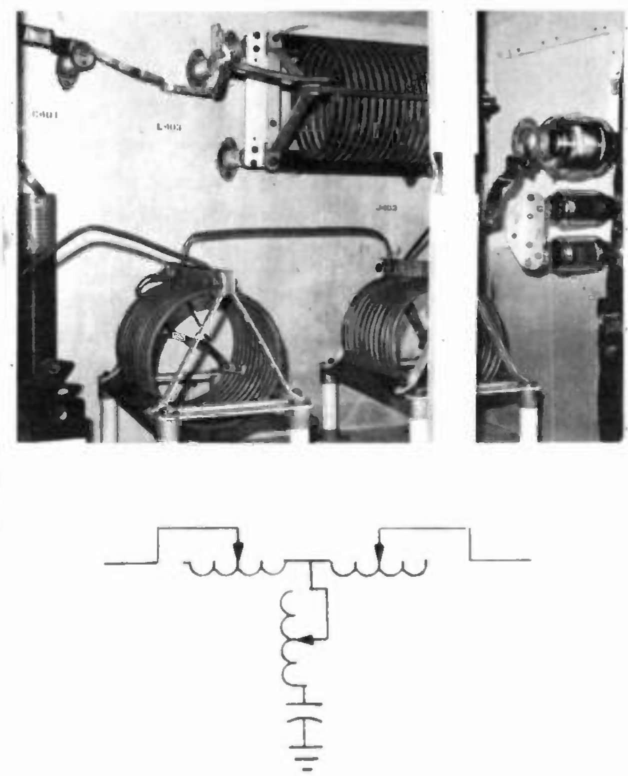


FIGURE 3

shapes.

The photo and schematic in figure 3, to the left, show a typical Tee network tuning box. Generally speaking, the input and output coils will be used to control the J factor and the shunt leg will control your resistance, as seen by the incoming feed line. There is some interaction between all of the components and proper methodology must be followed when setting one of these up. We have many other items to cover and we won't be getting into methodology, so I'm just mentioning it at this point.

If you know the tower height and frequency, you can predict the resistance and reactance. The tables shown in figure 4 can be used to reasonably predict these figures for any given tower height and frequency.

Let's take, as an example, KRVA's tower in Ft. Worth. We decided to use a 160' tower. This gives us a tower that is 94 electrical degrees tall. KRVA is on 1600, and a 90° tower (1/4 wavelength) is 153.7 feet. We are using a tower

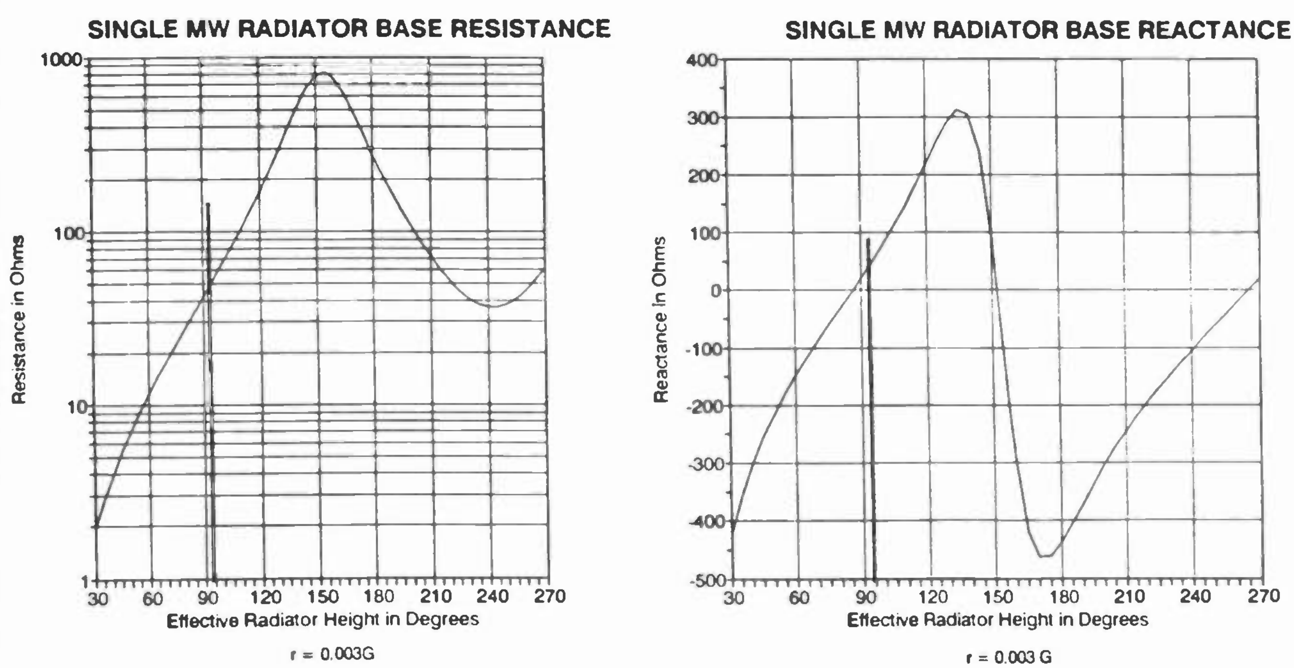


FIGURE 4
RESISTANCE AND REACTANCE VS TOWER HEIGHT IN DEGREES

GRAPHS HAVE 94° TOWER PLOTTED

that is 160 feet. This comes out to the 94 degrees mentioned earlier.

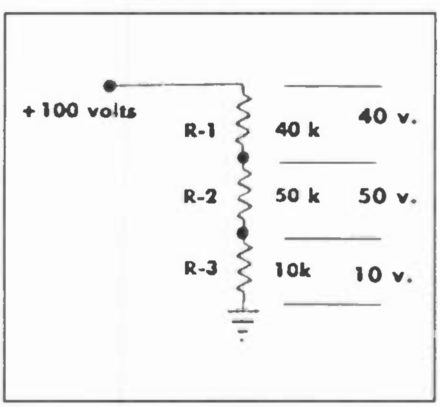
Now, looking again at figure 4, you can see where I have plotted a 94° tower. Now we know that the network can expect to see a an inductive reactance (+J) of about 25 and a resistance of 50 Ohms. This is true for non-directional only. When towers are closely spaced, there is interaction that changes the whole ballgame and you get what is known as mutual impedance. A full discussion of this is beyond the scope of this paper. Therefore, remember that we are talking about a single tower out in an open area and, when we get to directionalizing this tower by adding a second one, the impedances will be quite different. It is necessary, however, to have an understanding of matching a non-directional, to understand what the phasor is doing.

Now, returning to our non-directional example, one thing that becomes apparent from this example is that a tower, such as this, requires very little in the way of a matching network. The transformation from the 50 ohms J -0-, that the line wants to see, to the 50 ohms +j 25 that the tower exhibits is quite small.

This transformation ratio is very important in designing a station for performance. The less transformation you have to make, the easier it is to keep your bandpass broad; thereby allowing all of your audio to get into the tower and avoiding reflected power. Sometimes, however, in order to get the desired directional pattern it is necessary to deal with some less than desirable ratios. In these cases, additional networks or components can be used to lessen the effect.

Again, it is beyond the scope of this paper to get into these. However, knowing that this is the case, may help you understand what you see at your own station.

Before we get into the phasor, we need to take a quick look at a voltage divider. Using a simple string of resistors and Ohm's law, we can divide the voltage to get what we need, where we need it. It is readily apparent that the voltage across each resistor is proportional to the resistor value. We will see this same thing as a part of our phasor.



Now that we have talked about the individual actions taking place in one. Let's put it together and examine a simple phasor.

At this point we will take a look at a simple two tower array. We will examine the function of each component, or group of components, to see how they affect the performance of the array.

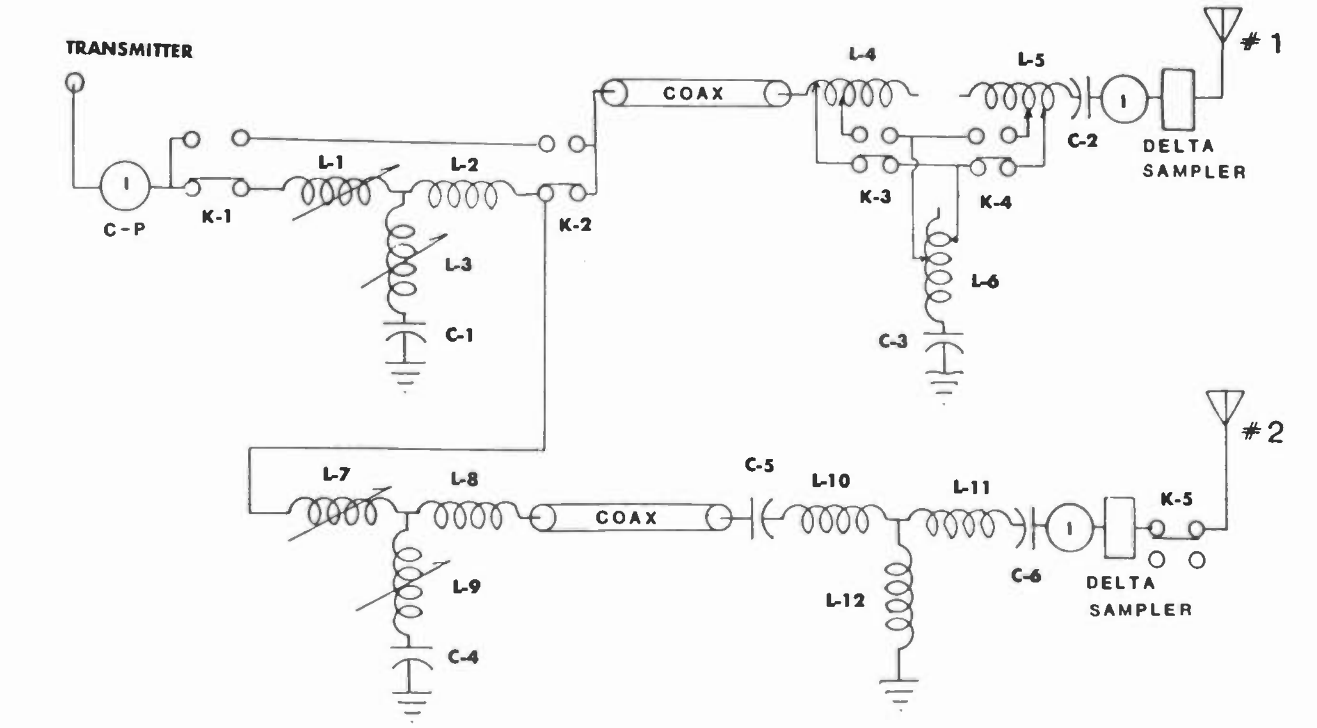
As you know, two towers is the least number you can have and still develop a directional pattern. This is simply because you must have power division and phase relationship in order to direct the signal. In our example we have a non-directional daytime operation and the two tower array at night. This is a very common setup to have NON-D day and DA at night.

On the next page is the full schematic of the phasor. We will be looking at different sections as we go along and this will be our reference figure for getting the whole picture.

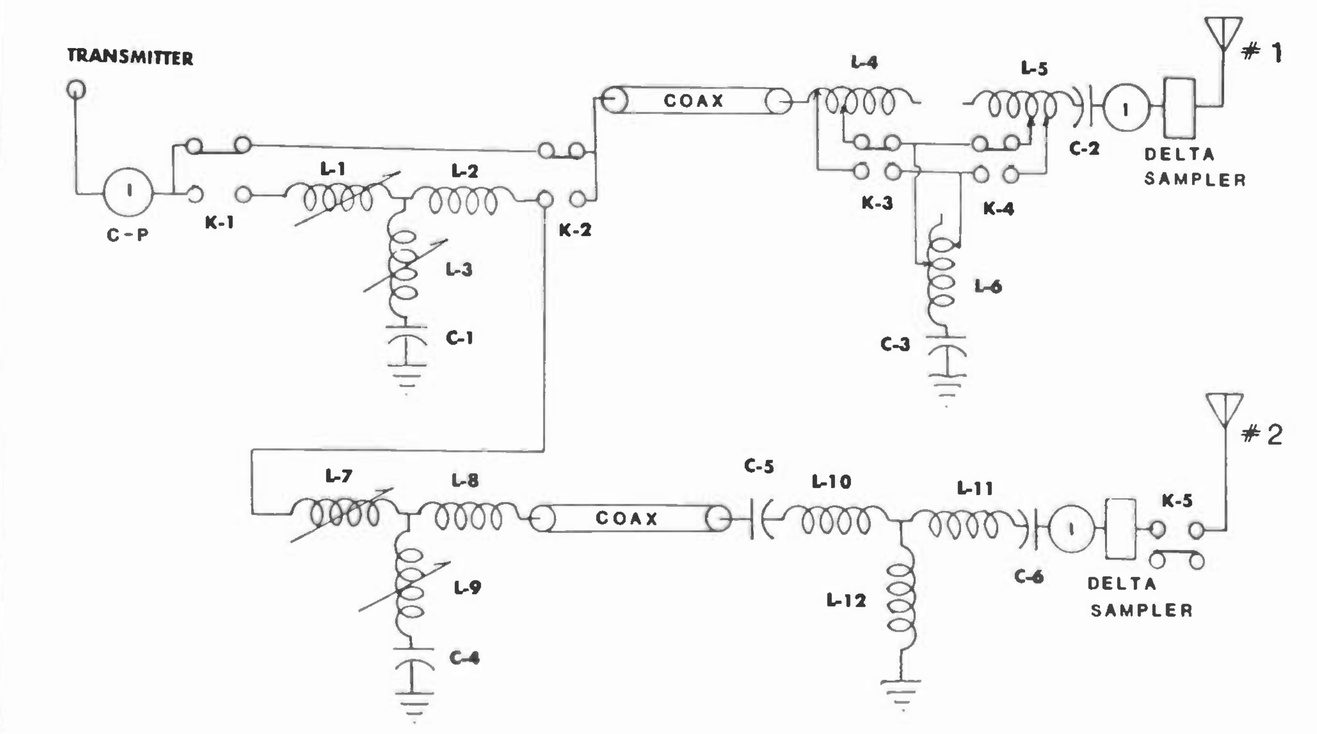
First, let's look at what we have when the array is in NON-D status. The bottom figure shows the phasor with the relays set for daytime operation. In this mode, K-1 and K-2 serve to delete the T network made up of L-1, L-2, L-3 and C-1. K-2 also deletes the feed to tower #2. K-5 "floats" tower #2 by leaving the tower totally above ground and not connected to any components.

In many arrays, this relay would instead, switch in a coil that would serve to detune the tower. The coil would be set to present an extremely high reactance and would be tuned for minimum current in the tower. This would give the tower the appearance of a giant resistor, as far as the RF is concerned, and its effect would then be negligible. Either way the object is to render the tower "invisible".

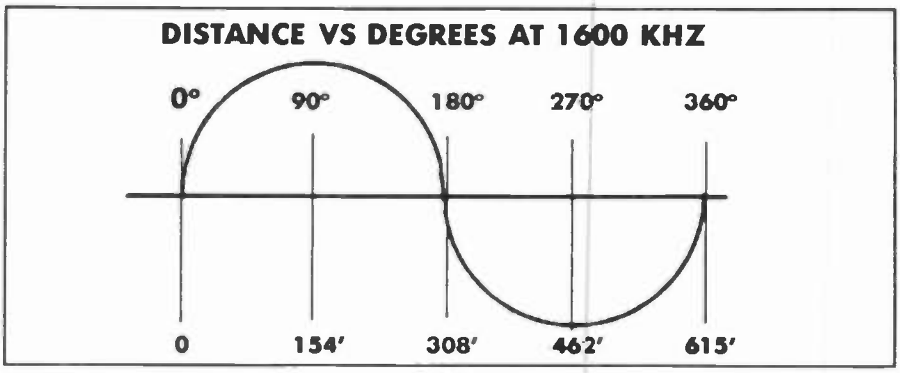
K-3 and K-4 switch in the two inner taps on L-4 and L-5 which are connected to the lower tap on L-6. This, along with C-3 forms a T network to match the 50



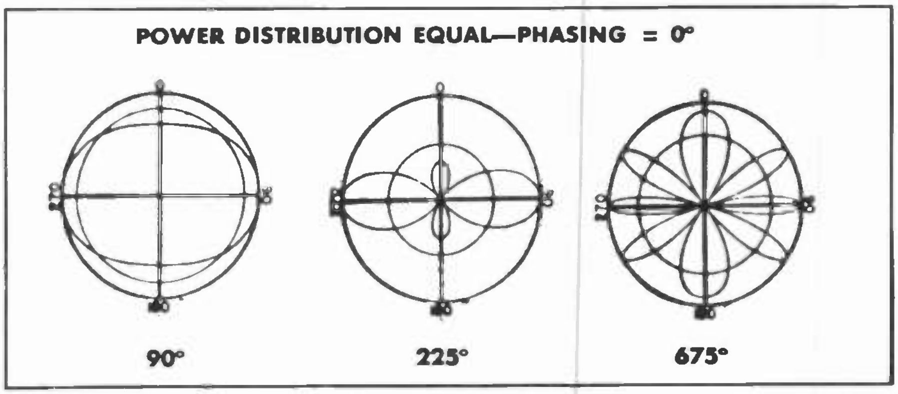
PHASOR SET FOR DIRECTIONAL OPERATION



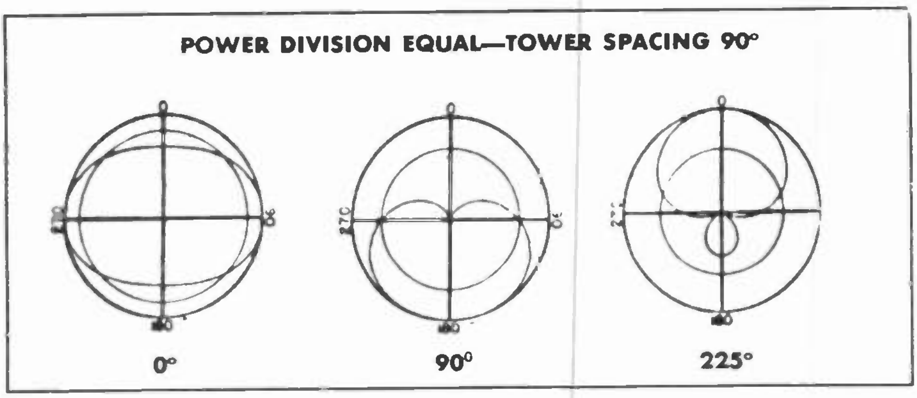
PHASOR SET FOR NON-DIRECTIONAL OPERATION



DISTANCE VS DEGREES AT 1600 KHZ



POWER DISTRIBUTION EQUAL—PHASING = 0°



POWER DIVISION EQUAL—TOWER SPACING 90°

Ohm output of the transmitter to the tower. In the NON-D mode we simply have the transmitter output connected to the feed line, connected to the T network, connected to the tower. All other components are switched out of the act.

Remember that schematics are drawn for simplicity and the taps on the coils may not actually be anywhere near the position, or even in the order, shown on the schematic. The tap shown nearer the center on the schematic might, in reality, be closer to the end than the one shown near the end of the coil.

C-2 and C-6 serve to offset part of the +j of the tower in order to make the network design more manageable. (Remember that +j is inductive, -j is capacitive, and they cancel each other) This gets the tranformation figures into an area where broad-banding is easier. Remember that you want to see as little shift as possible in the resistance and reactance as you move away from the center frequency. By offsetting some of the +j in the tower, it makes the networks easier to design for more efficiency in this area.

The transmitter, of course, wants to see 50 Ohms with -0-j. Although there is interaction when changing any of the components in the T network, generally speaking, L-6 will have more effect on your reactance and the top coils, particularly L-4 will have the most effect on your resistance.

As a part of understanding our simple, two tower, array we need to also look at some of the other items that affect the pattern besides phase and power distribution. Phase and power distribution are the two items that are controlled by the phasor. However, when the array was designed, a number of other factors were considered in order to achieve the desired shape.

Tower spacing has a tremendous effect on the pattern shape. This is considered when the pattern is designed. We will not attempt to get into designing an array here. What we want is an understanding of what goes into the design so that we can then understand what we see happening around us. For our purposes here, we will just look at a couple of examples of what spacing can do to the final shape.

The actual array, upon which this paper is based, has the two towers spaced 90 electrical degrees apart. This simply means that the distance between the two towers is one quarter of a wavelength at that frequency. Remember that radio comes off the tower as a sine wave and anytime the length of something, or distance, is referred to in degrees, it refers to the distance the wave would travel during that many degrees of its cycle. The top figure on the preceding page shows a sine wave with certain key degree markings. If we assume that we are looking at 1600 kHz, then a wavelength is 615 feet long (rounded off). Therefore, 90 degree spacing would mean that the towers would be placed 154 feet apart. 180 degrees would be 308 feet apart, and so on.

Taking a look at the effect spacing has on the pattern, look at the center figure on the preceding page. The patterns shown were all made with the power distribution being equal and the phasing being left at zero degrees. The only thing being changed is the tower spacing.

Now, to show the effect of the phasing interacting with the tower spacing, take a look at the bottom figure on the preceding page. In this instance, the power division is still equal between towers and the tower spacing is left the same. However, we are now feeding tower number two varying degrees out of phase in reference to tower one. With tower spacing and power division being left the same, the different patterns shown are created by changing only the phasing.

It becomes readily apparent that a large number of patterns can be developed simply by varying any one of the several variables. To conserve space, I am not going to show examples of changing power division, I think you get the idea. For all of these examples I used towers of the same height. You can get even further variation by having towers of uneven height. You can also see how a relatively slight shift in your DA parameters can result in a fairly large shift in your pattern.

Having digressed in order to pick up some necessary background, we will now switch the phasor to directional and climb back in it to see what each of the components is doing.

Looking at the upper schematic on page 232, we find two tee networks made up of L-1, L-2, L-3, and C-1 in the upper portion of the schematic (which we will call Tee 1), and L-7, L-8, L-9, and C-4 in the lower portion (which we will call Tee-2). These two networks combine their actions to do two things. One, the input of Tee 1 sets your common point impedance. It is affected, however, by any changes you make down the line and is always adjusted last. The other thing that these two networks do together is divide your power.

The two networks work very similar to the voltage divider circuits we looked at earlier. The resistance of the two networks are set such that the resistance, as seen looking toward each tower, will provide a circuit equivalent to two resistors in a voltage divider configuration. For instance, if the input of Tee 2 has 80 percent of the total resistance presented at the junction of Tee 1 and Tee 2, then tower #2 will get 20% of the power. This is then one quarter of the power in tower #1 (80/20). In this case your antenna monitor would show tower #1 as reference tower with 100, and tower #2 would read .250. In the case of more than two towers, the percentage of resistance for each tower will determine its share of the power. Your reference tower is, of course, always the tower with the most power and all other towers are referenced as a percentage of it.

The other two networks at the towers serve to match the towers to the lines. However, note C-5. This extra capacitor in the line to tower #2 serves to provide a phase shift that will get tower #2 in the ballpark of the desired reading. Then, by adjusting the combination of L-7 and L-9, you can get the phase fine tuned with L-7 providing phase control and L-9 being used to set your power divider back to its proper reading.

This concludes our look at this array. I hope that the explanations have helped in your understanding of what you see in your own situation. As always, your comments and suggestions are welcome.

TRANSITIONS: THE PATH FROM ANALOG TO DIGITAL

Tuesday, April 16, 1996
9:00 am -12:00 pm

Session Chairperson:
Jerry Whitaker, Technical Writer, Beaverton, OR

***VIRTUAL STUDIOS: THE APPLICATION OF AVIATION SIMULATION TECHNOLOGY TO LIVE BROADCASTING TECHNOLOGY**
Aviv Tzidon
RT-SET
Herzelia, Israel

MOVING PRODUCTION AND POST-PRODUCTION STUDIO DATA FASTER THAN REAL-TIME
Marc Friedmann
PRISA Networks
San Diego, CA

***THE TRANSITION TO COMPRESSION: LESSONS LEARNED**
Matthew S. Tietze
Public Broadcasting Service
Alexandria, VA

***APPLICATIONS OF HIGH SPEED DIGITAL NETWORKS IN FILM AND VIDEO POST PRODUCTION**
James Fancher
Pacific Ocean Post
Santa Monica, CA

INTERACTIVE TELEVISION: THE PRINCIPLES FOR A WORLDWIDE COMMON FAMILY
Louis Libin
NBC
New York, NY

***THE GROWTH OF THE DIGITAL VIDEO SIGNAL**
Jason Job
Advanced Audio Visual Systems
Sioux Falls, SD

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*Paper not available at the time of publication.

MOVING PRODUCTION AND POST-PRODUCTION STUDIO DATA FASTER THAN REAL-TIME

Marc Friedmann and Don Deel
PRISA Networks, Inc.
San Diego, CA

ABSTRACT

Economic methods are now available to move production and post-production data faster than real-time in networked environments. Using ANSI standard Fibre Channel serial interfaces on Silicon Graphics workstations and servers with optimized hardware and software, computer-to-computer and computer-to-disk communications have been demonstrated to transfer digital image data at a sustained throughput up to 600 megabits/sec. Incorporating these interfaces, production and post-production facilities are achieving order-of-magnitude improvements in response time when accessing and transferring large files.

THE NEED FOR FASTER THAN REAL-TIME DATA

The motion picture and television industries are currently in the midst of a major shift from traditional analog production techniques to new digital production techniques based on software tools developed for general purpose computing platforms. Like the traditional hardware based solutions which are rapidly being replaced, these tools are available in a performance hierarchy that extends from inex-pensive, general purpose software for mass produced personal computers, to high-end applications used by small groups of professionals, on special purpose workstations optimized for image rendering.

Software based tools are now routinely used for the most demanding motion picture and video projects, for a wide range of creative tasks including:

- Editorial - scripting, storyboarding, logging and nonlinear editing.
- Image synthesis - animation, painting, graphics and titling.
- Image processing - retouching, rotoscoping, color correction, morphing and wire removal.
- Multi-layer composition - special effects, traveling mattes and linear keying.

As digital production techniques proliferate and the number of computer-literate artists grows, data file sizes have increased exponentially. So has the need for broad-based artistic collaboration. Rapid and responsive communication between computer-based artists, editors and producers is becoming a critical factor as the ramp-up in digitally processed images accelerates.

Digitized film and video images used in production and post-production produces one of the computer industry's most demanding applications. At 1-40 megabytes per frame, even a clip of a few seconds in length rapidly grows to more than a gigabyte file (see Figure 1).

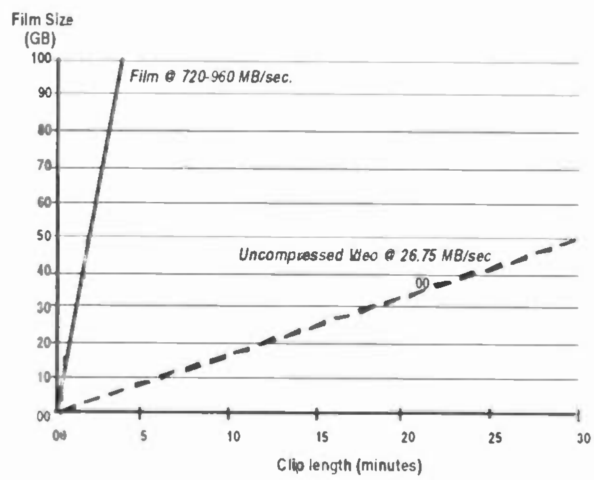


Figure 1. Digital Media File Size

EXISTING NETWORK APPROACHES

Moving files this size from central storage to an artist's workstation or among the artists collaborating on a project, using the current generation of digital networks, can introduce significant delays in the production process. The massive data transport requirements for film and video production overwhelm the capabilities of networks designed to handle email and printer sharing, such as Ethernet, Fiber Distributed Data Interface (FDDI) and Token Ring.

Cost-effective solutions for storage and sharing these massive image data files is critical to today's motion picture and television facilities. Currently, two system architectures prevail in post-production facilities that have begun the conversion to computer-based production tools.

The first, client-server architecture (see Figure 2) centralizes storage access through a large, high-speed processor. Local workstations access files through the server, which delivers them to limited local memory and disk. Centralized storage attached to the server frequently exceeds 1 terabyte. The physical interface between the server and storage must be as rapid as possible; this is typically facilitated using parallel SCSI data busses and striped disk arrays based on RAID techniques. Currently, fast and wide SCSI is used, yielding transfer rates approaching 20 megabytes per second (MB/s).

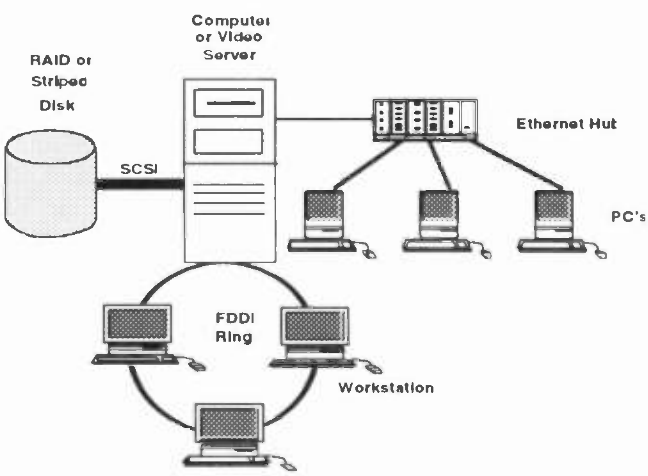


Figure 2. Client/Server and Networks

The second system architecture, cross-mounted disks, provide high speed local access to files on disks attached to each workstation, and let users share files between workstations, typically over an Ethernet or FDDI network. The server effectively is distributed across all network systems. Local storage is typically on 1-4 disks, representing 1-36 gigabytes of storage. While interface to these disks is through SCSI, disk access speed limits file transfers to a maximum rate of 3-4 MB/s. Since each workstation has access to all disks, simultaneous access to remote disks by multiple users can cause significant performance problems.

File access across the network is generally supported by one of three protocols: Internet Protocol (IP); File Transfer Protocol (ftp); or Network File System (NFS). While use of NFS is straightforward and offers flexible file access, its use of small packet sizes-typically <8 KB-and considerable management of each block by the operating system significantly reduces data throughput rates. Since ftp procedures are disk-to-disk transfers, they are limited to the throughput rate of the slowest disk interface involved. Applications seeking higher performance than NFS or ftp may use the UNIX Remote Procedure Call (RPC) directly, which operates as a much faster computer memory-to-memory transfer. Both ftp and RPC-based transfers, however, are ultimately limited by the transfer rate capability of the network.

Today's studios generally use standard Ethernet or FDDI for network communication. To improve responsiveness, some installations are beginning to experiment with switch-based Fast Ethernet. Ethernet and FDDI have been optimized as enterprise-wide networks, which are dominated by small message traffic rather than the large block transfers common with digital video files. Although specified line rates may range from 10 to 100 megabits per second (Mb/s), the combination of network and file system structures yields an observed actual throughput of one to 25 Mb/s under moderately loaded conditions. Transfer of a gigabyte file, common in production and post-production environments, at these data rates requires five minutes to two hours. As artists using workstations at post-production facilities continue to grow, the need for much higher speed networks is rising dramatically.

HIGH SPEED INTERFACES

Most significant among the new high speed is Fibre Channel. Fibre Channel is a scaleable interface standard from the computer industry defined to achieve high speed data transfer among personal computers, workstations, mainframes, disk drives, peripherals and display devices. With demonstrated sustained throughput over 600 Mb/s, it is the only computer network-oriented interface currently being deployed capable of moving studio data faster than real-time.

Approved as an ANSI standard, and using either loop or switch-based topology, it combines attributes of SDI-like channels with packetized computer networks over a serial interconnect capable of operating across campus-wide distances.

Having the ability to support multiple protocols simultaneously, Fibre Channel is a hardware intensive interface for environments involving a wide variety of computer, disk and studio equipment. Key technical features of Fibre Channel are summarized in Figure 3.

Feature	Fibre Channel
Line Rate	266, 531, or 1062.5 Mbits/sec
Data Transfer Rate	640-720 Mbits/sec
Frame Size	2112 Byte Payload
Protocols	SCSI, IP, ATM, SDI, HIPPI, 802.3, 802.5
Topology	Loop, Switch
Data Integrity	10E-12 BER
Distance	Local and Campus; up to 10 Km

Figure 3. Fibre Channel Features

The Fibre Channel structure is defined as a multi-layered hierarchy of functional levels. Five layers define the physical media and transmission rates, encoding scheme, framing protocol and flow control, common services and the upper layer application interfaces (see Figure 4).

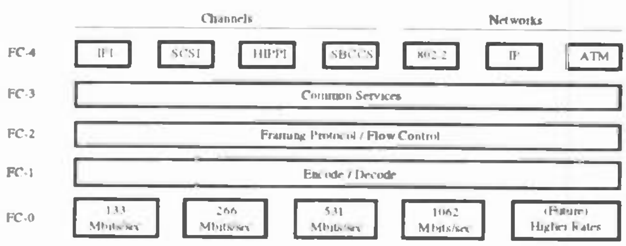


Figure 4. Fibre Channel Hierarchy

FC-0, the lowest layer, specifies the physical features of the media, connectors, transmitters and receivers, including electrical and optical characteristics, transmission rates and other physical elements of the standard (see Figure 5). Note that video coax and the 1300 nm single mode fiber found in broadcast facilities are incorporated in the standard. FC-1 defines the 8B/10B encoding/decoding scheme used to integrate the data with the clock information as required by serial transmission techniques. FC-2 defines the rules for framing the data to be transferred between ports, a look-ahead sliding-window flow control scheme, different mechanisms for circuit and packet switched classes of service, the error detection techniques, and means of managing the sequencing of data transfer. FC-3 provides common services required for advanced features, such as striping and hunt groups. FC-4 provides the seamless integration of existing standards, by accommodating a number of other protocols such as SCSI, TCP/IP, FDDI, HIPPI, SDI, ATM, Ethernet and Token Ring.

Fibre Channel combines the best attributes of a channel with those of a network through a simple technique: it provides a means to transfer data between a buffer at the source device (e.g., a video server drive) and another buffer at the destination device (e.g., a workstation or frame buffer). Fibre Channel ignores the data itself and how is formatted, and simply takes what is in the sending buffer and transports it to the receiving buffer at the full bandwidth of the channel. After initial handshaking, control of the rate of data flow is handled by the receiving device indicating the amount of available memory buffer available. This low-level flow control allows Fibre Channel to avoid any data loss due to congestion. Simple error correction is handled in hardware, much like a channel. If a data transfer fails due to an error then a re-try occurs immediately, without consulting system software, maintaining above real time performance.

Medium	Max. Dist.	Data Rate (Mbits/sec)	Signal
Single Mode Fiber	10 km	266, 531, 1062	LW Laser
50μm Multimode Fiber	2 km	266, 531, 1062	SW Laser
67μm Multimode Fiber	1.5 km	133, 266, 531, 1062	LW LED
Video Coax	100 m	133, 266, 531, 1062	ECL
Miniature Coax	35 m	133, 266, 531, 1062	ECL
Shielded Twisted Pair	100 m	133, 266	ECL

Figure 5. Fibre Channel Media

To accommodate on-line, off-line, video and computer needs, Fibre Channel defines four different classes of service (see Figure 6). *CLASS 1*, a circuit switched connection, functions much in the same way as today's SDI physical channels. No other devices can share the engaged link when a Class 1 connection has been established between two devices. *CLASS 2* is a connectionless, frame-switched link which provides guaranteed delivery with acknowledgment of receipt. As with traditional packet-switched networks, the path between two ports is not dedicated, allowing for shared use of the link's bandwidth. *CLASS 3* is a connectionless "datagram" service that allows data to be sent rapidly to multiple devices attached to the fabric, but no confirmation of receipt is given. By not having to wait for confirmation, Class 3 service speeds the time of transmission. However, if a single user's link is busy, the hardware will not immediately know to re-transmit the data. *CLASS 4* offers constant available minimum bandwidth or guaranteed latency, and is useful for isochronous applications, such as single or multiple streams of real time digital video.

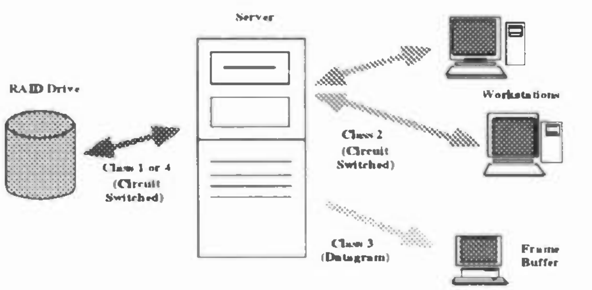


Figure 6. Fibre Channel Classes of Service

Fibre Channel supports a variety of fabric topologies. It is a closed system which relies on ports logging in with each other and the fabric to trade information on attributes so they can determine if they can exchange information. Possible fabric topologies include point-to-point, crosspoint-switched, and arbitrated loop. All classes of service can be supported by either the switch or arbitrated loop topologies.

FASTER THAN REAL-TIME STUDIO NETWORKS

Dramatic improvements in system response can be achieved in the post-production environment through cost-effective application of Fibre Channel interfaces and performance-enhancing software. The key to successful deployment is optimal use of Fibre Channel's variety of fabric topologies (see Figure 7). The Arbitrated Loop topology offers a low cost connection useful in post-production applications where transfer requests tend to be large and bursty. With up to 127 ports connected on a single loop, each port competes for a connection through a defined arbitration process. When arbitration is complete, two successful nodes obtain access to the entire bandwidth of the loop. A crosspoint switch may be used to connect workstations on arbitrated loops, centralized storage and other I/O devices. It offers the greatest total bandwidth and is applicable when high aggregate system load needs to be sustained.

Arbitrated loop topologies should be used in post-production workgroups (see Figure 8), which would consist of workstations configured with either cross-mounted disks or in a client-server

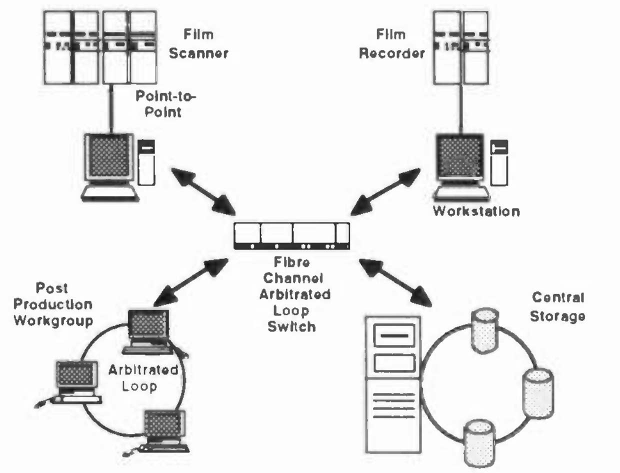


Figure 7. Fibre Channel Fabric Topologies

arrangement. Specialized interaces, such as D1 or archive tape, would be through a workstation or the server. For maximum performance, disk attach would be to a shared RAID via Fibre Channel. If workgroup storage is centralized through the server, then this disk would likely be a RAID array. To maximize uptime and reliability, a Fibre Channel hub should be included in the arbitrated loop configuration. This hub is a non-switching passive device used to ensure loop operation is maintained in the event a node is powered off or a cable is disconnected or broken. A hub could also be located in a remote wiring closet, simplifying system administration and reconfigurability.

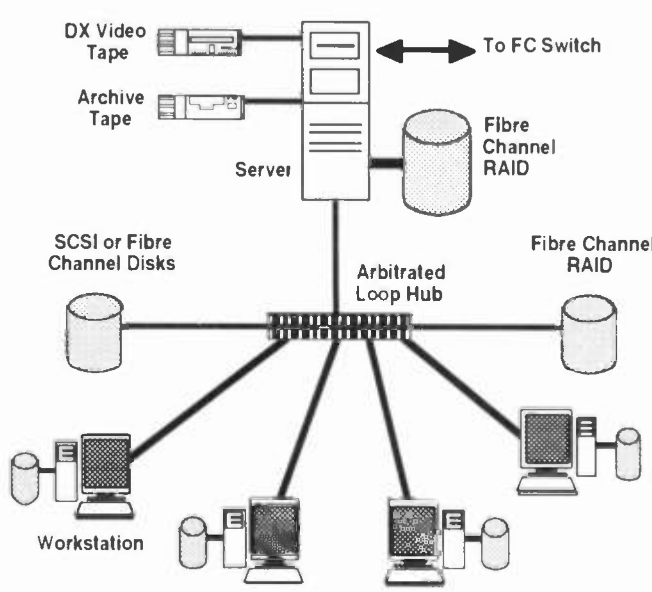


Figure 8. Post-Production Workgroup Based on Fibre Channel Arbitrated Loop

Central storage also benefits significantly from Fibre Channel arbitrated loop (see Figure 9). RAID arrays are available housing up to 36 nine-gigabyte (GB) SCSI drives. Configured using multiple SCSI channels linked via FC-AL, these arrays provide over 300 GB of storage with transfer rates approaching 100 MB/s. Alternatively, using newly available 4 GB drives with native Fibre Channel interfaces, up to 126 Fibre Channel drives may be connected in a loop topology within the cabinet, producing a RAID offering 0.5 terabytes (TB) of storage. In 1996, 9 GB Fibre Channel drives will be available, pushing a single RAID over 1 TB storage capacity. With up to 126 RAIDs on an arbitrated loop, over 100 TB of storage will be accessible at speeds approaching 100 MB/s through a single port on a server. As with the work group topology described previously, a hub can be incorporated to ensure maximum up-time and reliability.

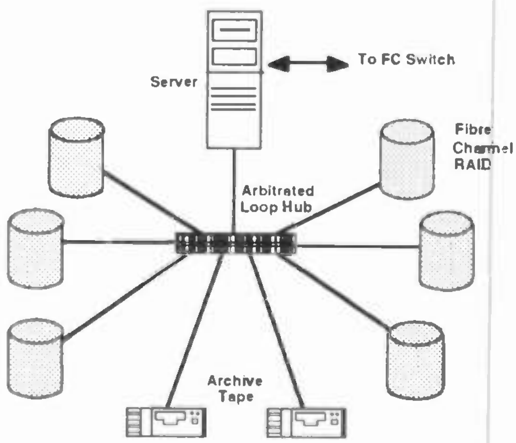


Figure 9. Centralized Media Storage

Specialized interfaces and protocol conversion, required to transport SMPTE-259M and other digital video bit streams, will be incorporated in the Fibre Channel switching fabric as the market develops. This will permit a topology with similar advantages to the centralized multi-level (audio, video & data) matrix switching products used in many post-production facilities today.

FIBRE CHANNEL ADAPTERS FOR STUDIOS

The most popular servers and workstations used to work with digitized film and digital video are made by Silicon Graphics, Inc. On the higher-performing end of the spectrum, Challenge DM, L, and XL machines are used as servers, and Onyx systems are used as workstations. These systems offer a 64-bit SGI proprietary computer bus called HIO, capable of 320 MB/s operation. For desktops, SGI offers the Indigo2 workstation which come equipped with a 267 MB/s, 64-bit GIO computer bus. These computer busses provide an excellent match for the performance capabilities of Fibre Channel.

Prisa Networks has designed its family of NetFX Fibre Channel adapters for SGI's GIO and HIO bus connections. The NetFX adapters provide one or two independent, fully capable Fibre Channel ports, allowing SGI systems to communicate with disk and network traffic simultaneously. Each port is capable of supporting point-to-point, arbitrated loop, and switched fabric Fibre Channel topologies, and providing Class 1, 2, 3 service. Copper or fiber optic media can be used by the ports, at the standard speeds up to 1062 megabits/second.

The NetFX Fibre Channel ports have been designed to support an average sustained data transfer rate in excess of 70 megabytes/second (560 megabits/second) for multi-megabyte transfers. Delivered performance is highly dependent upon the specific mix of large block data transfers and small message traffic, which trade off against each other. Because of per-transfer handling overheads, large block data transfers result in higher data throughput rates than small message traffic.

NETFX HARDWARE

The hardware for the NetFX Fibre Channel adapter includes Gigabaud Link Modules (GLMs), Fibre Channel Controllers, FC Controller Interface Logic, Local Memories, BiDi FIFO and BiDi Register Structures, and a Host Bus Interface Logic section (see Figure 10).

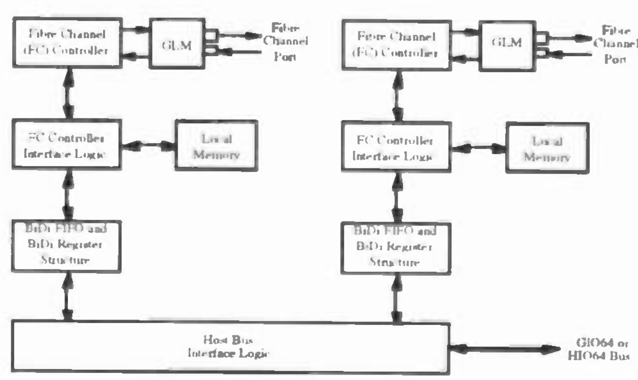


Figure 10. NetFX Hardware Block Diagram

Gigabaud Link Modules (GLMs) perform high-speed 20:1 parallel-to-serial and 1:20 serial-to-parallel functions and allow each port to communicate using either copper or fiber optic media. Since different GLMs are required to support specific Fibre Channel media combinations, each port on the NetFX adapter has a GLM socket that allows it to be configured independently.

The Fibre Channel (FC) Controller performs the high-speed FC-1 and FC-2 functions and assists with some of the FC-4 functions. Each FC Controller connects directly to its GLM and handles the lower-level communications functions in hardware. Most of the FC Controller's interactions with the host system are via direct memory access (DMA) operations, which are used to access commands, deliver status, and perform data transfer operations. Memory structures are used for communications between the NetFX adapter and the host system to minimize the number of interrupts that must be handled by the host operating system.

Each FC Controller Interface Logic block manages requests made by its FC Controller for DMA activities, and handles requests from the host system for accessing the port's Local Memory, FC Controller, and GLM. DMA requests from the FC Controller are directed to either the Local Memory or to the host memory, depending upon the referenced address. Requests from the host system are for accessing data structures in the Local Memory, for accessing the control and status registers in the FC Controller, and for exercising control and status functions over the GLM.

Local Memory is provided for each FC port to enhance overall system performance. Each port's local memory can be accessed both by the FC Controller and by the host system. By placing most of the memory data structures for commands and status in local memory, DMA operations to and from host memory are largely limited to data transfers.

The BiDi FIFO and BiDi Register Structures are used by the FC Controllers and the Host Bus Interface Logic as communications buffers. The BiDi FIFOs are used to "pipeline" DMA data transfers and sustain high data throughput rates by minimizing the performance impacts introduced by bus arbitration and host memory access latency.

The Host Bus Interface Logic manages communications via the connection to SGI's GIO or HIO bus, and includes special provisions for improving data throughput rates by pipelining all DMA Write and Read operations to and from host memory. This helps to minimize the performance impact caused host memory access latency.

NETFX SOFTWARE

The NetFX Fibre Channel adapter supports a hierarchical software facility that includes low-level drivers, protocols, network management support and diagnostics (see Figure 11). NetFX software has been optimized for very high sustained data throughput while supporting high transaction rates for small-message traffic. Implemented using standard interfaces provided by the SGI IRIX operating system, the Hardware and N_Port Drivers allow the host operating system software to communicate with the NetFX hardware. Protocol software enables the operation system to communicate with attached Fibre Channel devices. Facilities for SNMP-based network managers are supported, as are diagnostics, to maintain the network and verify the correct operation of the hardware.

NetFX I/O and network software facilities are made available to the operating system and to user-level applications by the protocols. I/O-oriented protocols for encapsulating SCSI traffic are provided for communicating with attached Fibre Channel I/O devices, such as disk drives. Both SCSI Initiator and SCSI Target mode operations enable software to initiate or receive SCSI commands, perform data transfers, and receive or return responses.

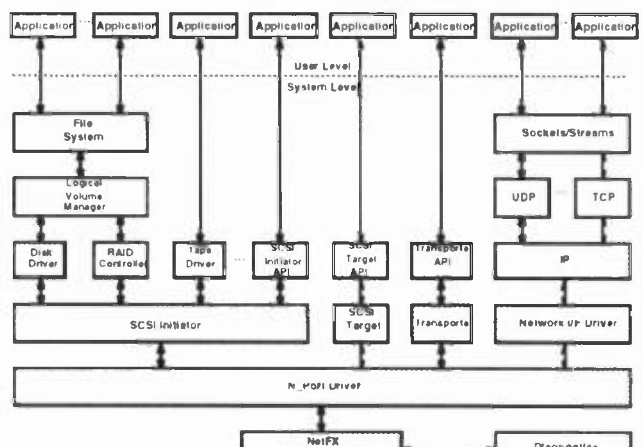


Figure 11. NetFX Software Hierarchy

A network-oriented protocol for encapsulating Internet Protocol (IP) is also made available to the operating system. This software allows the operating system to send and receive IP traffic over

Fibre Channel for standard data communications protocols, such as TCP, UDP, NFS, SNMP, and Telnet.

NETFX TRANSPORTER

Prisa has developed a protocol called the "Transporter" which optimizes transfers of large blocks of data between computers that are connected by Fibre Channel. This protocol is ideal for moving digitized film and uncompressed digital video data between different computers very quickly; in the case of digital video data, it can make these transfers happen faster than real time.

Transporter treats data transfers between computer systems as memory-to-memory I/O operations, rather than as the more traditional data communications networking operations used in conventional local area networks, such as Ethernet and FDDI. Transporter avoids many of the software overheads and inefficiencies associated with standard IP-based data communications protocols by utilizing the "native" capabilities of Fibre Channel, most of which are implemented in hardware for maximum throughput efficiency.

Fibre Channel has the ability to interconnect multiple systems, as do conventional networks. It also allows large data transfers to proceed at nearly the full speed of the communications media and incorporates hardware-level flow control and error detection capabilities. These characteristics are typical of conventional I/O channels. Transporter takes advantage of these capabilities by implementing network-style functions in an I/O-style fashion.

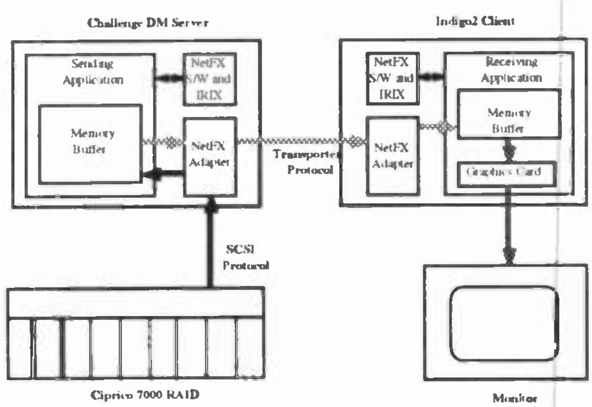


Figure 12. Throughput Test Setup

NETFX THROUGHPUT

An SGI Challenge DM, Indigo2 workstation and Ciprico 7000 Series RAID were connected together with NetFX Fibre Channel adapters to run throughput tests. Application-level programs were used to play a digitized video file directly from the Ciprico RAID to the monitor on the Indigo2 via the Challenge DM (see Figure 12), and actual throughput was measured.

Sustained transfer rates of 60 MB/second (480 megabits/second) was measured for the end-to-end, application-to-application throughput. This transfer rate is approximately twice that needed for real-time digital video. Peak file transfer rates over 70 MB/s have been observed.

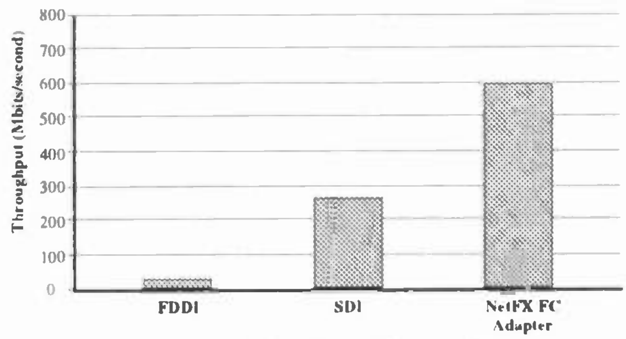


Figure 13. Relative Throughputs

The differences between the sustainable throughput rates that are possible using FDDI, SDI, and the NetFX Fibre Channel adapters are dramatic (see Figure 13). The sustainable throughput rate shown for FDDI is a typical number for moderately loaded conditions, and the rate shown for SDI is the line rate.

CONCLUSION

Using Fibre Channel, it is possible to move uncompressed digital video around the studio environment at faster-than-real-time speeds. Fibre Channel adapters have been created for the computer systems most commonly used in studios, and throughput measurements show sustained data transfer rates of 480 megabits/second between these systems, with peak file transfer rates up to 600 megabits/second achievable. By incorporating NetFX adapters, studios can achieve order-of-magnitude improvements when accessing and transferring large digital video data files. The speed of data movement will become critically important as the industry ramp-up of digitally processed images accelerates, and Fibre Channel adapters provide a solution to this growing need.

INTERACTIVE TELEVISION: THE PRINCIPLES FOR A WORLDWIDE COMMON FAMILY

Louis Libin
NBC
New York, NY

INTRODUCTION

Broadcasting services are the most important mode of information delivery in the world today, outnumbering computers, giving up-to-date information vital to many activities of the public and providing coverage by a unique mix of terrestrial and satellite that leaves few corners of the globe without coverage. The television services are of particular importance. There are some one billion television receivers in the world today and in many developing countries, the number of television receivers is several times the number of telephone sets, making television in these areas of even greater importance for the delivery of information.

Broadcasters have always shown great initiative in adopting new technologies and have led the way in developing digitally based methods for the production, distribution and transmission of radio and TV services. The powerful processors that are in everyday use for picture manipulation and video servers cause many computer systems to appear lethargic in comparison. The introduction of digital broadcasting to the home will create an immense information network that will dwarf the best efforts of telecommunications in terms of locations served and capacity. Broadcasters already have the capability and depth of resources to produce content for the Information Highway and have an essential role to play in its implementation. The next step is to ensure that the protocols and standards for broadcast delivery are compatible with the open, interoperable concepts of the Information Highway. The major element currently missing is an appropriate mechanism to achieve the level of interactivity needed for broadcasting services.

Interactive TV could be of value not only to broadcasters (in providing instant audience reaction, for example), but also to operators of other services outside broadcasting which could have important commercial or societal implications. Broadcasters and program providers should see interactive television as a challenge and opportunity. The challenge will be to create program material in an imaginative way for an interactive environment. The opportunity will be that of sharing in this important new service.

The production of programming that makes use of interactivity is somewhat more complex than for conventional non-interactive material, and some additional costs will be incurred here, but are not considered to be a significant burden, in comparison with other major production elements.

This paper presents the requirements and some possible solutions.

THE INTERACTION CHANNEL

The basic requirement of a worldwide common family of interactive services calls for a definition of the basic interaction channel.

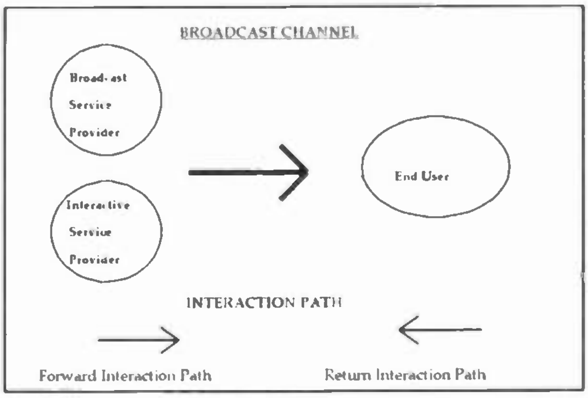
The first requirement of an interaction channel is that the user be able to respond in some way to the interactive service. The response may take the form of "voting" for a particular participant in a show, "purchasing" goods demonstrated/advertised in a shopping channel program, etc. This would be achievable within a 1-way (reverse direction) narrow-band path.

A higher level of interactivity might require that a consumer, who has made a response to an interactive service, be sent an acknowledgement. This might be the case where the consumer has made a credit card purchase from a shopping channel via the basic interaction channel. That consumer would expect to receive an acknowledgement that his credit card transaction had been accepted. This level of interactivity would require a two-way interaction channel, one in the reverse direction, the other the forward direction.

A further level of interactivity would occur where in response to information in the interactive service, the consumer requests further information on particular topics from the source of the service, or from a central database via the source of the interactive service. This would require that the forward channel be broadband. In this particular example, the reverse path would only need to be a narrow-band one, but it is likely that applications will arise whereby the consumer will need to make a broadband response/contribution to the interactive service and also receive a broadband "answer" from the service source.

Reference Model for Common Interaction Channel

A reference model for system architecture of narrowband interaction channel in a broadcasting scenario (interactive services) is defined. In this reference model, two channels are established between the Service Provider and the User.



Broadcast Channel: a unidirectional broadband point-to-multipoint channel which may include video, audio and data established from the service provider to the users. It may include the forward interaction path.

The Interaction Channel is generally established between the service provider and the user for interaction purposes. It is formed by the Return Interaction Path and the Forward Interaction Path.

The Return Interaction Path (return Channel) establishes a communication channel from the User to the Service Provider.

The Forward Interaction Path establishes a communications channel from the service provider to the User. This channel may not be present in all implementations.

THE REQUIREMENT FOR INTERACTIVITY IN BROADCASTING

Interactivity may mean different things. To no two people does interactive television mean the same thing. To no two people does "interactivity" mean the same thing. Very few in the media, when discussing "interactive television" understand the difference between computers and television. These are different media; they are merging, but they are different in one important aspect. Computers are non real-time machines and television sets (broadcasting and cable) are real-time devices. One can interact with any computer. In fact, a computer requires that interaction, but at one's own time. On the other hand, a television receiver does not require much interaction, and the consumer is at somewhat the mercy of the broadcaster or cable operator for the correct path to follow.

Interactive broadcast television is at the junction of analog and more importantly, digital, enabling technology and consumer familiarity and appreciation for more personal services.

The following are examples of Program Related Interactivity:

Program Guide
Description of the programs available from a certain program provider. The scheduled programs may be customized to a viewers need.

Subtitling
Subtitles to be inlaid in the image at the receiving system.

Program Summary
A summary of the main program. The summary is updated during the program.

Home Shopping
Interaction from the viewer with a mainstream program. The viewer could send a message to the central office to buy the merchandise.

Audience Polling
Viewers vote for the program outcome or answer questions to obtain a polling result.

Gambling or Gaming
Placing bets on races and games

Auctions
Placing bids on goods and services

Multi-choice Pgm
Choice of one program from a multi-channel program (ie. camera positions in sporting event)

Additional Info
Viewer controlled add-on data or information related to program.

Near VOD
Staggered video transmission on different channels for near instantaneous response.

All the program related interactivity uses listed above have the advantage of their relation to the program material - very little data needs to be "wasted" on introduction. Other than relation to program, these uses are, by their nature, extremely diverse and are only the beginning of a dynamic user application requirement list. The following chart is the beginning of a list that will define the data rate required for various uses.

Interactive Services/Data Rate

	L=Low M=Medium H=High
Video On Demand + (Near Video On Demand)	L
Videoconferencing + (Audioconferencing)	H
Home Shopping	M
Home Banking	M
Games	M
Gambling	M
Distance Learning	M
Data Downloading (See Appendix A)	H
Program Guide	L
Subtitling	L
Program Summary + (Additional Information)	L
Auctioning	M
Audience Polling	L
Multi-Choice Program	L
Share Dealing/Stock	M

INTERACTIVE ALTERNATIVES

The return path necessary for real broadcast television interactivity, can be effected in a number of ways, including radio links, either terrestrial, satellite or hybrid, telecommunications links using either wired or radio connections or over cable distribution systems through the direct implementation of return channels. The return channel is an integral part of the broadcast interactive television system. This is the portion that will allow the home viewer to respond and interact with the transmitted information.

In most cases, the return path will be operated on a "multi-point to point" basis, linking many users of the broadcast service back to a central point. In addition, some local interactivity will also be present at each user site. The orderly use of the return path in a multiplex will be controlled by command information sent along a forward data path associated with the program.

Multiplexing techniques, such as Time-Division Multiple Access (TDMA) are well developed in the field of cellular radio, as are the techniques of spread-spectrum transmission, such as Code-Division Multiple access (CDMA) used in the emerging PCS environment. Using these techniques, a number of users can share a common return path without interference.

In the consideration of the return path for interactive uses, a number of factors need to be considered, such as the amount of data to be transmitted, the duty cycle of the data "burstiness", and especially the allowable latency (the apparent delay time for transmission) and the set-up time (the time needed to establish the connection from the user to the multiplexer, a significant period of time in switched telecommunications networks). It should be noted that the latency may be a variable, dependent on the number of interactivity requests currently being processed.

For radio-based systems, a number of areas of the spectrum can be considered, including the existing TV bands, compatible use of the current UHF allotments for cellular radio or spectrum allotted for personal communications in the 1.5 to 2.0 GHz region. In the USA, a band at 218-219 MHz has been allotted specifically for broadcast TV interactive links. The return links will be very low power and likely would use spread spectrum techniques to reduce further any problems of interference.

The integration of the interactive return path into developing systems for other telecommunications applications has merit in terms of a sharing of the costs of development, implementation, operations and usage, but may impose operational and access constraints that are undesirable for this purpose, such as excessive overhead or delay.

Bandwidth

In many cases, bandwidth will depend not only on data rate, but also on the specific choice of transport mechanism and frequency band. In any over-the-air applications, the system parameters will most likely follow the bandwidth parameters of the systems already in mass production. For example, where it is proposed to use the existing cellular mobile network, some of the main parameters of that mobile system may well be appropriate to the interaction channel.

Modulation

The modulation and coding will depend mainly on the system used for the interaction channel, and should be selected, if possible from existing methods so as to utilize mass produced components, particularly where more expensive solutions are proposed of the interaction channel

Structure of the Interactive Service

To ensure openness of access for users and services and of maximum commonality among interactive services and customer premises equipment, the protocols and interfaces used in the interaction path should be common, open and standardized. Similarly, the communications should be arranged in layers, independent of each other to the extent possible and enabling of future development (extensible).

Transport Mechanisms

The interaction channel is an integral part of the interactive television system. This is the portion that will allow the home viewer to respond and interact with the information contained in the service. In principal, several possibilities exist for the interaction channel.

In terrestrial over-the-air applications, it may be possible to use the existing receiving antenna and cable. It may also be possible that a special transmitter/receiver unit could be incorporated into the television set. It is also possible that a simple transmitter and antenna, similar to those used for cellular mobile systems, may be a more cost-effective solution. Also, VSAT satellite-based systems could be attractive solutions, especially for users connected to SMDTV networks.

Locally Interactive Systems

Locally interactive systems operate without a return path to the program source by selecting among a number of redundant transmissions, either in a single channel or on two or more channels, to effect some degree of interactivity. For some learning applications, this technique can be highly effective and can be augmented with PSTN techniques. One example of such a system is shown below.

Options for the Home

A number of possibilities currently exist and others are in various stages of development and implementation. The choice for any particular application or situation may involve one, or more, of them and no single one is likely to meet all the possible needs. They include:

- Dedicated, low-power digital transmitters (possibly also including a similar receiver) included in the TV receiver, or in a set-top unit, communicating to cell receivers serving small geographical areas, such as a community or village. Techniques used for digital broadcasting, such as COFDM, may be appropriate and operation within unused TV spectrum space is likely to be feasible without interference, as in the U.S. 218-219 MHz allotment for interactive TV-related services. The cost of such an approach is a concern, given its current lack of synergism with other services and potential loss of economies of scale. However, in view of the enormous installed base of TV receivers in the world, this may not be a serious concern, if interactivity is widely implemented. Studies continue.

- The Public Switched Telecommunications Network (PSTN), either in a wired implementation or in a wireless form such as Personal Communication Systems (PCS), Local Loop Bypass (LLB), Cellular radio, radio trunks or the future FLMPTS network. Such approaches are attractive in that they use the PSTN infrastructure synergistically and thus offer good opportunities for developing countries. Serious attention would be needed in regard to the set-up and administration aspects of the network, in view of the intermittent, short-burst nature of much of the TV-related traffic.

- Satellite resources, such as Low Earth Orbiting Satellites (LEOS), Mobile Satellites (MAST), or VSAT technologies. The satellite approach is attractive, giving excellent coverage of wide geographical areas with minimum infrastructure. Some are already in use for commercial applications of a distributed nature, generally mobile. The cost and appropriateness for this application require further consideration.

Another important consideration is the level of service required for the specific interactive application. Interactivity may exist at a number of levels , from full interactivity with symmetrical channels in the two directions (games for example), through limited interactivity in which the return channel is very much smaller in capacity compared to the forward channel, to local interactivity, in which the interaction is only time. Games need response times of a few tens of milliseconds for proper interaction while transactional interaction, shopping or banking for instance, can accept slower response time, but demands higher security. It is thus essential that the needs for interactivity be matched with the capability of the return channel in performance and cost.

FURTHER DEVELOPMENTS OF BROADCAST INTERACTIVITY

Study Group 11, ITU Working Party 11C prepared a draft new question on Interactive TV Broadcasting": Doc. 11/216 Rev.2. Broadcasting systems will probably also be used for other purposes, so it seems to be of great advantage to incorporate the broadcasting receiver into interactive systems. For such purposes, the return channel is an important part of the system;

good isolation would be required between the received and transmitted signal and, therefore, good frequency separation would be needed.

With the development of such interactive systems on one side, and modern communication technologies on the other (including cellular digital radio communication, low earth orbit satellites (LEOS) links, etc.) a common technical approach should be sought.

THE FUTURE OF BROADCAST INTERACTIVITY

It is necessary to continue developing a better understanding of the technologies that can deliver Broadcast Interactive Services to and from the home, and how they can best be applied to a growing range of viewer needs. This learning should not focus strictly on technical issues; but should address the broad range of human factors that can help foresee how people will use this new technology in their day-to-day lives.

The work of Study Group 11 in advancing the development of this technology has, thus far, been based on the needs of entertainment and business applications but it is now clear that some of the most valuable benefits of it lie in the sphere of Distance Education.

CONCLUSIONS

It seems to be of great advantage to incorporate the broadcasting receiver into interactive systems. For such purposes, the return channel is an important part of the system; good isolation would be required between the received and transmitted signal and, therefore, good frequency separation would be needed.

The global development of interactivity for television and the impacts that it could have on the development of the wider telecommunications infrastructure, is clearly a matter for the ITU in all its Sectors and can bring benefits to all its members globally, but particularly in developing countries. The current activities need to be augmented and accelerated to ensure that developments and innovation worldwide will meet the needs of all countries, through studies, recommendations, conferences, and a clear program of consultation among technologies, broadcasters, and the full range of affected users. This process would benefit considerably from a series of demonstration and pilot projects in those areas where the new strategies and approaches are considered necessary, thus ensuring that the actual and future needs of the developing countries are well identified and tested before the creation of recommendations that may be based on the needs of others.

With the development of such interactive systems on one side, and modern communication technologies on the other (including cellular digital radio communication, low earth orbit satellites [LEOS] links, etc.) a common technical approach should be sought. Broadcast Interactive Television holds out to our society the promise of tremendous personal freedom and control over our communications and entertainment.

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RADIO FACILITIES DESIGN: RADIO'S MISSING DIGITAL LINK

Tuesday, April 16, 1996 1:00 - 5:00 pm

Session Chairperson:
Jerry Whitaker, Technical Writer, Beaverton, OR

***DEDICATED VERSUS GENERAL PURPOSE HARDWARE, THE DEBATE CONTINUES**
Steve Epstein
Broadcast Engineering Magazine
Overland Park, KS

REPLACING AND ENHANCING THE PUBLIC RADIO SATELLITE SYSTEM: WHAT WE LEARNED
Gregory P. Monti
ABC Radio Networks
New York, NY

***ISDN FOR BROADCAST: THE REAL WORLD AND WHAT WE'VE LEARNED ABOUT IT**
Steve Church
Telos Systems
Cleveland, OH

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Greater Media, Inc. WPEN/WMGK Radio
Philadelphia, PA

THE NEXT GENERATION RADIO STATION ELECTRONIC COMMUNICATION SYSTEM
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T1 DIGITAL STL: DISCRETE VS. COMPOSITE TRANSMISSION
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Intraplex, Inc.
Westford, MA

INTERCONNECTING THE DIGITAL CHAIN
Jim Hauptstueck
Harris Broadcast Equipment
Richmond, IN

*Paper not available at the time of publication.

REPLACING AND ENHANCING THE PUBLIC RADIO SATELLITE SYSTEM: WHAT WE LEARNED

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ABSTRACT

The Public Radio Satellite System, the audio network that serves noncommercial radio stations in the United States, and the producers and distributors who sell programs to those stations, was put on the air in 1979. As its tenth year of service approached, National Public Radio's Distribution Division was faced with system obsolescence and with a need to upgrade channel capacity, to improve data bandwidth and to take advantage of the personal computer and reduced-bit-rate audio "revolutions" to construct a replacement system. There are lessons here for anyone managing large projects in political engineering organizations.

WHAT NPR DOES

National Public Radio is a private, non-profit corporation invented by public radio stations in 1970 to act as an aggregator of funds and as a provider of services. Besides being a program production house and a trade association, NPR operates a satellite audio network called the Public Radio Satellite System (PRSS). The PRSS serves NPR's own programming divisions and other producers and distributors of audio programming as well as serving 415 downlink radio stations and 23 stations with uplinks. The PRSS handles about 30,000 individual program audio transmissions per year, ranging in length from five-minute newscasts to five-hour operas to 24-hour format services.

THE 1979 SATELLITE NETWORK

Design efforts for the original PRSS network began in the mid-seventies. NPR was the first U.S. radio network to convert from leased landlines to satellite transmission, with the audio channels beginning service in Summer, 1979.

Audio was sent on twelve analog FM satellite channels in a single-channel-per-carrier format. Analog noise reduction used 3:1 companding from dbx. Each satellite channel was mono, so two channels were required for stereo feeds. The single channel per carrier mode allowed any channel to originate at any of 17 (now 23) uplink sites, including the master site at Washington, DC. Receivers made by Coastcom, Tectan, Microdyne, McMartin, Harris, Scientific-Atlanta and Micro Phase were compatible with the system and were widely used by stations. This system was turned off on January 3, 1996.

Data support for printable messages, program schedule and real-time numeric cues was sent over a 1,200 bit per second audio FSK channel riding on a separate, uncompanded FM channel. The computer that received and sorted this data and provided printer and other control outputs was designed prior to the personal computer era. The computer had 28 kilobytes of volatile RAM, no keyboard, no mouse, no monitor and no disk drive.

WHY REBUILD NOW?

Our spacecraft, Westar 4, reached its scheduled end of life in 1991 and long-term replacement was required. On the ground, there were about 3,000 analog audio receivers in use. At the time, some were approaching 10 years old. The specialized computers and interfaces could no longer be produced economically for new downlinks joining the system. 1980-era integrated circuits were becoming unavailable.

A grant of $49 million was made available by the U.S. Congress through the Corporation for Public Broadcasting. This grant was approved prior to, and was not affected by, subsequent threats of public broadcasting cutbacks.

NEW DIGITAL AUDIO CHANNELS

We originally considered replacing the analog system with another analog system, but expanding it to allow for up to 24 programs to be sent at once. Our space segment deal gave us the room to do that with two full C-band transponders on Galaxy 4. Some space segment was also available for lease to outside parties to help defray the costs of operating the noncommercial system.

Compressed bit rate technology

The appearance of digital signal processor chips and bit rate reduction algorithms in 1989 and 1990 encouraged us to avoid another analog system. But, to be acceptable, digital had to sound better than the companded analog.

We received bids for a digital audio transmission system from Dolby, California Microwave, ComStream, Micro Phase, RE America, Scientific Atlanta, and Wegener. Each vendor was allowed to propose its own audio bit rate reduction algorithm. Listening tests proved that all of them sounded better than the existing analog system.

Our choice of ISO/MPEG Layer 2 (actually MUSICAM) rate reduction was based on our perception that it was becoming a world standard. By mid-1992, many terrestrial digital audio broadcast systems had been proposed which would use it. We attempted to avoid one step of transcoding at each radio station by sending the satellite audio in the same, or in a similar, format.

What we bought

ComStream proposed, and NPR accepted, a three-part configuration for the modulator (SCPC digital exciter). They paired their standard CM701 variable rate modem, configured for transmit-only, with their DAC700 MUSICAM encoder-decoder box. The audio coding technology was licensed from MUSICAM USA (formerly Corporate Computer Systems). We required that a special control and monitor communications port using RS-485 electrical standards be added to the modulator so that it could be intermixed on a computer communications bus with boxes made by other vendors. ComStream proposed a separate NPR Control Module which would reside in one of the option slots of the modem. It would have a small processor to communicate with the RS-485 bus and would store and forward certain configuration and addressing information necessary for operation with the custom receivers that would complete the link.

Lesson learned: Allow extra time, extra antacids and extra headache medicine for the data protocol negotiation sessions between software developers who work for two different companies, each of whom does things their own way.

Demodulator input frequency
At the time we signed our deal in late 1993, ComStream had already had a broadcast audio receiver on the market for two years. It used an L-band frequency input intended for use with a low-noise block converter (LNB) at the antenna. This has been a common design among both professional and consumer receive-only equipment for several years and is less expensive than using an intermediate frequency near 70 MHz. However, in order to pre-configure our stations' antenna sites in 1990 and 1991, we had to make the 70 MHz (52 to 88 MHz) versus L-band (950 to 1,450 MHz) choice then. At the time, networks using the Scientific-Atlanta's DATS system were retaining their 70 MHz I.F. while upgrading the channel-end receivers to reduced-bit-rate coding. High stability "digital ready" LNBs were not yet available. Several of our downlinks had I.F. cable links several hundred or several thousand feet long and 1,500 MHz cable amplifiers were not available in 1991 (and may not be available now). In addition, most of our station downlinks wanted to keep in service some analog demods using the 70 MHz frequency.

We selected the tried-and-true 70 MHz I.F. system, even if it would be more expensive in the long run. This is always a problem with advancing technology. At what point do you yell, "Stop!," and accept the technology that is available at the time?

Custom receiver
ComStream's existing receiver also had no front panel controls and only a few basic indicators. The design intent was to either change receiver settings through an in-band control channel over the satellite from the network headquarters or to have the downlink user plug in a dumb terminal or a personal computer running communications software to read status or to make adjustments. While the trend among broadcast equipment manufacturers is toward moving the front panel off onto a user's computer, we had to have a front panel to allow station user control over the channel received.

Audio programs would be transmitted on the PRSS at MUSICAM rates of 256,000 bits per second for discrete stereo and 128,000 bits per second for mono. These provide near-contribution-quality links that can be used both for inbound program-assembly feeds and for outbound network distribution. An auto-bauding feature in the receiver automatically discovers and adjusts to the incoming channel data rate each time the unit is retuned. The feature also supports 64,000 and 192,000 kb/sec rates in case of future market demand for those rates.

A separate, data-only mode switches the MUSICAM decoder out of the circuit to allow reception of a 64 kb/sec broadcast data channel for Satellite Operating Support System, described below. This is a substantial improvement over the previous 1.2 kb/sec data channel. It also allows the same demod to be used for audio or data in case of individual unit failures.

The PRSS must serve two groups of customers, stations and producers of programming. Producers had asked for several years for a way to know or control which stations received and used their audio programs. We had our vendor implement a conditional access feature in each receiver so that program-by-program, station-by-station subscriptions could be offered. A second mode allows all receivers we know about to receive all audio programs. In both modes, audio remains unavailable to the public at large and unavailable to stations we don't know about or who don't have the proprietary receiver.

Each receiver is equipped with analog left and right audio outputs, and with an AES-EBU digital output and with an output in the MUSICAM reduced-rate format. All outputs are available simultaneously.

Lessons we learned: We didn't specify a specific temperature rise, a limit on power consumption or a limit on mechanical noise output for each receiver. Audio and R.F. specifications were paramount. This resulted in a one-inch-diameter fan through a gridded cutout on the back panel to cool each one-unit-tall receiver. At many stations, who had set up a separate satellite recording room without any microphones years ago, this was not a problem. At others, demods were to be located in the air control or production room, in some cases within inches of a microphone. Stations had to alter their plans once the receivers arrived.

Backlight lesson: We selected a two-line liquid crystal character display for the front panel while viewing sample displays in a brightly-lit laboratory. We picked the one with the most contrast. This display did not have a backlight feature. To meet the Part 15 emitted R.F. requirements, each receiver had to have a conductive plastic window over the display, which reduced visibility further. The display proved difficult to read in some stations' studio lighting schemes.

RDAT remote control lesson: Analog recorders will continue to roll, and will stay in the record mode, without a usable input signal. RDAT recorders, when using their AES-EBU input ports, may or may not stay in record mode without AES-EBU sync. Some RDAT recorders will drop out of record mode if the AES-EBU signal ceases and will not return to recording until a human operator restarts them. In some cases, stations have had to program the automation software to produce continuous play and record contact closures for the entire duration of the feed to assure that a dropout of AES-EBU signal during uplink site switching or during receiver retuning will not cause the recorder to stop.

OTHER R.F. COMPONENTS

Uplinks

To comply with the "two-degree spacing" requirement of section 25.209 for C-band satellites, our older uplink antennas were retrofitted with an Andrew design that re-used the existing foundations and tripod mounts but added a two-degree-compliant reflector and feed element.

Upconverters: Since each uplink site could be sending a mix of channels on both transponders simultaneously, each uplink required upconverters that could translate separate 70 MHz bands to two different 6 GHz C-band transponders simultaneously. A design, suggested by LNR Communications, grouped some common components and some frequency-specific components into a single upconverter chassis. The result was a one-rack-unit-high converter with two 70 MHz inputs. Each input would be converted to its own transponder frequency near 6 GHz and both 6 GHz bands would appear combined at a single output port. The cost savings over purchasing separate upconverters for 21 uplink sites was substantial. Instead of four $11,000 upconverters per site (including backup units), we bought two $14,000 dual upconverters, a savings of over $300,000.

Power amplifiers: Existing Varian Associates 75-watt and 125-watt uplink power amplifiers, some now 16 years old, are still performing well. A slow capital-replacement program was already underway so we did not replace them as part of this project.

Downlinks

The C-band receive only antennas in the system vary in age from 1 to 16 years. Some are aluminum and some are fiberglass laminate. With the price and diameters of antennas falling, and with several years of useful life remaining, these units were kept in service and not replaced, with only a few exceptions.

To fix signal margin problems at two sites in Alaska, we upgraded 5-meter aluminum or fiberglass antennas to 9.3 meters. While this was expensive, it was important to maintain long-term reliability at two towns with low look angles through hostile weather, Kotzebue (on the west coast of Alaska near Russia) and at Barrow (on the north coast).

Downconverters: The new system would occupy two C-band transponders and each downlink would need to receive both transponders simultaneously. Once the 70 MHz decision was made, Scientific-Atlanta did a custom manufacturing run of 500 C-band to 70 MHz downconverters for us to downconvert the second transponder. These converters had a port which tapped an intermediate stage where the downconverted C-band transponder was near 250 MHz.

Intrafacility links: At about half the stations, the C-band 4 GHz signal runs directly into the studio rack where the demods would be located. In those cases, the second downconverter and its I.F. output wiring were simple to add. At some stations, the distance from the antenna to the demods was too long for an economical 4 GHz run, so the second downconverter was located, with the first one, in a rain-proof shelter under the antenna. The second 70 MHz spectrum was carried on a second cable to the demodulators.

Where adding a second cable was not feasible (about 40 sites), the signal from the special 250 MHz tap on the new downconverters was sent through the same cable by passively combining it with the 70 MHz spectrum from the first downconverter. Then, a custom downconverter from Watkins-Johnson was installed to demultiplex the two spectra and to shift the 250 MHz down to 70 MHz to drive the receivers.

At a handful of sites, the radio station studio was located at a place where the satellite was either not visible from any land or buildings the station controlled, or was swamped with C-band interference — and no suitable cable link route was available from an alternative site nearby. At these sites, audio is demodulated at remotely-controlled satellite receivers at a distant downlink dish site and is re-encoded and transported to the studio using leased-line or microwave T-1 digital links. At the studio end, the audio is decoded to analog one last time for use. This required that the MUSICAM audio be decoded to analog at the dish site and immediately re-encoded into MUSICAM again for terrestrial transport.

Lesson learned: Any digital audio from satellite demods which are receiving data from multiple, unsynchronized uplink sites cannot be directly sent through a synchronous transmission medium like T-1. The bit rates from the individual sources will vary. Even if the variations are only a few parts per million, one or more of the sources will eventually run too fast or too slow compared to another. Data frames, which may or may not relate to MUSICAM frames, will be dropped or omitted.

We explored several options which would have required custom design of equipment that could multiplex audio data from asynchronous sources, pass it through a synchronous data link, and reconstruct a gap-free encoded audio signal with the same part-per-million error as it had at the source. An additional problem was that our satellite channels could change at random times between the 64, 128, 192 and 256 kb/sec data rates. Any synchronizing device developed would also need to handle those variations. NPR could not accept the risk of such a custom development project for two uplinks and two downlinks, so we fell back on the "Universal Synchronizer"™— an intermediate step of analog audio.

Lesson not learned: Most radio station owners, when picking a studio site, will check for good roads and parking, good STL line of sight to the transmitter site, good power and good telco service. The factor missing from consideration may be line of sight to your network's satellite.

SATELLITE OPERATING SUPPORT SYSTEM

IBM was brought in as prime contractor on our custom software and integration project known as the Satellite Operating Support System (SOSS). Subcontractors who specialize in broadcast process control, scheduling and billing, and multi-user messaging systems were hired to do the development work.

Processors and coprocessors

The final System Technical Center (satellite master control hub) will use reduced instruction set computing (RISC) processors running AIX for the multi-user database and messaging servers. AIX is the IBM version of UNIX. It was chosen to handle the large number of multiple, simultaneous in-house, dial-in and internet users of the messaging system.

User workstations and real-time automation processors at the System Technical Center and at the uplink and downlink radio stations are running the OS/2 operating system. Workstations at the satellite time booking center will also use OS/2. It provides multi-threaded processing and pre-emptive tasking to prevent any one application from crashing the system. In addition, those PCs controlling satellite or broadcast equipment contain a serial port coprocessor (ARTIC) card so that real-time commands are sent out instantaneously regardless of what the main processor and disk drive are doing.

Serial control bus

Broadcast and satellite equipment are controlled by the each site's PC using a four-wire RS-485 serial control bus. Custom-made general purpose interfaces (GPIs) and network message alert boxes interoperate on the same bus as the ComStream modulators and demodulators.

The general purpose interfaces allow user programming of four momentary or maintained contact closures per box for control of tape recorders, alarms, and other devices. There are also four user-programmable, optically-isolated inputs per box. Programming of inputs and outputs is done with simple point-and-click editing in on-screen dialog boxes.

Two I.F. signals
Since two 70 MHz lines are entering each station's demod rack from the downconverters, and since any program transmission could be on any channel on either transponder, each demod must be independently switchable between transponders, both manually and under automation. For each group of four demods, a custom-designed bank of four A/B switches provides the routing of I.F. signals. The I.F. signal path is passive, but each box is actually a two-input, four-output matrix switcher for 70 MHz signals.

Schedule and messages
For uplink and downlink stations, the program schedule is delivered as a database. Database records are repeated periodically in case they are missed. Stations then use a custom-developed editor to electronically mark which transmissions they want to use and to assign demods and tape recorders (or equivalent storage devices) to them. Additional, local events can also be added to the schedule which allows each station to make the PC into a basic automation system which can be used for play-out to local air as well as for network recordings.

Our text messaging system, called DACS, allows producers and stations, as well as NPR staff to send addressable messages to all stations, to groups of stations or to individual stations. The typical load is currently about 200 messages per day, growing at about 20% per year. Message content includes a text copy of the program schedule, schedule change announcements, program content listings (including detailed timings and audio cues), promotional items, fundraising items, etc. A dial-in interface allows outside parties to upload messages to our server for broadcast and also allows stations with downlink failures or personnel not located at stations to retrieve the day's messages.

The DACS system also supports text or binary file attachments. This allows software upgrades, graphics files, executables, spreadsheets and word processing documents to be sent by NPR or by independent producers to stations or from stations to each other.

Since desktop computers have notoriously inaccurate clocks that drift several seconds per hour, the satellite data channel also provides a once-per-minute time synchronization signal which sets the PC at each uplink or downlink to match the Eastern Time being used by the Technical Center's automation system. This keeps the local PC clocks within one OS/2 time slice, or 32 milliseconds, of Eastern Time. Of course, the time signal actually arrives at each downlink delayed by the much larger 260-millisecond transit time of the satellite hop. But the audio program undergoes the same delay. Time can be displayed at each radio station in local time or in Eastern Time at the station's option.

Computer lessons learned
Assume lemon equipment: When one ships out 440 personal computers to radio stations, there are bound to be 10 to 15 lemons in the batch. After replacement of many modules and boards, they can be fixed eventually, but not in a week or three.

Assume little or no computer knowledge: Our Technical Support Desk, with three full-time employees, has received over 10,000 calls for help during the first three months after the computers shipped. Some stations have called dozens of times per station with questions like:

- I don't have time to read the manual. Tell me what I really need to know.
- Question: How do I delete a file in OS/2? Answer: Open an OS/2 window and do it the same way you do it in DOS. Question: How do you delete a file in DOS?
- What's a backslash?
- (From 3,000 miles away) I'm holding two cables in my hand. I made them up today. One works. The other doesn't. Why is that?

Specify everything in the contract: During writing of Statements of Work for many custom contracts, we often wrote detailed requirements of how functions and features would work. During the editorial process, many of those specifics were removed as being "unnecessary details" which the contractor would not need to know to quote the job. Today, those details are reinserted into the Statement of Work and they have a new name: "change orders."

Magnitude of change orders: In a custom, software-hardware integration project, assume that change orders will be between 100% and 150% of the original, fixed-price contract cost. This is in addition to the original contract, so the total will be 200% to 250% of the fixed-price quote.

The lowest bidder is not always the least expensive alternative: It may fit the capital budget for the project, but what are the support costs and operational inconveniences when the manufacturer doesn't stock adequate parts, modules and whole spare units?

Have a beta test period: In order to meet a January, 1996, cutover date to the new system, we did not undergo a formal beta testing period for the custom software systems with actual end users. Conducting a beta test period would have spawned a long list of changes which we could not afford and would have had to selectively kill. It also would have busted the schedule. With hindsight, we should have allocated money and time and done beta testing anyway.

Overcome the fear of a camel: Many NPR managers feared that, if too many parties were involved in the design or specification of a hardware item or a software application, that it would end up being "a horse designed by a committee." This is certainly a reasonable fear. We did invite end-user radio station personnel to comment and to make suggestions on functions and features we should implement, but those consultations used the "unaided recall" method, i.e., stations could make their opinions known to us on a blank slate, but could not react to designs or draft requirements under consideration. We should have overcome that fear of designing a camel and should have invited at least a small sample of station chief engineers and operations managers to review the design progress. Many of the problems now identified could have been caught and fixed before delivery to all 440 sites.

CONCLUSION

Except for some of the automation system, booking system and message server functions at the System Technical Center in Washington, the new Public Radio Satellite System is delivered to the end-user downlink stations and is now on the air. The process took far longer than anticipated. Back in 1988, we thought we'd be done by the end of 1993. The next time the system is rebuilt, there will be new funding challenges, new business requirements, new technologies, and new lessons to be learned.

PRACTICAL CONSIDERATIONS OF USING ISDN AUDIO CODECS AS BACKUP STUDIO-TRANSMITTER LINKS

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ABSTRACT

Integrated ISDN audio codecs designed primarily for facilitating remote broadcasts using inexpensive phone company switched digital services have been commonly available since the autumn of 1994. Now as ISDN deployment by telephone companies becomes more universal, an increasing number of radio stations may find themselves in a position to utilize these codecs for an entirely different service: that of a backup audio studio-transmitter link. While the fiscal benefits of such an implementation can be significant, successful installation and maintenance of such a backup system will require the careful consideration of a number of factors other than the merely economic.

This paper undertakes to discuss the advantages and pitfalls of using ISDN audio codec devices in the novel role of emergency studio-transmitter link, provides brief descriptions of the operational characteristics of the component devices, shows a payback analysis of a simple ISDN system compared to microwave and leased equalized line links, describes additional economic vs. reliability tradeoffs that may be considered in certain situations, and offers a starting point for broadcast engineers planning the installation of such a backup STL system.

INTRODUCTION

Many radio stations routinely operate with some sort of STL (most often consisting of dedicated microwave transmitters, receivers and associated antennae and transmission lines) to convey their program audio from their studio site to their remotely-located transmitter site. To achieve a high degree of critical path redundancy, some stations maintain backup microwave system components and/or make arrangements with the local telephone company to lease full-time equalized program lines connecting studio to transmitter. Other stations, weighing the significant costs of redundant program link facilities against their admitted infrequency of use, elect to forgo backup components and leased telephone lines entirely and rely solely on their primary STL systems.

But now ISDN audio codecs can offer an alternative that has never existed before for the alternate program line quandary – since these codecs are capable of providing high-quality bi-directional stereo audio between wired locations at a moment's notice using telco facilities which cost (in most cases) a fraction of the expense of equalized lines, it is reasonable to consider their use as backup STL systems.

The economic benefits of such a system are indeed impressive: stations which currently lease program lines may find that the savings in monthly phone line charges (for ISDN vs. equalized lines) pays for the cost of acquiring the codecs in two years or even less. Other stations unable to justify backup program line costs in the past may now for the first time have access to an economical alternative. And stations already using ISDN codecs for remote broadcasts are in a particularly good position to make use of this technology.

Successful installation of such an ISDN codec system, however, requires careful consideration of a number of factors other than the merely economic — proper audio and control interfacing, the potential for audio degradation caused by multiple passes through perceptual coding algorithms, initial and continued ISDN circuit reliability, and automatic periodic testing of the system — to name but a few.

BACKGROUND

At this point, some cursory descriptions of the technologies involved may be helpful to the reader.

ISDN stands for Integrated Services Digital Network, and actually describes a family of switched data services offered by the phone company. In its simplest form, called the ISDN Basic Rate Interface or BRI, the phone company brings to your premises a single pair of wires which carries two independent, bi-directional 64 kbps data paths. With the appropriate dialing and interface equipment (which you supply) you can make and receive calls to other ISDN subscribers on either or both of these data paths.

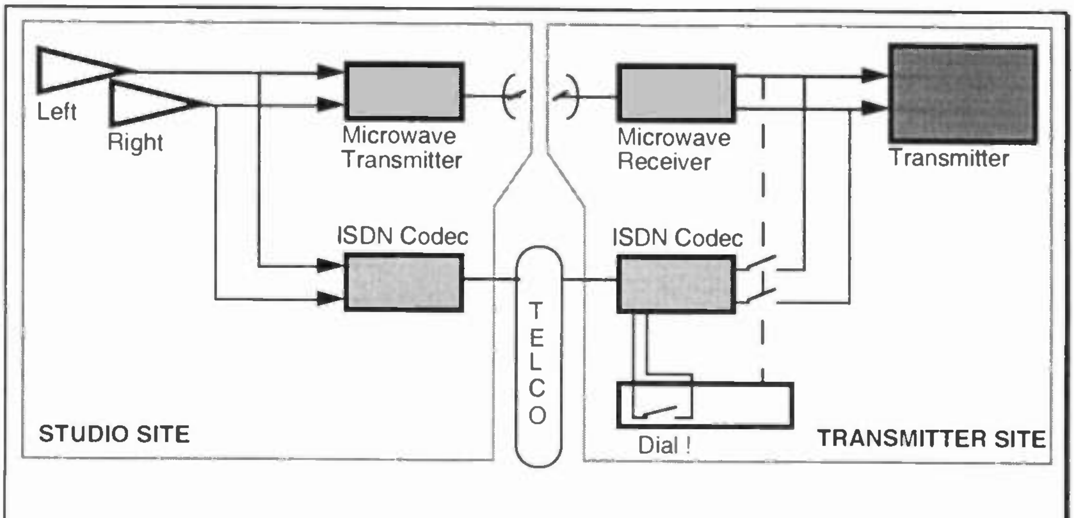

Figure 1. Simple Backup ISDN Codec STL Block Diagram

Generic dialing and interfacing equipment has existed for several years to service the more conventional data transmission needs of banks, department stores, financial markets and other pioneer early users of ISDN. But it is only comparatively recently, as ISDN service has become more generally available, that several companies have developed and now market broadcaster-specific ISDN dialing and interface equipment. Known colloquially as "codecs", these all-in-one devices are designed specifically to accept stereo or two-channel broadcast quality audio, digitize and process it so that it can be sent over the ISDN line's 64 kbps data paths, and dial a companion device at the other end of the link.

The companion codec then automatically answers an incoming ISDN call, receives the data stream from the distant codec and reassembles that data into broadcast quality audio. And since the ISDN data stream is bi-directional, these codecs are actually designed for two-way operation, so that stereo broadcast audio can be sent in *both directions* simultaneously.

HOW WOULD IT WORK?

The simplest and most basic layout of ISDN codecs configured for backup STL use would have one codec permanently installed at the studio site, connected to a dedicated ISDN line. A second, identical codec would be permanently installed at the transmitter site, connected too to its own dedicated ISDN line. See Figure 1.

Because ISDN is a switched service, and most customers are billed for each minute whenever a connection exists, the typical backup STL ISDN installation would have the system idle until acually needed to avoid additional charges. This implies a need to quickly connect whenever the system is to be tested or is genuinely needed.

ISDN codecs have the ability to store call setup information in their internal memories – the equivalent of speed dial numbers. Most codecs intended for broadcast use also provide as standard equipment (or at least as a nominal cost option) a number of outside world status inputs and end-to-end contact closures.

Using these features, the transmitter site codec can be programmed to automatically dial the studio site codec whenever a simple contact closure is made, and to end that call when the closure is removed. The contact closure could be asserted by a silence sense, by the microwave STL receiver squelch contacts, or by the station's intelligent transmitter remote control system. The studio site codec would always be fed the station's program audio, and would remain continuously ready in an "autoanswer" mode.

If equipment at the transmitter site detected the failure of the primary studio-transmitter link, the ISDN codec at the transmitter would dial its twin at the studio, connect generally in one second or less, and immediately begin receiving the station's program

	Purchase Price	Installation Cost	Monthly Cost
ISDN Codecs (set of 2)	$ 9,960	$ 340	$ 86
Equalized Lines – 15k Stereo	0	3,000	545
Microwave Transmitter/Receiver *Redundant Antennae Required?* *Antenna Space Rental?* *Redundant Transmission Line?* *Double-Hop Involved?*	9,950	0	0

Table 1. Relative Costs of Backup STL Methods

audio, which would then presumably be routed by external switching components to the transmitter.

More exotic ISDN codec configuration schemes are possible as well, and will be discussed later.

ADVANTAGES OF BACKUP ISDN CODEC STLs

Modern well-engineered, well-installed and well-maintained microwave STL systems tend to be quite reliable – but not so reliable that most stations with any sort of on-air revenue stream to protect haven't seriously considered some sort of backup. For it is nearly inevitable that at some point during the useful life of the system, something will happen to take it off the air. Outright equipment failures, unusual weather phenomenon such as thermal inversions and heavy precipitation, and even damage to the microwave antenna and transmission line by lightning strikes or careless tower riggers are unfortunate facts of life for these systems that can lead to seriously degraded performance or even complete outages that extend for significant periods of time.

So if backup studio-transmitter links are essential for many stations, what are the advantages of using ISDN codecs for this application?

Cost. Again depending on the rate policies of the local telco provider, the ISDN equipment can be far cheaper to operate over its lifetime than permanently installed leased equalized phone lines, and the cost for using ISDN codecs is even competitive with the purchase of backup microwave link equipment.

Table 1 shows the initial cost relationships between various backup STL systems, which each purport to be able to pass broadcast-quality stereo audio from one site to another. The equipment costs shown on the table are recently quoted list prices for typical ISDN codecs and for typical congested-region microwave equipment, which is designed for use in spectrum crowded metropolitan areas. The table's telco-related charges are derived from our company's most recent quotes and/or invoices from Bell Atlantic for service in the Philadelphia area.

Note that the microwave system listed in the table consists only of a redundant transmitter and receiver, and assumes this backup equipment will share the primary microwave system's existing transmission line and antennae.

From the data in this table, it is possible to project out the cumulative station costs for using the various backup STL systems. See Figure 2.

The chart helps put the nature of the cost outlays of the various systems into perspective. For example, it is evident that leasing equalized lines from the phone company has a much lower initial cost than buying either redundant microwave components or ISDN codecs, because with phone lines there is no large equipment purchase involved. But the chart also makes it fairly clear that initial costs notwithstanding, both the codec and the microwave systems quickly pay for themselves in greatly reduced or non-existent monthly charges compared to equalized lines. Assuming new installations of both systems, for example, the cost of running the codec system becomes less than the cost of installing and keeping equalized lines after just sixteen months.

What if a station already has backup leased telephone lines and would consider replacing them with an ISDN codec system? Without the initial $3,000 telco equalized line installation charge to consider, the payout of the monthly equalized line takes an additional six months – resulting in a twenty-two month interval for the cumulative equalized line charge to exceed the cost of the ISDN codec system – still a respectable less-than-two-year payback on the original equipment investment.

While it is trivially easy to make a compelling argument preferring ISDN codecs to equalized lines on an economic basis, this same argument cannot be made against a backup microwave system. The graphs clearly depict the significant financial advantage that led broadcasters to embrace microwave links in the first place: once you own the equipment, the link is basically free. A high-quality microwave transmitter and receiver start out at a price nearly identical to a set of ISDN codecs, but for the microwave gear there is no significant usage charge throughout the life of the equipment. This slight price advantage may be nullified for backup microwave systems, however, in the case where a station wishes complete redundancy for its microwave system. Such redundancy entails the purchase and installation of a second transmission line and antenna at both the transmit and receive sides of the microwave link. Such a fully redundant system might allow a station to survive lightning or tower rigger damage to an antenna or line, and, with sufficient height diversity of antennae, might even effectively mitigate the effects of thermal inversions and weather fades. If the price of purchasing and installing additional runs of transmission line and microwave antennae are significant, or if any additional building or tower rental fees required to install second runs of line and antenna exceed the monthly charge for ISDN service, then ISDN codecs might once again appear economically preferable.

The slight economic advantage of redundant microwave equipment disappears almost entirely if a double-hop is part of the path, because in that case two complete redundant microwave transmitters and receivers would be required to achieve redundancy, doubling the initial cost. Projecting out the payback periods using the cost figures in Table 1, the cumulative cost for such a redundant double-hop microwave system would exceed the ongoing cost of using ISDN codecs for more than nine years.

Reliability. Even if a microwave STL system consisted of completely redundant components – transmitters, receivers, transmission lines, and antennae, by putting all of its "eggs" in one "basket", that is, by relying on the common terrestrial RF path the system components share in common, outages can still happen. Aside from weather phenomenon mentioned before that can cause serious reception problems, the failure of AFC or modulation-limiting circuits in the microwave transmitters of your spectrum neighbors can result in impressive amounts of interference on your assigned frequency, which you may be powerless to control. Like conventional equalized phone lines, ISDN codec systems offer a completely different logistical means of delivering audio from the studio to the transmitter, one that is unlikely to be disturbed by the same problems that occasionally befall microwave paths.

The ultimate reliability of any system of redundant components or multiple paths will be enhanced further by careful installation planning. Electrical service feeds should be chosen so that the failure a single circuit breaker will not interrupt power to components in more than one path. Where possible, wires or transmission lines connecting components of differing systems should not run in the same bundles.

Quality. On their face, ISDN codecs promise to deliver audio with near CD-quality – certainly a rival to the very best analog equalized lines the phone company has ever managed to install. Frequency response for typical codecs is specified to be flat out to 20 kHz, noise is -90dbm, and distortion is practically unmeasurable.

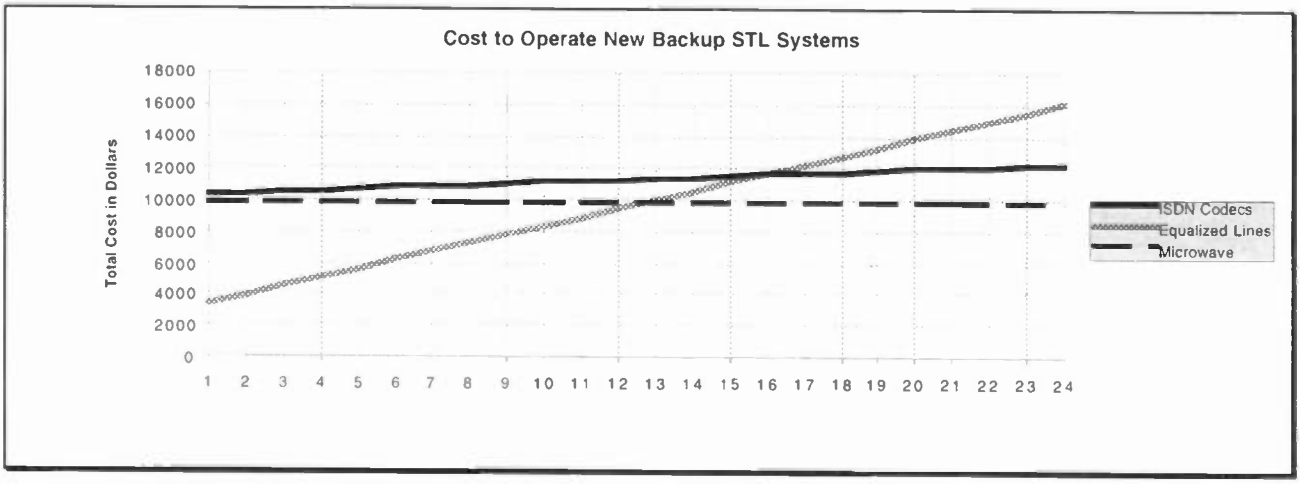

Figure 2. Costs of using various STL systems

Aside from managing the ISDN connection with the telephone company's central office, the codec's other primary task is to convert analog audio to digital and to "package" this digital data for transmission through the ISDN circuit. This is not a trivial task. "CD-quality" is generally considered to be 16-bit resolution sampled 44,100 times per second per audio channel. Simple arithmetic tells us that the data path therefore for CD-quality should be able to handle 16 x 44,100 x 2 = 1,411,200 bits per second. Recall however that the data carrying capacity of ISDN is only two 64 kbps channels, or 128,000 bits per second – less than one-eleventh of the necessary capacity.

The codec reconciles these widely disparate data rates through the use of "perceptual coding". Human psycho-acoustic phenomenon are exploited to allow the codec microprocessor's algorithm to analyze incoming audio, then discard those roughly nine out of ten sampled bits that contain redundant or irrelevant data. What remains after this "distillation" process is a set of data small enough to fit through the limited bandwidth of the ISDN circuit, yet large enough to allow a decoder at the other end to reconstruct an approximation of the original audio, which is, in theory, largely indistinguishable by humans to the original.

A number of coding algorithms are on the market, and the relative strengths and weaknesses of each are sometimes hotly debated by their proponents. Nearly all agree, however, that subjecting audio to this process is a drastic step, to be avoided when possible. (This is one reason that ISDN codecs are probably not a good choice as primary studio-transmitter links).

Therefore quality issues concerning perceptual coding may be magnified if a station is using contributory audio sources that themselves have *already* been through one or more bit-rate reduction coding passes before reaching the STL transmitter: two common examples of such audio sources are SEDAT network satellite audio and commercials delivered digitally to on-site terminals over dialup links by such carriers as Digital Generation Systems and Digital Courier. If one pass through a perceptual coder is tough on audio, what do successive passes do? Are slight errors in the encoding/decoding process of the first pass magnified into audibility by subsequent coding and decoding?

Nor can an engineer take too much comfort from the knowledge that audio passing through a backup STL ISDN codec will at least avoid still further coding passes downstream – since all of the proposed digital audio broadcast systems being debated now, arguably the future of radio broadcasting – rely on perceptual coding techniques to effect bit-rate reduction for the main transmitter/receiver link. Consumer-purchased DAB radios of the future *will* contain codecs.

There are unfortunately no hard and fast answers to the cascading algorithms question as of yet; engineers at stations with a significant reliance on already perceptually coded material are advised to proceed cautiously in this area, perhaps by arranging a rental or trial period of an ISDN codec under consideration so that independent listening tests can be made first hand.

Flexibility. Compared to the rigid point-to-point nature of microwave links and equalized phone lines, a backup STL ISDN codec system can offer some amazing flexibility, and also the opportunity to make reasoned tradeoffs in ultimate reliability in return for additional cost savings.

Consider the very common scenario where a single company owns to radio stations in a market, an AM and an FM for example, with studios at a common location but transmitters in different places. The layout of Figure 1 could of course be totally duplicated to provide backup STLs for each station. But since any single failure of a microwave system is a rare occurrence, how much rarer still would be multiple failures that effect each station at the same time?

Exploiting statistical probabilities, Figure 3 depicts an expanded version of the original block diagram for two stations, each with a primary microwave STL path, and each with a dedicated backup ISDN codec permanently installed at the transmitter sites. But the studio site in this case has only a single ISDN codec, shared by both stations. At either transmitter site, a failure of the main microwave STL link will cause the backup ISDN codec there to autodial the studio codec, and a unique permanent contact closure at the transmitter site codec, transmitted end-to-end to the studio site codec whenever the link is connected, will identify which of the sites is calling. This closure can then be used to switch the appropriate station's audio onto the studio site codec.

The economic advantages of this flexibility can be significant. Instead of purchasing four codecs and maintaining four ISDN circuits, an AM-FM combination like this needs only three. When shared thusly, the payback period for the purchase of the codec equipment compared to the use of dedicated redundant microwave components or leased equalized lines that was previously discussed above is shortened still further. And, if the unthinkable did occur – that is, the simultaneous failure of both stations' primary microwave links – the ISDN components shown could be reconfigured to provide appropriate *mono* program audio to each transmitter simultaneously (accomplished by having each transmitter site codec dial only one of the two 64 kbps data channels associated with the studio's ISDN line).

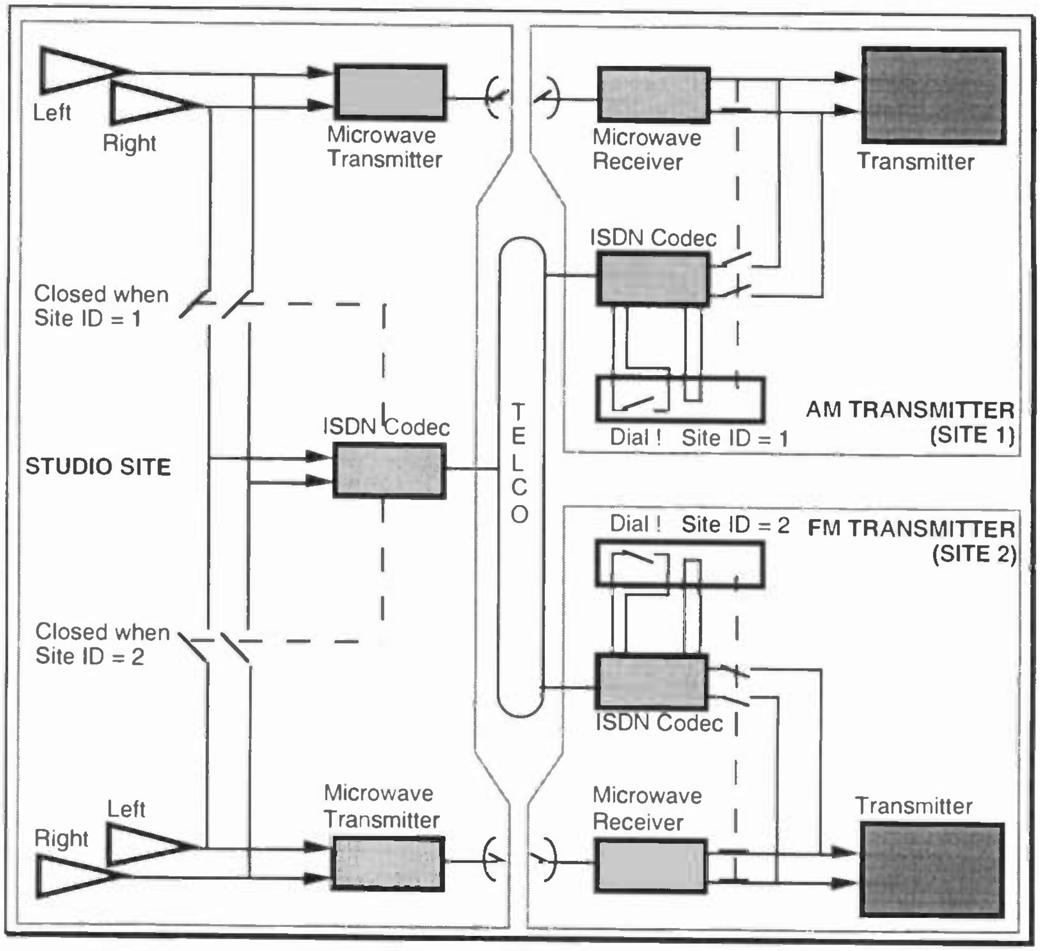

Figure 3. Block Diagram of Codec Installation for Two Commonly-Owned Stations

Those stations that already possess a pair of ISDN codecs for use with remote broadcasts are already "half-way there" toward implementing backup STLs. Since for remotes, one codec is usually permanently installed at the studio anyway, a station willing to, at worst, discontinue using the studio site codec for a remote broadcast would be able to use it for a backup STL, with the purchase of a single additional codec and ISDN line at the transmitter site.

Further economies of scale are obviously possible for duopoly stations that maintain more than two transmitter sites serviced from a common studio site, with the understanding that multiple simultaneous primary failures (hopefully extremely rare occurrences) may leave some stations without program material.

The easy interconnectability of ISDN codecs will even allow engineers to devise more elaborate plans to keep their stations on the air.

Disaster Planning. Some stations, particularly in major metropolitan areas, have entered into reciprocal agreements where each station agrees to allow the other to use spare facilities at its studio location should a fire, bomb threat, etc. make the other party's studio temporarily uninhabitable. A noble goal in principle, but one often stymied by a significant economic obstacle: the not inexpensive need to arrange for program links from each studio to the counterpart's transmitter. Since complete studio outages may be, if anything, even rarer than multiple STL failures, it can be difficult to economically justify the additional expense of maintaining spare microwave transmitters or leased equalized lines at a neighboring station.

Here again, the relatively low expense and flexibility of ISDN codecs can make such a reciprocal agreement at last financially palatable. When each station as party to a reciprocal agreement maintains compatible ISDN codecs for its own use as backup STLs at its studio and transmitter sites, these same codecs could easily serve to provide a quickly-arranged emergency STL path for a *guest* station that found itself ousted without warning by a disaster at its regular studio.
Nor are emergency studio possibilities limited merely

to other stations in your market. For example, arrangements could be made in advance with local recording studios for emergency facilities usage. More and more recording studios today have installed ISDN codecs themselves to enable recording sessions with distant talent, so the purchase cost of additional codecs and ISDN circuits may not even be necessary

Often a given transmitter site will have one or two emergency studios set up and ready to operate; with ISDN codecs installed, an FM station with a studio emergency might well get its feed from an emergency studio at its companion AM station's transmitter site. Or, a commonly owned group of radio stations may undertake to provide backup to each other in the event of serious studio emergencies – a station in Philadelphia could be operated in an emergency from its sister in Boston.

The point is, with an ISDN codec installed and able to feed the transmitter, an engineer faced with a major studio disaster has access to a much greater number of ways to get back on the air.

ASSESSING CANDIDACY

Aside from the quality caveats mentioned above, there are a few other potential disadvantages to using ISDN codecs as backup STLs that should be considered.

- Will it work when you need it? It is stating the obvious to point out the more complex a system is, the more likely it is not to perform as expected. While leased equalized lines may be low-tech and pricey, there's something undeniably comforting about having a continuous run of copper from your studio to your transmitter. In contrast, proper operation of a backup STL ISDN codec system depends on the proper operation of several pieces of very sophisticated electronic equipment, some of which (like the telephone company's central office switches) are completely outside of your control. For our part, we have basically found that once ISDN lines are up and once any installation problems have been corrected, they tend to be very reliable.

- Promptness of repair. Pricey equalized lines do buy you a certain measure of clout with the phone company. When you call the repair service to complain about a broadcast line outage (at least in the Bell Atlantic service area), you generally get immediate and round-the-clock attention until the problem is cured. On the other hand, inexpensive ISDN circuits are nearly on the same basis as POTS lines when it comes to repair priority – you may have to argue with a repair supervisor to get after-hours attention to a line problem.

- Coding delay. The perceptual coding algorithm operating in today's ISDN codecs actually delays program audio by a small but noticeable time – somewhat less than 0.5 seconds. Radio announcers monitoring the station off the air may become confused and disoriented by the delay, and may need to switch their headphones to monitor the program feed of the console rather than the off-air feed of the mod monitor. If you suspect the backup link will be used often, consider installing an automatic headphone monitor switch which operates, perhaps with a codec contact closure, whenever the link is on the air. A more elaborate switching setup might switch to the non-delayed program feed only when the link is up <u>and</u> the announcer's mike is turned on. This will allow normal off-air monitoring of music and other program material.

- Outboard equipment. Many stations use composite audio microwave STL systems. Since ISDN codecs are capable only of sending two discrete channels of audio, a separate stereo generator and an appropriate switching device may be required in addition to the ISDN codec system. Of course, this same additional equipment would also be required to use leased equalized lines which also provide only discrete audio channels.

Here are two sides of the same coin, a dilemma faced constantly by broadcast managers:

1. Stay on the air at all costs.
2. Spend no money at all.

Trying to strike a balance between these two edicts can be a significant challenge, especially at major stations when the stakes are high. If engineers ruled the world, no number of backup systems would be too many. But since financial responsibility is rightfully a worthy goal, stations contemplating the installation of ISDN codecs for backup STLs need to perform some careful examination of priorities to determine where along the above decision line they stand.

ISDN codecs are easiest to justify from a financial point of view when they would replace existing and expensive leased equalized lines.

Stations in major metropolitan markets, even if they already possess some primary STL system redundancy, may decide the relatively small cost of these systems, together with their flexibility for use with alternate studio disaster planning, makes good sense.

Stations operating without any backup to their primary STL, or stations that for whatever reason experience frequent failures of their primary STL, as well as stations whose revenue streams would be seriously affected by long outages, are the ones most likely to benefit from backup system installation.

Those stations that already use ISDN codecs for remotes are also in a good position to benefit economically from their system's use as backup STLs, since the payback period for the equipment purchase will be shorter.

ASSURING SYSTEM INTEGRITY

Once the decision is made to deploy ISDN codecs for backup STL purposes, do all that's possible to guarantee the system will work when it is needed.

- Make all cables connected to the system carefully – particularly the cables that connect to the ISDN phone line. Bit-error rates of ISDN connections can climb (and the codecs can fail to properly connect) due to poor continuity in the data line. Make sure all connections, including bridging clip connections on telco-supplied punchblocks, are clean and tight.

- Use an uninterruptible power supply to run the codecs and all associated devices at both the transmitter and studio sides of the connection. The UPS will help assure power line spikes do not lock up the systems' microprocessors and prevent the units from working.

- Ensure the system is regularly tested. Some remote control systems can be programmed for "time of day" functions, and can be set to automatically log status indications. If possible, consider programming the transmitter remote control to dial up the studio site ISDN codec at least once per day – and to either log, or preferably sound an alarm – if ever the units fail to successfully connect.

- Go to the trouble of actually trying the units on the air for a few minutes at least weekly Make sure audio levels from the codec device are properly matched to the transmitter, and that the link sounds acceptable on the air.

- Install some explicit way (an alarm chime, a light, automatic paging, etc.) for you or for the announcer to know that the backup link has come up and is on the air. This way, you'll be able to investigate and correct on a timely basis any problems with your primary STL system.

CONCLUDING WORDS

The flexibility, economy, and reliability of broadcast ISDN codecs, together with the increased availability of inexpensive ISDN service from regional phone companies, combine to make these devices worthy of consideration as backup STL systems. Stations which embrace the technology may simultaneously enhance their ability to remain on the air in a variety of failure situations, and save significant amounts of money compared to more traditional equalized line backup links.

ACKNOWLEDGMENTS

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THE NEXT GENERATION RADIO STATION ELECTRONIC COMMUNICATION SYSTEM

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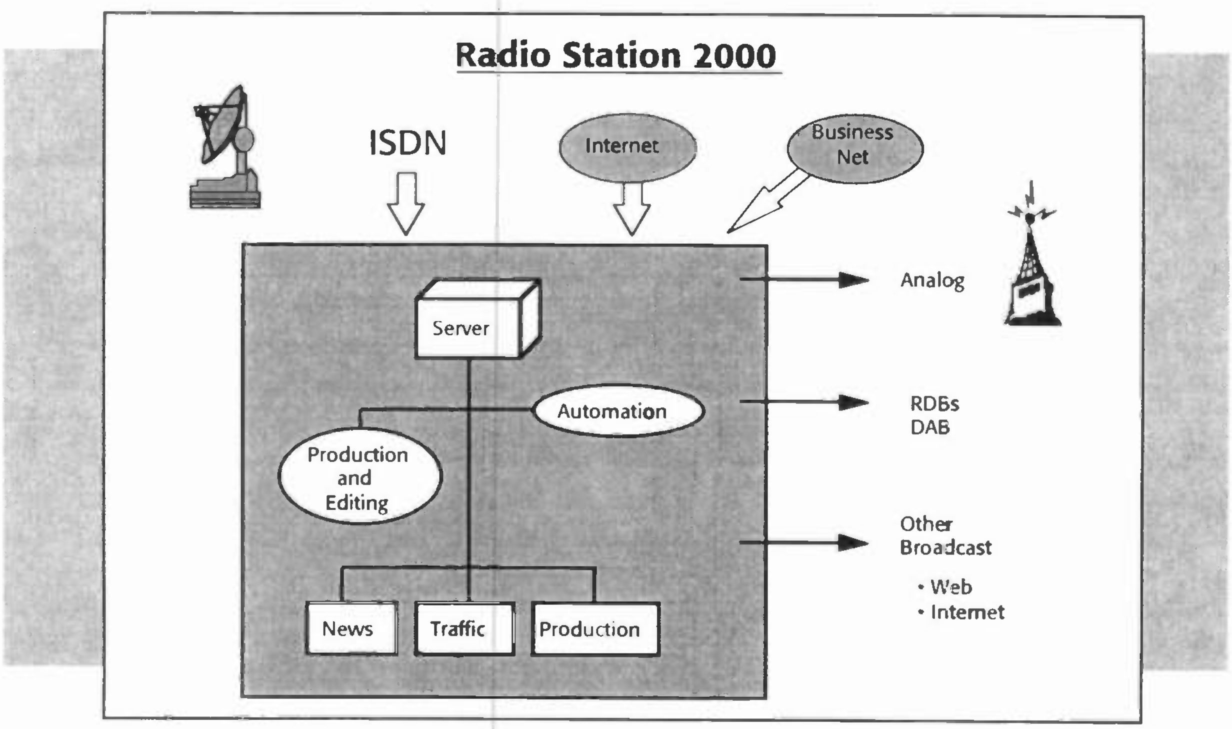
ABSTRACT

By the year 2000 the majority of radio stations will move from analog, tape-based studio operations to digital, network-based systems. These technologies will include digital production and editing, digital receipt of audio and automated internal and external processing, resulting in stations that are fully networked for all station functions. The automation of these operational functions also supports the automation of today's manual logs and business processes. The next generation radio station will apply and take advantage of multimedia, communications and digital technologies that are already becoming affordable and widely available now.

What will it take to achieve this radio station 2000 vision? First and foremost, it will take technologies and solutions that permit a station to make choices in how it operates. Systems must be interoperable, able to communicate to each other and extensible to incorporate future technologies. This is the basic premise of the choice ABC Radio Networks and DG Systems have made for deployment of digital technologies to network affiliates. This selection is significant as stations have more and more communication options, such as the Internet, direct broadcast satellite (DBS) and personal communications services (PCS). Broadcast radio will be faced with extreme competitive pressures as advertisers realize an expanded selection of broadcast media to choose from. Radio's adoption of open systems is essential for the application of digital technologies. The technologies enable the support of creative production and expanded products for listeners, while offering improved services to advertisers.



In applying these technologies, a class of products and services is emerging in the industry to address the growing complexity of radio station operations that will merge these independent technologies into a coherent and easily accessible set of networked solutions. To reach this next level technical challenges must be overcome, such as creating standard methods of information exchange, access and delivery and integrating both existing and new technologies.

A set of industry standards have already begun to emerge that define how electronic information will be exchanged, and how that information will automate and simplify paper-based transactions in the broadcast industry.

At the core of this solution are high-speed communications technologies, including digital satellite, ISDN, the Internet and messaging networks. Integrating these systems with advanced software, easy-to-use applications, powerful client/server technology and today's analog broadcast tools will bring a new level of capabilities to broadcast radio.

Vision of the Industry

As recently as a few years ago, the convergence of computing, multimedia and communications technologies in the radio broadcasting industry was an abstract theory. Today, there are more than 4,000 radio stations which have begun to make that theory a reality by utilizing a digital, multimedia network for the delivery and receipt of advertising, music and other content.

The next generation radio station will build on this foundation, integrating ever more powerful, faster and easier to use technologies. This development holds the potential to dramatically revolutionize radio station programming, operations and marketing.

Radio is today much less an industry of disparate stations and operations than it was even a decade ago. As the industry embraces digital technology, this trend will continue. An increasing reliance on digital communications enables information sharing within companies and broadcast groups and even throughout the industry, creating instantaneous links between reps, music companies, advertisers and stations.

At the station operations level, the implications are profound: fast access to content, simplified audio storage and playback, automated affidavits, scheduling and programming systems and immediate access to information online will be integrated into increasingly streamlined user interfaces, promising greater ease of use and efficiency.

For example, in a typical day:

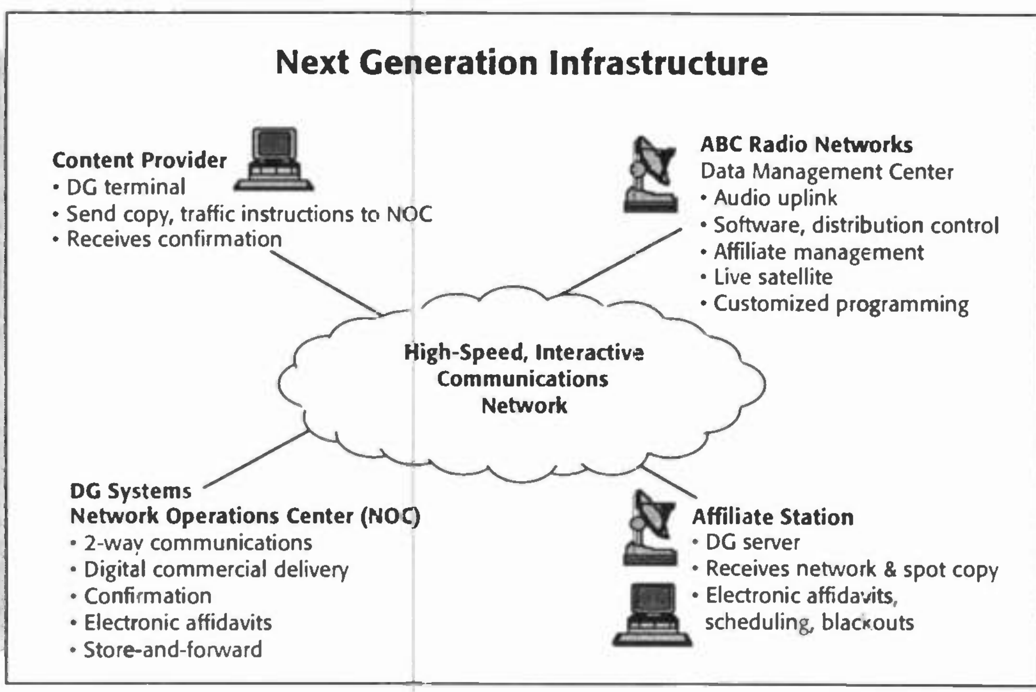
- A station may have new programming automatically downloaded into computer overnight, requiring no attendance, manual assistance, or sorting through tapes;
- A program director may sample audio and compile a playlist, including commercials and news, from a PC;
- A station may have schedules downloaded automatically to a PC, which can be accessed in multiple departments throughout the station;
- A station may receive local and national news updates via the audio server and produce its own local news program using the news sources;
- A station may produce specialized promotional information and electronically transmit it to a sister station;
- A station may receive network commercials that are localized to its specific market or region;
- A station may join a live program in progress at a point in the program chosen by the station.

Vision of the Architecture

In order to achieve these benefits, new systems must fit smoothly into today's radio station environment. The technology involved includes hardware, software, communications and multimedia, all working together in a client/server architecture. The solution being developed by ABC and DG Systems for affiliates stations throughout the United States is an example of how these technologies can be integrated into existing environments, yet also enable future extensions. This vision was demonstrated at the last NAB, and trials in radio stations are now underway.

The new infrastructure we are developing combines communications expertise with unique product and programming capabilities. By integrating landline and satellite communications, affiliates gain the fastest, most efficient and cost-effective audio and information at their fingertips. For example in this solution, DG Systems' Network Operations Center provides communications routing such as high-speed modems, ISDN lines and digital satellite communications working in conjunction with ABC's live satellite feeds. Because some programming requires live transmission, while other content, such as commercials or short-form programming can be delivered in a store-and-forward system, the synchronization of technologies results in simplified operations for the affiliate station. As a result, affiliates have the option of receiving live broadcast audio when needed, and point-to-point, customized audio when it is most appropriate.

The combination of landlines and digital satellite, managed from a central Network Operations Center, provides bandwidth-on-demand for each delivery. Each station is assured of receiving its designated audio and data files on time, regardless of the volume being transmitted over the network. With the unique bandwidth-on-demand capability of digital satellite and landline transmissions, commercials and many other audio files are downloaded directly into a server within the station. Affiliates no longer need to tie up resources waiting for commercial satellite feeds — receiving commercials no longer requires an attendant, and scheduling is automatically transmitted and stored for access as needed.



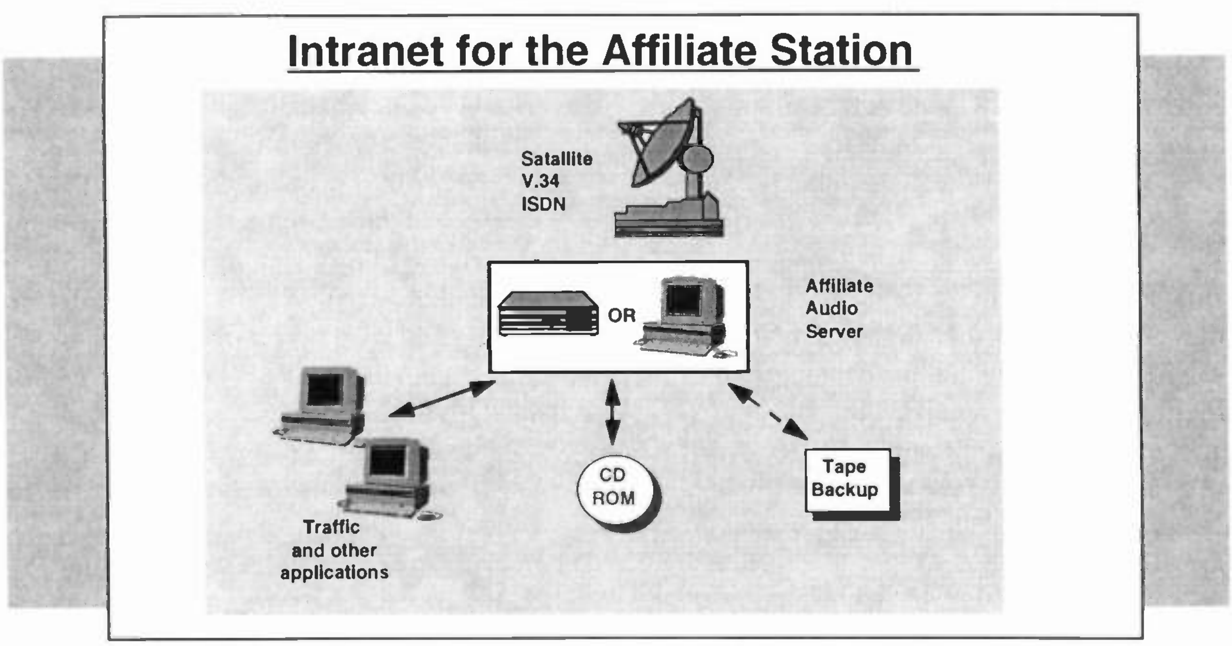
In addition, this system enables two-way communication. This capability allows each network affiliate to not only receive essential scheduling and programming information, but also to send its own audio files to sister stations or communicate with the network online. Affiliates will also use the system to electronically transmit information such as affidavits and station pledges to the network head-end.

The Foundation

Serving as the brain of this example is a powerful server, which in our case is called the Affiliate Audio Server. Data, audio, images and graphics files are downloaded to this server throughout the day, bringing timely news, programming, commercials and operations information to the affiliate in digital format. The system also includes powerful, yet simple-to-use, software tools so that personal computers and workstations can access content from the server across a local area network (LAN) from any point throughout the station.

It is essential that this server is flexible in order to easily migrate to future applications and technologies over time. Stations must consider issues such as investment protection, training and upgrade installations when justifying a transition to a next generation system. For example, ISDN provides ample bandwidth today. However, when technologies such as asynchronous transfer mode (ATM) and frame relay become more cost-effective and accessible, stations will want to migrate without the headache of massive upgrades. The server must fit into any environment that a station chooses. Previously, affiliates have had to choose from selected proprietary systems, resulting in limited integration and automation within the station. Moving forward, open systems will become more important as stations begin to electronically share information within a station, among a group of stations and with advertisers, partners and suppliers.

This networked server architecture represents an enormous step forward for digital delivery. In the past, digital commercials or music singles were typically accessed in a single location. In the next generation station, client/server technology provides multiple points of access throughout the station. This enables others to review audio files remotely, as well as integrate the system into current traffic and scheduling systems. Users will have multitasking capabilities to allow recording files in the foreground while



also sending files simultaneously in the background.

The use of standard communication protocols is critical to ensure that the server will interoperate with existing and future station applications over both local and wide area networks. Standards such as TCP/IP, HTTP and FTP enable Internet connection, which will continue to grow in importance for the radio industry. The client/server solution must also integrate easily into existing LAN environments in order to provide the level of access and automation desired. Each affiliate makes choices unique to its format, geography, listeners and needs. A great level of flexibility and adhering to standards is required to bring digital technology to the entire industry. The foundation of the next generation station is essentially an optimized intranet solution, an increasingly common use of Internet tools in organizations today. Access points are distributed throughout the LAN within the affiliate station, with the simple look-and-feel of an Internet World Wide Web browser. At the same time, an external infrastructure managed by DG Systems and ABC, or other content providers, is tapped to continually provide valuable, updated information to the affiliate network. While the affiliate has access to an international, dynamic network, it also retains the security, privacy and cost efficiency of its own LAN.

The server must meet additional demands of the affiliate such as high-quality digital audio, multiple network standards and increasing storage capacity. In addition, the affiliate server can be connected to external mass storage devices, CD ROM, tape drive for backup, printers or other devices. The server brings a high level of flexibility and integrates easily into existing production and operations systems.

The applications available throughout the affiliate station are just as important as the flexibility and power of the server. In order to simplify operations within the affiliate station, the client/server system presents data to each department or user that is specific to their needs. For example, a traffic manager can easily view just the scheduling information, while the programming manager will utilize the same system to support news and talent.

The interface for these applications is an Internet browser. By relying on industry standard technologies, this solution provides consistent presentation of data and easily runs in Macintosh, Windows or UNIX environments. In the future, emerging standards such as Sun Microsystems' Java programming language will bring an even greater level of interactivity to applications such as electronic affidavits.

Together, this system provides a standard, end-to-end solution for networks and affiliate stations. The network head-end has a two-way communications method to transmit digital text, audio, images and graphics to affiliate stations. The system provides affiliate stations with automated, consistent tools to manage schedules, blackouts, programming and other essential content.

Communications Network

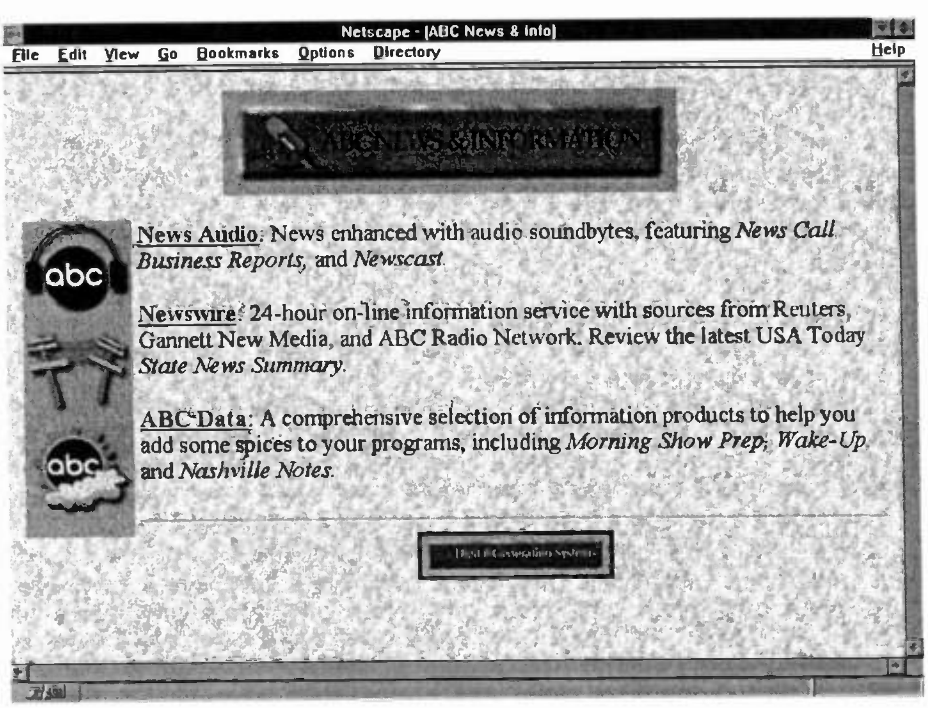
The benefits of the next generation station are not possible without a scalable, well-managed network. Stations need the assurance of a system that can easily adapt to both current and future network technologies. Existing communications systems such as ISDN have already been tapped to begin offering digital delivery. However, emerging fiber-optic networks such as ATM will eventually bring greater bandwidth and multicast deliveries. While many stations have begun to consider the Internet as a tool to reach greater audiences, it can also be utilized for additional storage and messaging between talent, advertisers, reps, sister stations and more. Digital satellite technology will also play a significant role in effective communications for the next generation radio station. Although not every station will need each of these technologies, the server-based communications system is built to easily meet the changing needs of any affiliate and incorporate emerging technologies.

By integrating these capabilities, the network will offer the greatest level of reliability. Today, if a live satellite feed fails, the station must end the program broadcast. As mission-critical information and content is carried on the digital network, stations must have a fault tolerant system with no down time.

A combination of leading-edge network capabilities ensures the most efficient use of the system and guaranteed delivery to affiliates. For example, the system is able to automatically determine the best transmission path through the network to each affiliate, which decreases congestion. Unlike analog satellite, with which you purchase fixed amounts of bandwidth, digital satellite provides bandwidth-on-demand. This dynamic bandwidth utilizes only the necessary amount of the network, so that the remainder of the connection can be utilized for additional transmissions.

The advanced capabilities and high level of network management also enable prioritization for affiliates and content. The delivery of worldwide breaking news, for example, must take priority over a previously scheduled morning drive time sound bite. In the event of a network failure, this system also allows a simple rerouting of files that guarantees delivery to the affiliate.

This backbone system is managed from a Network Operations Center, which is essential to ensure two-way communications, delivery confirmation and elimination of contention for individual destinations. The management of connection and congestion are also essential to ensure that dissimilar devices can communicate across the network. From the Network Operations Center, operators are able to access servers remotely, update software tools and maintain both the local and wide area links to each affiliate. Adhering to industry standards and extensive network management is key to the success of this communications network. Without it, affiliates would not receive the same level of customization, automation or simplified operations that will enable them to remain competitive.



Data Management Center

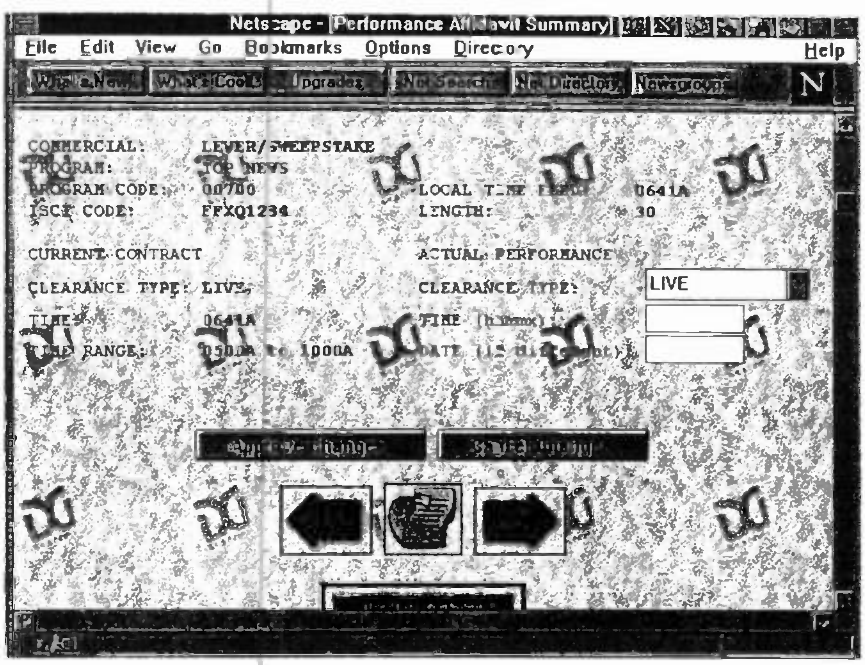
The Data Management Center is a virtual head-end, coordinating, customizing and managing content for the affiliate stations. It provides an application protocol interface (API) which acts as a conduit between external systems, such as existing mainframes that maintain databases of affiliates, and the communications network that delivers essential data, audio, images and graphics to each station.

Using the end-to-end network solution, the Data Management Center can now transmit content electronically, resulting in faster, higher quality delivery to each affiliate. Affiliates can use the client application to request, select and listen to audio files updated throughout the day. For example, an early morning manager might access ABC files such as Morning Show Prep or Wake-Up to add to the drive time program. A mid-afternoon manager may access Newswire updates and the USA Today State News Summary for a current news program.

As you can see in the sample screen shot, users browse and click through simple dialog boxes. With this graphical, consistent interface users locate the specific information they need for a particular job, yet all data is stored in the same server for easy delivery and access. Delivery of audio in a digital file permits the station to further edit, develop and assemble local programming using the station's own digital production and editing systems.

ABC also composes programming and commercial information each day that is delivered electronically to each affiliate. Affiliates have traditionally received commercials via bulk-feed, schedules via fax and affidavits via mail. Managing functions such as affidavits and blackouts with a paper-based system resulted in long lead times, misplaced documents and any number of errors. By contrast, ABC can now leverage the Data Management Center applications to customize programming and content for its affiliates, including regionalized copy, blackouts and electronic scheduling.

The automated affidavit application electronically captures essential data onto the server. This process saves enormous amounts of time and increases data integrity. By using the two-way communications feature of the network, stations can enter and transmit the data online, resulting in compensation as much as three months faster than the manual process. The Data Management



Center also utilizes the two-way messaging capability to gather feedback from affiliates and confirm delivery and airplay of commercials.

Some programming, such as live, syndicated talk-shows or sporting events, still needs to be delivered via live satellite feed. Commercials, however, are delivered more efficiently using the store-and-forward digital network. The Data Management Center can now synchronize the live feeds with the digital system. The server within the station is equipped with a net cue management card that distinguishes between the two delivery methods and provides the affiliate with a live program and commercials seamlessly.

Another benefit for network affiliates is the elimination of commercial bulk-feed. Both national and spot commercials can now be sent digitally to the Affiliate Audio Server. With no bulk-feed tapes to review, copy management and commercial insertion is drastically simplified for the affiliates.

Looking Forward

Today, we've looked closely at the next radio station electronic communication system, its components and benefits, from the perspective of our work in creating and implementing this vision. From here, digital technology will only bring more and more benefits to the radio industry, not only stations, but advertisers, talent, and consumers, as well. With end-to-end automation and communications, the stations themselves essentially become content factories, taking digital media from various sources, then packaging and delivering it to growing audiences.

As the Internet becomes the next mass media tool, stations are quickly learning its value for marketing and expanding its reach. Once security concerns are resolved, the Internet will also become a valuable repository for radio sound bites, music and news archives and advertising communications. Today's digital technology has already begun to explode physical boundaries and will continue to enable remote station management, international radio broadcasting and other global applications. Addressability will bring greater power and more accurate marketing to advertisers.

Once again, radio is renewing itself to become an ever greater medium for immediate access to and exchange of information and entertainment - the next radio station electronic communication system is the logical first step and an important foundation for our industry.

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T1 DIGITAL STL: DISCRETE VS. COMPOSITE TRANSMISSION

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Abstract

T1 digital transmission is rapidly becoming the STL medium of choice for many radio stations. This paper looks at the advantages and disadvantages of using a T1 circuit to transmit discrete stereo audio versus composite audio.

The issues addressed are those of T1 digital bandwidth, coding limitations, signal-to noise ratio, system headroom, stereo separation, and the ability of the T1 circuit to carry other channels such as remote control, telephone, data, SCA, and SAP.

T1 Digital STL

For years, broadcasters were limited in their choice of STL equipment to two main options: 950 MHz radio, and equalized analog telephone lines. Today a third approach, T1 digital transmission, is rapidly becoming the system of choice for many radio stations.

The decision to use T1 may be based on several factors:

- Necessity. The station has no line-of-sight to the transmitter, or frequencies are unavailable.
- Sound quality. A digital STL can sound better than even the best analog systems.
- Cost. A single leased T1 line can cost less than multiple leased analog lines.

Whatever the reasons, a broadcaster who decides to use T1 transmission must then choose what type of system to implement.

Digital T1 STL systems can be designed either to transmit **discrete** left and right channel stereo (Figure 1), or to transmit a **composite** stereo signal (Figure 2).

Figure 1: Discrete Digital Audio STL

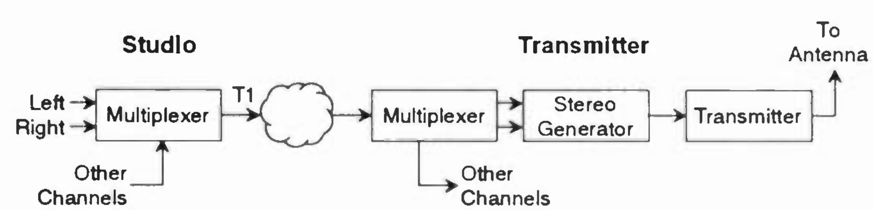
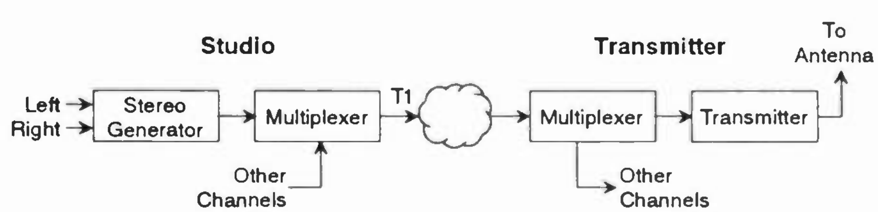


Figure 2: Composite Digital Audio STL



As the preceding figures illustrate, in a discrete transmission system the stereo generator is located at the transmitter site, while a composite transmission system places the stereo generator at the studio.

There are sometimes good reasons for having the stereo generator in the studio. For example, many manufacturers combine an audio signal processor and stereo generator in the same chassis, and the broadcast engineer may wish to make frequent adjustments to the audio processor.

However, broadcasters should realize that a composite digital STL system can introduce compromises in four important areas:

- Dynamic Range and Signal-to-Noise Ratio
- System Headroom
- Stereo Separation
- Channel Capacity

Let's take a look at why this happens.

Dynamic Range and Signal-to-Noise Ratio

Analog-to-digital conversion requires a sampling frequency at least twice as high as the analog bandwidth; digitizing a radio program signal whose highest frequency is 15 kHz (15,000 cycles per second) requires at least 30,000 samples per second.

In actual practice, 15 kHz audio is generally sampled 32,000 times per second, or oversampled at a multiple of 32,000 to simplify the implementation of an aliasing filter.

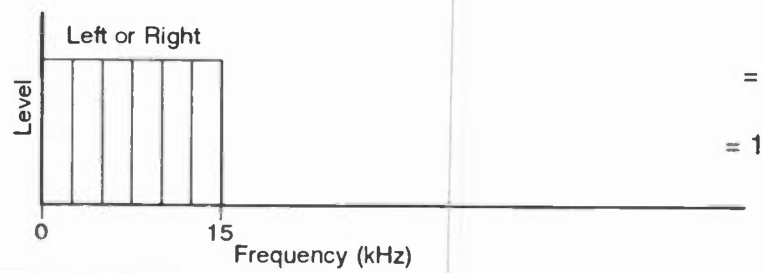
To maintain CD-quality audio, each sample must be at least 16 bits. Multiplying these numbers shows us that linear, uncompressed digitizing of discrete 15 kHz stereo produces a total of 1,024,000 bits per second (1.024 Mbps), as illustrated in Figure 3. A 1.024 Mbps signal fits into a T1 circuit with plenty of room for additional channels.

A composite stereo signal occupies 53 kHz of bandwidth, and the entire 53 kHz must be encoded as a single analog signal. This requires a sampling rate of about 112,000 samples per second. With 16-bit coding, we get a total of 1.792 Mbps (Figure 4).

The problem is, a T1 circuit runs at 1.544 Mbps, with a payload capacity of 1.536 Mbps, so it cannot transport a 1.792 Mbps signal. To get composite stereo onto a T1 line, the sampling resolution must be reduced from 16 bits to 13 bits or less, which results in a worse signal-to-noise ratio and a 12 to 18 dB decrease in dynamic range.

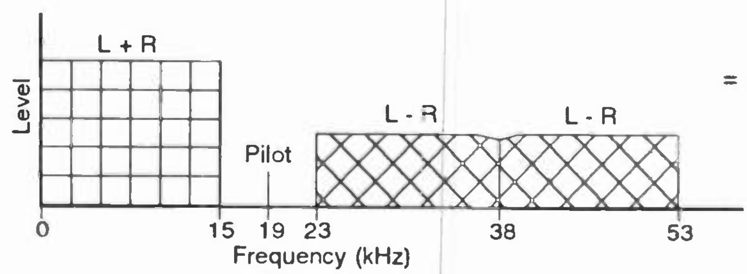
Also, the higher bit rate required by a composite signal means less room in the T1 circuit for other channels, as described in detail later in the section on Channel Capacity.

Figure 3: Discrete Stereo Encoding



32,000 (samples per second)
x 16 (bits per sample)
= 512,000 (bits per second)
x 2 (channels)
= 1,024,000 (bits per second)

Figure 4: Composite Stereo Encoding



112,000 (samples per second)
x 16 (bits per sample)
= 1,792,000 (bits per second)

System Headroom

Digital audio encoders have a limited amount of headroom on their analog audio inputs. Any signals above the maximum headroom level must be sharply limited to prevent overmodulation and severe distortion.

On a composite system, the L + R signal contains the peak levels of both the left and right channels. When both channels have high peaks at the same moment (a common occurrence), the signal needs more limiting than either the left or right channel of the same audio on a discrete system (Figure 5).

The result is a further decrease of up to 6 dB in the effective dynamic range of a composite system compared to a discrete system.

Stereo Separation

No audio transmission system has an absolutely flat frequency response. The variation over the analog audio bandwidth is typically something like ±0.5 dB, meaning that the transmission system could create as much as a 1.0 dB error in relative levels between the lower frequencies and the higher frequencies.

On a broadcast audio signal sent to the transmitter using either an analog STL or a discrete digital STL, one decibel of difference is barely perceptible to the listener. In fact, a decibel is, by definition, the smallest difference in loudness that the human ear can detect.

However, as we saw in Figure 4, when the analog audio is converted to composite stereo before digital encoding, the lower frequencies contain the L + R component and the higher frequencies the L - R. Even one decibel of variation between these components on the STL link makes a big difference in stereo separation when they are reassembled at the listener's radio receiver.

The reason lies in the way the receiver uses these two signals to reconstitute the original left and right channels:

$$(L + R)/2 \;+\; (L - R)/2 \;=\; \text{Left channel}$$
$$(L + R)/2 \;-\; (L - R)/2 \;=\; \text{Right channel}$$

For a composite signal with a 1 dB differential in power gain between the L + R and L - R components (a voltage ratio of about 0.9), the effect is to introduce a portion of each channel into the other:

$$(L + R)/2 \;+\; 0.9(L - R)/2 \;=\; 0.95L + 0.05R$$
$$(L + R)/2 \;-\; 0.9(L - R)/2 \;=\; 0.95R + 0.05L$$

This means that the stereo separation is degraded to 20Log(0.95/0.05), or about 26 dB, while an unimpaired broadcast signal has separation ranging from 40 to 60 dB. Discrete transmission has no such potential effect on stereo separation, because the stereo generator is located at the transmitter, after the digital STL link.

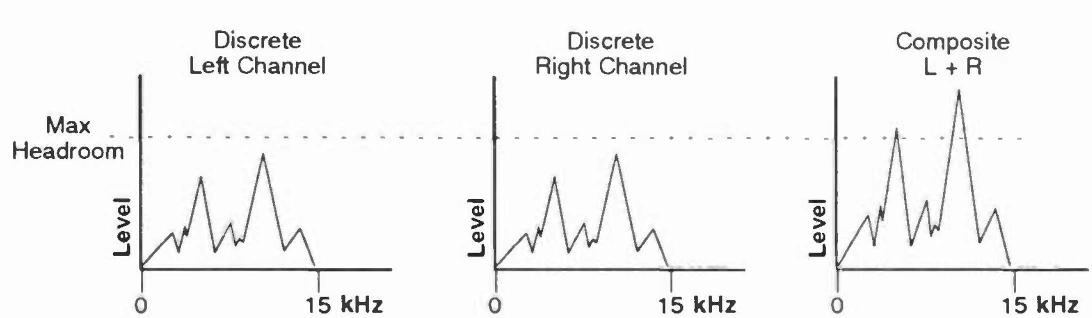
A six degree phase differential between the L+R and L-R signals can be shown to create similar results; that is, stereo separation is reduced to 26 dB on a composite transmission system, but is unaffected on a discrete system.

Channel Capacity

One of the advantages to a broadcaster in using a T1 digital STL is the ability to multiplex several signals together. SCAs, SAP channels, transmitter remote controls, data, and voice can all be combined with the broadcast audio signal for transmission on a single T1 circuit. And the duplex nature of T1 lets you use the same system for both STL and TSL channels.

The payload portion of a T1 circuit consists of

Figure 5: System Headroom, Discrete vs. Composite



twenty-four 64 kbps channels called time slots, for a total of 1.536 Mbps (24 x 64 = 1536). As we've noted earlier in the discussion of dynamic range, the discrete STL approach using 16-bit linear coding produces a 1.024 Mbps digital signal.

To this, Intraplex adds some overhead for error mitigation bringing the total up to 1.152 Mbps, or three-quarters of the payload capacity of the T1 circuit. This leaves one-fourth of the T1, or six time slots, available for other STL or bi-directional channels.

A composite system, even using smaller audio coding samples, has less bandwidth available. Table 1 shows the amount of extra bandwidth remaining in the T1 circuit at several sample sizes.

Composite vs. Discrete: The Bottom Line

Table 2 sums up the relative advantages of T1 digital STL systems.

Table 1: Digital Audio Encoding - Sample Size vs. T1 Bandwidth Usage

System Type	Samples per Second (for 15 kHz Stereo)	Sample Size	Digital Transmission Rate	Number of T1 Time Slots Used for Broadcast Audio	Number of T1 Time Slots Available for Other Channels
Discrete	32,000 x 2	16-bit	1.152 Mbps	18	6
Composite	112,000	16-bit	1.792 Mbps	Not usable	--
		15-bit	1.680 Mbps	Not usable	--
		14-bit	1.568 Mbps	Not usable	--
		13-bit	1.456 Mbps	23	1
		12-bit	1.344 Mbps	21	3

Table 2: Composite vs. Discrete T1 Digital Audio Transmission

Feature	Discrete Transmission	Composite Transmission
Location of stereo generator?	Transmitter	Studio
CD-quality digital audio coding with full 16-bit accuracy?	YES	NO 13-bit or less
Greater than 90 dB dynamic range and signal-to-noise ratio?	YES	NO up to 18 dB lower
Full headroom even with peaks on both channels simultaneously?	YES	NO requires up to 6 dB more limiting
Full 40 to 60 dB stereo separation even if frequency response varies?	YES	NO may be reduced to 26 dB
Ability to carry multiple additional channels without compromising audio quality?	YES	NO must further reduce the number of bits per sample
Click and pop noise removal through digital error mitigation?	YES (with Intraplex)	NO bit errors can cause noise

INTERCONNECTING THE DIGITAL CHAIN

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With all of today's interest in digital audio, we see a lot of questions still unanswered. What products really work together in the digital chain? How do we interconnect them? How can we gain the most efficiency for our capital expenditures? Will my people know how to use these products? Let's take an indepth look at what is available for the digital chain that works today. We will discuss connectivity between different products, and look at many of the aspects that are paramount for your success in the conversion from the old inefficient analog to the new digital chain.

When we look at the integrating a digital chain in any facility, we ask ourselves why we want to go through the transition. Most always, the initial answer for the new digital chain is audio quality. Achieving the high level of audio quality that digital audio affords us can give you a tremendous edge over the competition. After we have commited ourselves to the task at hand, we must sit down and define all of the associated parameters. We then step back and ask ourselves what equipment we want to incorporate into our chain what benifits it will add once it is integrated.

We also seek out other areas of concern where a digital chain can add new efficiency to our operation. These areas can often offset and sometimes even pay for the cost of the transformation itself. Some of these areas where efficiency can be added are found in the transfer of scheduling information bi-directionally around the facility. Having production instantly available on all stations within the location without dubbing. Allowing a single operator to do their jobs so much more efficient, that they can produce and do a lot more of what we pay them to do. The administrative aspects of the new technologies can give us instant access to mission critical information that we had to spend countless hours gathering before. Remote control capability, walk away, etc., etc., etc.. There are a lot of new advantages for digital.

These types of new efficiencies can be hard to determine on your own. It is much easier to enlist the knowledge of individuals that know what the new digital equipment is capable of. You do not want to risk putting ourselves out there on the bleeding edge. There are a lot of sources available to answer your questions including distributors, peers, and manufacturers. To accurately tackle the problems of integration on your own is achiveable, but will take a little time doing research before you can make a reasonable decision. After you have spent your valuable time seeking solutions, you will probably find out that some of the parameters have changed. Today's hardware and software technologies are literally changing daily. Do not let the challenge of change get you down. Even though it can be like attempting to stand on top of a basketball, the rewards for the successful far exceed the efforts taken.

Until we humans can speak and hear digitally, we will continue to have analog equipment at the beginning and end of any audio chain. After we have selected the types of equipment we are planning to use in our chain, we need to look at their inputs and outputs. We want every piece of gear to have digital I/Os if possible, If we can get both digital and analog I/Os we are automatically ahead of the race. That gives us the ability to use the equipment in analog environments when and where we need to.

You will find that the majority of digital equipment such as CD players, DAT machines, MD , DCC, digital editors, and hard disk systems all have some sort of digital I/O standard today. Most digital inputs and outputs on source equipment are either AES/EBU or SPDIF.

AES/EBU is the professional balanced standard that has been accepted by the majority of broadcasters worldwide. This standard is a serial data stream consisting of combined left and right channels ranging between 16 to 24 bit audio information. The signal is self clocking for synchronization between devices allowing easy installation.

SPDIF is the unbalanced data stream mostly found in equipment geared for the consumer industry. The data formats of the two standards are incompatible with each other. Due to customer preferences in equipment, we must interface the two types of standards. For that reason, we have devices for converting the two formats. Regardless of which type of I/O we have, we ultimately can use them.

Our analog equipment that we want to use, or have to use, must be converted to digital at some point. The console or switching system we want to use in our chain becomes very important in helping, (or hindering) the blending together of all of these different types of equipment. We want to make sure that the console can give us analog inputs where needed. It should also have the capability to accept both AES/EBU and SPDIF inputs. Most consoles also offer multiple analog and digital output busses. They are capable of performing the tasks of routing switchers as well as console functions.As you can already see, doing a little Q&A on the console will save us a lot of grief.

Depending on the type of source equipment, we will have different sample rates to contend with. While CD players are generally 48k sampling rate, DAT players can vary between 48K and 32K. The majority of broadcasters use 32K sampling rates giving an audio bandwidth of 15 KHz. In a lot of cases, the cost of storage medium such as hard drive storage space, will dictate lower sampling rates. FM Transmitters broadcast 15 KHz audio bandwidth. Why pay for higher quality storage and audio flow that will not reach the end user. Fortunately, most of the professional digital consoles have built-in capability to do sample rate conversion. If the console we decide on does not, we will have to buy individual A/Ds, and sample rate converters for each input that is different from the consoles standard. In this way we match all of the different sources to a single locked in standard for the remainder of our digital chain.

At this point we should mention some of the concerns pertaining to the actual audio wiring of devices. Most audio AES/EBU connections are 3 pin XLR connectors. The video industry generally uses B&C connectors. Consumer grade devices mostly use RCA type connectors. By asking what types of connectors are on the equipment we purchase, we can easily find and connect our equipment together. Where we run into trouble is in attempting to reuse our existing cable. Digital audio data streams require 110 Ohm impedance cable. You can not reuse our old XLR-XLR cables for interconnecting AES/EBU devices. Even if you are looking at a connection that is a short distance, you need to replace the wire with 110 Ohm cable. We can run up to distances of 300 feet with the correct impedance. It is remarkable how many calls we get where equipment does not work and the problem is the cable. Digital audio needs digital cable.

From this point it is a matter of taking the AES/EBU output from the console to the input of our digital processor. From there we connect the AES/EBU output to either a digital STL, ISDN, or what ever means we digitally get our signal to the digital exciter input of our transmitter. There are a few other items to keep in mind such as our EAS ENDEC system, RBDS, subcarriers, and etc.. Some of them will stay analog some will be digital. Take each system on it own merits and determine the cleanest signal path available given the technology you have to work with.

If we are using compression algorithms in our digital chain, such as in the DSTL, hard disk system or elsewhere, we should attempt to standardize on a single algorithum. If we are using MPEG ISO LAYER II on our digital STL, we should also use try to use it in our hard disk system. Today's algorithms are very good but, we do not want to go in and out of any compression schemes more than a few times. It can get worse trying to go through a variety of different types and styles. It is a good idea to do an actual audio test to determine the final audio product quality. If it sounds OK after multiple generations, multiple algorithms, then go forward with it. We perceive digital as being higher quality audio. It is. That does not mean that we cannot make it sound bad. We can do that too.

Actual audio flow in one of our main concerns. Another entire aspect we are faced with revolves around our digital audio hard disk / automation systems. Here we get into the different aspects of data networking. A lot of the new efficiencies for our stations are found in this area.

There are three basic ways to transfer audio from one device to another. The first is real time analog playback and record. We have done that for years. Next is AES/EBU digital audio transfer. When we use AES/EBU we increase the audio quality, but still have to do the transfer in real time. The newest, most efficient way is to do direct audio file transfer. This method takes a digital file and transfers it much faster than real time. If we network our production room multi-track editor directly to our storage playback system, we get tremendous time savings. We no longer have to dub audio files into a different system. When the production room saves the audio file, it is instantly available on our on air workstation. This capability is not widely used in all equipment. It is worth a look for this type of capability to insure maximum efficiency for our staff.

There are two main styles of networking available. We have pier to pier and the central file server styles.

Pier to Pier networking is usually cheeper in smaller applications. This style requires storage space on each individual workstation. In larger applications, putting storage space in each machine can get expensive. Audio and scheduling information is transferred behind the scenes. Since priority is given to On Air audio playback, audio file transfer can take considerable time.

The central file server style starts out a little more expensive, yet can offer some very real benefits. When we need a lot of audio storage with distribution to multiple points simultaneously, this is the way to go. You can record an audio file one time and have it instantly played back in several places at the same time.

Connections for networking usually consist of one of two choices. Twisted pair 10 Base T or thinnet. In existing buildings where twisted pair wiring already exist, or in installations where the configuration is complex, we often use 10 Base T. In this style we use hubs for tieing together the different workstations before they go back to the file server. The hub cost makes 10 Base T more expensive, but it seems to be accepted more due to its wide usage in the existing computer industry. Thinnet uses standard RG58 50 Ohm coaxial cable with B&C connectors. This style is very easy to install. Most of the time it comes down to the type of networking offered by the manufacturer of the hard disk system. You need to review your particular application to determine which style you are comfortable with.

There are some other advantages worth our consideration that networking affords.

We can interconnect most any traffic and billing system and or music scheduling systems directly to the digital audio system. Today we also have manufacturers of these systems giving us the ability to do bi-directional file transfer with automatic discrepancy report generation.

We can integrate our news departments to our On Air facilities allowing instant news updates and truly paperless studios.

A lot of audio systems can give you an administrative workstation to allow engineering and management instant access to everything. Complete system security is also a must and can be administered from this workstation.

As we look at all of these changes and capabilities it can seem very confusing. We know the advantages of going to digital. We have to keep in mind that it is a lot of simple steps that when put together becomes complex. Break the overall stations requirements and systems down and determine the best way of individual interconnection. You feel as though you are covering new ground but, there are a lot of others that have been down the path already. Try to learn from their mistakes and use their experiences to help save yourself a little time and a lot of trouble.

TELEVISION FACILITIES DESIGN: THE NEWSROOM TALKS TO THE FILE SERVER

Tuesday, April 16, 1996
1:00 - 5:00 pm

Session Chairperson:
Marvin Born, WBNS-TV, Columbus, OH

HARD DRIVE STORAGE TECHNOLOGIES FOR THE DIGITAL NON-LINEAR EDITING FIELD
Bryan Robertson
Quantum Corporation
Milpitas, CA

INTEGRATING AN AUTOMATED ELECTION SYSTEM INTO YOUR FACILITY
Michael D. Rich
Media Computing, Inc.
Phoenix, AZ

A BROADCASTER'S GUIDE TO MANAGING FACILITY DESIGN AND CONSTRUCTION
Bice C. Wilson, AIA
Meridian Design Associates
New York, NY

***DIGITAL MAX**
Paul Berger
CBS
New York, NY

***IMPLEMENTATION OF AN ALL-DIGITAL TELEVISION PRODUCTION AND NETWORK DISTRIBUTION FACILITY**
Jay Adrick
Harris Corporation
Florence, KY

Matt Scalici
The Golf Channel
Orlando, FL

***INTEGRATING NEWSROOM COMPUTER SYSTEMS AND SERVER-CENTERED NEWS PRODUCTION**
Stevan Vigneaux
Avid Technology, Inc.
Tewksbury, MA

***FIBER CHANNEL HIGH PERFORMANCE STORAGE FOR VIDEO DELIVERY**
Bill Moren
Ciprico, Inc.
Plymouth, MN

*Paper not available at the time of publication.

HARD DRIVE STORAGE TECHNOLOGIES FOR THE DIGITAL NON-LINEAR EDITING FIELD

Bryan Robertson
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Abstract

In the last two years, hard drive manufacturers have made attempts at developing hard drives with features for the digital audio/visual market. These drives, marketed under the terms "AV Drive" or "AV Ready", were created through changes in the mode page operating parameters. This paper discusses the non-linear editing environments in which AV drives are used, what AV features are essential to each environment and what features are in development that will create higher performance, more cost effective drives.

Digital Editing/Animation Environments

The non-linear editing and animation markets cover several application areas including broadcast quality (D1), component (BetaSP), corporate or industrial video (JPEG), animation (JPEG), and multimedia (AVI, QuickTime). These markets constitute a wide range of performance and I/O expectations.

At the high end of the spectrum, broadcast quality (D1) uses uncompressed data rates of 20.95MB/s (sustained CCIR-601 full resolution) per stream. In coming years, HDTV will result in substantially higher data rates depending on the resolution.

Component editing, found in news rooms where the BetaSP format is common, uses either Avid Technology's AVR-26 or AVR-27 specifications at 3.5MB/s or 3.9MB/s respectively.

Corporate or industrial editing, which focus on JPEG, typically have a wide range of data rates. Typically in the 2MB/s for moderate quality JPEG to 10-12MB/s for high quality.

Animation workstations, used in news rooms or for special effects, typically use Motion-JPEG with transfers rates in the 1-8MB/s range depending on the quality used.

The multimedia market uses a variety of formats (AVI, QuickTime, etc) with data rates in the 450kb/s to 1MB/s range.

Market	Format(s)	Data Rate(s)
Broadcast	D1	20.95MB/s
Component	AVR-26 / 27	3.5-3.9MB/s
Corporate or Industrial	JPEG	2-12MB/s
Animation	JPEG	1-8MB/s
Multimedia	AVI, QuickTime	<1MB/s

Figure 1 - Non-linear editing data formats

Sustaining Throughput

The key in all of these areas is sustainability. In any application of digital video, the most important issue is the ability of the drive (or drive array) to sustain the desired throughput.

This is particularly true in composition applications where multiple tracks of video and audio data are being processed simultaneously. For example, assume the following:

Two video tracks @ 3.9MB/s each (AVR-27)
Eight audio tracks @ 176kb/s each (44.1khz)

The total transfer rate of all 10 tracks is about 9.2MB/s. In theory, any wide-SCSI drive should be able to deliver 9.2MB/s. However, in the aforementioned example the issue of latency is introduced.

Latency is, by definition, a delay. There are several areas in the operation of a drive which result in delays. The time required to execute a SCSI command is a delay. The time required for the drive to seek to a sector is a delay. There are lots of delays.

A hard drive is made from one or more flat disks called platters. These platters are coated with a magnetic material which, when polarized, provides electromagnetic peaks that are detected as 0's and 1's (bits). In general, each side of a platter is capable of storing data. A concentric circle of these peaks make a track. All the tracks in the same position (top and bottom) on different platters combine to form a cylinder. In each track, a series of bits that make 512 bytes (and some housekeeping bytes) is called a sector. Each platter used to record data has one read/write head and only one head can be used at any one time. Therefore, only one head can read one sector at any one time.

Data is read from the outside edge of a platter to the inside. Because more bits move under the outer data zone (OD) than the inner data zone (ID) the OD has better overall performance. Data read or written from sector to sector, track to track, in order is called sequential reading or writing. Anything else is called random reading or writing.

Now, in understanding how latency affects non-linear editing, consider how the data is read from the drive. A video file may be located in the outer data zones where transfer rates are high. The audio files that go with it may be located at the inner data zones where transfer rates are low.

The critical latency factor here is seek time, which is the average time it takes the head to get to the sector which is to be read or written. By moving back and forth across the disk (seeking), each file adding additional seeks, the overall time to transfer all files increases.

If the random seek time is 8.5ms, you may get no less than 85ms of latency just to read two video and eight audio files. The latency will result in lost frames or miscued audio. In the aforementioned example, if the video file is requested first and then the audio files, the drive will complete the request for the video file first. If there's not enough time to process the audio, it gets dropped or miscued.

Which brings one back to the question of how many drives are required to deliver the 10 streams. Is the answer 1 drive? No. It will take 2 drives, each of which will process half the data. It will take one drive per stream for the video data and four streams of audio data. Each drive would be required to deliver a sustained rate of 3.9MB/s for video and 704kb/s for audio, well within each drives performance level. This might still cause problems if the inner data zone performance is significantly lower than 4MB/s.

Additional latency occurs in track switches, the time is takes to get from one data track to another. These switches add 1ms for each switch. If the average track holds 128k of data, the number of switches is 30 for 3.9MB of video data and thus 30ms in additional latency. Overall, the drive experiences 115ms of latency if it tried to read a video file and all the aforementioned audio files. This is over 1/10th of a second. That's part of why two drives are needed.

Another aspect to consider when calculating the performance of a drive in this environment is sector mapping (also called reallocation). The typical AV drive today has one or two spare sectors per track of data. However, if those sectors are used, the drive will remap any new defective sectors to another track in the same cylinder. The time taken to seek to the new sector in a different track in the same cylinder is called Sequential Head Switch Time. This time adds 1ms or more to the overall time required to read the sector. If the data overlaps into another cylinder, the time increases to 1.5ms or more.

All of these delays are acceptable in the implementation of a normal PC or workstation. Since these systems typically depend more on CPU speed than drive performance, these delays aren't noticed. For AV applications they are a problem. To combat these design issues in editing applications, drive manufacturers have made some changes in how the drive operates.

Making an AV Drive

AV drives today consist of changes made in basic design and mode pages. The bulk of the changes have taken place in the mode pages. Mode pages contain information about how the drive operates in relation to SCSI protocols. Most settings can be changed by the user but most are done at the factory. The features shown in Figure 2 show the major changes that have been made by AV drive manufacturers.

Feature	Setting
Read Reallocation (ARRE)	Off
Write Reallocation (AWRE)	Off
Read Continuous (RC)	On
Read Retries	0
Write Retries	0
Hidden Retries	0

Figure 2 - Major mode page AV changes

The only change made to the hardware that is not reflected in the mode pages is the elimination of thermal calibrations (T-Cals). This feature is discussed later in the article.

Read and write reallocation have been turned off because of the aforementioned additional latency times introduced by mapping out defective sectors. This impacts data integrity however. By not reallocating defective data, you rely on ECC correction in the firmware to fill in where the media left off. In mission critical applications, this can leave you with gaps in otherwise perfect data. Also, even ECC correction takes time which may result in delays if done often enough.

Read continuous is a function in SCSI which tells the drive to read all the data requested by the system without regard for errors. Unless the drive encounters a hard error, a physical problem with the drive, the data will continue to be read. Again, this compromises the integrity of the data. This function doesn't effect writes though, only reads.

If a drive attempts to read or write a sector and fails, it will retry the read or write. The retry setting tells the drive how many times to make the retry attempt. The typical drive has up to 8 of these retries. Each retry is another rotation of the disk which takes .12ms or more depending on the rotational speed. Again comes the question of latency.

Hidden retries are retries embedded in the drives firmware. They are invisible to the user and can't be changed without changing the firmware. If a drive has these, it may still lose data due to the added retry latency.

Thermal calibration is the act of repositioning the heads to compensate for thermal distortion in the media. This keeps the heads aligned with the tracks to ensure data is read or written correctly. Embedded servo technology has replaced dedicated servo technology as a method of eliminating T-Cals.

Dedicated Versus Embedded Servo

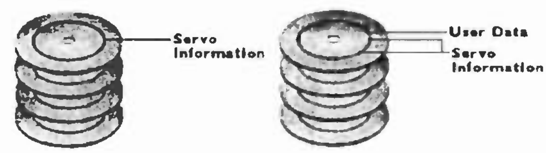

Figure 3 - Embedded versus dedicated servo

Embedded servos are special magnetic signatures located across the top and bottom surfaces of all the platters in the drive. By aligning the heads with these signatures during normal operations, the heads are always in the correct location, regardless of thermal expansion. Some drives still perform a T-Cal even with embedded servos, but do so when there is no data being transferred. Of all the aforementioned features, this is the only one which doesn't impact the integrity of the data.

Other changes made to current AV drives include modifying the prefetch settings and buffer full/empty ratios. Prefetching is when the drive continues to read data into the buffer ahead of what a read command requested. Since most reads are sequential, this improves performance.

The buffer full/empty ratios are concerned with how much data is read into the buffer. IBM and Quantum AV drives, for example, ignore these settings and adjust the values depending on the previous request size and transfer length. Drives that don't auto-adjust require these settings to be manually optimized for their intended environment. Auto-adjustment is ideal for AV since video and audio transfer sizes and lengths are different.

These features aren't the only way to improve delivery of AV data in non-linear editing environments. There are new technologies coming that improve performance without compromising data integrity.

AV for the Future

Making "AV Ready" drives for the future involves more than just patching existing technology. The next generation AV drives will be required to handle even higher performance requirements, greater capacity and do it all while ensuring data integrity. They will do this all by the end of 1996, if not sooner.

The features in Figure 4 show some of the improvements, some evolutionary, some revolutionary, that drive manufacturers are implementing in their drives to further benefit the digital non-linear editing market.

Feature	Benefit
Areal Density	Higher capacity
Zero Latency Reads	40% faster reads
Zero Latency Writes	40% faster writes
In-line sparing	Reduced latency
Fibre Channel	Wider bandwidth
Larger cache	Faster I/O

Figure 4 - Next generation AV features

The drive industry has been increasing capacity at an explosive rate. A drive which cost $800 five years ago and held 80MB of data now holds 4GB at the same price. The growth comes from increased Areal Density, the number of bits per square inch (see Figure 5).

The increase in areal density provides several benefits which eliminate problems found in existing AV or non-AV drives. The first benefit is that it decreases the number of disk platters required for a certain capacity. For example, a drive that once needed 10 platters to reach 4GB now requires 4. This cuts the number of heads 60% and reduces the chance of a head switch (a latency problem in existing drives) when reading a data stream.

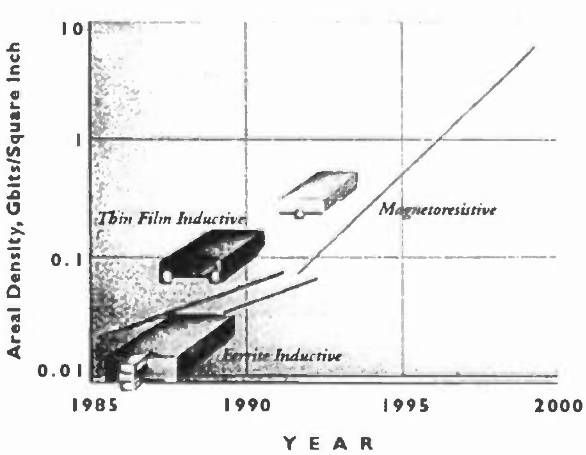

Figure 5 - Growth of areal density 1985-2000

The increased capacity also increases performance, an obvious benefit. By not having to switch heads as often, and by being able to read more data per track, the throughput goes up. This applies to the entire medium. This is essential when looking at the sustained transfer rate across all surfaces. The data rate at the ID is often substantially lower than the OD, enough so in existing drives that they are unsuitable for AV. Increasing areal density improves performance at the ID to a level that sufficiently exceeds current AV requirements.

Zero Latency Reads (ZLR) and Zero Latency Writes (ZLW) are methods of reading data in sequential sectors and eliminating head and cylinder switch times. Head and cylinder switching is a major latency issue in editing environments, especially in composite work.

In zero latency, the drive reads data into the cache as soon as the head reaches the target track (even if the first sector arrived at isn't the requested sector). When the target sector arrives, the read is processed with the chance that some of the data already in the cache is part of the read. This is based on the assumption that most data is stored sequentially. In AV environments where the data is primarily sequential, the data for the next operation is already in the cache. This provides a substantial performance increase depending on how much cache is available.

In-line sparing is another feature which decreases the chance of added latency. When a defective sector is found during an operation, most drives remap the sector to the last one or two sectors in the current track. This results in at least an additional ½ rotation of the disk to get to the new sector and read or write the data. In AV, this is unacceptable.

In-line sparing is a method of mapping defective sectors sequentially in a track and then renumbering all the other sectors in that track to reflect the new placement. For example, suppose sector 10 is defective. If the next available sector is 14, then sector 10 is marked bad and the data moved to 14. This differs from current methods in which the sector would be mapped out to number 150 (or some similar number) which is located at the end of the track.

The introduction of fibre channel drives will greatly improve overall performance beyond anything else implemented. With data rates of 100MB/s in the first generation, fibre channel drives will be able to support more complex editing functions like composition with fewer drives. In addition, the higher rates will allow editing with higher resolution formats such as D1 or HDTV, with fewer drives and at lower cost.

Fibre channel is an implementation of SCSI which provides improved throughput (up to 400MB/s by 1998) via a new serial communications architecture. This differs from the existing SCSI which is a parallel architecture. The new design allows for up to 126 devices per initiator (versus 8 now) over greater distances (up to 1km with fiber optics). This design means that a two drive array of fibre channel drives will do what a five drive wide-SCSI array does today.

The last improvement, increased cache size, benefits raw performance. Since cache sizes vary between vendors and drive models, no one cache size can be stated as better than another. However, the larger the cache size, the more margin for error in the rest of the read or write process. By having data in the cache at all times, the drive can still be delivering data even when an error occurs, and thus deliver uninterrupted video or audio.

Conclusion

Today's AV drives have proven valuable in advancing the reliability and performance of storage in digital editing. However, as integration of digital systems into all phases of the editing process continues, the need for improved performance and reliability will increase. In response to this, the drive industry is working on essential technologies and features which will make this integration possible. The key features of tomorrow's AV drives include increased areal density (capacity), zero latency reads and writes, the introduction of fibre channel, and increased cache sizes. All these features improve drive performance, lower relative costs and promise essential data integrity.

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INTEGRATING AN AUTOMATED ELECTION SYSTEM INTO YOUR FACILITY

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With the level and sophistication of automation at facilities increasing every year, it is getting harder and harder to integrate new systems with current systems. This paper is designed to help engineers become better skilled at identifying and resolving potential problems with automation systems in general and conflicts between/among "co-existing" automation systems in particular.

This paper is divided into three sections. The first describes steps that should be performed BEFORE the purchase of an automated system. The second section discusses techniques that can be used to help make the integration of multiple automation systems as seamless as possible to the users. The last section has suggestions on how to use your automated election system for other automation processes throughout the year.

WHAT TO DO BEFORE THE PURCHASE OF AN AUTOMATION SYSTEM

In essence, what you need to do is figure out what you already have and what you need to purchase. A good place to start is to take some time to assess your facility's current hardware and software.

Let's start with the broadcast equipment.

Take an inventory of all your broadcast equipment currently being controlled or, in the future, may be controlled by automation. Make a separate entry in your inventory list for each piece of equipment. Next to each entry, indicate if the equipment is

a) currently controlled by an automation system and if so, which automation system,
b) to be controlled exclusively by the new automation system, or
c) to be controlled by the existing AND the new automation system.

At this point, you're ready to make a list of the various functions you wish to automate for each piece of equipment.

Let's use VTRs for an example. Do you need to automate just transport control or is a higher level of sophistication required? A vendor (or you) might define transport control as PLAY, FAST FORWARD, REWIND, RECORD and STOP. Is that enough, or might you need the ability to position the tape to a particular SMPTE time code? Will you need the automation system to be able to detect the presence of a tape (for cassette machines) or tape threaded (for reel to reel machines)?

In the case of automated recording, can the automation system you're thinking about detect, and then act upon, the status of the record interlock? Do you need the ability to randomly pause recording and start up recording again with an assemble edit? Might you need to automate the recording of a program longer than the length of a tape? If so, you need the ability to overlap recording on two VTRs, and also need the ability to sync two VTRs for playback. Do you need redundancy, i.e., record on more than one tape machine at a time? What about the ability to detect and then act upon VTR failures? Etc., etc..

After you are done with the broadcast equipment, it is time to assess the computer systems. Take an inventory of all your computers. Make a separate entry in your inventory list for each computer. Next to each entry, indicate if the computer is

a) currently used by an automation system, and if so, which automation system,
b) to be used exclusively by the new automation system, or
c) to be used by the existing AND the new automation system.

Also next to each entry, indicate the computer's

a) operating system, including version,
b) amount and speed of each type of memory,
c) type, size and the manner in which each storage device (hard disk, tape drives, floppy disks, etc.) are connected to the computer,
d) a complete description of each peripheral device (keyboards, modems, printers, video displays, CDs, I/O cards, remote terminals, etc.).

If you have a computer network, it will also need to be inventoried, including

a) network operating system, with version,
b) manner in which the computers are attached to the network,
c) a full description of ALL equipment related to the network including model numbers and versions.

You also need to give some thought to redundancy - without it you could get into a lot of trouble. For example, if your Local Area Network server has a 10 cent capacitor short out in its power supply, in less than a second, you could lose communication on your LAN and permanently lose all the data on the server's hard disk! There are redundant power supplies available, redundant servers, redundant LAN connections, etc.

I like to define a fully redundant system as a system which does not have a single point of failure. In other words, there is no one item which, if it fails, could prevent your system from operating in a normal manner. The only major disadvantage to redundancy is price. The price benefit ratio decision is a tough decision to make - but it needs to be made.

Before you sign an agreement with your automation vendor, give some thought to how you plan to handle elections - what level of automation do you desire/need. How are you going to get election information from the outside world into your facility's system? How many automated ingesters will you require? How many manual remote inputs are needed? Once the information is in the system, how many people need to see the information in real time? How are you going to get the information on air? How many character generators and related equipment are going to be used?

One advantage to elections is the results typically do not start coming in until the evening. And by a nice coincidence, most facilities' heavy daily computer requirements are just that - during the ***day*** - not in the evening. You should be able to get "double duty" out of a lot of your computers. During the day they do one job, but on election night, the same equipment is used for a completely different application. A couple of our clients on election day, wait for the day staff to go home, then go all over the station and gather computers which they bring to "election central". They make floppy disks which log the computers on to the LAN as special election users and make sure they don't write anything to the local hard disk. They use the computers like crazy all night long and then return them to their rightful owners before they come in the next day.

Something else to consider - how will you give help to the users on the system? Will you use some type of an automated help desk? Can this help desk be augmented to help during the special case of election productions? Might the help be some form of intra-LAN communication? Might it be telephone support? Or might it be some combination of the above? In any case, keep in mind that you most likely will have to move computers and/or terminals around for election night coverage - plan for that possibility.

Now that you have a written inventory of all the equipment you wish to use in your automation system, including a detailed description of the functions to be automated, take this document, with the help of legal counsel, and physically attach it to your contract with the automation vendor as an exhibit. Have your counsel amend the automation contract so that the newly created exhibit is used to both clarify the automation work to be performed with the equipment to be used and as a test for proof of performance.

HOW TO MAKE THE INTEGRATION OF THE NEW AUTOMATION SYSTEM AS PAINLESS AS POSSIBLE

There are many different possible combinations and permutations of current hardware/software systems to be integrated with new hardware/software systems. Let's look at four cases and their possible integration solutions.

Case 1

Current system is a mainframe or mini computer system with dumb terminals to be integrated with a new system based on PCs in a Local Area Network (LAN).

Possible integration solution: Add to and/or replace some of the dumb terminals connected to the current system with PCs. The PCs are configured in such a manner that they actually have two physical data connections: one connection to the mini or main frame and the other connection to the LAN. The PC is loaded with some form of multi-tasking software (such as Microsoft Windows, for example) and runs two tasks simultaneously. Running in the first task is a dumb terminal emulator.

This emulator is designed in such a manner that it "appears" to the mini or main frame as the same old dumb terminal it is used to "talking" with and has a screen virtually identical to the one the operator is used to looking at.

Running in the second task is your new automation application. The operator taps what is known as a "hot key" to instantly switch between the mini/main frame emulator and the automation application. Note: NEITHER of the tasks are ended before switching to the other task - they can be put in a form of "suspended animation" or may even be configured to "run in the background". In that way, when the operator switches between tasks, they are brought right back to the place they left the task. By the way, you may be able to have both tasks displayed on the screen at the same time, if you so desire.

Case 2

Current system is on a LAN running under an old version of DOS and for one reason or another, the system cannot be upgraded to a current version of DOS; the new system is a multi-tasking system on a LAN using the latest version of DOS or Microsoft Windows 95, etc. and will not work with the old version of DOS you currently have.

Possible integration solution: Use some form of multi-tasking, as described above in Case 1, using the version of the operating system that comes with your new automation system.

When the PC is turned on and automatically starts the first task, it can also run a program call SETVER which can be configured to fool your current automation system into believing it is still running under the old version of DOS.

Case 3

Current system is not really a system but rather a bunch of unrelated PCs, each of which are running a different automation program physically connected to the equipment it is controlling and the new system is LAN-based.

Possible integration solution: Use a variation of the solution presented in Case 2 above, but rather than run the same application in the first task on each of the PCs, you would run the application which is specific to the hardware that is attached to that particular PC.

Case 4

Current system is LAN-based using Macs with a LAN topology NOT compatible with the new system which is PC-based. To make the integration more of a challenge, management has told you "not to replace the current LAN since it works just fine and don't spend money to fix something that is not broken!"

Possible integration solution: Most current Network Operating Systems (NOS), such as Novell's, are capable of running multiple computer operating systems simultaneously on the same LAN server. Most current LAN servers are capable of using one network interface card for one type of topology while simultaneously using another network interface card for a different type of topology.

Researching an election automation system can also make the transition from manual to automation less painless for the users. The more sophisticated election automation systems, such as Media Computing's ANGIS Pro, not only control multiple character generators of different models and/or manufacturers simultaneously, they can also control other related equipment. I'll explain what I mean using 2 examples.

First example - We have clients which have very good ratings during prime time and don't wish to upset some of their viewers by pre-empting regular programing. On the other hand, some of their viewers think the election is very important and should be covered in depth. The solution, the client rents one of the UHF stations in their market for election night and does the in depth coverage on the "U". At the same time, on the "V" they have brief cut-ins and updates. The CG used on the "VHF" station and the one used for the "UHF" station can be of different manufacturers, but the station need only maintain one election database for both types of simultaneous coverage.

Second example - The station's art director has decided to use different still store backgrounds based on the:

a. Number of candidates to be displayed,
b. Office they are running for (governor, mayor, etc.),
c. Winner projection for a particular race

Additionally, the character generator transitions will be based on the previous display (i.e., use transition A from governor's to mayor's race, but transition B from judge race to judge race).

With the selection of a more sophisticated election automation system, this is all possible. The system can "decide" which background is to be used and/or which transition is to be triggered using IF-THEN-ELSE logic based on dynamic data contained in the database. Since computers can literally send out hundreds of keystrokes a second to the devices being controlled, a quality automation system is capable of doing functions which are humanly impossible or at least EXTREMELY difficult for a human!

You know once you have two "unrelated" automation systems, someone in management on one of the systems is going to want information which is only available on the other system. So now what do you do?

Look for something both systems have in common, i.e., can one system send the information out a communications port such as RS-232 or RS-422 and the other system able to input information via its communications port? Do both systems have a file format in common? Etc.

Here's a good example to help make the point. One of our clients has a Mac-based scheduling system put into service years before we installed a custom automation system. It appeared the systems had nothing in common, until we discovered that the scheduling system can print a copy of the schedule. Well, it turns out that when the Mac prints, it uses an RS-232 port. So the way we got the Mac system to "talk" to the PC-based system is by having the Mac system "print" a report and the PC system "look" like a printer to the Mac system.

Another problem you might run into with two automation systems is a lot of broadcast equipment, in general, and many character generators, in particular, are designed to be controlled by one and only one "master". One way to handle this situation is to install a computer running arbitration software having both automation systems as inputs and a single output to the character generator. The arbitration software keeps track of which commands come from which automation system, so it can return the CG's response to the proper automation system.

HOW TO USE YOUR ELECTION SYSTEM FOR THE DAILY AUTOMATION OF MANUAL TASKS

The more sophisticated election automation systems are capable of being used daily for non election tasks. In the case of weather, you can have your graphic artist design character generator templates, assuring the use of the proper station look (correct fonts, colors, drop shadow, etc.) during regular business hours. Then, during early morning news or late night news, your regular weather people can sit down at a computer terminal and enter the current weather information by answering simple prompts like "current temperature", "today's high", "wind direction", etc.

The same technique used for weather can be done for national sports. But how do you handle local high school sports - which can involve scores for forty teams or more? Well, unlike a lot of character generators which are operated in a single user mode, an election system is designed to handle multiple, simultaneous inputs. So you can have multiple people in the sports department entering scores with the help of interns and/or administrative support staff.

Another advantage of using an election system for your sports scores is that someone sitting at a computer terminal can update the latest score, even if that score is on air while it is being updated. This eliminates the sportscaster from being put in the embarrassing position of having to say "the score on the screen is wrong, the latest score is...".

You can also use your election system for telethons. As far as an election system is concerned, there is very little difference between counting votes and counting dollars - it becomes a one candidate race without percents.

With an optional telephone call counter system added to your election system, you can poll your audience and show instant percents with bar graphics, i.e., dial this number if you agree or that number if you disagree. Remember, this all can be done without the necessity of having a graphic artist available after hours.

With optional automated wire interface software, you can further automate many of the tasks described above by stripping out the desired information from the wire service(s).

With some research, you will be able to find an election system capable of doing non election tasks at no additional cost - they are out there, just look for them.

A BROADCASTER'S GUIDE TO MANAGING FACILITY DESIGN AND CONSTRUCTION

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ABSTRACT

Tremendous change in the Broadcast Industry is necessitating the design and construction of many new facilities. Many organizations lack staff with experience in leading and managing such an undertaking. This paper will serve as a primer enabling both staff and management to administer and lead the process with confidence. We will walk through a typical project, describing crucial tools and processes, and highlighting classic pitfalls.

THE CHALLENGE

It's not enough that your staff has been cut, your budget is tighter every year, and you're working overtime to integrate ever changing technology into your facility while struggling to keep the old stuff going for just another year. On top of it all, your lease will soon be up and it's time to relocate the entire station operation to a new leasehold.

Since you have been managing most of the major purchases and capital projects for the station, you are nominated to manage the move. In your spare time. Spare Time?

Where do you begin?

You may be a General Manager, a Program Director, Operations Manager, or a Chief Engineer. These days the in-house project leader for a construction effort can be any of these. In any case, you are being called upon to take responsibility for a major transformation in the shape of your organization during a period of tremendous change. Chances are that you are not trained in such undertakings. Chances are that, after the latest staffing cuts, there is little or no institutional memory left in your organization which can give you the benefit of past experiences.

The good news is that there are several proven methodologies and decision making processes which you can use to assure the most beneficial outcome of the project ahead.

In this paper we will address several areas: how to set up the project, how to look for real estate, how to manage budgets and schedules; and we will offer some insights into the design process.

SETTING UP THE PROJECT

The Construction Enterprise

We look at a major capital project as an ephemeral, intrapreneurial new business start up. Ephemeral because this business is intended to last just long enough to make one product. Intraprenurial because it is an in-house high risk undertaking. New Business because its mission is substantially different from the core business of your company. And at the outset it is very much a start up.

The mission of this business will be to plan, design and build your new facility. Your job is to be the CEO of this business. You need to build a staff, and you need to devise a business plan.

Staffing the Construction Enterprise

The staffing of the project will include a Board of Directors made up of representatives of the various departments of your organization. The Chairman of that Board will typically be your corporate manager. The Board's role will be to establish policy and provide guidance to shape the day to day actions of the line staff, what we call the Project Team.

The rest of the team will typically be consultants and contractors hired for the project's duration. These consultants are brought on board to leverage your talents and those of the form givers in your organization. The best product will often come if these consultants have an ongoing "partnering" relationship with your company.[1] In this way it is sometimes possible to outsource the maintenance of an institutional memory.

A team which has experience in projects like yours can bring considerable added value to your project. They can offer the benefit of their long-term observations of the pattern of change in the industry and the benefit of their ongoing research into design solutions expressing this change. If no complete team with these qualifications is available in your region, then it is critical that at the least you find an architect and a broadcast systems designer who can bring such experience to your project.

The key team members include your architect and consulting engineers for structural and mechanical/ electrical engineering, your real estate agent, your real estate lawyer, your broadcast systems designer, and your construction manager or general contractor. Each of these representatives will speak for the interests of their aspects of the project. One of your most challenging tasks will be to guide and assist your architect in his role of balancing these sometimes divergent interests into one coherent vision for the future.

Delegate - You Still Have a Full Time Job

Don't forget that you will probably continue to have your traditional responsibilities within the company throughout this process. Your success will depend on your ability to articulate your needs, and then to manage the efforts of others in bringing them to fruition. One key role you will play is as the liaison between the Project Team and your in-house "Board". It is essential that you keep your board involved and maintain their support and ownership of the evolving product through regular project reviews.

The Importance of Writing a "Program"

The "business plan" of the Construction Enterprise is what's called your "Program". The core of your program is based on translating your organization's business plan into a statement of the physical implications of how your organization does business. Certain technical aspects of this document will be generated by your consultants. By examining your business plan, they will establish performance criteria for spatial area and volume, mechanical and electrical systems, project budgets, schedules, and lease criteria, among many other considerations.

The majority of our past clients have not had a current business plan at the outset of their projects. Your design professional must therefore have the ability to lead you through a policy making process which will establish the success criteria for the project.

One key piece of our project delivery methodology is the knowledge that if you carefully define the concepts which will establish the success criteria for the project, the details will follow. If you fail to build broad consensus on these success criteria, or don't get participation from the full spectrum of interests at the outset, the odds are stacked against your project. There's an old carpenter's dictum - "Measure Twice, Cut Once." It is in devising your program that you take the measure of your organization. This work establishes the "genetic code" which will inform all the myriad decisions required on your project.

This effort requires more than just listing physical criteria. Just as a business plan must articulate the values and principles which underlie the sales goals and product definitions, so your program must be qualitative as well as quantitative.

What kind of working environment are you committed to creating for your staff? What is the image you wish to project to visitors? Is your organization hierarchical or horizontal? Will every-one get an office? No one get an office? What have you learned from your existing facility that you do (or don't) want to replicate? How do you see your organization ten years from now? (OK, try three years.) How will your organization take advantage of this opportunity to transform itself and integrate change in this era of change?

Establishing Communications Protocols
As you begin to understand the complexity of the process you are undertaking, you'll also begin to appreciate the need for for clear lines of responsibility and communication. The Construction Industry has many standard means of communication - Programs, Meeting Minutes, Proposal Requests, Change Orders, and many more. There is no need to reinvent the wheel. There is every reason to avoid recreating the Tower of Babel. What is needed is to delegate responsibility for managing the flow of this information. It is also important to document which team member carries each of the many responsibilities which must be coordinated for a successful project.

Other tools which are becoming invaluable are e-mail and a standardized menu of software which is commonly used among the team. Virtually all the products of our firm are at one time or another electronic files. The ability to hang them on a piece of e-mail and instantaneously distribute them to all members of the project team for development and feedback greatly expedites our work.

THE TYPICAL PHASES OF A PROJECT

A typical construction project has seven phases:

- Pre-Design
- Schematic Design
- Design Development
- Construction Documents
- Bidding and Negotiation
- Construction
- Commisioning and Debriefing

In the **Pre-Design** phase, you will assemble your team, write your program, establish a preliminary budget and schedule, and begin the real estate search.

In **the Schematic Design** phase, the team will explore schematic design options for potential spaces, assist in negotiating leases, and eventually agree on a schematic design for a chosen space.

In the **Design Development** phase, the approved schematic design will be fleshed out, exploring the various systems and finishes in greater detail, maturing the design and tightening up your grasp of a realistic budget and schedule.

Once the design is matured, you move into the **Construction Documents** phase. During this period, the team will convey the approved design in an industry standard format including drawings and specifications. These documents will be used to bid and purchase the project in the marketplace, and will guide the contractor in the construction phase.

During the **Bidding and Negotiations** phase, your construction documents will go out into the market-place, and a market price will come back. Your team will assist you in adding the final members of the Project Team - the Sub-Contractors who will actually build it.

The **Construction** phase is largely self explaining. Sometimes time is saved prior to this phase by the pre-purchase of long lead time items such as HVAC equipment, light fixtures and imported materials.

One often neglected phase is **Commissioning** . In this phase, systems are started up and tuned, and the members of your organization are oriented to the new space and its various systems. As-built drawings and operating manuals are archived, and maintenance contracts are initiated for systems requiring them. This period is also an opportunity for debriefing and disbanding the staff of the Construction Enterprise. Much can be learned in this debriefing process which will add to the institutional memory of your company.

THE REAL ESTATE SEARCH

Your Program is an invaluable tool in the search for new real estate. It is amazing how many clients come to us with a signed deal, but without having generated a program by which to judge the suitability of their chosen leaseholds. The program is the means to filter out only those possibilities which truly suit your organization. Just the process of asking all the questions involved in writing the program will prepare your intuition for judging. Another common mistake is the selection of space strictly on the basis of lease cost and square footage. All leaseholds of a given size are not equal. It is essential that the professionals on the project team assess each building to see whether it offers the services required, such as power and acoustical separation, and whether it accommodates the most effective groupings and adjacencies of departments within its given configuration. The hidden costs related to bringing these services into line with your needs can make a huge difference in the viability of apparently similar leaseholds.

It is also essential that your architects do a preliminary layout of any space which becomes a serious contender.

THE MULTIPLE CONTRACTS IN ANY PROJECT

A single construction process typically has multiple sets of contract documents for the various aspects of the work. Each of these has certain appropriate design professionals and contractors. A list of typical contracts sets, with their related professionals and contractors, is included in Table 1.

Contract	Design Professional	Contractor
General Construc-tion	Architects and Consulting Engineers	General Contractor
Office Furniture	Architects, Interior Designers	Furniture Dealer
Broadcast Furniture	Architects, Broadcast Furniture Fabricators	Broadcast Furniture Fabricators
Broadcast Systems	Broadcast Systems Integrators	General Contractors, Electrical Contractors, Broadcast Systems Contractors
Hazardous Materials Removals	Specialized Consultants	Specialized Contractors

Table 1. Typical Component Contracts

Clearly the weaving together of all these disparate contracts is one of the largest challenges facing the project team. In the past this was often the responsibility of an in-house facilities management staff. Another person who commonly carries this responsibility is the Architect. One metaphor which has traditionally been used for their role is as the conductor of an orchestra, keeping the efforts of the various segments in balance.

A new industry is developing to replace the in-house facilities staff - the Project Manager. While not licensed professionals, these consultants often bring a semblance of an institutional memory of the construction process to the table.

MANAGING YOUR BUDGET

Your program is also a critical tool in laying the groundwork for your project budget. Budgeting is one of the most mystified aspects of the construction industry. With a good program and a little diligence, there is no reason for this mystery.

1. Establish costs for your core program.
2. Cost out unique site and project specific aspects.
3. Keep room open in your budget for special items, identity.

Table 2. The Steps to a Budget

The first step in generating a realistic budget is to use the square footage numbers generated by your program to project a basic budget based on square foot costs. These costs are quite knowable for any given marketplace. Your design professionals and general contractors will be able to help you establish numbers for your project. Some rules of thumb for different kinds of space are indicated in Table 3.

The next piece of your budget building process is to identify costs unique to your project or the building under consideration. Your professionals will know which of these costs are included in the square footage costs used for your project. Some typical unique costs might include the items listed in Table 4.

TYPE OF SPACE	TYPICAL COST RANGE
Premium Office Space (Conference Rooms, Reception areas and other high finish areas)	$80 - $125 and up
Typical Office Space	$30 - $60
Technical Space (Studios, Rack Room, Mechanical Rooms)	$150 - $250

Table 3. Costs of Space

The synthesis of your base square-foot-cost-based budget and these unique costs should provide a reliable preliminary construction cost.

This method will help you to responsibly plan for special design features which are outside typical construction costs. These features, which are often the first victims of a budget crisis, can be monitored and maintained in the budget when you have confidence that your budget is on target and that you know their real costs.

Acoustical Separation to Neighboring Tenancies
Adequate and Suitable Power
Emergency Power
Hazardous Material Removal
Rectification of Existing Conditions not Complying with Building Codes
Special Program Requirements not Expected Under Typical Square Foot Costs
Costs for Antenna Systems, Cable Risers, Grounding System Risers
Structural Reinforcement Required for New Construction
Special Construction and Finishing Sytems

Table 4. Examples of Unique Costs

Once your budget is established, it becomes an ongoing tool in the design process. As you proceed with design, each decision must be analyzed in its light. Typically the budget is updated as you move through each phase of the project. If you find yourself discussing any new unique conditions, make sure to add them to your list and establish a cost for them. It is also essential to continually question past budget decisions, looking for vestigial items which are inconsistent with current thinking.

It is often healthy to begin a budget pruning process about 80% of your way through the design development phase. At this stage you should have a good sense of where the overall budget is going, you have typically accumulated a few vestigial items, and it's usually about time to begin the process of slowing down the momentum of adding cost to the project.

As you move into the bidding and purchasing process, it is important to realize that there will be some things you cannot anticipate, or which you know are distinct possibilities but cannot yet quantify. It is essential to maintain a contingency fund outside the contracted construction cost. You can also control these late breaking costs through the use of Alternate Bids, Unit Prices and Allowances.

Alternate Bids involve the development of alternate designs and the solicitation of pricing for both alternates, allowing you to select between them at your leisure. Unit prices can be requested from each of your trades and establish a fixed basis for pricing additions to and deletions from the project scope. Allowances are essentially contingencies built into the contract cost. They are useful for such things as access doors, fire stopping and other costs which are typical but unquantifiable until the project is underway. What is not used of such allowances is bought back out of the contract when all is said and done.

One strategy which we have used with great success over the years is to bring a General Contractor on board early in the design process and to give them the responsibility to provide ongoing market feedback for the evolving design.

SCHEDULE MANAGEMENT

Schedule Management ranks right behind costs in terms of fear and mystification for clients. It is amazing to us how many clients and contractors have effectively abandoned hope of managing their schedules. The first step is for the team to commit to jointly managing the schedule.

Scheduling, like budgeting, must begin the moment you decide to move. Each policy and design decision must be analyzed in light of both schedule and budget. One scheduling issue typical of broadcast projects incorporating so many component contracts is the need to weave together the milestones and critical paths of the various component contracts. This is typically the responsibility of your in-house project manager, your architect and your general contractor.

The litigious way to manage schedules is to use liquidated damages, which penalize contractors for missing deadlines. This strategy creates a huge amount of hostility and paperwork for all members of the project team. Every delay by every party needs to be documented so that when delays happen, responsibility can be apportioned.

Our preferred, Partnering-based way to manage schedules is for the project team to work jointly to establish a base schedule and for them then to monitor any changes at weekly project meetings. They can jointly develop strategies for minimizing negative impacts from problems which arise. The responsibility for leading this effort usually rests with your architect until the bidding and negotiation phase and moves to the general contractor thereafter.

There are two basic components in a project schedule: the critical path of design and construction tasks which must be accomplished, and the planning for long lead times related to certain items and materials.

An example of the critical path of tasks is the need to create and refine coordination and shop drawings for each of the trades which must go up above the ceilings before any of those trades can be released for fabrication and installation. Another classic limitation on completion of acoustically sensitive spaces is that there are many layers of construction and only so many people who can fit into a given room at one time.

The second component of schedule management is long lead-time items. At a very early point in the design process it is essential to identify any long lead-time items and plan ahead for their acquisition. It is common that some of these items are purchased before the bidding phase if a contractor is on board or if your organization has the sophistication to do so. Some typical long lead items are listed in Table 5.

HVAC Equipment
Electrical Switchgear and Power Systems
Anything Imported
Light Fixtures
Custom Fabrications such as Casework
Acoustical Specialties
Furniture

Table 5. Typical Long Lead Items

At the end of the bidding and negotiation phase, it should be the general contractor's responsibility to establish a project schedule which they are committed to administering. Any changes in that schedule must be accomplished as Change Orders, just as changes in project cost are accomplished by Change Orders. As project scope is changed, it becomes the GC's responsibility to negotiate changes in schedule as well as cost. If this method is diligently followed, the team will have a solid sense of a realistic schedule throughout the project.

Several classic pitfalls of scheduling include:

- Not allowing adequate time for the client's in-house review of various decisions. It is essential that the client commit to limitations on these periods if the schedule is to be knowable.
- Not all weeks are equal. December and August are half months at best. The week of the NAB is always a hard one to get clients to make decisions. Plan for these lapses.

- A failure on the part of the client organization to build a healthy consensus on policies as they develop. This often gives rise to unanticipated eruptions of design change at random points in the process.

BALANCING TIME, QUALITY AND COST

Figure 1 shows one of the key management challenges in the design process - finding an appropriate balance among time, quality and cost.

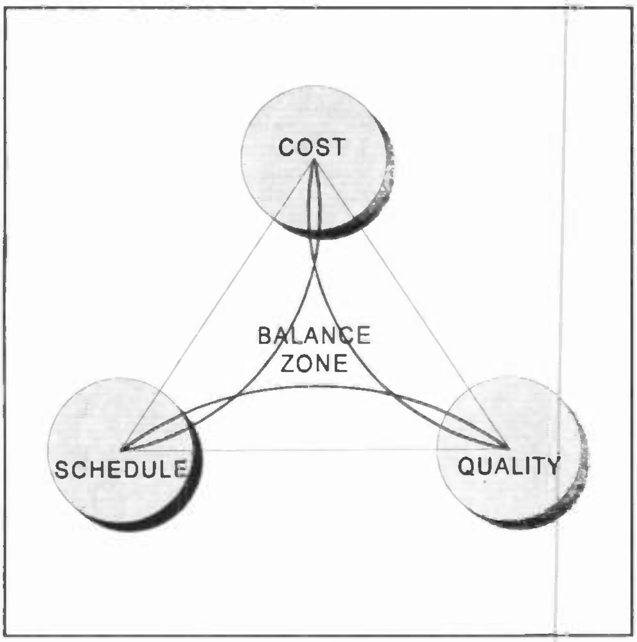

Figure 1. Essential Project Parameters

It is relatively easy to achieve any two of these three. If you want to control costs, you often have to choose between schedule and quality. If you're in a hurry, you sometimes have to choose between meeting your budget and your quality standards. If quality is crucial, it will cost you either time or money

One of the arts in these undertakings is keeping your project in balance while sacrificing none of these aspects.

THE DESIGN PROCESS

We will end this paper with a few observations on the design process. Some aspects of this process are wonderfully empirical: How many staff members must be accommodated? How big is a given piece of equipment? Others are subjective and situational: What color scheme will best enhance your business? What feeling should your reception area convey?

The infrastructure of a design is generated by those empirical form givers. The empirical then defines the potential for the subjective.

It is the subjective, the ideas and ideals expressed in a design that gives it vitality. There's a biblical saying - "Without vision the people perish." If the design of your facility is to bring the fullest benefit to your organization, it must also transcend the merely empirical needs and express something of the spirit of both the place it occupies and the organization itself.

Lastly, trust in the fact that design is a process. You won't know all the answers until it is over. There will be some issues which you feel you've addressed, but which keep popping up asking for attention. That is the process telling you that the issue needs further resolution.

Your guides to this process are your design professional and the other members of your project team and the methodology they will bring to their work.

SUMMARY

The process of design and construction requires you to question fundamental issues about the shape of your business - even if your only decision is to faithfully replicate exactly what you currently have.

As the in-house project manager, your most essential roles will be:

- To leverage your time and abilities by careful selection of the members of your project team.
- To ensure that your organization focuses its talent on the creation of a program and the definition of success criteria for the project.
- To shepherd the consensus building process as policies are established and design solutions are taking shape.
- To assist the project team in making value judgments balancing budget, schedule and quality.

We hope that the insights offered by this paper have helped to demystify the process and that they will enable you to garner benefit for your company from any construction projects you manage.

A list of references you might find valuable for future study follows.

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EUROPEAN BROADCASTING UNION: DIGITAL VIDEO BROADCASTING IN EUROPE

Wednesday, April 17, 1996
9:00 am - 12:00 pm

Session Chairperson:
George Waters, European Broadcasting Union, Geneva, Switzerland

OPEN, INTEROPERABLE AND MARKET DRIVEN THE DVB PROJECT
David Wood
European Broadcasting Union
Geneva, Switzerland

DIGITAL TERRESTRIAL TELEVISION IN EUROPE THE DVB-T SPECIFICATION
Ulrich Reimers
Technische Universitat Braunschweig
Braunschweig, Germany

***DVB RECEIVERS**
Hopes and Realities
Helmut Stein
Nokia Consumer Electronics
Dusseldorf, Germany

***THE DVB APPROACH TO CONDITIONAL ACCESS**
David Cutts
France Telecom/DVB
London, United Kingdom

EUROPEAN PATH TO DIGITAL TELEVISION THE MULTIMEDIA DIMENSION
Sven Olof Ekholm
Sveriges Television AB
Stockholm, Sweden

*Paper not available at the time of publication.

OPEN, INTEROPERABLE AND MARKET DRIVEN: THE DVB PROJECT

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Abstract

In 1993, a cross industry group, the DVB Project, was establish to agree specifications for digital television broadcasting. The work of this group is well advanced, and specifications for satellite, cable and terrestrial digital television are now available. The paper explains background to the project and gives its objectives and future plans. The DVB systems are in use in a number of parts of the world, and have been submitted to the DAVIC Project and the ITU-R and ITU-T for world-wide standardisation.

Introduction

The DVB Project is a major success story. It will provide the television future for many countries of the world, possibly for the next fifty years. A confluence of the right events, and the right history, made it happen. The result is a unified set of specifications for tomorrow's digital television broadcast systems.

The project aimed at developing future-proof but practical systems for digital television broadcasting. The three by-words of the Project have been the same since the project began - open, interoperable and market-driven.

The systems for digital television broadcasting need to be **open** in the sense that the specifications are published, and they are available to all manufacturers to produce without prejudice.

They must be **interoperable** in that they should use the same general architecture and baseband coding, transport, and error correction schemes for all the different media delivery means. This means that much of the receiver circuitry will be common to the different delivery media (satellite, cable, terrestrial, etc.). The systems should use the maximum number of generic elements and, only where unavoidable, application-specific elements.

The systems should be **market-driven** in the sense that they should accord with what is likely to be saleable, rather than with what engineers think the public ought to be interested in.

There are currently about 200 Member organizations in the DVB Project. Most are European, because of its history. But a good number now come from the Far East, North America, and Australasia. The DVB Project Office is organized by the EBU from its Headquarters in Geneva, Switzerland; and the DVB Steering Board is managed by staff at the Federal Department of Communications in Germany.

The DVB Project is organized in a tree structure. There are groups (Modules) which deal with commercial matters, and a Module dealing with technical issues. Sub-groups reporting to the Modules tackle the more detailed questions. The Modules report regularly to a Steering Committee.

The DVB process is that , first, commercial groups decide what - broadly speaking - the systems need to do. They then pass the baton to the technical groups to develop the solutions. Finally the Commercial groups check what has been developed, to see if it really does what they want. This formula has been used with each of the delivery system specifications developed to date, and the consensus view is that it works well.

If you don't learn from your mistakes

The DVB Project began in Europe. This may have been lucky. Europe has arguably made more mistakes in the past than many other regions about television standards development. Now at least it could learn from them.

The continent was split with PAL and SECAM in the nineteen sixties - without the European public really ever being able to see a serious difference in picture quality. In the early eighties, the MAC system was developed for satellite broadcasting. It was a technical triumph, but designed without enough regard for the non-technical worlds of costs, politics, and promotion. The HD-MAC system was also at the cutting edge of technology in the second half of the eighties. But it took too long to develop, and lacked involvement by enough programme service providers.

When the DVB project started in 1993, at least it didn't have to repeat the same mistakes.

The DVB plan was to develop, in a single organizational structure, a set of future-proof but pragmatic specifications for digital television, covering the range of delivery media.

The systems are made up of **generic** (or common) components and **application-specific** components. The main application specific component is the modulation system, because this needs to be tailored to the particular characteristics of the delivery media.

The systems considered are given below:

DVB-S
A satellite broadcasting system for the 11/12 GHz bands

DVB-C
A cable transmission system focused on 8 MHz cable channels

DVB-CS
A transmission system for SMATV.
DVB-T
A terrestrial broadcasting system for the UHF bands.

DVB-MS
An MDS broadcasting system (>10 GHz).

DVB-MC
An MDS broadcasting system (<10 GHz).

It clearly made sense to use existing worldwide standards for the system components wherever possible. Where they were not available, they had to be devised, with the intention of encouraging the rest of the world to use them too.

The universally applicable parts of each system, developed in the project, are given below.

DVB-SI
The Service Information system

DVB-TXT
A Fixed-format teletext delivery system

DVB-CI
The Common Interface for use in Conditional Access and other applications

The specification for an Interaction Channel for interactive television is now being prepared.

The Container concept - the key to future proofing

Each of the delivery systems is designed as a data container which can carry any type of data the service provider wishes.

Nevertheless, the baseband coding systems will normally be MPEG (audio) and MPEG-2 (video).

The MPEG systems presented the Project with what it needed at exactly the right time - the most advanced digital compression system available, for both sound and pictures, and furthermore a system which had worldwide agreement.
This meant that the two projects, MPEG and DVB, could feed off the success of each other, to stimulate even more the manufacture of the integrated circuits needed for DVB receivers.

The same circuits would go into all sorts of audio-visual products, thus ensuring lowest costs and widest availability, coupled with the most advanced digital compression techniques available.

The project has issued guidelines for Europe on minimum requirements for first generation satellite and cable receivers. It will probably do so also for terrestrial receivers. These are not standards, but identify the minimum elements, taken from the MPEG toolkit etc., needed for first receivers.

The guidelines specify the conventional quality member of the MPEG family. If, in future, there is a market for HDTV, or even some other type of service, this can be included in the receiver guidelines.

The market for new technology

The need for a market-driven approach was the result of burned fingers. The engineers now realized that, before designing a new broadcast system, it was necessary to decide what the system should do for the public, and how much it should cost, to be successful.

The price factor is critical. Some years ago, the EBU carried out some market research to evaluate the public reaction to high-definition television.

HD-MAC pictures from an Olympic Games were displayed at various public sites throughout Europe. We found that, in spite of the large public appeal of HDTV for sports programming, there was still a relatively low ceiling on the price that viewers would be prepared to pay for equipment to receive this new technology. This ceiling price was, in fact, well below the likely retail costs of HDTV receivers at that time. It was clear that the hour of HDTV broadcasting had not yet come.

At the start of the DVB Project, there was considerable experience in the UK, France, and Germany on the manufacturing and marketing of analogue set-top boxes for satellite reception. This experience gave clues to the amount the public might be willing to pay for a much wider choice of channels.

The known public appetite for an ever-increasing choice of channels, coupled with the prospect of a low-cost set-top box which would work in conjunction with an existing television receiver, gave much impetus to the Project.

One of the other lessons learned from the past was the need to shorten dramatically the time taken to market a new system. The development of a new system needs to be done in months instead of years, if the risk of being overtaken by new technology - before the product even reaches the shops - is to be removed.

Another critical factor is that all actors in the programme value added chain must be consulted in the development of a new system. They must all believe they have something to gain by moving to the new system. The content provider, the programme service provider, the carrier, the consumer electronics manufacturer, and the viewer, must all be 'on board' and convinced they have a lot to gain by moving to the new technology.

Development priorities

When the Group was first set up, it seemed that the pathfinder digital television system would be based on terrestrial transmissions. However, it soon became clear that this was not to be the case. For many reasons, digital satellite television was going to lead the way.

The world of terrestrial television broadcasting is a very complex one, there are frequency congestion and cohabitation problems to consider. Terrestrial frequency channels are often regarded as a precious national resource, and their use is bound up in legislation and governmental decrees. Finding the channels to broadcast digital terrestrial television is a major headache, almost everywhere.

Furthermore, the practical environment of terrestrial television broadcasting is a complex one. It is a world where the public expects, almost universally, to receive services via a rooftop antenna and, very often, via just a "rabbit-ear" antenna as well.

To develop a digital terrestrial television system was certainly a key target for the DVB project. But it was going to take time and effort.

DVB-S - the first specification

Bearing the above in mind, it was clear that the digital terrestrial system should not be the first job to be tackled. The first target was digital satellite broadcasting - DVB-S. The system required here was an easier one to develop, and to introduce from a legislative viewpoint.

The satellite specification was begun and finished in about six months. The DVB-S system combined MPEG coding and transport with a straightforward QPSK modulation system. The data capacity is selectable, according to the transmitter bandwidth, transmitter power, and the target antenna size. The target shop price for the set-top receiver box was 450 USD.

In this system, a popular combination for Europe of error correction rate and transponder bandwidths leads to a digital delivery package, running at about 39 Mbit/s. This can carry between six and ten conventional-quality television channels - depending on how critical you are about quality.

DVB-C and DVB-CS - the partners of DVB-S

A cable partner for the DVB-S system was clearly important. It seemed most efficient to design the system to be capable of taking all the services provided by a single satellite transponder and transcribing them on to a single terrestrial cable channel. The DVB-C system, centred on the use of 64-QAM modulation, was developed.

Southern Europe has a preponderance of localized cable systems. These are called SMATV systems and are distinct from cable networks which cover large areas - CATV systems.

After extensive practical tests on SMATV networks, the SMATV system was agreed. This effectively allows re-transmission on the SMATV network of either the DVB-S or DVB-C formats, depending on the operating environment of the SMATV network.

The technical characteristics of the DVB-S and DVB-C systems were delineated after extensive practical trials within the EuroImage Project. The results of which are freely available.

DVB-T - the terrestrial system

By now, it was the turn of digital terrestrial television to take centre stage within the DVB Project, although research into the best technical parameters needed had been continuous over the previous two or three years

Digital terrestrial television did indeed prove to be a complex technical issue. A range of technical parameters have to be chosen. The final choice will have an impact on the receiver costs, its capabilities, and when it could be in the shops.

Some parts of the world need a digital terrestrial system earlier than others, and some countries have more acute frequency-congestion problems than others. This has led the DVB Technical Module to conclude that the best outcome for digital terrestrial services would be to allow for two options for the modulation system:

2k OFDM plus QAM

This option is aimed at those who need to start services very early; it uses a less-sophisticated modulation scheme than the second option;

8k OFDM plus QAM

This more-sophisticated option is aimed at those who need large-area Single Frequency Networks (SFNs).

Receivers that are designed to receive the 8k/QAM system will also need to be able to receive and interpret the 2k/QAM broadcasts.

Conditional Access

The DVB project has discussed the subject of Conditional Access at length. A package of measures have been agreed. They include a Common Scrambling algorithm (the 'Superscrambler'). A Common receiver interface, based on the PCMCIA interface has been developed, which can be used to connect different proprietary CA sub-systems to the same IRD.

Moving forward

A worldwide standards group, DAVIC, was formed some time after the DVB Project was launched. We could be forgiven for thinking that this may have been prompted partly by the success of the DVB Project.

DAVIC is drawing together - and specifying where necessary - the systems for all types of audio-visual delivery media, including broadcasting. Several DVB proposals have been successfully submitted to DAVIC, and the ITU.

The fundamental formula of the DVB systems is to use MPEG Baseband Coding and Multiplexing, RS and, where needed, Convolutional Code FEC, coupled with a modulation layer that is the best fit the technical and economic requirements of the service.

Have we done all the right things, at all the right times? Inevitably not. But the record will show that, overall, it was a success, and an effective vehicle to launch the digital television revolution in several parts of the world. It will bring benefits to the consumer electronics manufacture in terms of component costs and market size. It will bring benefits to the programme service provider in terms of maximizing the supplier base and providing for a fast take off.

All that will add up to considerable benefits for the viewer, in terms of cost and choice. This is not unreasonable. After all, in the end, he or she pays for everything.

Ackowledgements

Very many engineers and others have contributed to the success of the DVB Project and it would be impossible to name even those who most deserve recognition. Nevertheless, we can note that both Prof. Ulrich Reimers

and Dr. Mario Cominetti have already received well-deserved awards for their efforts. Furthermore, the staff of the DVB Project Office - Lou Dutoit and Peter McAvock - have worked tirelessly on the project.

DIGITAL TERRESTRIAL TELEVISION IN EUROPE - THE DVB-T SPECIFICATION

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It is well known around the globe that the European DVB Project has developed numerous technical solutions for the coding and transmission of digital television [Wood 1996]. For various reasons, but mainly because a considerable market pull first and foremost demanded solutions for satellite, cable and (S)MATV, the design of a system for terrestrial transmission of DVB signals was postponed until the period of 1994/1995. In November 1995, the Technical Module (TM) of the European DVB Project was able to finalize what is called the "Common 2k/8k Specification," alias DVB-T (Terrestrial). Some key features of DVB-T will be described below.

1 Requirements to be met by the terrestrial system

One of the important achievements of the DVB Project is the common understanding of all of its members to agree so-called "Commercial Requirements" for important individual development projects first, and only then to start the system design according to these requirements. It is therefore important to consider some of the key requirements before diving into the specification:

(1) DVB-T needs to resemble DVB-S (Satellite) and DVB-C (Cable and SMATV) as much as possible in order to enable the manufacture of multi-standard receivers at the lowest possible cost.

(2) The conceptual model of "data containers," which has also been used for DVB-S and DVB-C, needs to be supported. The "data container" is the result of channel coding and modulation and caters intuitively for the fact that DVB provides for quasi-error-free transmission of all sorts of data within each individual channel on cable, satellite, terrestrially etc. DVB has thus designed systems for data broadcast. An interesting side-effect of the conceptual model is that DVB can very flexibly be used for radio, HDTV, SDTV and every conceivable mixture of such services per container - as long as the capacity of the container (e.g. 39 Mbit/s in DVB-S) permits.

(3) The size of the data container for DVB-T should be as large as possible. It should primarily be tailored for 8-MHz-channels.

(4) The system needs to be designed in such a way that full-area coverage can be achieved when using stationary rooftop aerials. The reception by portable receivers (when operated in a stationary mode) should be supported. There is no requirement for mobile reception.

(5) The system must provide the capability to operate Single-Frequency-Networks (SFNs). In SFNs adjacent transmitters may use identical frequencies if they transmit absolutely identical data containers.

(6) The type of technology selected must meet the 1997 deadline when both services and the first sales of inexpensive consumer receivers based on DVB-T are scheduled to begin.

(7) "Hierarchical Modulation" needs to be included as an option.

In broad terms, the requirement for maximum commonality with DVB-S and DVB-C as well as the demand for SFNs automatically lead to the block diagram of the encoder side of DVB-T as depicted in **Figure 1**. The shaded elements of the diagram are those common with DVB-S [Reimers 1994]. They include mainly the channel coding ("C"), whereas the following elements relate to the kind of modulation which was selected (Orthogonal Frequency Division Multiplex "OFDM"). DVB-T is therefore termed a specification using "COFDM."

2 An introduction to COFDM

COFDM is a well-known concept, much liked in many parts of the world (e.g. in Canada, Europe, Japan) and currently being introduced via Digital Audio Broadcasting (DAB) in many parts of Europe [Saito 1996]. The major benefit of COFDM is the fact that the serial baseband bit stream which needs to be transmitted is distributed over many, closely spaced individual carriers (...frequency division...). The individual set of such carriers which is processed at a given time is called a "COFDM symbol." Due to the large number of carriers which all handle part of the baseband bit stream in parallel, the duration of such a symbol (T_S) is considerably larger than the duration of one bit of the bit stream. In reality, the duration may be up to 1 ms. Thus the receiver can - before starting to evaluate each individual symbol - wait until channel conditions have become stable, which, for instance, means that all echoes have been collected. COFDM therefore turns echoes from destructive signals which cause intersymbol interference into constructive signals adding to the energy of the direct transmission path. To define a certain part of the symbol duration during which the receiver pauses before starting the evaluation, the "Guard Interval" (duration: T_g) is introduced.

One type of echo which is typical for SFNs is the signal from some adjacent transmitter broadcasting an identical COFDM symbol. This signal cannot be distinguished from a classical echo and therefore can be evaluated like an echo if it arrives at the receiver during T_g. The definition of the guard interval therefore has an immediate influence on the type of SFN one would like to support. The larger T_g, the larger the maximum allowed distance between transmitter sites. On the other hand, one would like to keep the length of the guard interval as short as possible because, in the terminology of information theory, it is a non-used part of the channel capacity.

There is another balance to be taken into account here. The "useful" part of the symbol duration (T_u) which remains after the receiver has ignored the incoming signal during T_g is directly related to the spacing of the individual carriers within one symbol. In fact, it is just the inverse. Therefore, if for example the guard interval was chosen such that a 60 km distance between adjacent transmitters in an SFN would cause no problems (T_g = 200 µs) and if the total symbol duration was selected as 1 ms,_then the spacing of the individual carriers would become 1.25 kHz (1/ 800 µs). In one 8 MHz channel something like 6000 carriers would be present in parallel in order to carry their individual share of the baseband data stream.

3 Selecting the parameters for COFDM

There are a number of key decisions to be taken when a COFDM system is to be designed. They include the choice of the number of individual carriers per symbol, the selection of a "Guard Interval," the choice of the modulation scheme for the individual carriers, the method of synchronization, and many more. In the process of designing DVB-T, the selection of the number of carriers proved the most complex issue. The reasons for this are quite straightforward. Several members of the DVB project are convinced that nationwide SFNs are the key to the success of terrestrial DVB in their respective countries. These members therefore implicitly requested values for the length of the guard interval like 200 µs. They automatically had to accept that the number of individual carriers per channel would be in the order of 6000. Due to the fact that the OFDM modulation (see Fig. 1) is most efficiently implemented by employing an "Inverse Discrete Fourier Transform - IDFT" in the encoder and that the demodulator in the home receiver then would require an integrated circuit able to perform a corresponding real-time "Discrete Fourier Transform - DFT," the choice of that DFT chip becomes crucial. The IDFT is performed in chips which compute up to some power of two. $2^{13} = 8192$ = "8k" is the value closest to 6000. Thus selecting a value for T_g = 200 µs means that an "8k" demodulator must be included in the receiver.

Unfortunately, the evaluation of an "8k"-based systems proposal resulted in the awareness that the first-generation consumer hardware might not be able to meet the availability requirement nor to meet the cost requirement described above.

Representatives of several countries in which meeting the 1997 deadline for the introduction of a system is more important than the requirement for the support of national SFNs insisted on a less complex solution. This solution would, for example, support a guard interval of duration T_g = 50 µs and would consequently result in approximately 1500 carriers in an 8 MHz channel. Such a signal would be generated using a "2k" IDFT. After careful evaluation of the options a "Common 2k/8k Specification" [TM 1545] was designed which supports both solutions.

Figure 2 shows the transmission frame for DVB-T. The similarity of the "2k" and the "8k" variants is demonstrated by the fact that at first glance the frame structures for both variants look identical. In both cases 68 consecutive symbols are grouped in one frame. A structure of "Scattered Pilots" and "Continual Pilots" is used for time and frequency synchronization. The "Transmission Parameter Signalling (TPS) Pilots" carry all data necessary to inform the receiver about parameters like length of the guard interval, code rate of the convolutional code and type of modulation of the individual carriers.

4 Modulation of the OFDM carriers

The individual carriers may be modulated either by QPSK, by 16-QAM or 64-QAM.

Selecting a certain type of modulation directly affects both the available data transmission capacity in a given channel as well as affecting the robustness with regard to noise and interference. On the other hand, the choice of the code rate of the convolutional code can be used to fine-tune the performance of the system.

Table 1 shows one aspect of the system performance that can be achieved as a function of the type of modulation and the code rate. The table indicates the net-data transmission-efficiency in an 8 MHz channel for all possible combinations of modulation and code rate. The values shown include all relevant effects; for instance, the loss of efficiency effected by the different sorts of pilot signals is included. Both variants of DVB-T, the "2k" as well as the "8k" system, show identical performance.

Code rate	QPSK	16-QAM	64-QAM
1/2	0.62 Bit/s per Hz	1.24 Bit/s per Hz	1.87 Bit/s per Hz
2/3	0.83 Bit/s per Hz	1.66 Bit/s per Hz	2.49 Bit/s per Hz
3/4	0.93 Bit/s per Hz	1.87 Bit/s per Hz	2.80 Bit/s per Hz
5/6	1.04 Bit/s per Hz	2.07 Bit/s per Hz	3.11 Bit/s per Hz
7/8	1.09 Bit/s per Hz	2.18 Bit/s per Hz	3.27 Bit/s per Hz

Table 1: Net-data transmission-efficiency of DVB-T in an 8 MHz channel (non-hierarchical modulation).

"Hierarchical Modulation" can be selected instead of the aforementioned types of modulation. In the case of hierarchical modulation, the baseband data stream is split into a high-priority data stream and a low-priority data stream. The high-priority data stream is protected by a more powerful variant of the convolutional code (for example, code rate = ½) and used to determine the quadrant of the complex signal plane in which the modulated individual carriers can be described by their respective I and Q values, the constellation diagram. The low-priority data stream is protected by a less powerful variant of the convolutional code (for example, code rate = 5/6). This data stream is then used to modulate the individual carriers in such a way that around the positions in the constellation diagram determined by the high-priority data stream a "cloud" of constellation points is defined by the low-priority data stream. In good C/N conditions each individual point of the constellation diagram can be identified and both data streams can be detected. If the C/N ratio decreases, the constellation points in the cloud can no longer be detected, but it is still possible to identify in which quadrant of the complex plane a constellation point is located. Thus the high-priority data stream can still be evaluated.

5 Conclusions

DVB-T is one of the latest results produced by the European DVB Project. It describes a powerful and versatile system for terrestrial transmission of all sorts of digital signals, for example, of TV programs. Based on the specification, the design of encoders and decoders has started in several countries. It is expected that first transmissions will take place in 1996 and that by the end of 1997 full services will have started at least in the United Kingdom. The UK wishes to use the "2k" variant of DVB-T. It is expected that before the end of the decade, for example, in Spain, first services based on the "8k" variant will have begun. Receivers able to decode "8k" signals will also be able to decode "2k" signals.

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[Wood 1996] David Wood: NAB Presentation 1996, "Open, Interoperable, and Market Driven - The DVB Challenge."

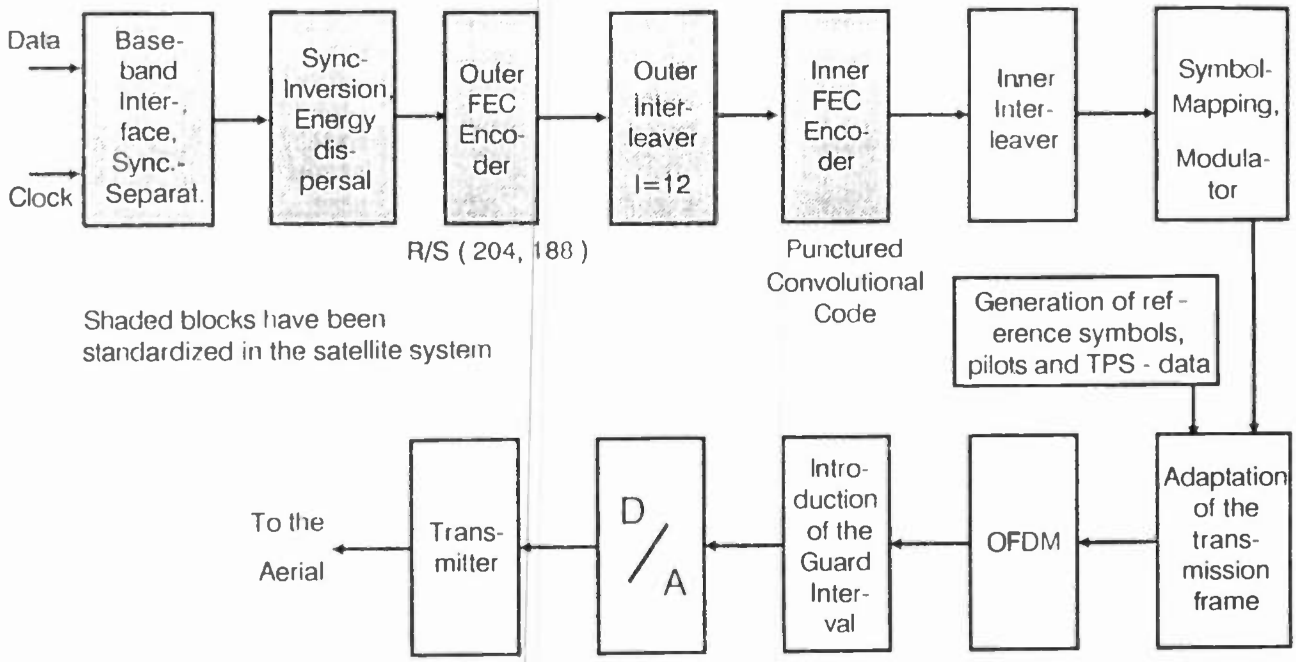

Figure 1: Block Diagram of the Encoder for Terrestrial DVB (without Hierarchical Modulation)

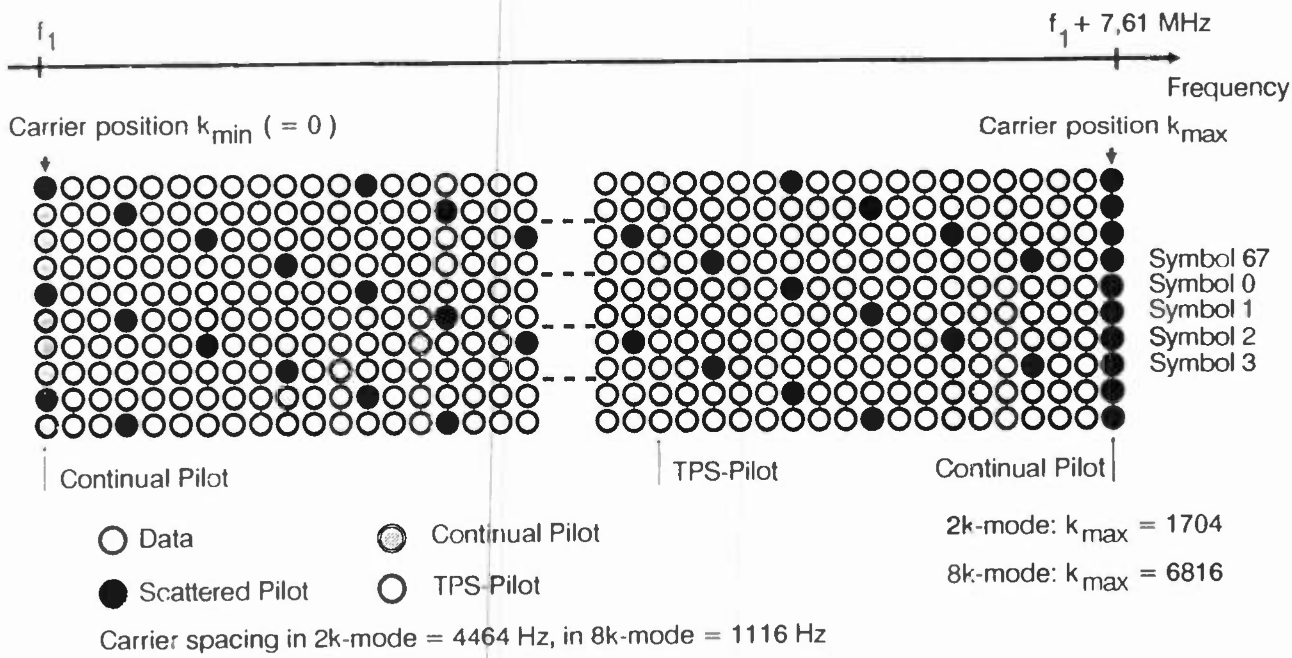

Figure 2: Transmission Frame for Terrestrial DVB (DVB-T)

EUROPEAN PATH TO DIGITAL TELEVISION THE MULTIMEDIA DIMENSION

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ABSTRACT

What is the situation of standardisation of multimedia in Europe?
What is going on in the field of multimedia?
What is the so called Multimedia portfolio?
Can multimedia be "the killer application" for digital TV?
How and when will the market respond?
How will multimedia reach its costumers?

MULTIMEDIA IN EUROPE

The European Telecommunication Standard Institute (ETSI) has identified 161 work items in the field of multimedia. The standardisation work is done in 16 projects. The work is almost finished.

The European Union supports the development of multimedia services through this so called ACTS Programme which contains of almost 100 different projects, most of which are joint ventures with many actors.

In the following you will find some examples of those projects (supported by the European Union). There is a lot more going on in different countries and groupings but the following will give some ideas of the European activities.

SMASH - Storage for Multimedia Applications and Systems at the Home

Main Objectives

The main objective is to show the technical feasibility of an integrated storage unit that is connected to the network and other consumer equipment and enables to down load, store and retrieve information efficiently for all multimedia applications.

Participants:

Philips Consumer Electronics (NL), Philips Italy (I), DeutscheThomson-Brandt (D), Olivetti (I), Tandberg Data (N), University of Essen (D), Technische Universiteit Delft (NL).

PANORAMA - PAckage for New OpeRational Autostereoscopic Multiview systems and Applications

Main Objectives

The ultimate goal for future telecommunications is highly effective interpersonal information exchange. The effectiveness of telepresence. In this concept it is crucial that visual information is presented in such a way, that the viewer is under the impression of actually being physically close to the party with whom the communication takes place.

The PANORAMA project will perform research in the areas of hardware and software development of a multiviewpoint autostereoscopic system towards the use in telecommunication. Beside the usage for 3-D visual communication, several other 3-D imaging applications can be benefit from the proposed system, due to the overlapping of many components.

Participants:

Siemens AG (D), AEA Technology (UK), CCETT (F), Deutsche Telekom AG (D), Heinrich Hertz Institut (D), Intracom (GR), TU Delft (NL), Thomsom Broadcast System (F), University Hannover (D), University Thessaloniki (GR), CSELT (I), Politechnico di Milano (I).

AMUSE - Advanced Multimedia Services for Residential Users

Main Objectives

The global objective of the project is to carry out experiments in the provision of interactive and distributive multimedia services to real residential users. These services will be demonstrated in field trials carried out in different islands, using an end-to-end ATM infrastructure (from Terminal Equipment to Multimedia Server), through the Core Network and various types of Access Networks.

Participants:

Isaltel (I), Instituto de Engenharia de Sistemas e Computadores- INESC (P), Autor-Tecnologias Multimédia (P), Portugal Telecom/CET (P), CSELT (I), Deutsche Telecom-FZ (D), GPT (UK), IDEA (B), National Technical University of Athens (GR), OnLine Media (UK), Siemens AB (D), Sirti (I), Siemens Nixdorf (D), Swiss PTT (CH), Telecom Italia (I), University of Paderborn (D), University of Stuttgart (D), Videotime (I), Post and Telecom Iceland, University of Iceland, Nyherji (IC).

KIMSAC - Kiosk-based Integrated Multimedia Service Access for Citizens

Main Objectives

Public users of "ordinary" everyday computer and telecommunication services are overwhelmed in two significant ways: 1) the selection of the most appropriate service, given their particular needs at a point in time; and 2) having selected a service, the difficulty of operation (use) of that service. KIMSAC addresses these problems by asserting the need for some form of mediation between users and services. This mediation is conceptualised by KIMSAC in terms of Personal Service Assistant (PSA). A PSA operates on behalf of a user in an analogous manner to a human personal secretary.

Participants:

Broadcom Eireann Research Ltd (IRL), Centro Studi e Laboratori Telecomunicazioni Spa (I), Trinity College Dublin (IRL), Digital Equipment Ireland Ltd, Social Welfare Services (IRL), Cap Sesa Telecom (F), Teltec Ireland, Swedish Institute of Computer Science, Imperial College of Science, Technology and Medecine (UK), Information Technology Centre (IRL), Foras Aiseanna Saothair (IRL).

SICMA - Scalable Interactive Continuous Media Server- Design and Application

Main Objectives

The general aim of the SICMA project is to design a scaleable server for the delivery of images, data and continous multimedia information and to demonstrate its efficiency by applying it to a relevant application, namely the so called "Virtual Museum". The server will be used within various test-beds showing the potential to serve a large number of users under various conditions. It is the aim of the project to design a cost efficient server making interactive access to various forms of media feasibel this way.

The server will fully realise the DAVIC standard. Special emphasis will be put on the connection of a number of servers within one lager network, realising a distributed information network in this way.

Partners:

Parsytec Computer GmbH (D), University of Paderborn (D), Natural History Museum (UK), Cap Gemini SpA (I), Multimedia Systems Centre (GR).

MOMUSYS- Mobile Multimedia Systems

The main objective of the project MObile MUltimedia SYstems is to develop and validate the technical elements necessary to provide new audio-visual functionalities for mobile multimedia systems. Such functionalities are being identified in the context of SIO MPEG-4 which will become the standard for coding of audio-visual information in multimedia systems. Most of these functionalities ar unsupported or insufficiently supported by the available or emerging standards. The new audio-visual functionalities, which include content manipulation, content scalability and content based access, must be achieved with algorithms that provide very efficient compression on the one hand and robustness against transmission errors on the other. To that purpose new ways of communicating data and even coding methods between terminals must be developed, going away from simple syntax definitions to a more generic language. (ITU-R TG 8/1, ETSI SMG 5).

CINENET - Cinema Films and Live Events via Satellite and Cable Networks

Main Objectives

The development of High Definition television systems together with digital transmission has created the possibility of electronic cinema-like film and video distribution.

The CINENET project seeks to "make it happen" by distributing over satellite and cable networks:

- Cinema quality digital movies, sports, live events and business presentations.
- To viewing halls, cultural centres, hotels, universities and cinemas, and eventuall to individuals.

Participants:

Alcatel Espace (F), Barco Industries (B), Centre Commun d'Etudes de Télédiffusion, CCETT (F), Eutelsat (F), HD-Thames (UK), Heinrich Hertz Institute, HHI (D), HISPASAT (E), Institut für Rundfunktechnik, IRT (D), Thomson Broadcast Systems (F), Vidéo Transmission Haute Résolution, VTHR (F).

INTERACT - Interactive Television Return Channel Standardisation and Trials

Main Objectives

The identification of market requirements for the introduction of interactive television and data services using DVB platforms.

The characterisation and specification of the physical return channel for cable and terrestrial broadcast networks. The development, manufacture and experimental trials of an asymetric UHF return channel system.

Participants:

Independent Television Commission (UK), Barco (B), SAT (F), Bosch (D), Nozema (NL), TDF (F), RAI (I), Retevision (E), Thomson Broadcast Systems (F), European Broadcasting Union (B), CCETT (F), Integan (B), University of Ghent (B), Alcatel (E), Universidad Politecnica Madrid (E), Sagem (F).

SMASH - Specific MPEG4 Architectures Software and Hardware

Main Objectives

Objective of the project is to implement, on a suitable hardware platform, the tools and algorithms of the forthcoming MPEG-4 standard. The scope of the project will be to evaluate different coding schemes among those that will form the upcoming MPEG-4 standard. The end user, service provider and system vendor requirements will be analysed in order to indentify and specify the key costs and technical objectives that must be met so as to achieve commercial viability. High performance flexible and programmable MPEG-4 terminals will be realised to demonstrate the features functionality of the system by choosing suitable MPEG-4 applications to show its main advances with respect to already available tools and methods.

Participants:

SGS Thomson Microelectronics Group ST CASA (I), Centro Studi e Laboratori Telecomunicazione (I), Heinrich Hertz Institut für Nachrichtentechnik Berlin GmbH (D), SGS Thomson Microelectronics Ltd (UK), University of Hannover LFI (D), Ecole Polytechnique Federale de Lausanne (CH), Forschungs-zentrum Informatik (D), Ecole Nationale Superieure des Telecommunications ARECOM (F), France Telecom CNET (F), Studer Professional Audio AG (CH), Siemens AG (D), Loboratorie d´electronique Philips LEP (F), Telenor AS (N), Matra Communication (F), Consorzio per la Recerca sulla Microelettronica nel Mezzogiorno CoRiMMe (I).

SETBIS - Set Top Box for Interacitve Services on Demand

Coordinator: U. Barth, Stuttgart

The SETBIS project aims at the provision of tools to assess the user acceptance of interactive services on demand and broadband communication by application-driven pilot projects.

To provide such acceptance test tools, a high-performance set top box for interactive services on demand and broadband communication will be specified and realised. The key point in the set top box project is to determine the required level of functionality and to realise this in order to fulfill the user and content provider demands.

Partners:

Alcatel SEL AG (D), Fraunhofer Gesellschaft/IAO (D), Isatel SIT (I), GSM Gesellschaft für Software Management mbH (D), Loewe Opta GmbH (D), Seleco (I), UI Design (S), Kaufhof Holding AG (D), University of Florence, Dipartimento di Ingegneria Elettronica (I).

OPARIS-oD - Open Architecture for Interactive Services on Demand

Coordinator: Dr. H.J. Matt, Stuttgart

The OPARIS project addresses the overall architecture for future Interactive Services-on-Demand systems (ISoD). In particular the architecture, functionality & environment for ISoD media servers and the architecture, functionality & environment of ISoD media switching centre for distribution of information and for interaction with real users.

DIGISAT - Advanced Digital Satellite Broadcasting and Interactive Services

Services for a digital broadcasting scenario will be identified, as well as their requirements, with a double service orientation:

Broadcasting services (with associated near-interactive capabilities).

Interactive services (based on network with return channel capabilities).

Teletext Services will be studied as an specific example of data services.

The technology required to establish the **return channel** via satellite will be developed as a key target of the project.

SMATV grouping, this innovative concept allows the concentration of the traffic from SMATV headend concentrators existing in a block of buildings into a single terminal point to link to the satellite path.

VSAT configuration, which transports via satellite the information up to the broadcaster premises.

Participants:

HIPASAT (E), Portugal Telecom, Philips (F), RAI Research Center (I), RETEVISION (E), SAT (F).

ISIS - Interactive Satellite multimedia Information System

Main Objectives

ISIS intends to demonstrate the technical and economical feasibility of Satellite Interactive Television services in the framework of Multimedia Applications in the furture European Telecommunication Scenario.

ISIS will design and develop a prototype communication system abel to provide typical TV end-users with a large amount of new services in addition to traditional TV distribution services. One of ISIS keywords is "Interactivity", that is the possibility for TV end-users to dialogue with service providers for a continuous adaption of the service with actual user needs.

Participants:

Alenia Spazio (I), Nuova Telespazio (I), Eutelsat (F), Hispasat (E), IBM Semea (I), Intracom (GR), Philips LEP (F), RAI (I), Space Engineering (I), University of Salzburg (A), SBP (I), Normarc (N), Community of Mediterranean Universities (I), University of Florence (I), Balkan Press Ltd (GR).

ATLANTIC - Advanced Television at Low bitrates And Networked Transmission over Integrated Communication systems

The main objectives of this project are:

- To assemble and demonstrate the essential building blocks of a programme distribution chain based on compressed video signals from the input of a studio to the final display
- To pursue international standardisation, where appropriate, for the technology demonstrated by the project.

Participants:

BBC (UK), Centro Studi el Laboratori telecommunicazioni (I), Ecole Nationale Superior des Telecom (F), Ecole Polytechnique Federale de Lausanne (CH), Fraunhofer Gesellschaft (D), Insitito de Engenharia de Sistemas e Computadores (P), Snell & Wilcox (UK).

OKAPI

Open Kernel for Access to Protected Interoperable interactive services

Main objectives

OKAPI intends to develop a trusted kernel, which appears like a distributed Operating System. This OKAPI Kernel will ensure interoperability, openness, equity, users privacy and evolutivity for opening the European multimedia market to every potential service provider and user. A major target is to achieve through the kernel a maximum degree of commonality for the future European Multimedia Services Distribution System.

Participants:

Université Catholique de Louvain (B), Fraunhofer Institute Darmstadt (D), University of Thessaloniki (GR), Ecole Polytechnicque Fédérale de Lausanne (CH), Deutsche Bundespost Telekom (D), Pastel /Mons/ (B), Université de Liège (B), CSELT (I), EBU rtbf (B), Philips (NL), CCETT (F).

SOMMIT - Softeware Open MultiMedia Interactive Terminal

Main Objectives

The overall goal of the Project is to define an open interactive multimedia terminal architecture, application and delivery media independent, that allows a full software implementation. The architecture will be validated by developing a terminal that will make use of a tool-kit of modules (covering part of the architecture) which could be reused in different instantiations of terminals depending on the actual time, manufacturer and product.

Participants:

CSELT Centro Studi e Laboratori Telecomunicazione SpA (I), CCETT (F), IBM Deutchland, Philips Research (NL), Sony Telecom Europe (BE).

DIANE - Design, Implementation and operation of a Distributed Annotation Environment

Main objectives

Distributed Multimedia Services envisaged today in most cases distinguish sharply between service and content provider on one side and consuming users on the other side. Only a few applications have been realised strengthening the role of end-users both as content provider and consumer. Existing multimedia authoring systems are either application specific or allow solely combinations of media generated entirely by a user, as it is the case for multimedia mail. The goal of DIANE is to develop a multimedia service removing these deficits.

Participants:

Kapsch Aktiengesellschaft (A), IPVR University of Stuttgart (D), European Centre of Excellence for Parallell Computing (A), Sistemas y Tratamiento de Información SA (E), Hospital General de Manresa (E).

THE MULTIMEDIA PORTFOLIO

The European Broadcasting Union (EBU) has identified five categories and services:

- Mainstream programme or related add-ons
- Helper data/signals
- Down-loaded information
- Receiver-dependent down-loading files (specific brand needed)
- Personal or closed user group services.

Throughout the classes of indentified services, the characteristics of a given service may be defined by a number of parameters. These may be regarded as indicating the technical system, the type of access and the likely timescale for implementation. The services may also be classified under the level of activation by the viewer, and if the interaction is locally or distant.

Mainstream programme or related add-ons

1. Programme guide

Description of the programmes available from a certain programme provider. The scheduled programmes may possibly be extracted from a list to be customized according to the viewer´s need.

2. Subtitling

Subtitles to be inlaid in the image at the receiving system. A number of different languages could be transmitted at the same time.

3. Resumé

A summary of the mainstream programme. The resumé is updated during the course of programme.

4. Programme description

A description of the ongoing programme for the benefit of visually impaired viewers (e.g. Audetel).

5. Home shopping

Interaction from the viewer with a mainstream programme. The viewer could send a message to the "shop" or a central office to buy the goods.

6. Audience polling

The viewers vote for the outcome of a programme or answer questions to obtain a polling result.

7. Gambling/auctioning

Placing bets, bids etc. on races, games , ...

8. Multi-choice programme

Choice of one programme from a multi-channel programme package (e.g. camera position in sports game, quiz games).

9. Extra information

Viewer controlled add-on data or information relating to the main programme.

10. Near video-on-demand

A video programme is transmitted several times simultaneously on different physical channels. The start time of the programme is staggered from channel to channel (e.g. a start every 20 minutes). A feeling of a near-instantaneous response in then obtained by the viewer.

Helper data/signals

11. PDC

A system to remotely control recording of a programme on domestic VCRs. This may also be used during normal viewing to provide the name of the programme, duration remaining and to control channel changing.

12. Avertising indication

Signal that indicate advertising and avoid recording them if not wanted.

13. RDS-like label

Signal that indicate or interrupt the pro-gramme or VCR recording from an important message.

14. Anti-copying

Signal that controls the possibility of recording.

15. Navigation Information

Data that helps the viewer to navigate amongst the many different information sources. This is not the same as Service Information (SI).

16. Newspaper

An electronic newspaper that is constantly updated.

17. Non-real-time quiz games

Quiz game or similar, to be used locally.

18. Teletext

An information service which is transmitted for viewing on any suitable-equipped receiver capable of decoding data according to the Teletext standard.

19. Educational progammes

Multimedia programmes to be used separately or in conjunction with the main programme. They could also link with printed papers and exercises.

20. Personal ads

Advertisements that are placed by the viewer via a feedback channel into the transmission information source and subsequently boadcast to all the viewers.

21. Locally processed data

Executable files which could generate a programme in the receivers.

22. Traffic Information

Informations about traffic, road conditions, maps, etc.

Receiver dependent downloading files (specific brand needed)

23. Computer games

Downloaded games to a receiver which has a certain amount of processing power. The games could also be played in a two-way communication environment.

24. Downloaded CDs

CD information (audio, photo, CD-Interactive, CD-ROM) which could be recorded locally.

25. Downloaded software

Upgrades of computer programmes, databases or programme related software for interactive services.

26. Operating Systems update

Updating the home receiver operating system.

27. Addressed Display/unit

Information that is addressed to a certain peripheral unit (e.g. printer, fax).

28. Updating stored data

Updating a price list that is accompanied by the main information on another storage device such as optical disk, etc.

29. Multi-player games

Computer games that are multi-cast to a number of users.

Personal or closed user group services

30. Video-on-demand

On-demand service from a video jukebox. The video is played out whenever the viewer whishes. Could also provide music-on-demand with a sound only music jukebox.

31. Interactive shopping

Active search and shopping in a guide or "shopping mall".

32. Independent data services

Independent data services distributed to a closed user group and not related to the programme in any way.

33. Home banking

Making bank transactions via a dial-up service.

34. Telecom services

Traditional telecom services as well as new services such as Videophone, Videofax.

35. Business services

"Multi-cast" business services to a closed user group. Could for instance be "Telesurgery" for the medical profession.

36. Customized information

The user sets up an information profile for retrieved or received data.

CAN MULTIMEDIA BE "THE KILLER APPLICATION" FOR DIGITAL TV?

The change over from analogue to digital need heavy investments for all concerned; the broadcasters, the net operators and the public. The most difficult part of the turn over is to make the public willing to put away the old analogue equipment and buy new digital tools. What will be the driving force?

First of all the TV public looks at programs and not at channels and services. The ideal situation is the whole family together looking at the same programme, as we did long ago, sitting by the fireplace listening to storytellers. In the new world every member of the family will have their own choice of programmes and services and they could of course still be a happy family, but it also could be the opposite.

How the public will meet the new situation is very difficult to say, but for sure, this is a question of generation. People in the middle and older ages will not find it easy to use all the possible services of the new multimedia world, but the youngsters will find it a natural part of the daily life. The question is when they will be able to spend enough money on the services making it profitable for the media deliverers to set up and handle a service. Even if the easy answers is not to be found the future is digital.

TECHNICAL REGULATORY ISSUES

Wednesday, April 17, 1996
9:00 am - 12:00 pm

Session Chairperson:
Dane Ericksen, P.E., Hammett & Edison, San Francisco, CA

DISASTER RECOVERY FOR BROADCAST FACILITIES
Richard Rudman
Westinghouse/CBS
Los Angeles, CA

FCC TOWER REGISTRATION
Robert D. Greenberg
Federal Communications Commission
Washington, DC

*THE IMPACT OF ENVIRONMENTAL, HEALTH AND SAFETY LAWS AND REGULATIONS ON THE BROADCASTING INDUSTRY
Linda D. Kelley
Westinghouse Electric Corporation
Pittsburgh, PA

PANEL ON UNATTENDED STATION OPERATIONS
Panelists: Ken Brown, Capital Cities/ABC Inc., New York, NY; Bob Greenberg, Federal Communications Commission, Washington, DC; Christopher Imlay, Booth, Freret & Imlay, Washington, DC; Frank Lucia, Federal Communications Commission, Washington, DC

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*Paper not available at the time of publication.

DISASTER RECOVERY FOR BROADCAST FACILITIES

Richard Rudman
Westinghouse/CBS
Los Angeles, CA

ABSTRACT

Nature, weather, terrorist attacks, arson, utility outages, or even simple accidents that get out of hand are indelible facts of modern life. Broadcasters must keep their people and systems safe and operational, or risk becoming disaster victims themselves. While broadcast engineers should take realistic precautions ahead of time, it is impossible to prepare for every eventuality.
This paper outlines some of the many considerations involved in disaster recovery. The approach that this paper takes puts life-safety considerations first. The assumption is that a calm, well-trained, intact professional staff is itself a primary recovery resource.

DISASTER PLANNING-GET EXPERT HELP

Disaster planning and recovery is at once an art, a science and a technology. Disaster planners have their own professional groups and certification standards. If the resources are available, hiring a professional or obtaining professional advice is a wise course of action. If you contract for a comprehensive program, you will avoid a major pitfall of today's broadcast industry when we often are trapped fighting brush fires and find little or no time for planning brush clearance. No matter who develops your disaster recovery strategy, an essential part of the process will be a written plan that will be periodically tested.

Preparedness 101

The private sector can learn from government emergency professionals. They recognize that PREPAREDNESS is only the starting point of a continuous cycle.

The second step is the actual RESPONSE to an emergency. RECOVERY comes third after immediate life-safety issues and other dangers are under control. The emphasis is restoring things to what passes for normal. The last phase is MITIGATION when the lessons of the emergency are studied. These lessons are applied as the circle is completed to another round of PREPAREDNESS.

What do you prepare for?

Disaster planners talk in terms of REALISTIC RISK. Simply put, it means you do not spend as much time and effort planning for floods if your primary risk is earthquakes and your facility is on high ground. If you have lived in your area for sometime, you will already have a fair idea of your greatest risks. If new to a region, research is a must. Be wary of advice from those who have lived there for a while. They may overlook something you consider to be a significant risk. For instance, there are no 150 year-old transmitter sites. Long-term risks like 100 or 150 year floods may not have even been considered when your site was designed.

How do you prepare?

Disaster planners will also tell you that there are certain common elements to many types of preparedness. Both floods and earthquakes can isolate victims from utilities, supermarkets, and emergency medical assistance.

Responsibility for preparedness

Local emergencies can easily be triggered by preventible failures in air supply systems, roof leaks, or uncoordinated telephone, computer or AC wiring changes. Seemingly harmless acts such as employees plugging electric heaters into the wrong AC outlet have brought down entire facilities.

Broadcast engineers should have responsibility (if not direct supervision of) the environmental infrastructure of broadcast facilities. This includes studios and transmitter sites. Without this critical element that tightly couples oversight to responsibility, critical electronic systems will be at the mercy of who or what does control the environment. Without responsibility, essentials such as proper installation practices, preventive maintenance, training and testing may fall by the wayside.

STARTING THE PLANNING PROCESS

Listing realistic risks

Disaster recovery will be an easier task if some level of preparedness is in place and periodically tested. The process begins with A REALISTIC RISK ASSESSMENT. Your list should identify hazards based on local conditions.

You should include the likelihood that threats from present or former employees who may hold grudges could turn into violent actions. Listeners and viewers sometimes get mad at broadcasters. Consultation with both the programming and security departments is strongly recommended.

Are there any nearby man-made hazards such as airports or chemical plants? What about nearby buried pipelines? What about the construction of your facility?[1] What hazardous materials are stored on the premises? Don't forget the obvious ones like compressed gas cylinders and fuel for your emergency generator.

RISK ASSESSMENT AND BUSINESS SURVIVAL

Once you have compiled your list of realistic risks, decide what you can do to protect your people and the facility from them. Preparedness lives or dies on top management commitment. Without high level support, funding and time to do proper preparedness will be hard to come by. Write a thoroughly researched report to top management in non-technical language. Relate risks to time off the air, potential for injury or death to staff or audience, and potential damage to community image. What would 24 or more hours off the air do to drive time ratings?

Adding your skills

Part of this process will rely on your broadcast engineering trouble shooting skills and senses. Do not overlook the obvious. Computers, transmitters, or telephone equipment depend on cool air. How long can they continue to operate during a heat wave when the one central building air system has failed?

Take practical precautions

Preparedness should be considered in the same light as an insurance policy. The greater the risk, the higher the premium. Some communications facilities in earthquake country have three diesel generators and three air handling systems. Quake-resistant buildings sometimes rest on huge rubber isolators at the base of each supporting column.[2] These isolators can protect the structure and its contents from the most violent types of lateral movement during earthquakes. Proper seismic design and preparedness may be expensive or impossible for existing facilities, but may add no more than 10% to construction costs of a new building.

The Business Resumption Plan

A formal written Business Resumption Plan (BRP) is just as important as an organization's written disaster plan. Some experts argue that the written disaster plan and the BRP should be formulated, tested and updated in concert. The Prime Directive: When disaster strikes, the first concern of an organization must be for the safety of its employees, customers, vendors, and visitors.

The BRP is activated only when the basic requirements of life safety have been fulfilled. A BRP is a template to perform your damage assessments, salvage operations, emergency restoration and, if needed, relocation of critical functions.

The focus of Business Resumption Planning is maintaining or resuming core business activities following a disaster. The major goal of any business BRP is resumption of production, delivery, customer service and cash flow. In broadcasting, it begins with getting back on the air if you are down, or making sure you can stay on the air under adverse conditions when your facility is impaired.

What a BRP can do for you

A comprehensive and well designed BRP can accelerate RECOVERY, thereby saving time, money and jobs. A good BRP is an insurance policy, if not a major investment. The following steps are the backbone of a comprehensive BRP:

- Conduct a Business Impact Analysis (BIA).[3]
- Promote employee buy-in and participation.
- Seek input starting at the lowest staff levels.
- Build a recovery strategy.
- Build a validation process and testing procedure.
- Assure a Continuous Update of the BRP.[4]

AN OUNCE OF PREVENTION

While the title of this paper uses the word "RECOVERY", the reader will not find a specific list of things to do that will magically prevent fatalities or injuries that have already taken place, or reconnect wiring to equipment lying on the floor in pieces. If faced with rehabilitating a facility where there has been no preparedness, competent broadcast engineers will eventually get the facility back on its feet. If the reader is faced with such a prospect, your main resource will be your ability to reach out to suppliers and other who you will need to begin the rebuilding process. Emergencies sometimes make this more difficult since others who have planned ahead may have written agreements in place with the very assets you may need on short notice. Roads may be impassible, limiting delivery possibilities. Recovery in such cases is not impossible. It just takes a lot longer. There is a lesson here you can convey to your owner or general manager.

Workplace safety: a beginning

Employers must assure safety in the workplace at all times. Some states like California have passed legislation that mandates that most employers identify hazards, and protect their workers from them. Natural emergencies create special hazards that can maim or kill. A strong foundation of day-to-day safety will lessen the impact of major emergencies. For instance, assuring that plate glass in doors has a safety rating could avoid an accidental workplace injury, as well as compounding injuries during a major disaster.

OUR SPECIAL HAZARDS

Special and dangerous hazards are found in the broadcasting workplace. Tall equipment racks are often not secured to floors, much less secured to load-bearing walls. Preventing equipment racks from tipping over during an earthquake will avoid crippling damage to both systems and people. Bookcases and equipment storage shelves must be solidly secured to walls. Certain objects should be *tethered*, rather than firmly bolted. While securing heavy objects is mostly common sense, consult experts for special cases. Don't forget seismic-rated safety chains for heavy objects like TV studio lights and large speakers.

Computers and monitors are usually not secured to work surfaces. A sudden drop from work station height would ruin the day of most computers, video monitors, and their managers. An industry has sprung up that provides innovative fasteners for computer and office equipment. Special Velcro® quick-release anchors and fasteners can support the entire weight of a personal computer or printer, even if the work surface falls over.

Bolting work stations to the floor, and securing heavy equipment with properly rated fasteners helps address major seismic safety issues. G forces[5] measured in an upper story of a high rise building during the Northridge quake were greater than 2.7 times the force of gravity. A room full of unsecured work stations could do a fair imitation of a slam dance, even at lower accelerations. Cables can pull loose, monitors can implode or become lethal projectiles, and delicate electronics can be smashed into scrap. Even if you do not live in a region where there has been recent seismic activity, consider that some parts of the earth have been given a *long overdue* rating by respected seismologists. Maybe you will be the only one on your block to take such warnings seriously. Maybe you will be the only one left operational on your block!

Glass houses

Many local building codes now call for shatterproof glass for sliding doors and windows in homes and businesses. Large plate glass windows, including those we use in our studios, can become killers during major emergencies. A moment magnitude 7.4 earthquake can lethally hurl dagger-like shards from untreated plate glass windows through the air.[6] Extreme wind pressures during hurricanes can have a similar effect. The solution: Purchase shatterproof glass, or retrofit. Several companies manufacture a special film coating that can be applied to existing non-safety plate glass. This film will hold the glass together to reduce the danger of flying fragments.

Safety maintenance

Maintaining safety standards is difficult in any size organization. A written safety manual that has specific practices and procedures for normal workplace hazards as well as the emergency-related hazards you identify is not only a good idea, it could lower your insurance rates. If outside workers set foot in your facility, prepare a special Safety Manual for Contractors. Include in it installation standards, compliance with Lock-Out/Tag-Out[7], and emergency contact names and phone numbers. Make sure outside contractors carry proper insurance, and are qualified, licensed or certified to do the work for which you contract.

RESCUE PLANNING FOR DIRE EMERGENCIES

When people are trapped and professionals can't get through, our first instinct may be to attempt a rescue. Professionals tell us that more people are injured or killed in rescue attempts during major emergencies than are actually saved. Experts in Urban Search and Rescue (USAR) not only have the know-how to perform their work safely, but have special tools that make this work possible under impossible conditions. The *Jaws of Life*, hydraulic cutters used to free victims from wrecked automobiles, is a common *USAR* tool. Pneumatic jacks that look like large rubber pillows can lift heavy structural members in destroyed buildings to free trapped people.

You as a broadcast engineer may never be faced with a life or death decision concerning a rescue when professionals are not available. Those in the facilities you are responsible for may be faced with tough decisions. Consider that your planning could make their job easier, or infinitely more difficult. Also consider recommending USAR training for those responsible for on-line management and operations of the facility as a further means to ensure readiness.

THE RECOVERY PROCESS

The disaster has happened. Fatalities have been identified and the injured are being given first aid. What happens next?

Facility integrity: roof, walls and floor

A qualified person must first inspect the facility to make sure that it is safe. If you have just experienced the full effects of a hurricane or significant earthquake, damage may or may not be obvious. Most construction codes provide only that a structure must be built strong enough to allow evacuation without loss of life or serious injury. You may be faced with the ultimate irony of a building that meets all local code provisions for an earthquake but may be rendered unsafe by the actual event.

Circumstances beyond your control

Serious and obvious damage may result in a revocation of your building occupancy permit. You may not even be allowed in the facility to remove broadcast equipment. Do you have a contingency plan in place for an alternate studio site? A remote truck may be your only resort. Consider installing a locked access box outside the building where you may be able to connect to your microwave or phone company-provided studio transmitter ling (STL).

Basic assumptions

Assuming your facility is intact, or you have a contingency plan to get a signal on the air, what other considerations come into play during RECOVERY? Your critical considerations can range from little things like employee morale, to big things like erecting an emergency antenna or getting your computer system working again.

Outside Plant Communications Links

Your facility may be operational, but failure of a wire, microwave, or fiber communications link could be devastating. All outside plant links discussed below presuppose proper installation. For wire and fiber, this means adequate service loops (coiled slack) so quake and wind stresses will not snap taunt lines. It means that the telephone company has installed terminal equipment so it will not fall over in a quake, or be easily flooded out. A range of backup options are available.

Outside Plant Wire

Local telephone companies still use a lot of wire. If your facility is served only by wire on telephone poles, or underground in flood-prone areas, you may want what the telephone industry calls *alternate routing*. Alternate routing from your location to another Central Office (*CO*) may be very costly since the next nearest *CO* is rarely close. Ask to see a map of the proposed alternate route. If it is *alternate* only to the next block, or duplicates your telephone pole or underground risk, the advantage you gain will be minimal.

Essential Service designation

Most telephone companies can designate as an ESSENTIAL SERVICE a limited block of telephone numbers. Such lines are restored before businesses and homes not considered critical to emergency recovery. Lines so-designated are usually located at hospitals, utility companies and police headquarters. Broadcast news departments are often considered in this category by government emergency managers. Contact your phone company representative to see if you can qualify.

Public pay telephones are essential

Inform your news departments that pay phones in public areas are at the top of the priority list for restoration by phone companies. This is not true of private pay phones, or pay phones NOT located in public areas.

Dial tone trivia

Most broadcast engineers are aware of the basics of the telephone system in use in the United States. What we may not know is that the luxury of getting dial tone the instant we pick up a telephone instrument is not the norm in many parts of the world. The lesson: If you pick up an instrument properly connected to an outside line and detect side tone[8], you will eventually get dial tone. Jiggling the hook switch or hanging up puts you at the end of the queue waiting for available dial tone. Patience is a not just a virtue you need to teach to those having to make emergency calls. It is a necessity.

Microwave links

Wind and seismic activity can cause microwave dishes to go out of alignment. Quake-resistant towers and mounts can help prevent alignment failure, even for wind-related problems. Redundant systems should be considered part of your solution. A duplicate microwave system might lead to a false of security. Consider a non-microwave backup such as fiber for a primary microwave link. Smoke, heavy rain and snow storms can cause enough path loss to disable otherwise sound wireless systems.

When a major link failure does occur, you will have hopefully recorded ahead of time azimuth and elevation readings that will enable more rapid restoration. A stock of emergency restoration hardware, including brackets, bolts, and short tower sections is a wise precaution. An alternate STL path, even telephone lines, can likewise be prudent insurance.

Fiber Optics Links

If you are not a fiber customer today, you will be tomorrow. Telephone companies will soon be joined by other providers (including Cable) to seek your fiber business. You may be fortunate enough to be served by separate fiber vendors with separate fiber systems and routing to enhance reliability and uptime. Special installation techniques are essential to make sure fiber links will not be bothered by earth movement, subject to vandalism, or vulnerable to single point failure.[9]

Fiber should be installed underground in a sturdy plastic sheath called an interliner.[10] This sheath offers protection from sharp rocks or other forces that might cause a nick or break in the armor of the cable, or actually sever one or more of the bundled fibers. Systems that only have aerial rights-of-way on utility poles for their fiber may not prove as reliable in some areas as underground fiber. Terminal equipment for fiber should be installed in quake-secure equipment racks away from flooding hazards. Fiber electronics should have a minimum of two parallel DC power supplies which are in turn paralleled with rechargeable battery backup.

Sonet® technology is a proven approach you should look for from your fiber vendor. This solution is based on topology that looks like a circle or ring. A fiber optics cable could be cut just like a wire or cable. A ring-like network will automatically provide a path in the other direction, away from the break. Caution! Fiber installations that run through unsealed access points such as manholes can be an easy target for terrorism or vandalism.

Satellite

Ku or C Band satellite is a costly but effective way to link critical communications elements. C Band has an added advantage over Ku during heavy rain or snow storms. Liquid or frozen water can disrupt Ku Band satellite transmission. So can thick smoke. A significant liability of satellite transmission for ultra-reliable facilities is the possibility that a storm or major fire could cause a deep fade, even for C Band links. Another liability is short but deep semi-annual periods of "sun outage" when a link is lost while the sun is focused directly into a receive dish. While these periods are predictable and last for only a minute or two, there is nothing that can prevent their effect unless you have alternate service on another satellite with a different sun outage time, or terrestrial backup.

EMERGENCY POWER

UPS, or Uninterruptible Power Supplies, are now common in the broadcast workplace. From small UPS that plug into wall outlets at a personal computer work station, to giant units that can power an entire facility, they all have one thing in common — batteries.

Batteries are not immortal

UPS batteries have a finite life span. Once exceeded, an UPS is nothing more than an expensive door stop. UPS batteries must be tested regularly. Allow the UPS to go on line to test it properly. Some UPS test themselves automatically. Routinely pull the UPS AC plug out of the wall for a manual test. Some UPS applications require hours of power, while some only need several minutes.

Powerful perceptions

While UPS provide emergency power when the AC mains go dead, many are programmed with another electronic agenda: Protect the devices plugged in from what the UPS thinks is bad power. Many diesel generators in emergency service are not sized for the load they have to carry, or do not have proper power factor correction. Computers and other devices with switching power supplies can distort AC power wave forms. The result: Power degraded to the point that a "smart" UPS will not trust it. The solution may be as easy as using a less educated UPS if the equipment being protected can tolerate it.

The wrong side of the power curve

After an UPS comes on line, it should go back to sleep after the emergency generator picks up the load and charge its batteries. If it senses the AC equivalent of poison, it stays on, or cycles on and off. It's battery eventually runs down. Crash! Your best defense is to test your entire emergency power system under full load. If an UPS cycles on and off to the point that its batteries run down, you must find out why. Consult your UPS manufacturer, service provider, or service manual to see if your UPS can be adjusted to be more tolerant. Some UPS cycling may be unavoidable with engine-based emergency power when heavy loads like air conditioner compressors cycle on and off.

Off to a good start?

Technicians sometimes believe that starting an emergency generator with no equipment load is an adequate weekly test. If your generator is diesel-driven, this causes a condition known as WET STACKING. Wet Stacking occurs when a generator is run repeatedly with no load or a light load. When the generator is asked to come on line to power a full equipment load, deposits that build up during no-load tests prevent it from developing full power under load.The cure is to always test with the load your diesel has to carry during an emergency. If this is not possible, obtain a resistive load bank to simulate a full load for an hour or two of hard running several times per year.

There no fuel like an old fuel

Fuel stored in tanks gets old. Old fuel is unreliable. Gum and varnish can form. Fuel begins to break down. Certain forms of algae can grow in diesel oil. Fuel additives can extend the storage period, and prevent algae growth. A good filtering system, and a planned program of cycling fuel through it, can extend storage life dramatically. Fuel chemical composition, fuel conditioners, and the age and type of storage tank all affect useful fuel life.

Fuel facts

There are companies that will analyze your fuel. If necessary, they can filter out dirt, water, and debris that can rob your engine of power. The cost of additives and fuel filtering is nominal compared to the cost of new fuel plus hazardous material disposal charges for old fuel. Older fuel tanks can spring leaks that either introduce water into the fuel, or introduce you to a costly hazardous materials clean up project. Your tank will be out of service while it is being replaced.

Avoiding crank(case) calls

While you are depending on an emergency generator for your power, you would hate to see it stop. A running generator will consume fuel, crankcase oil, and possibly radiator coolant. You should know your generator's crankcase oil consumption rate so you can add oil well before the engine grinds to a screeching, non-lubricated halt. Water-cooled generators must to be checked periodically to make sure there is enough coolant in the radiator. Make sure you have enough coolant and oil to get the facility through a minimum of one week of constant duty.

Generator maintenance

Most experts recommend a generator health check every six months. Generators with engine block heaters put special stress on fittings and hoses. Vibration can loosen bolts, crack fittings, and fatigue wires and connectors. If your application is super-critical, a second generator may give you a greater margin of safety. Your generator maintenance person should take fuel and crankcase oil samples for testing at a qualified laboratory. The fuel report will let you know if your storage conditions are acceptable. The crankcase oil report might find microscopic metal particles; early warning of a major failure.

Mission impossible?

Mission dictates Need. Need dictates Reliability. If the design budget permits, a second or even third emergency generator is a realistic insurance policy. When you are designing a facility you know must never fail, consider redundant UPS wired in parallel.[11] During major overhauls and generator work, as well as your source for major emergencies, make sure you have a reliable nearby source for reliable portable power. High power diesel generators on wheels are common now to supply field power for events from rock concerts to movie shoots. Check your local phone directory under GENERATORS-ELECTRIC or EMERGENCY LIGHTING EQUIPMENT. If you are installing a new diesel, remember that engines over a certain size may have to be licensed by your local air quality management district, and that permits must be obtained to construct and store fuel in an underground tank.

AIR HANDLING SYSTEMS

People and equipment crash when they get too hot. Clean, cool, dry and pollutant-free air in generous quantities is critical for modern communications facilities. If you lease space in a high-rise, you may not have your own air system. Many building systems often have no backup, are not supervised nights and weekends, and may have uncertain maintenance histories.

Air Defense

Your best protection is to get the exact terms for air conditioning nailed down in your lease. You may consider building your own backup system, a costly but essential strategy if your building air supply is unreliable or has no backup. Several rental companies specialize in emergency portable industrial-strength air conditioning. An emergency contract for Heating Ventilating and Air Conditioning (HVAC) that can be invoked with a phone call could save you hours or even days of downtime. Consider buying a portable HVAC unit if you are protecting a super-critical facility. Don't forget to have several hundred feet of inexpensive flexible ducting and a lot of duct tape on hand in advance.

Putting a damper on bad air

Wherever your HVAC comes from, there are times when you need to make sure the system can be forced to recirculate air within the building, temporarily becoming a closed system. Smoke or toxic fumes from a fire in the neighborhood can enter an open system. Toxic air could incapacitate your people. With some advanced warning, forcing the air system to full recirculation could avoid or forestall calamity. It could buy enough time to arrange an orderly evacuation and transition to an alternate site.

Emergency actions

If no recirculation function is designed into the system, be prepared to shut it down for as long as you can. This may buy you time to force the outside dampers closed, or allow the outside source of polluted air to dissipate.

WATER HAZARDS

Water in the wrong place at the wrong time can be part of a larger emergency, or be its own emergency. A common mistake is locating a water heater where it could flood electrical equipment when it finally wears out and begins to leak. Unsecured water heaters can also tear away from gas lines during wind storms or earthquakes, leading to a potential explosion or fire. The water in that heater might be missed for another reason. It could have been a source of clean emergency drinking water.

Floods

Your facility may be located near a source of water that could flood you out. Many businesses are located in flood plains that see major storms only once every 100 or 150 years. If you happen to be on watch at the wrong time of the century, you may wish that you had remembered to stock a large supply of sand bags. Remember to include any wet or dry pipe fire sprinkler systems as potential water hazards.

Water and electricity create a special life safety hazard to you and your staff. Don't trust rubber gloves, rubber rain wear, or other protective gear. Always play it safe and disconnect power when working around flooded areas.

EMP

The EMP (Electromagnetic Pulse) phenomenon that is usually associated with nuclear explosions can disable almost any component in a communications system. EMP energy can enter a component or system coupled to a wire or metal surface directly, capacitively, or inductively. Some chemical weapons can produce EMP, but on a smaller scale.[12]

FEMA and EMP

The Federal Emergency Management Agency (FEMA) has been involved in EMP protection since 1970 and is charged at the Federal level with the overall direction of the EMP program. FEMA provides detailed guidance and, in some cases, direct assistance on EMP protection to critical communications facilities in the private sector. AM, FM and TV Transmitter facilities that need EMP protection should discuss EMP protection tactics with a knowledgeable consultant before installing protection devices on radio frequency circuitry. EMP devices such as gas discharge tubes can fail in the presence of high RF voltage and disable the facility they are supposed to protect.

ALTERNATE SITES

No matter how well you plan, something still could happen that will require you to abandon your facility for some period of time. Government emergency planners usually arrange for an alternate site for their Emergency Operations Centers. Communications facilities can sign mutual aid agreements. Advanced planning is the only way to assure access telephone lines, satellite uplink equipment, microwave, or fiber on short notice. If management shows reluctance to share, respectfully ask what they would do if their own facility is rendered useless.

SECURITY

It is a fact of modern life that man caused disasters must now enter into the planning and risk assessment process. Events ranging from terrorism to poor training can cause the most mighty organization to tumble. The World Trade Center and Oklahoma City terrorist bombings are a warning to us all. Your risk assessment might even prompt you to relocate if you are too close to a potential ground zero.

Hostile takeovers

Federal Communications Commission (FCC) rules still state that licensees of broadcast facilities must protect their facilities from what amounts to hostile takeovers. Breaches in basic security have often led to serious incidents at a number of broadcast facilities throughout the county. It has even happened at major market TV stations. Since it is much easier to prevent such incidents than to recover from them, here are a few suggestions:

- Approve all visits from former employees through their former supervisors.
- Escort non-employees in critical areas.
- Secure roof hatches from the inside, and have alarm contacts on the hatch.
- Check for legislation that may require a written safety and security plan.[13]
- Use video security and card key systems where warranted.
- Keep fences in good repair, especially at unattended sites.
- Install and test alarms at unattended sites
- Limit places bombs could be planted.
- Consider limiting the number of outside windows or using bullet resistant glass.
- Plan a safe way to shut the facility down in case of invasion.

WORKPLACE AND HOME: HAND-IN -HAND PREPAREDNESS

A critical facility deprived of its staff will be paralyzed just as surely as if all the equipment suddenly disappeared. Employees may experience guilt if they are at work when a regional emergency strikes, and they do not know what is happening at home.

Guilt and bad decisions

The first instinct of many employee who are at work when emergencies hit is to go home. This is often the wrong move. Blocked roads, downed bridges, and flooded tunnels are dangerous traps, especially at night. People who leave work during emergencies, especially people experiencing severe stress, often become victims.

An answer to guilt

Encourage employees to prepare their homes, families and pets for the same types of risks the workplace will face. Emergency food and water and a supply of fresh batteries in the refrigerator are a start. Battery-powered radios and flashlights should be tested regularly. If employees or their families require special foods, prescription drugs, eye wear, oxygen, over-the-counter pharmaceuticals, sun block or bug repellent, remind them to have an adequate supply on hand to tide them over for a lengthy emergency.

Workplace and home basics

Heavy objects like bookcases should be secured to walls so they will not tip over. Secure or move objects mounted on walls over beds. Make sure someone in the home knows how to shut off natural gas. An extra long hose can help for emergency fire fighting, or help drain flooded areas. Educate employees on what you are doing to make the workplace safe. The same hazards that can hurt, maim, or kill in the workplace can do the same at home.

Contact!

An excellent preparedness measure is to identify a distant business, relative or friend who can be the emergency message center. Employees may be able to call Aunt Tilly from work to find out if their family is safe and sound. Disasters that impair telephone communications teach us that it is often possible to make and receive long distance calls when a call across the street will not get through. Business emergency planners should not overlook this hint. A location in another city, or a key customer or supplier may make a good out-of-area emergency contact.

On the road

Personal and company vehicles should have emergency kits that contain basic emergency supplies. Food, water, comfortable shoes, and old clothes should be in this kit. If their families are prepared at home or on the road, employees will have added peace of mind that may sustain them until they can get home safely.

Expectations, 9-1-1, and Emergencies
Television shows that show 9-1-1 saving lives over the telephone are inspirational. During a major emergency, resources normally available to 9-1-1 may be unavailable. Emergency experts used to tell us to be prepared to be self-sufficient at the neighborhood and business level for 72 hours or more. Some now suggest a week or longer. Government will not respond to every call during a major disaster. That's a fact.

Managing Fear
Anyone who says they are not scared during a hurricane, tornado, flood, or earthquake is either lying or foolish. Normal human reactions when an emergency hits are colored by a number of factors, including fear. As the emergency unfolds, we progress from fear of the unknown, to fear of the known. While preparedness, practice, and experience may help keep fears in check, admitting fear and the normal human response to fear can help us keep our cool.

Some people find that an effective way to prepare mentally is to review their behavior during previous personal, corporate, or natural emergencies.They consider how they could have been better prepared to transition from normal human reactions like shock, denial and panic, to abnormal reactions like grace, acceptance, and steady performance. The latter behaviors reassure those around them and encourage an effective emergency team. Grace under pressure is not a bad goal.

Other "normal" reactions
Most people experience a rapid change of focus toward one's personal well being when an emergency strikes. "Am I Ok?" is a very normal question at such times. Even the most altruistic people have moments during calamities when they regress. Even the best of us can temporarily become selfish. Once people realize that they do not require immediate medical assistance, they can usually start to focus again on others, and on the organization.

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Federal Emergency Management Agency (FEMA), "Electromagnetic Pulse Protection Guidance, Vols. 1-3," Washington, D.C. 20472

Handmer, John and Parker, Dennis, "Hazard Management and Emergency Planning," James and James Science Publishers

Rothstein Associates, "The Rothstein Catalog on Disaster Recovery and Business Resumption Planning," Rothstein Associates. (914-923-4636)

FOR FURTHER INFORMATION

Associations/Groups

The Association of Contingency Planners, 14775 Ventura Boulevard, Suite 1-885, Sherman Oaks, CA 91483

BICEPP, Business and Industry Council for Emergency Planning and Preparedness, PO Box 1020, Northridge, CA 91328

The Disaster Recovery Institute, 1810 Craig Road, Suite 125, St. Louis, MO 63146

Earthquake Engineering Research Institute (EERI), 6431 Fairmont Avenue, Suite 7, El Cerritos, CA 94530

National American Red Cross, 2025 E Street, N.W., Washington, D.C. 20006

National Center for Earthquake Engineering Research, State University of New York at Buffalo, Science and Engineering Library-304, Capen Hall, Buffalo, NU 14260

National Coordination Council on Emergency Management (NCCEM), 7297 Lee Highway, Falls Church, VA 22042

National Hazards Research & Applications Information Center, Campus Box 482, University of Colorado, Boulder, CO 80309

Business Recovery Planning

Harris Devlin Associates, 1285 Drummers Lane, Wayne, PA 19087

Business Recovery Planning, cont'd.
Industrial Risk Insurers (IRI), 85 Woodland Street, Hartford, CT 06102

MLC & Associates, Mary Carrido, President, 15398 Eiffel Circle, Irvine, CA 92714

Price Waterhouse, Dispute Analysis and Corporate Recovery Dept., 555 California Street, Suite 3130, San Francisco, CA 94104

Resource Referral Service, P O Box 2208, Arlington, VA 22202

The Workman Group, Janet Gorman, President, PO Box 94236, Pasadena, CA 91109

Life Safety/Disaster Response
Caroline Pratt & Associates, 24104 Village #14, Camarillo, CA 93013

Industry Training Associates, 3363 Wrightwood Drive, Suite 100, Studio City, CA 91604

Special Emergency Supplies
Extend-A-Life, Inc., 1010 South Arroyo, Parkway #7, Pasadena, CA 91105

Worksafe Technologies, 25133 Avenue Tibbets, Building F, Valencia, CA

Velcro ® USA, PO Box 2422, Capistrano Beach, CA 92624

Construction/Design/Seismic Bracing
American Institute of Architects, 1735 New York Avenue, NW, Washington, D.C. 20006

Geotechnical/Environmental Consultants
H.J. Degenkolb Associates, Engineers, 350 Sansome Street, San Francisco, CA 94104

Leighton and Associates, Inc., 17781 Cowan, Irvine, CA 92714

Miscellaneous
Data Processing Security, Inc., 200 East Loop 820, Forth Worth, TX 76112

EDP Security, 7 Beaver Brook Road, Littleton, MA 01460

ENDUR-ALL Glass Coatings, Inc., 23018 Ventura Blvd., Suite 101, Woodland Hills, CA 91464
Mobile Home Safety Products, 28165 B Front Street, Suite 121, Temecula, CA 92390

Commercial Filtering, Inc., 5205 Buffalo Avenue, Sherman Oaks, CA 91423 (Fuel Filtering)

ENDNOTES

1. Information from the Northridge earthquake could rewrite seismic building codes for many types of structures. The Northridge quake showed that some high rise structures thought to be quake-safe are not. Designers should be aware that seismic building codes usually allow for safe evacuation. They do not embody design criteria to prevent major structural damage. Quake-safe is not necessarily quake proof.

2. These rubber shock mounts are sometimes called *base isolators*. They are built using laminated layers of Neoprene® rubber and metal. The new Los Angeles County Emergency Operations Center is designed for 14 inches of lateral movement from normal, or a bit more than 28 inches of total movement in one axis. The entire structure floats on 26 of these base isolators, designed to let the earth move beneath a building during an earthquake, damping transmission of rapid and damaging acceleration.

3. A proper Business Impact Analysis (BIA) should yield the answer to the question of what level of damage your operation can sustain and still operate. When the cost of protecting against an emergency exceeds the benefit derived from protection, the law of diminishing returns sets in.

4. Thanks to Mary Carrido, President, MLC & Associates or Irvine, CA. for core elements of Business Resumption. It is far beyond the scope of this chapter to go into more detail on the BIA process.

5. One *G* is of course equal to the force of earth's gravity. A acceleration of 2 *G's* doubles the effective force of a person or object in motion, and nullifies the effectiveness of restraints that worked fine just before the earthquake. Force = (Mass) X (Acceleration). Seismic accelerations cause objects to make sudden stops — 60 to zero in one second.

6. The Richter Scale is no longer used by serious seismic researchers. *Moment magnitude* calculates energy release based on the surface area of the planes of two adjacent rock structures (an earthquake fault) and the distance these

structures will move in relation to one another. Friction across the surface of the fault holds the rocks until enough stress builds to release energy. Some of the released energy travels via low frequency wave motion through the rock. These low frequency waves cause the shaking and sometimes violent accelerations that occur during an earthquake. For more on modern seismic research and risk, please refer to the bibliography for this chapter.

7. **Lock-Out/Tag-Out** is a set of standard safety policies that assure that energy is removed from equipment during installation and maintenance. It assures that every member of a work detail is clear before power is reapplied.

8. If you blow or talk into the telephone transmitter and hear yourself in the receiver, what you hear is called side tone. Telephone systems provide a small amount of feedback so users can judge their voice levels. Side tone depends on DC voltage on the line, an almost 100% sign that the line is connected all the way back to your first phone company Central Office. Connection beyond that point will of course depend on the integrity of the rest of the public switched network (PSN).

9. Single point failure can occur in any system. Single point failure analysis and prevention is based on simple concepts: A chain is only as strong as its weakest link, but two chains, equally strong, may have the same weak link. The lesson may be make one chain much stronger, or use three chains of material with different stress properties.

10. Interliners are usually colored bright orange to make them stand out in trenches, manholes and other places where careless digging and prodding could spell disaster.

11. Consult the vendor for details on wiring needs for multiphase parallel UPS installations.

12. FEMA publishes a three volume set of documents on EMP referenced in the bibliography. They cover the theoretical basis for EMP protection, protection applications, and protection installation. It is listed in the bibliography at the end of this paper.

13. California Senate Bill 198 mandates that California businesses with more than 100 employees write an industrial health and safety plan for each facility, addressing workplace safety, hazardous materials spills, employee training, and emergency response.

FCC TOWER REGISTRATION

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ABSTRACT

This paper will review the Federal Communications Commission's (FCC) new rules for registering towers in the United States. For the purpose of this paper, the terms antenna structure and tower are interchangeable. These terms refer to any structure that is itself an antenna, or has an antenna on it, and is subject to FCC rules.

INTRODUCTION

On November 28, 1995, the Commission adopted a Report and Order (R&O) in WT Docket No. 95-5, which created new rules that will streamline the Commission's antenna structure clearance process. These new rules will replace the current clearance procedures, which apply to licensees and permittees using antenna structures requiring Federal Aviation Administration (FAA) notification, with a uniform registration process for the owners of antenna structures. The Commission also updated Part 17 of the rules entitled "Construction, Marking, and Lighting of Antenna Structures" to be consistent with FAA air safety recommendations and revised the rules to make antenna structure owners primarily responsible for structures that require painting and/or lighting. These changes unify federal guidelines for painting and lighting antenna structures and designate a single point of contact for resolving lighting outages. These actions will decrease burdens on the public and the government by requiring fewer filings with the Commission. We feel these actions will expedite the processing of applications requiring FAA notification.

This proceeding was initiated by the FCC's Tower Standardization Team to explore ways to streamline the Commission's antenna clearance process, to reduce administrative burdens on the public and the Commission, and to ensure safety in air navigation. The Tower Standardization Team was created in 1992 at the author's suggestion when the call came from the Chairman for ideas for Total Quality Management (TQM) teams. Fifty commenters and reply commenters, representing structure owners, Commission licensees, lighting manufacturers, publishers, consulting engineers, and governmental agencies, responded in this proceeding.

HIGHLIGHTS OF THE REPORT AND ORDER

The highlights of this Report and Order are as follows:

- Registration will reduce the number of entities responsible for antenna structure painting and lighting from 900,000 licensees to 75,000 owners.

• Registration will reduce the number of filings concerning minor changes to site data by a factor of twelve.

• Registration will reduce the time it takes to process antenna structure-related applications and notifications requiring consideration of painting and lighting specifications. There were approximately 17,000 such applications and notifications in 1993.

• Electronic filing procedures will decrease the possibility of processing backlogs by permitting owners to register immediately upon receipt of an FAA "no hazard" determination for the antenna structure.

• Updating Part 17 to be consistent with the latest FAA air safety recommendations will unify federal guidelines concerning the painting and lighting of antenna structures.

• Industry and the Commission will save millions of dollars each year.

BACKGROUND

The Commission and the FAA each have the statutory responsibility of ensuring that antenna structures do not present a hazard to air safety. Section 303(q) of the Communications Act of 1934, as amended (Communications Act), mandates that the Commission "require the painting and/or illumination of radio towers" in cases where there is a "reasonable possibility" that an antenna structure may cause a hazard to air navigation. Similarly, Section 1501 of the Federal Aviation Act authorizes the FAA to require that persons proposing to erect a structure provide notice to the FAA, when such notice will promote air safety. In its efforts to promote safe air commerce, the FAA periodically publishes Advisory Circulars, two of which set forth its recommendations for painting and lighting of structures.[1] In 1992, at the request of the FCC, Congress amended Sections 303(q) and 503(b)(5) of the Communications Act to: 1) make antenna structure owners, as well as Commission "licensees and permittees," (licensees)[2] responsible for the painting and lighting of antenna structures, and 2) provide that non-licensee antenna structure owners may be subject to forfeiture for violations of painting or lighting requirements as specified by the Commission.[3]

Since the late 1950's, the Commission has worked in concert with the FAA to promote air safety through the antenna structure clearance process. Currently, each applicant proposing to construct or alter an antenna structure that is more than 200 feet (60.96 meters) in height, or that may interfere with the approach or departure space of a nearby airport runway must notify the FAA of proposed construction. The FAA determines whether the antenna structure constitutes a potential hazard, and may recommend appropriate painting and lighting for the structure.[4] The Commission then uses the FAA's recommendation to impose specific painting and/or lighting requirements on subject licensees.[5]

Approximately 75,000 of the 500,000 antenna structures in the United States require notification to the FAA. On average, there are 12 tenant licensees on each of these 75,000 antenna structures. Currently, each licensee on a structure must notify the Commission of changes in painting and lighting, overall structure height, or corrections to site coordinates. In 1993 alone, licensees filed over 17,000 such notifications. The Notice of Proposed Rule Making (Notice)[6] in this proceeding sought to reduce antenna clearance processing delays by decreasing the number of redundant filings. Thus, in the Notice the Commission proposed to replace the current clearance process with a uniform registration procedure that applies to antenna structure owners. Requiring owners, rather than tenant licensees, to register and notify the Commission concerning changes to these antenna structures will (1) reduce certain filings by a factor of 12 to 1, (2) eliminate thousands of duplicative notifications from licensees each year, and (3) reduce administrative burdens for 900,000 licensees nationwide. The preponderance of commenters, representing antenna structure owners as well as licensees, support the registration procedure substantially as proposed in the Notice.

In the Notice, the Commission further proposed to revise Part 17 to reflect recent changes to the two FAA Advisory Circulars and implement statutory language holding antenna structure owners primarily responsible for compliance with the Commission's painting and lighting specifications. These actions would unify federal painting and lighting guidelines for antenna structures and increase safety of air navigation by providing a single point of contact by which the FAA and the Commission may quickly resolve emergency lighting outages. A majority of the commenters agree that these actions will help to ensure the safety of pilots and aircraft passengers nationwide.

ANTENNA STRUCTURE REGISTRATION
Who must register

Proposal. In the Notice, the Commission proposed to require the owner of each antenna structure requiring FAA notification to file a registration application with the Commission for each applicable existing or proposed antenna structure. The Commission proposed to register each structure meeting the registration criteria, including structures that make up AM arrays.[7] We asked whether this proposal should be extended to include all antenna structures, regardless of height above ground level or proximity to nearby airports. Further, the Commission asked whether the Commission should register voluntarily painted or lighted structures.

Decision. Based on the overwhelming support exhibited by all sectors of the telecommunications industry, the Commission will replace our current antenna structure clearance procedures with a uniform registration applicable only to the antenna structure owner. The Commision anticipates a 12 to 1 reduction in filings related to changes in structure height, painting, or lighting. A single entity will be responsible for notifying the Commission of changes to the structure, rather than each tenant licensee on the structure.[8] Thus, registering antenna structures will reduce economic and administrative burdens on the public and the Commission. Further, the registration database will increase air safety by allowing the FAA and the Commission rapidly to identify a single responsible entity during lighting outages. This registration requirement does not duplicate information collection burdens already imposed by the FAA. In fact, the FAA does not retain or compile notifications for future reference. Therefore, we cannot fulfill our statutory responsibilities under the Communications Act or improve the antenna structure clearance process using data already provided to the FAA. We intend, however, to work in concert with the FAA to unify public burdens related to clearing potential hazards to air navigation.

The final rules require the owner of each antenna structure requiring FAA notification to register the structure with the Commission. The Commission sees little benefit, however, in broadening the original proposal to include all antenna structures. The initial goal of this proceeding was to streamline the antenna clearance process that affects only 75,000 of the

500,000 antenna structures in the United States. Only those structures that meet FAA notification criteria have been identified as potential hazards to air navigation. The remaining structures are either less than twenty feet above existing terrain, shielded by nearby obstructions, or otherwise well below established flight paths of aircraft. The Commission will not register structures that do not meet the FAA notification criteria.

The final rules also require registration for each subject antenna structure, including structures that make up AM directional arrays. In order to streamline the antenna clearance process, we must individually register each structure. By using a "group" registration, as suggested by one commenter, we would merely create a new set of administrative burdens which would slow down the processing of structure modifications. For example, structures in an AM array may routinely be located hundreds of meters apart, vary in height, and have different painting and lighting requirements. By registering each applicable structure within AM arrays, we can easily identify the painting and lighting requirements for each structure. Registering a "group" of structures, however, would require a detailed filing, describing the layout of the array and the FAA recommendations for each applicable structure. The "group" would not benefit from the expedited processing of future modifications to the registration because there would be no automated means to discern exactly which structure within the array requires modification. Further, registering "groups" of structures would lengthen and complicate the registration form and effectively prevent electronic registration for these structures. In making this decision, we are aware that the owners of approximately 1,800 AM arrays will be required to file multiple registration forms. We do not believe, however, that this is an unreasonable burden when balanced against the public interest benefits of ensuring safety in air navigation, streamlining our clearance process, speeding the disposal of simple modifications, and obtaining an accurate database for air safety purposes. We also note that completing the one-page Form 854 by the structure owner and processing by the staff requires a modest amount of time, and does not require an accompanying fee.[9]

Consistent with our stated policy, the Commission will only require registration of antenna structures that require FAA notification. Thus, not every structure within an AM array would necessarily be registered. Further, the registration number issued to structures within a given array would not be related numerically or otherwise. There is no compelling air safety concern to offset the significant administrative burdens associated with providing special registration numbers for antenna structure arrays. Notwithstanding our registration requirements, we will continue to solicit the coordinates for the center of the array when a licensee files for a construction permit using FCC Form 301, since we use the center coordinates for calculating signal coverage.

When to register

Proposal. In the Notice, the Commission proposed to require all new structures meeting FAA notification criteria to be registered prior to construction. For example, a prospective owner would notify the FAA of proposed construction, receive a determination from the FAA, and then register the structure with the Commission prior to construction. Additionally, the Commission asked whether the registration process for existing structures should be implemented initially by geographic region, by antenna structure height, upon license renewal, or any other logical grouping that would expedite implementation.

Decision. After July 1, 1996, all new antenna structures meeting the notification criteria must be registered with the Commission. However, for existing antenna structures, the Commission agrees with a majority of the commenters that prescribing filing windows by region is the simplest method to register these structures. Therefore, the Commission is requiring owners to register existing structures by state, in accordance with the filing windows set forth. In this manner, existing structures will be registered over the two year period between July 1, 1996, and June 30, 1998. Notwithstanding the filing windows, owners registering existing antenna structures that require re-notification to the FAA (e.g., owners correcting an error in site data) must register immediately upon receipt of the new FAA determination.

Further, the Commission will require that each applicable antenna structure be registered prior to construction. Proposed antenna structures that are determined by the FAA to present a potential hazard to air navigation must be lighted during construction. The Commission's registration process will be the federal government's only method of requiring such safety lighting, as the FAA does not have statutory authority to mandate the painting or lighting of antenna structures. Further, the timely registration of all applicable antenna structures is essential to the integrity of the new registration database. The Commission intends to eliminate the possible processing delay mentioned by the commenters by permitting owners to register electronically. In cases where a structure owner wishes to begin construction immediately upon receipt of the FAA determination for the structure, the electronic filing capability will enable the owner to register the structure with the Commission and receive a registration number within minutes. By providing an electronic filing option, we can satisfy the concerns of the personal communications service industry regarding speed of service, while registering antenna structures in a timely manner.

How to register

Proposal. In the Notice, the Commission attached a draft copy of revised FCC Form 854 "Application for Antenna Structure Registration" and sought comment on ways to streamline the form and expedite processing, such as permitting electronic filing. The Commission also asked whether a registration fee should be imposed in order to recover the administrative costs associated with processing applications, administering the new registration database, and providing database access to the public and other government agencies. And finally, the Commission asked whether a registration renewal process would be necessary in order to maintain the integrity of the data.

Decision. Although some of the same site information is required on FAA Form 7460-1 (FAA form) and the revised FCC Form 854, the forms are incompatible to the extent that neither agency can adopt the other's form in its current format. In our case, using the FAA Form would significantly delay or possibly prevent the processing of registration requests. For example, we seek to obtain information concerning the owners of antenna structures. The FAA form, however, does not specifically request information about the structure owner. Additionally, the questions on the FAA form are, for the most part, not compatible with electronic filing, which will be a major component of our new registration process. Notwithstanding the difficulties involved in developing a dual-use form to satisfy the information gathering needs of both agencies, the Commission will continue to work closely with the FAA, site owners organizations, and other interested parties to seek ways to decrease the public burden related to antenna structure registration. One possible solution might be to combine the forms to fulfill the needs of both agencies. At present, however, the Commission will use revised FCC Form 854 as the sole means to register antenna structures.

Based on the overwhelming support of the commenters, the Commission will permit owners, at their option, to register electronically. Permitting owners to file electronically will eliminate processing delays by registering antenna structures within minutes of filing an application. Not all owners will have similar needs when filing FCC Form 854, thus the Commission intends to permit owners to file electronically or in the traditional manner via mail. Using either filing method, the owner of a proposed structure could begin construction as soon as a registration number is obtained. Detailed instructions on how to register electronically will be described in a future Public Notice and published in the Federal Register.

The Commission will not require owners to pay a registration fee. Although there may be significant start-up costs involved in implementing the registration process, we agree that a nominal registration fee could deter some owners from registering structures, would reduce the speed of service in processing registrations, and would complicate electronic filing procedures. Instead, the Commission will seek to recover administrative costs through the economic benefits of the streamlined system and by charging a nominal fee to those who wish to access the antenna registration database. The site information contained in the new database will be of considerable interest to law firms and communications engineers. Therefore, after a majority of the existing structures are registered, we may consider permitting on-line queries of the antenna registration database through a "900" telephone number on a fee-per-minute basis. Prior to providing a "900" telephone number service, the Commission will initiate a notice and comment proceeding concerning applicable fees. Such fee-per-minute charges would only apply to value-added access to the database, such as on-line status reports for registration applications. Database searches and copies of the entire database, however, would continue to be available through the Commission's copy contractor or free of charge through the Internet. Further, we will continue to investigate the creation of other tools, such as providing the registration database in CD ROM format. This means of recouping the administrative costs would allow the users of the registration information, rather than the suppliers of the information, to fund the initiative.

Finally, the Commission is not requiring owners to renew their registration periodically. While it is important to maintain the validity of the site data contained in the registration database, there is little benefit in mandating a renewal process for tens of thousands of owners, when only a small percentage will need to update registration information in the next five or ten years. Therefore, we will not require owners to renew their antenna structure registrations on a periodic basis. Instead, owners must notify the Commission, using FCC Form 854, of any change of structure height, ownership, owner's address, or upon dismantling the structure.

SUPPLEMENTAL FILING / POSTING REQUIREMENTS

Proposal. In registering over 75,000 antenna structures, many owners will likely initiate site surveys to ensure the validity of site information. With the proliferation of inexpensive satellite-based locating devices such as Global Positioning System (GPS) receivers, some structures that may have been previously located using an area map can now be easily located to a higher degree of accuracy. In the

Notice, the Commission asked whether we should require owners to specify site latitude and longitude to the nearest second and structure height to the nearest meter. Further, the Commission proposed to require that the antenna structure registration number be displayed and be readily visible from the base of each registered structure.

Decision. Based on the commenters' support for a standard format, the Commission will request location data in terms of degrees, minutes, and seconds, and height data to the nearest meter. The antenna structure registration database will accept latitude and longitude data, in either the NAD 27 or NAD 83 datum, up to an accuracy of one second and height to one meter. Owners must specify which datum is used and may use surveying tools of differing accuracy, such as maps, GPS receivers, or GPS receivers with differential corrections to obtain site data. It is left to each owner, however, to evaluate the surveying method being used and round to the appropriate significant digit.[10]

The Commission also agrees with the commenters' view that we should facilitate the submission of accurate data by structure owners and provide a means for owners and licensees to correct inaccurate data. Therefore, in general, the Commission will not issue forfeitures to owners or licensees attempting to correct errant site data during registration. Commission authorizations and FAA determinations of "no hazard" to air navigation, however, are based on the originally submitted site data. Thus, changing the coordinates on tenant licensees' authorizations, depending on the magnitude of the error, may violate the interference protection criteria set forth in the rules or may invalidate the original FAA determination for the site. For example, a five-second error in longitude may have caused a broadcast station to be authorized too close to an airport, another broadcast station, a radio frequency quiet zone, or an international border.

Therefore, the Commission clarified the procedures by which owners and tenant licensees may ensure that correct site data appear in the Commission's registration and licensing databases. Owners must submit accurate site data without regard to the height or coordinates listed on tenant licensees' station authorizations. Existing structures will be assigned painting and lighting specifications upon registration based on site data from the original FAA determination for the structure or from site data referenced in the Commission's current antenna clearance database. Corrections of previously submitted site data of less than one second in latitude or longitude, or of less than a foot in height, will not require a new aeronautical study, and the structure will retain the previously assigned painting and/or lighting specifications. The FAA, however, requires a new aeronautical study for corrections in latitude or longitude of one second or more, or a correction in height of one foot or more. In this case, the owner must seek a new FAA determination prior to registration, and the structure will be assigned painting and/or lighting requirements based on the new FAA determination.[11] Tenant licensees should note any discrepancies in the site data appearing on the registration (FCC Form 854-R) and their station authorizations and notify the appropriate licensing branch.[12] Tenant licensees will not be required to submit a filing fee when correcting site data. In cases where a correction of site data for a tenant licensee would be in violation of the Commission's Rules for a particular radio service, the tenant license(s) involved may be required to take measures to avoid harmful interference, such as decreasing antenna height, reducing power, or employing a directional radiator. In general, however, we will not require tenant licensees to cease operations while the owner seeks a new FAA determination or while coordinating corrections with the individual licensing branches within the Commission.

The Commission agrees that antenna structures on large buildings or landmarks may require a different posting requirement than free-standing antenna structures. Therefore, in the case of antenna structures located on top of buildings, we will clarify the rules to require that the registration number be posted at the base of the antenna structure, not the base of the building. Further, the Commission will not require the registration number to be conspicuously posted at the base of antenna structures in cases where a federal, state, or local government entity determines that such a posting would detract from the appearance of a historic landmark. In such cases, however, the owner must make the registration number available to representatives of the Commission and the FAA or members of the general public upon reasonable demand.

STATUTORY CONSIDERATIONS

Proposal. In the Notice, the Commission asked whether the proposed antenna structure registration process would require any changes to the Commission's environmental rules, 47 C.F.R. §§ 1.1301-1.1319. Specifically, the Commission sought comments on whether our registration requirements constitute an "action" under the National Environmental Policy Act of 1969 (NEPA) or an "undertaking" under the National Historic Preservation Act (NHPA). Further, the draft FCC Form 854 attached to the Notice also included a proposed certification required under the Anti-Drug Abuse Act of 1988 (ADA). Currently, Commission licensees must comply with the provisions of NEPA and certify their status under the ADA prior to receiving a station authorization.

Decision. The Commission determined that registering a structure constitutes a "federal action" or "federal undertaking," such that the imposition of environmental responsibilities on the structure owner is justified. At the outset, the owner may be proposing to register and construct a structure at a location that significantly affects the quality of the human environment within the context of NEPA. The Commission believes that irreparable harm to the environment may be avoided by requiring owners to assume responsibility for environmental compliance at the outset. Moreover, the Commission believes that such a requirement will effectuate the implementation of federal environmental policies which require that environmental considerations be integrated into the early planning stages of authorized actions and undertakings.[13] This is particularly true here because the location of an antenna structure in a sensitive site area, as defined by Section 1.1307(a) of our Rules, will, in most situations, have the greatest effect on the environment. The subsequent application for an authorization on the structure is a federal action which may have little, if any, additional environmental consequences.

Further, the Commission will divide the responsibility to comply with our environmental rules between owners and tenant licensees, as appropriate. Clearly, a distinction exists between environmental responsibilities pertaining to structural matters and responsibilities pertaining to radio frequency radiation levels. In order to clarify the division of responsibility for compliance with our environmental rules between owners and licensees and to notify non-licensee owners of their new responsibilities in this regard, the Commission will revise FCC Form 854 to state which environmental concerns must be taken into account by the owner at registration and which concerns are the responsibility of the licensee filing for an authorization. In general, owners who file FCC Form 854 must also comply with federal environmental rules pertaining to the site at the time of registration, while radio frequency radiation levels at the site will be the responsibility of the tenant licensee(s). For new structures, under our environmental rules, a structure owner will be required initially to identify whether the proposed site is in a sensitive location as defined in Section 1.1307 of our Rules. If so, the structure owner must prepare and file an Environmental Assessment, which must be reviewed by the Commission staff prior to the structure's registration and construction.[14] For existing structures, tenant licensees presumably have complied with our environmental rules in locating their facilities. In the event that the owner is unable to register the structure due to the ADA, the first tenant licensee authorized on the structure would be responsible for registering the structure and complying with NEPA pertaining to the site, as well as radio frequency radiation levels, at the time of registration. Notwithstanding the procedures set forth above, licensees authorized on antenna structures not subject to registration would remain responsible for fully complying with the Commission's environmental rules.

In accordance with the ADA, revised FCC Form 854 will include an anti-drug certification, consistent with Amendment of Part 1 of the Commission's Rules to Implement Section 5301 of the Anti-Drug Abuse Act of 1988, 6 FCC Rcd 7551 (1991) (Anti-Drug Order). In the Anti-Drug Order, we adopted the APA definition of "license," which broadly construes a "professional and/or commercial license" to include "the whole or part of an agency permit, certificate, approval, registration, charter, membership, statutory exemption or other form of permission."[15] This broad definition of "license" clearly encompasses antenna registration. If the owner of a structure is denied registration because he cannot so certify, the Commission would not require existing tenant licensees to vacate the structure nor would future tenant licensees be precluded from placing antennas on the structure. Rather, the first tenant licensee authorized on the structure would be responsible for registering the structure and providing a copy of the registration (FCC Form 854-R) to the owner. The owner would still be responsible for painting and lighting the structure, providing a copy of the registration to all tenant licensees, and posting the registration number.

PART 17 UPDATE

Proposal. In the Notice, the Commission noted that parties intending to construct or modify an antenna structure may be required to seek a determination from the FAA as to whether the proposed structure is a potential hazard to air navigation. In cases where an antenna structure may pose a hazard to air navigation, the FAA may recommend painting and/or lighting in accordance with two of its Advisory Circulars, "Obstruction Marking and Lighting" (AC 70/7460-1H), August 1991, as amended by Change 2, July 15, 1992, and "Specification for Obstruction Lighting Equipment" (AC 150/5345-43D), July 1988. Part 17 of the Rules sets forth requirements for painting and lighting antenna structures, which generally reflect earlier versions of the two FAA Advisory Circulars. The Commission has the authority to specify painting and lighting requirements other than those listed in Part 17 in cases where current guidelines are inadequate to ensure air safety.[16] Pursuant to this authority, the Commission generally relies on the FAA's recommendation, and not Part 17 of the Rules, when prescribing painting and/or lighting for each licensee on a given antenna structure.

In order to update our rules in accordance with the most recent FAA Advisory Circulars, and streamline Part 17, the Commission proposed to incorporate by reference into Part 17 the painting and lighting recommendations contained in the Advisory Circulars listed above. Under our proposal, owners of antenna structures that received clearance prior to January 1, 1996, would retain the old painting and/or lighting requirements but would be required to comply with the Advisory Circulars within ten years. Further, each owner registering a new antenna structure would be assigned painting and/or lighting requirements referenced in Part 17 at the time of registration. The Commission proposed that, once a new or existing structure has been assigned painting and/or lighting requirements via the registration process, the structure could be maintained in accordance with the registration for an indefinite period.

Decision. The Commission will incorporate by reference FAA Advisory Circulars "Obstruction Marking and Lighting" (AC 70/7460-IH), August 1991, as amended by Change 2, July 15, 1992, and "Specification for Obstruction Lighting Equipment" (AC 150/5345-43D), July 1988, in Part 17 of the Rules.[17] If the FAA makes substantive amendments to either of the Advisory Circulars, the Commission must initiate a public proceeding prior to updating Part 17 of the Rules. Non-substantive changes could be handled by simply issuing an Order. Substantive changes, for example, include increasing the number of red beacons on all structures, while non-substantive changes could include updating an address or phone number listed in the documents. Our proposal is merely to incorporate by reference the current FAA painting and lighting recommendations into Part 17 of the Rules. This will not change, in any way, the current procedure in which the FAA recommends specific painting and/or lighting specifications for a particular structure. We will continue to use the FAA recommendation in meeting our statutory responsibility under Section 303(q) of the Communications Act to prescribe appropriate painting and/or lighting requirements for antenna structures.

The Commission will not require owners to update painting and lighting requirements unless specifically recommended by the FAA. At present, the FAA recommends painting and lighting for antenna structures prior to construction only, and thereafter does not recommend periodic updates. We agree with the commenters that requiring owners to update painting and lighting requirements in accordance with Advisory Circulars would place significant new economic and administrative burdens on owners. Therefore, the Commission will not require owners to comply with the new Advisory Circulars unless such action is specifically recommended by the FAA. For existing structures, Form 854R (antenna structure registration) will, in most cases, denote the specific painting and lighting requirements originally assigned to the structure. Owners may retain the original painting and lighting requirements indefinitely or may apply to paint and light in accordance with current FAA recommendations. For new construction or alteration of existing structures, Form 854R will reference the FAA Advisory Circulars found in Part 17 of the Commission's Rules at the time of registration.

NEW REQUIREMENTS FOR ANTENNA STRUCTURE OWNERS

Primary responsibility

Proposal. In the Notice, the Commission stated that the antenna structure owner would be primarily responsible for maintaining prescribed structure painting and/or lighting in accordance with Part 17.

Decision. After careful consideration of the comments, the Commission continues to believe that the antenna structure owner should have the primary responsibility for maintaining the prescribed painting and lighting of the structure, while a secondary responsibility should be imposed on the individual licensees on the structure. The Commission rejected the view that licensees on the structure should be relieved of all responsibility for maintaining the prescribed painting and lighting of the structure. The reason is twofold. First, in enacting Public Law No. 102-538, 106 Stat. 3533 (making structure owners as well as Commission licensees responsible for the painting and lighting of antenna structures and making non-licensee structure owners subject to forfeitures), Congress did not, in any way, suggest that licensees should be relieved of their responsibility to maintain the prescribed structure painting and/or lighting. Second, one of the Commission's primary responsibilities in this area is ensuring that antenna structures do not pose a threat to air safety. Thus, for compelling public safety reasons, the Commission must have means to ensure that prescribed structure painting and/or lighting is maintained at all times and that lighting outages will be promptly rectified. To this end, the Commission believes that continuing to impose a responsibility on licensees will make it incumbent on them to assure that the structure owner maintains prescribed painting and/or lighting, and, if necessary, take steps to maintain painting and/or lighting in the event of default by the structure owner.

The Commission emphasized that under normal circumstances, we will only look to the structure owner to maintain the prescribed painting and/or lighting. However, in the event the structure owner is unable to maintain the prescribed painting or lighting, e.g., in cases including but not limited to abandonment, negligence, or bankruptcy, the Commission would require that individual licensees on the structure undertake efforts to maintain painting and lighting upon request by the Commission. Additionally, if a tenant licensee has reason to believe that the structure is not in compliance or that the owner is not carrying out its responsibility to maintain the structure as required by Part 17 of the Rules, the licensee must immediately notify the owner, notify the site management company (if applicable), notify the Commission, and make a diligent effort to ensure that the antenna structure is brought into compliance. The Commission is not, however, requiring licensees to independently monitor the antenna structure. Instead, licensees must assume responsibility and take appropriate action if circumstances would lead a reasonable person to question whether the structure is being maintained. For example, if a licensee becomes aware that electrical power is no longer available at the site or has rental payment for antenna space returned due to unavailability of the owner, the licensee must take reasonable actions to ensure that the structure is immediately brought into compliance. Under these circumstances, any sanction that may be directed to a licensee will be determined on a case-by-case basis depending upon the magnitude of noncompliance, its length of time, access of the licensee to the structure and the diligence of the licensee to rectify the noncompliance with the prescribed painting or lighting or to alert the Commission or the FAA.

DEFINITION OF "OWNER"

Proposal. The rules proposed in the Notice defined the owner of an antenna structure.

Decision. For the purposes of Part 17, the Commission defined the owner as the individual or entity vested with ownership, equitable ownership, dominion or title to the structure. In this regard, where a party leases land and then builds a structure, during the term of the lease only the structure owner will be deemed to be an "owner" under the rules and have primary responsibility to maintain the prescribed structure painting and/or lighting. In the event the land owner acquires possession of the structure, the land owner would then become the structure owner.

The Commission stated that in situations where the legal owner is a financial fiduciary entity (such as a bank or mortgage company), the ownership entity may not be responsive, nor aware of the painting or lighting requirements. It is for this reason that we have included "equitable owner" and "dominion" within our definition of owner. By these terms, we mean the party in possession or control of the antenna structure irrespective of any mortgage or lien on the property or the antenna structure. In the event of acquisition by default, foreclosure, or other process that may result in a transfer of ownership to or among financial institutions, we will proceed on two fronts. First, to cover the immediate maintenance requirements of an antenna structure of uncertain or contested ownership, we expect licensees on the structure to assume their secondary responsibility to maintain the required painting and/or lighting. Second, upon ascertaining the identity of any new ownership entity, the Commission will inform and hold it responsible for maintaining the prescribed structure painting or lighting.

In none of these situations, however, will either the owner or licensee be relieved of responsibility for maintaining the prescribed painting and/or lighting of the structure. There is nothing contained in the 1992 revision of Sections 303(q) and 503(b)(5) of the Communications Act which authorizes making structure owners responsible for the painting and lighting of antenna structures, that suggests that a site management company without any ownership connection to the structure should fall within the definition of "owner." As such, there is no statutory basis to assess a forfeiture against a site management company. It is also our view that it would not be in the public interest to permit licensees to circumvent their secondary responsibility to maintain prescribed painting and/or lighting by merely entering into a contractual arrangement with the structure owner or site management company which precludes access to the structure. Notwithstanding private contractual arrangements, licensees and structure owners, are and will continue to be, held responsible for maintaining the prescribed structure painting and/or lighting. Any resolution concerning a failure to perform pursuant to a private contractual arrangement, including appropriate remedies or damages, are matters to be resolved in a local forum.

NOTICE TO OWNERS AND LICENSEES

Proposal. Both Section 503(b) of the Communications Act and Section 1.80(d) of the Commission's Rules, 47 C.F.R. § 1.80(d), require that non-licensee antenna structure owners be given notice of their painting and/or lighting obligations prior to the issuance of a forfeiture penalty. In the Notice, the Commission asked what form of notification would be sufficient to inform owners of their obligation to register, paint, and light their structures.

Decision. The Commission will provide notice regarding antenna structure registration and the specific roles of owners and tenant licensees through several methods. First, the Commission will publish a summary of this Report and Order and the final rules in the Federal Register. Second, we will contact site owners associations and communications industry trade publications in order to publish summaries of this action in various media. Third, we urge all tenant licensees to notify non-licensee antenna structure owners regarding the rules outlined herein. These three means of notification should be sufficient to ensure that all owners, including Commission licensees and non-licensees, receive notice of their new responsibilities. Sending notification letters to all existing licensees regarding these new responsibilities would not target non-licensee owners, would not provide a greater level of notification than the three methods described above, and would place a large administrative burden on the Commission. Further, because this is the first time the Commission has attempted to identify antenna structure owners, there is no means at the Commission's disposal to notify structure owners individually.

CONCLUSION

The actions in this proceeding will serve the public interest: (1) by expediting application and notification processing; (2) by streamlining regulations to ease the public and governmental burdens associated with processing certain applications; (3) by unifying federal regulations regarding antenna structure painting and lighting; (4) by increasing safety of air navigation; (5) by establishing that primary responsibility for antenna structure operation and maintenance rests with the owner; and (6) by saving industry and the Commission millions of dollars each year.

Author's Note: As team leader, I would like to thank the members of the Tower Standardization Team for their efforts and insights over a three year period. The Tower Standardization Team members are as follows: Lisa Stover, Steve Markendorff, and Roger Noel of the Wireless Telecommunications Bureau; Jim Voigt and George Dillon of the Compliance and Information Bureau; and Robert Greenberg, Robert Hayne and Sharon Bertelsen of the Mass Media Bureau.

Editor's Note: The opinions expressed by the author are not necessarily those of the Federal Communications Commission.

1.See, e.g., Obstruction Marking and Lighting (AC 70/7460-1H) released in August of 1991, as amended by Change 2, July 15, 1992, and Specification for Obstruction Lighting Equipment (AC 150/5345-43D) released in July of 1988.

2.Although there is a functional distinction between licensees and permittees, both entities assume the same responsibilities regarding painting and lighting antenna structures. Thus, from this point on, the word "licensees" shall confer the same meaning as the phrase "licensees and permittees," regarding secondary responsibilities for structure maintenance.

3.See Pub. L. No. 102-538, 106 Stat. 3533, enacted October 27, 1992. In essence, Section 503(b)(5) authorizes the Commission to assess forfeitures for violations of Section 303(q) of the Communications Act if the owner has been previously notified regarding specific painting and/or lighting obligations. This authority is now reflected in 47 C.F.R. § 1.80(d).

4.For instance, the FAA may recommend that the proposed structure be equipped with red beacons and/or white strobe lights and be painted with white and aviation orange bands. In addition, Part 17 of the Commission's Rules sets forth the requirements for painting and lighting of antenna structures, specifying the number, type and location of such lights based upon the height of the structure as well as the colors and pattern to be used in painting the structure.

5.In most cases, painting and/or lighting requirements are specifically listed on each subject licensee's station authorization.

6.Notice of Proposed Rule Making, WT Docket No. 95-5, 10 FCC Rcd 2771 (1995).

7.Many AM broadcast stations are authorized on the basis of a directional antenna. An AM directional array involves multiple antenna structures. The reference coordinates for such an AM station mark the center of the array, rather than the exact location of each antenna structure. Under the new rule, each structure in the array requiring notice to the FAA would be registered.

8.Licensees remain responsible for compliance with the administrative requirements of their radio services, including notification of changes in site data, if applicable.

9.Similarly, the effort required to complete and process Form 854 to reflect a change in structure height, change in painting and/or lighting requirements, or dismantlement of the structure is minimal.

10.Seven and one-half minute geological maps may yield accuracies within 1 second; GPS receivers may be accurate to 100 meters ($\approx$3.3 seconds), while GPS receivers using differential corrections may be accurate to 1 meter ($\approx$0.05 second).

11.Each antenna structure owner must provide a copy of the registration to all tenant licensees.

12.The Commission intends to revise all application forms permitting applicants to supply an antenna structure registration number or numbers. The Commission will release a Public Notice noting the revision of applicable forms.

13.See 40 C.F.R. §§ 1500.5(a), 1501.2 (regulations by the Council on Environmental Quality); 50 C.F.R. § 402.11 (regulations by the Fish and Wildlife Service); and 36 C.F.R. §§ 800.1(b), 800.3(c) (regulations by the Advisory Council on Historic Preservation).

14. An Environmental Assessment is the document that explains the environmental consequences of a proposed structure. See 47 C.F.R. §§ 1.1311-1.1312.

15. See 6 FCC Rcd 7551 (1991).

16. 47 C.F.R. § 17.22.

17. This incorporation by reference was approved by the Director of the Federal Register, in accordance with 5 U.S.C. 552(a) and 1 C.F.R. Part 51, to be effective 30 days after publication of final rules in the Federal Register.

FCC ANTENNA STRUCTURE REGISTRATION FILING WINDOWS
FOR EXISTING STRUCTURES CONSTRUCTED PRIOR TO JULY 1, 1996

FILING WINDOWS	STATES / TERRITORIES
July 1-31, 1996	MI, MT
August 1-31, 1996	AZ, HI, NC
September 1-30, 1996	AK, NM, NY
October 1-31, 1996	MA, MO
November 1-30, 1996	IL, WY
December 1-31, 1996	NV, OK, PR
January 1, 1997 - February 28, 1997	CA, OH
March 1-31, 1997	IA, VA
April 1-30, 1997	AS, GA, GM, GU, MP, VI
May 1-31, 1997	LA, ME, RI
June 1-30, 1997	CO, MN
July 1-31, 1997	NE, PA
August 1, 1997 - September 30, 1997	FL, IN
October 1-31, 1997	DE, KS, WA
November 1-30, 1997	NH, OR, WI, WV
December 1-31, 1997	AL, DC, MD
January 1-31, 1998	AR, ND, UT
February 1-28, 1998	ID, MS, SD, VT
March 1-31, 1998	KY, TN
April 1-30, 1998	CT, NJ, SC
May 1, 1998 - June 30, 1998	TX

HIGH QUALITY RF FOR TELEVISION

Wednesday, April 17, 1996
9:00 am - 12:00 pm

Session Chairperson:
Louis Libin, NBC, Inc., New York, NY

***THE KLYSTRODE IOT**
A TECHNICAL EVOLUTION
Andy Haase
Eimac Division of Communication & Power Industries
San Carlos, CA

A NEW 5" HELICALLY CORRUGATED COAXIAL CABLE WITH A 6 1/8" COAXIAL CABLE PERFORMANCE
Ron Tellas
Andrew Corporation
Orland Park, IL

60 KW DIACRODE UHF TV TRANSMITTER DESIGN, PERFORMANCE AND FIELD REPORT
Timothy P. Hulick, PhD
Acrodyne Industries, Inc.
Blue Bell, PA

HIGH POWER MULTIMODE FILTERS FOR ATV, DAB, AND COMMON AMPLIFICATION TV SYSTEMS
Derek J. Small
Passive Power Products
Gray, ME

STANDARD REFERENCE CURVES FOR ATV FILTER MASK
Paul D. Smith
Micro Communications, Inc.
Manchester, NH

AN INDUCTIVE OUTPUT TUBE AND CIRCUIT OPTIMIZED FOR DIGITAL ATV
Roy Heppinstall
EEV Limited
Chelmsford, England

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*Paper not available at the time of publication.

A NEW 5" HELICALLY CORRUGATED COAXIAL CABLE WITH A 6 1/8" COAXIAL CABLE PERFORMANCE

Ron Tellas
Andrew Corporation
Orland Park, IL

ABSTRACT

With rising limitations on tower space coupled with the need to multiplex two or more channels on a single coaxial transmission line, higher power capability—without increasing size—is required. A new type of transmission line has been developed which has virtually the same power handling as helically corrugated 6⅛" coaxial cable, but the dimensions and weight of 5" coaxial cable. This performance is achieved through the use of a high temperature dielectric of original design (patent pending) to minimize attenuation. Characteristics and test results of this cable and its connectors will be presented. System considerations and other applications, such as ATV, that can benefit from the advantages this new cable type has to offer will also be discussed.

CHARACTERISTICS

Cable Construction

Construction of the new cable is shown in Figure 1. It is similar to other continuously corrugated cables in that the inner and outer conductors are made of seam-welded and corrugated copper tubes. The inner and outer conductors are kept concentric and mechanically stable through the use of a spirally wound dielectric. The distance between inner and outer conductors is selected to give the cable an impedance of 50 ohms. If there was a method to keep the inner and the outer conductors concentric without a dielectric, the power that the cable could handle would be limited only by the temperature of the outer conductor. With a spirally wound polymer dielectric, however, the limit is that temperature of the inner conductor which is less than the degradation threshold temperature of the polymer.

Typically, coaxial cables uas a dielectric made of polyethylene (PE) or polypropylene (PP) which limits the temperature of the inner conductor to 212°F and 250°F, respectively. The only way to safely increase the average power handling without degrading the life of the cable is to use a polymer with a higher melting temperature than the typically used polyolefins. This,

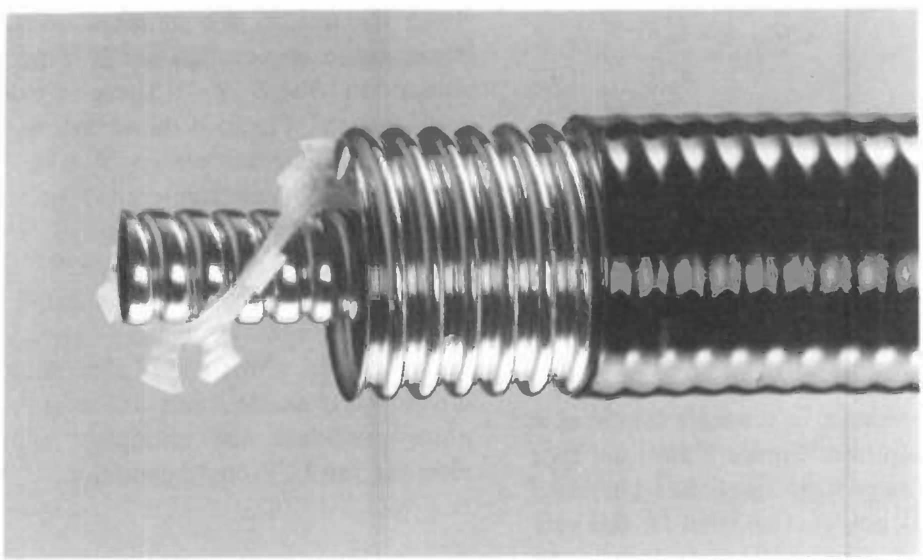

Figure 1: Cable Construction

in turn, will allow for a higher inner conductor temperature and provide a higher average power rating. Changing the dielectric material to ethylene propylene flouropolymer (FEP) allows the inner conductor to run safely at 302°F and thus allows the average power of the 5" diameter construction to exceed that of a similar 6 ⅛" diameter construction.

As mentioned previously, the dielectric must supply good mechanical concentricity of the inner to the outer conductors while not hindering the electric signal passing through. Since FEP has a higher dielectric constant and dissipation factor than PE or PP, special measures must be taken to ensure that the added losses and capacitive effects are compensated for without jeopardizing mechanical performance. The best solution to the problem results in the notched dielectric pattern shown in Figure 2. This approach warranted a design unique to the point that a patent was applied for. At the time this paper was written, the patent is pending.

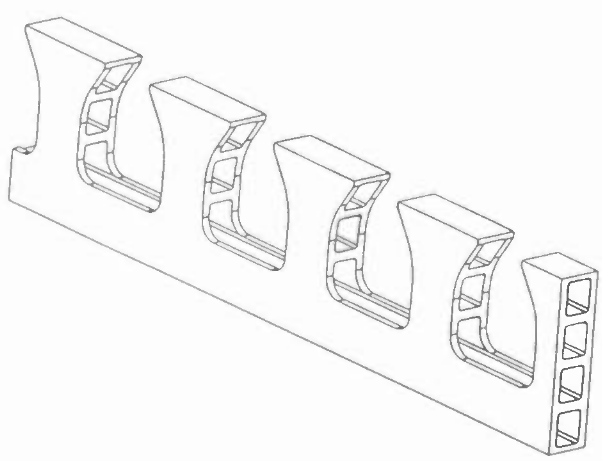

Figure 2: Notched Dielectric

Electrical Characteristics
Compensating the impedance of the cable for the new notched dielectric pattern of Figure 2 does not take place without affecting other important electrical parameters. The peak power is affected for this new cable when compared to similar 5" constructions. The reason for this is that the relative velocity of the cable was made greater than the existing 5" cables to eliminate any added attenuation. The distance between the inner and outer then had to be lessened, which caused the peak power to drop from 1890 kW to 1690 kW. The benefits of increasing the average power at the slight expense of decreasing the peak power is warranted for the television applications this cable was intended for.

Attenuation for a few representative frequencies in the broadcast band is shown in Table 1 for the new high power cable and for the 6 ⅛" cable. As can be expected, the 6 ⅛" cable has the lower attenuation. It is a fact of physics that bigger cables have lesser attenuation. However, this should not be the only criterion in selecting a transmission line for an application, as shown later in discussions of system considerations.

Table 1: Attenuation

Freq, MHz	New Cable, dB/ 100 ft	6⅛"cable, dB/ 100 ft
400	.171	.122
500	.197	.137
800	.269	.186
860	.283	.186

The connectors for this new cable are designed to achieve better than -38dB return loss per connector across the entire broadcast frequency band. When these connectors are attached to the cable, the transmission line system has an expected VSWR as shown in Table 2. These specifications are conservative. Figure 3 shows the measured VSWR for a 1050 ft length of cable with 6 ⅛" EIA interface connectors with test adapters to type N interface. As shown, the worst case measured VSWR for this transmission line system is 1.06 (30.7 dB return loss) over the entire domestic and international broadcast frequencies and is better than the published specifications. This figure also shows the cable's broadband characteristics, which makes the cable a good candidate for multiplex applications and planning for ATV implementation.

Table 2: Published VSWR

Freq, MHz	New Cable, Max	6 ⅛" cable, Max
<108	1.06	1.06
<230	1.08	1.08
<800	1.10	1.06
<860	1.10	1.10

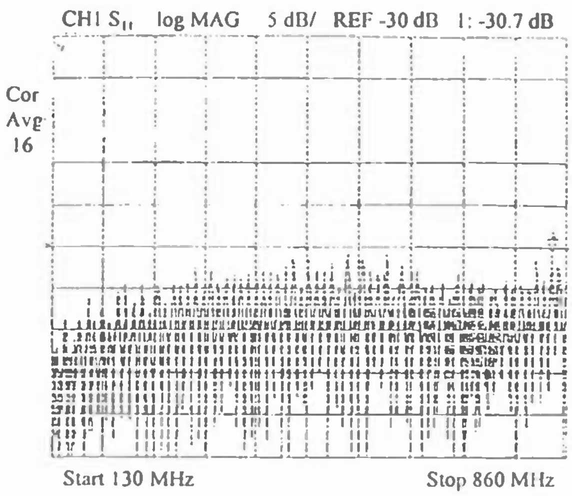

Figure 3: VSWR of 1050' Assembly

Mechanical Characteristics

As expected, the mechanical characteristics are similar to 5" cable construction. This gives the new cable a considerable advantage to the 6 ⅛" cable construction. The more significant advantages include less than half the weight and smaller bending radius. These advantages lead to savings in installation time since the cable is easier to route into a system and install. All of the mechanical characteristics are given in Table 3.

Table 3: Mechanical Characteristics

	New Cable	6⅛" Cable
Weight, lb/ft	3.4	7.3
Min Bend Rad., in (one bend)	36	47.2
Min Bend Rad., in (repeated)	50	78.7
Number of Bends, Min	15	—
Max Tensile Force, lbf	1000	1124
Bending Moment, lb-ft	200	—
Crush Strength, lb/in	240	—
Connector Weight, lb	29.6	55.1

SYSTEM CONSIDERATIONS

When choosing a transmission line, several factors must be considered. Of those factors, the most significant are efficiency, wind loading, shipping and installation.

Efficiency

Efficiency is a convenient way of expressing the power loss of a transmission line and is useful in selecting the appropriate line for an application. It is defined as the ratio of power delivered to the antenna to input power into the transmission line. It is easily calculated from the attenuation and length of the transmission line by applying the following equation:

$$eff = \frac{100\%}{10^{\frac{IL}{10}}}$$

$$\text{where: } IL = attenuation + 10\log(1 - |\rho|^2)$$
$$\rho = reflection\ coefficient$$

When selecting a transmission line, a good rule of thumb is that the efficiency should be better than 70%. Using the attenuation figures given in Table 1 and the above equation, the efficiency of the newly developed high power 5" cable and the 6 ⅛" cable was found for lengths of 300, 500 and 1000 foot transmission line systems. The results are shown in Table 4. Since the cables all had relatively low reflection coefficients, the contribution of reflection loss to insertion loss was disregarded in the analysis.

Table 4: Efficiency

Freq, MHz	New Cable, %			6 ⅛" cable, %		
	300'	500'	1000'	300'	500'	1000'
400	88.9	82.1	67.4	91.9	86.9	75.5
500	87.3	79.7	63.5	91.0	85.4	72.9
800	83.0	73.3	53.8	88.3	81.3	66.1
860	82.3	72.2	52.2	87.9	80.7	65.2

The data shows that the new cable is less efficient than the 6 ⅛" cable. However, one must understand the minimal effects this can have on the entire system. For a system with short to midrange lengths of transmission line, the effect is minimal. For a 300 foot system, the efficiencies differ by only a few percent. For longer lengths, the difference is greater.

When engineering a new system, or one in which both the antenna and transmission line are being replaced, this difference in efficiency can easily be removed through proper system planning. Take, for example, the 500 foot length of transmission line operating at 800 MHz. Table 4 shows that the efficiency is 73.3% and 81.3% for the new cable and the 6 ⅛" cable, respectively. Converting these ratios to decibels results in 1.35 dB and .90 dB, giving a difference in dB of .45 dB. If an omni-directional antenna has a numeric gain of 25, or 13.98 dBd, this .45dB difference in transmission line efficiency can be made up in the vertical gain of the antenna. The antenna, in this case, must have a vertical gain of 14.43 dBd. This corresponds to a numeric gain of 27.7, or 10.9% increase in vertical gain. This gain can easily be acquired by increasing the length of the antenna by 10.9%, or through antenna design techniques.

In the case where a directional antenna is used, the difference of 0.45 dB can also be eradicated through the use of the more sophisticated analysis tools now available to broadcasters. Using these tools of coverage contour and precise pattern shaping programs, the net effect could result in the same or better effective radiated power.

In an existing system, in which only the transmission line needs to be replaced, the only way to combat the decrease in efficiency is to increase the transmitter power. This method may be acceptable if other factors offset the higher electric bills that would occur.

Wind Loading

The effect that a cable will have on determining the total wind load is a rather involved procedure that is unique to each system. To determine the difference of the transmission line contribution to wind load, one must compare the difference in projected square area of one cable to the other. This difference becomes the ratio of the cable diameters. The total reduction in diameter the 5" cable will have as compared to 6 ⅛" cable is therefore:

$$100\left(1-\frac{5.2}{6.73}\right) = 22.7\ \%$$

This factor becomes more of an issue as tower space becomes more scarce and costly.

Shipping

The new cable is shipped on the same 12' 2" shipping reels used for other 5" cables. These reels hold up to 1150' of cable without connectors, or up to 1100' with attached connectors, enough for almost all installations. The size of the 6 ⅛" cable is such that it must be shipped on 13' 1" reels. Reels this large must be transported to the site on a special "low boy" trailer. These trailers are becoming increasingly difficult to locate, and the high load may require permits from the local department of transportation. Even so, the 6 ⅛" cable reels hold a maximum of only 722' without connectors, increasing the likelihood that two reels will need to be transported and the cable spliced.

Installation Considerations

As can be expected, the installation of the 5" cable is easier since it is lighter in weight, has a tighter bend radius and has connectors which are much easier to fit onto the cable. Experience with the 6 ⅛" cable shows that it is much harder to install due to its size and a tar-like flooding compound between the outer conductor and the jacket, to combat water migration. This flooding compound makes installation unsuitable in cold weather. In fact, the manufacturer of the 6 ⅛" cable recommends that installation not take place below freezing temperatures.

OTHER APPLICATIONS

As shown in Figure 3, the VSWR performance of the cable is relatively flat across the whole broadcast spectrum. This wide band feature is important for planning for the future. Since no flange spikes are present, a characteristic common to conventional segmented rigid coax, the cable can be used at any frequency. This allows for ease in multiplexing more than one signal at different frequencies onto the same transmission line. In planning for the advent of ATV, this wide band feature also allows for easy migration to new ATV channel allocations either by multiplexing or complete switch over of the system to ATV.

FIELD EXPERIENCE

Performance of the new transmission line has been proven under actual field conditions. The Dubai Broadcasting Authority required two runs of 170 m each for a UHF broadcast system, and awarded the contract to Alan Dick & Company Limited (ADC), Cheltenham, England. Before specifying the new cable, ADC used a computer model to run simulated tests to ensure that the cable characteristics were compatible with their broadband UHF panel antenna and met the required coverage. These being satisfactory, they preceded to run full power tests on a sample, which showed that the cable could carry the required power with no leakage. The Dubai Broadcasting Authority reviewed these test results, found them acceptable and purchased the cable.

The cable was then installed in Dubai in January 1996. The installation went smoothly and efficiently, following the instructions supplied. No performance problems occurred when the station was switched on, and all predictions were realized. The citizens of Dubai are now watching UHF TV programs transmitted through the new cable, and —we are told— the Dubai Broadcasting Authority is "very happy."

CONCLUSION

The new cable has been presented along with its mechanical characteristics, performance specifications, considerations for integration into a system and an example of actual use in the field. This cable is able give the benefits of a 6 ⅛" cable, but with the handling of a 5" cable. To better illustrate this point, the Table 5 contains the comparison of the new cable to a 6 ⅛" cable. As the table illustrates, the new cable has an average power rating that exceeds that of the 6 ⅛" cable at television frequencies and has handling characteristics typical to 5" cable. Significant differences can be seen in the bending radius and cable and connector weight. These attributes prove the new cable provides a 6 ⅛" coax in a 5" package.

Table 5: Comparison

Attribute	New Cable	6 ⅛" cable
Avg Power, 40°C:		
200 MHz	116 kW	95 kW
500 MHz	66 kW	58 kW
800 MHz	49 kW	45 kW
Min Bend Radius:		
Single Bend	36 in	47 in
Repeated Bends	50 in	79 in
Shipping Reel Size	12' 2"	13' 1"
Max Length	1150 ft	722 ft
Diameter over jacket	5.2 in	6.7 in
Cable Weight	3.4 lb/ft	7.3 lb/ft
Connector Weight	29.6 lb	55.1 lb

ACKNOWLEDGMENTS

I would like to thank Mr. Robert Leonard whose input to this paper was invaluable, and without his help, this paper may not have been written. I would also like to thank the management of Andrew Corporation for supporting the publication of this paper.

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60 kW DIACRODE UHF TV TRANSMITTER DESIGN, PERFORMANCE AND FIELD REPORT

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ABSTRACT

The pulsed modulation anode klystron set the power standard for the peak of sync visual power level in the United States in the early 1980s when 60 KW was reached with a single tube. A second tube provided 6 KW aural carrier power into a visual/aural diplexer. Still very low in power efficiency and increasingly expensive to operate, there have been challengers of the klystron to improve this situation boasting 60 KW peak of sync visual and 6 KW aural power in the single amplifier tube operating in Class AB or B. All have failed. The klystrode, which was the first attempt, no longer is supported by its manufacturer, and the IOT, the second attempt stops at 43 KW peak of sync power in combined 10% aural amplification---and continues to have serious life problems after five years in the field.

A relative late comer to the 60 KW scene, but one that has been well thought out and tested, is a new configuration tetrode which shows outstanding visual and aural performance with no excessive hotspots in the 14 pound tube or its ceramics or the cavity assembly in which it operates. Given the name *Diacrode*® because of its double ended tetrode construction, the power output is essentially double that of its predecessor in the family of high power UHF tetrodes developed by Thomson Tubes Electroniques.

This paper presents the design highlights of the highest peak of sync power (60KW with 10% aural) combined amplification single tube UHF television transmitter in the world. It also presents performance data, power consumption and a number of photographs showing a real transmitter. A special up to the minute report is presented on field performance of systems installed.

SOME HISTORY

The klystron UHF television broadcast transmitter set the precedent for 60KW of visual power back in the early 1980s. Generally this was accomplished with one tube for the visual and another of the same kind for the aural. Although the klystron has always boasted long tube life, it is terribly inefficient (even with pulsing) and consequently very expensive to operate. With so many klystron transmitters in the field, the setting was fertile to *find a better way*. The klystrode was the first klystron replacement of any consequence because it moved the class of operation from A to AB --- a definite step in the right direction, but fraught with life problems, it was soon replaced with another supplier's equivalent, the inductive output tube, or IOT. The IOT has been successfully marketed and well supported by the OEMs using it, but it has had its share of performance and life problem. After five years, it is showing a slow average life improvement as its manufacturer identifies solutions to its problems. Another player is the MSDC which shows significant improvement in power efficiency over the klystron, but has been given little support by the industry in general. For one thing, with its multiple collector layers, the power supply becomes very complicated and equally layered.

It is no secret that most uhf television transmitter manufacturers have opted for the IOT and dozens are on the air, but even if the tube provided the reliability expected of it, it still has the drawback of not being able to provide the 60KW power level set by the klystron when used in combined amplification. When the klystron reached 60KW, it did it by amplifying only the visual signal. Since then, the trend has been away from separate visual and aural transmitters, so the real challenge has been to do the same thing with one tube in common amplification. All tubes to date fall short of this goal, that is, all but one.

INTRODUCING THE *DIACRODE*

Physics dictates just how far the state of an art may go. In the case of the uhf tetrode, this limit appears to be around 55 KW peak envelope power without undue compression and distortion. If two major signals are being amplified by a tube like this and their relative power levels are that of the NTSC format, i.e. aural 10% of peak of sync, then the highest peak of sync signal possible with a tetrode is 31.5 KW in common amplification. To go any higher, more than one tube is needed or a different approach taken. A different approach is the path taken with the Au60D. With a tetrode *like* tube, the physical limit leading to power handling capability of the tetrode can actually be doubled without changing any of the tube voltages except the filament. Figure 1 shows the new approach in the form of a comparison between the tetrode

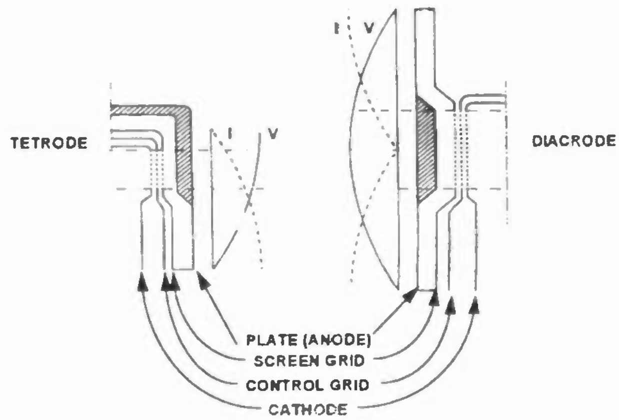

Fig 1. Cutaways of the tetrode and diacrode are compared. The diacrode is not actually twice as high as the
tetrode, but electrically it looks that way because of the role of the upper reflective cavity on the diacrode. Notice that RF current peaks above and below the diacrode center while on the tetrode there is only one peak at the bottom.

and its advanced concept double power tetrode or Diacrode as it has come to be known. Actually the diacrode is not complete without an external cavity connected across its output elements, i.e. between the anode and screen grid. This is a small cavity, dc blocked which rests on top of the tube. It is a quarter-wave transmission line measured from the top of the cavity to the vertical center of the tube. It is short circuited at the top reflecting an open circuit at the vertical center of the tube. The upper cavity and diacrode are shown graphically in Figure 2. This places an RF open circuit or current minimum at the vertical center of the tube, a current maximum at the base of the tube, like the tetrode, but a second current maximum above the tube at the cavity short circuit. This second current maximum is impossible in the plain tetrode since in the tetrode the RF current must go to zero at the top end away from the base end where no connections are made. If no connections are made, this is an open circuit condition and the current must go to zero at the top. With two current maximums, the RF power capability of the diacrode is double that of the equivalent tetrode while element voltages are the same.

TH 680 Diacrode

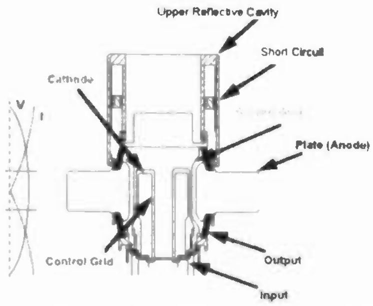

Figure 2. All Diacrode elements are shown with the effects of the upper reflective cavity. Its purpose is to reflect an open circuit into the vertical center of the tube since it is a quarter-wavelength short circuit at its far end. This causes a current minimum at the tube vertical center, precisely the opposite effect of that of the tetrode. Double current and, consequently, double power duty is achieved with the diacrode with a current peak at the top and bottom of the tube allowing double output power while keeping the voltages the same as the tetrode.

power than the 32,000V IOT in common

TABLE 1

ATTRIBUTES OF THE TH-680 DIACRODE IN 60 KW PEAK OF SYNC OPERATION WITH 10% AURAL POWER

➡ -1 dB BANDWIDTH	12.8 MHz
➡ UNCORRECTED IN-BAND IMD	-52 dB
➡ GAIN	15 dB
➡ ANODE VOLTAGE	8500 VOLTS
➡ SCREEN GRID VOLTAGE	600 VOLTS
➡ CONTROL GRID VOLTAGE	-120 VOLTS
➡ PLATE VOLTAGE AT IDLE, GRAY, BLACK	3.0, 9.1, 11.2 AMPS
➡ SCREEN GRID CURRENT AT GRAY	125 mA
➡ OPERATING FREQUENCY	470-860 MHz
➡ FILAMENT	8.8 VOLTS AT 180 AMPS
➡ WEIGHT	6.5 KG (14.3 lbs)
➡ SIZE	DIA. 8.2" HT 6.5"
➡ COOLING	HYPERVAPOTRON (WATER)

This is not to say that twice the power comes from doubling the efficiency of the tetrode, but it is to say that the efficiency is the same as governed by the class of operation (class AB) and the dc current is twice that of the tetrode. All other properties and aspects of the diacrode are the same as the tetrode. Gain is typically 14.5 dB, heater or filament voltage is twice that of the tetrode (10 volts at 180 amperes), but anode voltage remains at 8500 volts while screen grid voltage is essentially the same at 600 volts and control grid bias voltage is the same at about -120 volts and adjusted for 3 amperes of anode idle current. So here we have a tube that only requires 8500 volts and produces 50% more amplification and because of this low voltage, doesn't require a crowbar circuit or a substitute for it. It is the only technology that stands up to the precedent set by the klystron and does it in true class AB

efficiency. At this time, there isn't any other choice without using more than one tube. Just as an aside, keep in mind that this tube only weighs 14 pounds, and replacing it doesn't involve detuning any cavity sections since the tube sits external and above all that. Its expected life is greater than 20,000 hours and has a price tag of $25,500. Do a little arithmetic of your own or take a look at the figures presented in Table 2 to find out the savings over any other choice over the life of the transmitter operating at 60KW. You'll be pleasantly surprised!

TABLE 2

SOME TUBE COST COMPARISONS

CONSIDER 24 HOURS PER DAY FOR 20 YEARS: (175,200 HOURS)

TRANSMITTER	TOTAL COST	COST VS DIACRODE
DIACRODE	$219,000	
IOT	$665,760	$446,760 ADDITIONAL
MSDC	$464,280	$245,280 ADDITIONAL

CONCLUSION: TUBE SAVINGS ALONE WILL PAY FOR THE DIACRODE TRANSMITTER AS COMPARED TO AN IOT TRANSMITTER. ALTHOUGH THE MSDC ANALYSIS IS MUCH BETTER THAN THE IOT, IT IS STILL TWICE THE COST OF A DIACRODE SYSTEM.

CONSIDERING THAT THE BACKUP SYSTEM SUGGESTED FOR THE DIACRODE IS THE 2500 WATT DRIVER, AND THAT TWO DIACRODES WITH A MAGIC TEE MAY BE MORE ATTRACTIVE, REPLACING A PAIR OF DIACRODES IS STILL LESS EXPENSIVE THAN REPLACING A PAIR OF IOTS.

THE TRANSMITTER

A full 60KW uhf television transmitter has been developed around the diacrode providing actually 63KW peak of sync and 6.3KW of aural power to allow for output filter losses. Coupled with new solid state driver amplifiers capable of twice the power for a given amount of heat sink real estate, the 60KW transmitter only occupies four more inches of floor length than the 30KW tetrode transmitter. All other dimensions are the same. Of course the power supply is larger, but comparable to any other 60KW supply whether dry or oil filled. The heat exchanger is twice the size of the 30KW transmitter because twice the heat is generated. It may use a single distilled water loop for indoor operation or a double loop for an outdoor installation. Drive required before the interstage bandpass filter is about 2500 watts peak of sync which is a respectable power level using the PA bypass option should the PA need maintenance unlike the 400 watts or so available if the IOT is used. A block diagram of the Au60D diacrode amplifier is shown in Figure 3 while the complete transmitter is shown in Figure 4.

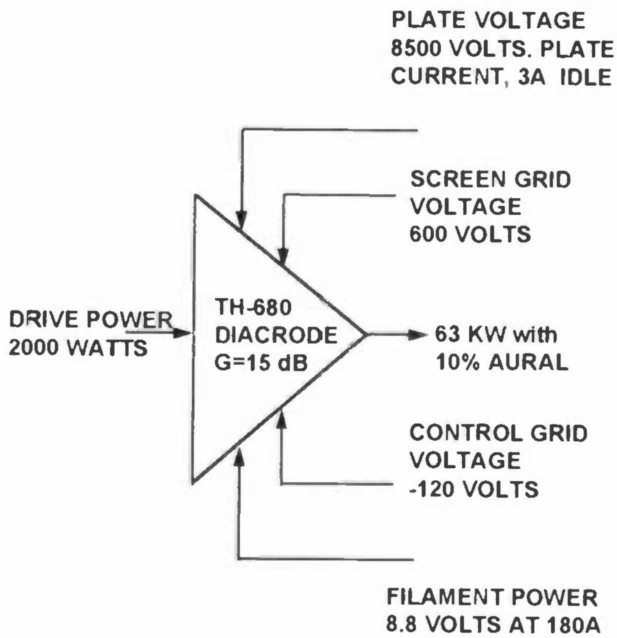

Figure 3. Diacrode support voltages and currents are shown

You don't need another tube to give you minimum backup power, but should you want it, the analysis given in Table 2 should convince even the casual reader that the diacrode is far more economical than any other choice.

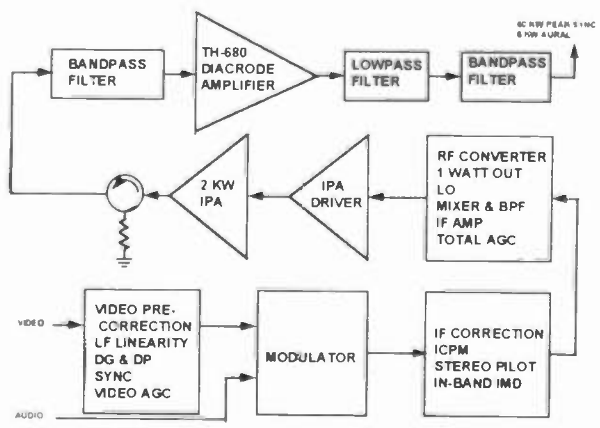

Figure 4. The Au60D Block Diagram

The TH-680 diacrode is easy to work with and a snap to replace. Requiring only about 20 man-minutes to change, one person can do it without disturbing the cavity tuning since all tuning sections are below the tube and its socket. Small Teflon water cooling hoses are connected to the two small copper fittings exiting the tube between the control grid and screen grid rings. These hoses come up from under the tube into the socket area after passing through the cavity section. The screen grid requires about a half gallon per minute of water flow. The anode is cooled by 15 GPM of water flow through much larger hoses as shown in Figure 5.

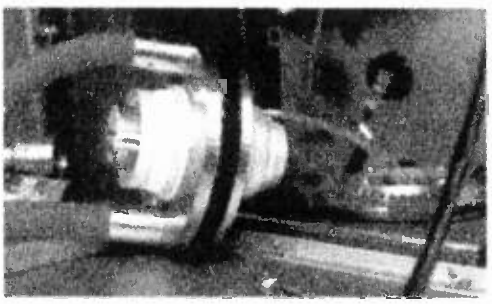

Figure 5. The fourteen pound TH-680 diacrode is shown on its side outside the socket with anode hoses connected. It is easily installed by one person. With no magnets or cavities surrounding it, tuning is not disturbed during tube removal or replacement. With more than 20,000 hours of life expectancy at a full 63 KW peak of sync power and 10% aural, the Au60D is the most economical approach ever taken by any transmitter manufacturer.

Au60D TRANSMITTER PERFORMANCE

Of primary concern is the performance of the transmitter at full power. The photographs given in figures 6 through 16 were taken of waveform monitor and spectrum analyzer presentations on the first Au60D transmitter built for KASY channel 50 in Albuquerque New Mexico. Tests were conducted prior to shipment at the Acrodyne facility in September 1995.

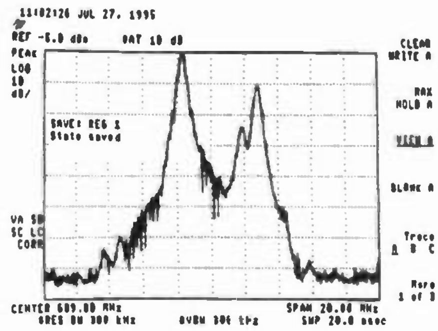

FIGURE 6. A 10 dB ratio is shown between peak of sync and aural carrier levels. Left and right out-of-band IM levels are immeasurably small.

Au60D Ch50
Ser# 3106

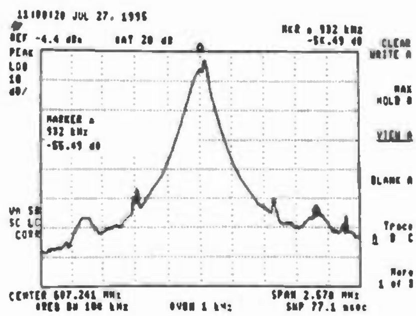

FIGURE 7. A good measure of overall transmitter performance is in-band IMD levels. The -920 KHz IM is 56 dB down while the +920 KHz component is down 55 dB below peak of sync.

Au60D Ch 50
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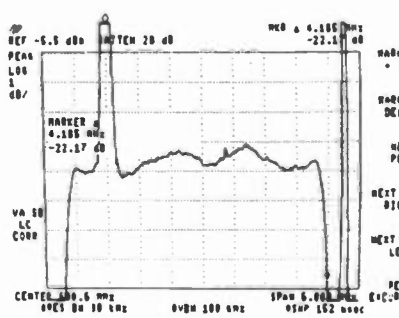

FIGURE 8. Bandpass flatness of the transmitter and all RF filters (BPF, LPF and driver BPF) is shown to be within 1 dB peak to peak.

Au60D Ch 50
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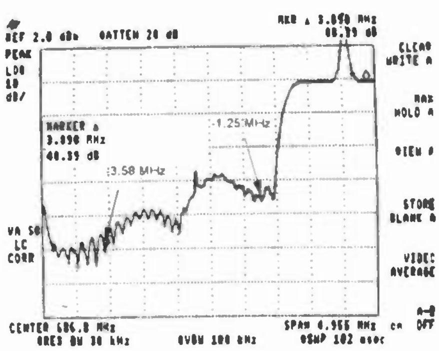

FIGURE 9. The -3.58 MHz level is down nearly 50 dB in the Au60D Transmitter. The -1.25 MHz component is down 35 dB.

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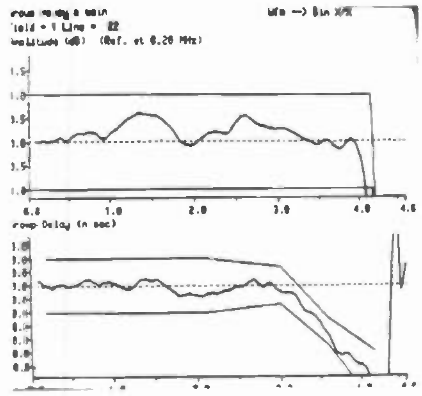

FIGURE 11. Group Delay and Gain are well within specification for the 60 KW IF visual and aural combined transmitter.

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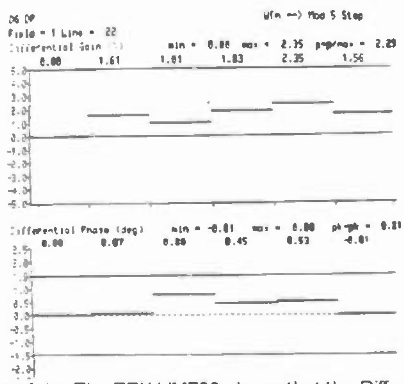

FIGURE 10. The TEK VM700 shows that the Differential Gain (top) is 2.35 % while the Differential Phase is 0.8 degrees (bottom).

Au60D Ch
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Ser# 3106

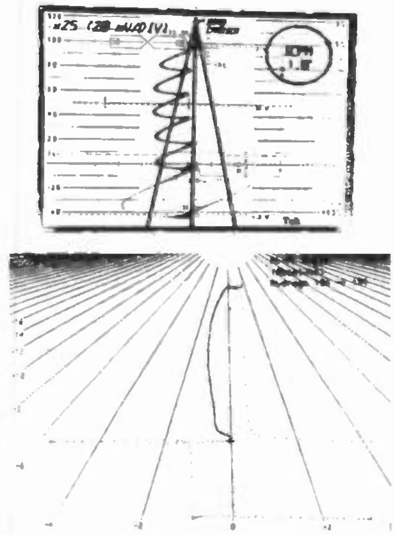

FIGURE 12. The TEK 1480 and VM700 both show ICPM to be 1.8 degrees.

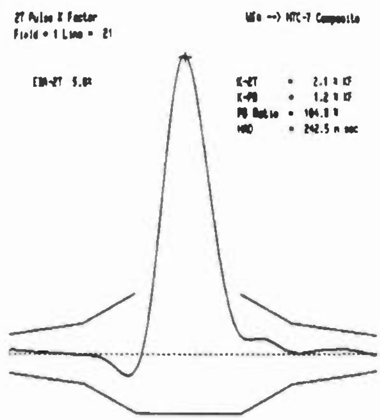

FIGURE 13. 2T performance is shown for the Diacrode 60 KW Transmitter.

Au60D Ch
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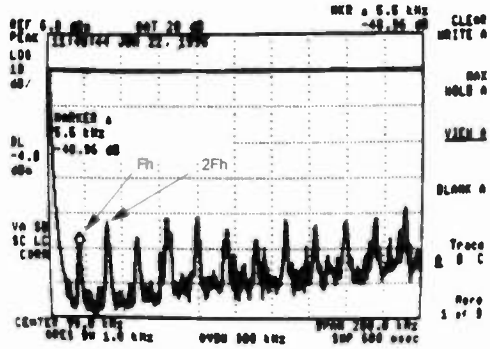

FIGURE 15. Stereo pilot protection is guaranteed by the residual FM audio baseband component at 15.7 KHz down by 48 dB and that at 2Fh down by 42 dB relative to 25 KHz deviation. All residual modulation out to 200 KHz is less than 7%.

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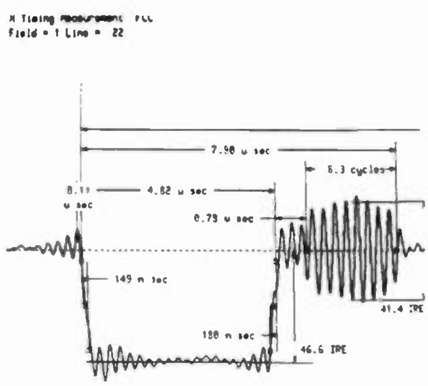

FIGURE 14. The sync pulse and color burst time domain waveforms are shown to be well within limits. The sync level is 40 IRE, nearly identical to the peak to peak burst. The monitor is a TEK VM700.

Au60O Ch
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Ser# 3106

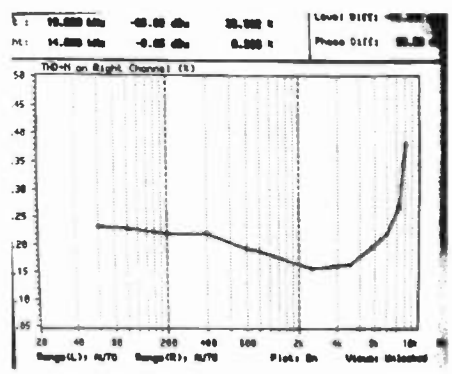

FIGURE 16. Total Harmonic Distortion is shown to be a maximum of about 0.37% at 15 KHz. The TEK VM700 gives the results.

Au60D Ch
50
Ser# 3106

THE HEAT RUN

While on the subject of performance, survival is also a performance measurement. The first transmitter was subjected to a 24 hour heat run from 7:00 AM of 21 September 1995 until 7:00 AM the next day operating at full blanking level picture (0% APL) with 10% aural power at 63 KW peak of sync. There was 60 KW of peak of sync power into the dummy load after the output lowpass and constant impedance bandpass filter. Ambient room temperature was 38°C or 100°F. There was no failure nor did any tuning or correction adjustments change except for a thermal fault in one of the solid state driver amplifiers. The room was cooled a few degrees and the test continued.

SOME COMPARISONS

The klystron may have set the standard in the early 1980s, but no equipment supplier builds klystron transmitters anymore. Comparisons must be done between present technologies. Table 3 lists some very vital information about IOT and diacrode transmitters which must be taken seriously in the decision process when purchasing a new transmitter.

Since both the IOT and the diacrode operate in class AB, it is to be expected that both types will have about the same power efficiency and power consumption. The gain of the diacrode is significantly less than the IOT requiring a higher power driver, but the advantage is that when the PA is out of service for any reason, the driver provides significantly more power as a back-up.

TABLE 3

A COMPARISON
60 KW COMBINED AMPLIFICATION

	DIACRODE	IOT
HIGH VOLTAGE	8.5 KV	32 KV
NUMBER OF TUBES	1-TH680 W/ 2.5KW BKUP	2-IOT7340 W/ REDUNDANCY & MAGIC TEE
CLASS OF OPERATION	AB	AB
IDLE POWER	25.5 KW	25.5 KW
PLANT CONSUMPTION 50% APL, 10% AURAL	1.33 W/W	1.33 W/W
GAIN	15 dB	21 dB
SIZE	D8.2"XH6.5"	D9.0"XH29"
WEIGHT	14.3 lbs	50 lbs
TIME TO CHANGE	20 MAN-MINUTES	3 MAN-HOURS
TUBE COST	$25,000	2X$34,175
TUBE LIFE	29,000 MAX RECORDED 20,000 EXPECTED BASED ON TH563	??????????
COOLING	15 GPM (NO GLYCOL)	18 GPM
KNOWN PROBLEMS	NONE	AFTER 5 YEARS THERE ARE STILL PROBLEMS WO CONFIRMED SOLUTIONS
OVERALL DYNAMIC PERFORMANCE	← SAME →	

The diacrode is much smaller and lighter than the IOT and, therefore, easier to replace. Also tuning is not disturbed when changing a diacrode. And perhaps the most important thing of all is that only one diacrode is needed to reach 60 KW.

FEATURES OF THE Au60D

The 60 KW diacrode transmitter boasts a single PA tube to provide full visual and 10% aural power on any UHF channel with less than 86 KW ac power consumption at 50% average picture level (APL). Further, since the anode or plate voltage is quite low, only 8500 volts, no crowbar circuit is needed because only about one-fifth of the energy is stored in the diacrode power supply compared to that of the IOT.

With seven dB of peak power overhead, the diacrode can provide 12 KW of average power 8-VSB according to tests run by Thomson Tubes in their plant in

Thonon France with excellent results. Technical information about these tests is not within the scope of this presentation, but is available from Acrodyne.

WHAT'S IN THE FIELD?

The transmitter at KASY on channel 50 was installed in December 1995 at the rather high RF dense location on Sandia Crest near Albuquerque New Mexico. The altitude at the Crest is nearly 12,000 feet so the very first diacrode transmitter in the world is located at one of the highest altitude locations. Nothing special was done to the transmitter for this altitude except that an oversized blower was installed external to the transmitter. The Au60D in its entirety (less power supply and heat exchanger) are seen in Figure 17. The power amplifier cabinet is the left most while the upper half of the center cabinet shows the PA meter and control panel and the right most cabinet is the entire 2500 watt driver with exciter. Figure 18 shows the lowpass filter rising vertically alongside the left of the transmitter while the constant impedance bandpass filter is horizontally suspended from the overhead. The 1-1/2 inch water pipes to the heat exchanger are seen in the upper forefront. The heat exchanger is in a small enclosed room by itself with intake air through one outside wall and exhaust air through a wall perpendicular to it. Although this installation uses a pure water to air heat exchanger, a pure water to water/glycol mix (indoor) may be used connected to a water/glycol to air heat exchanger (outdoor).

Figure 17. The world's first 60 KW single tube combined amplification diacrode UHF TV Transmitter installed at KASY, Albuquerque, NM (12,000 feet) operates on channel 50.

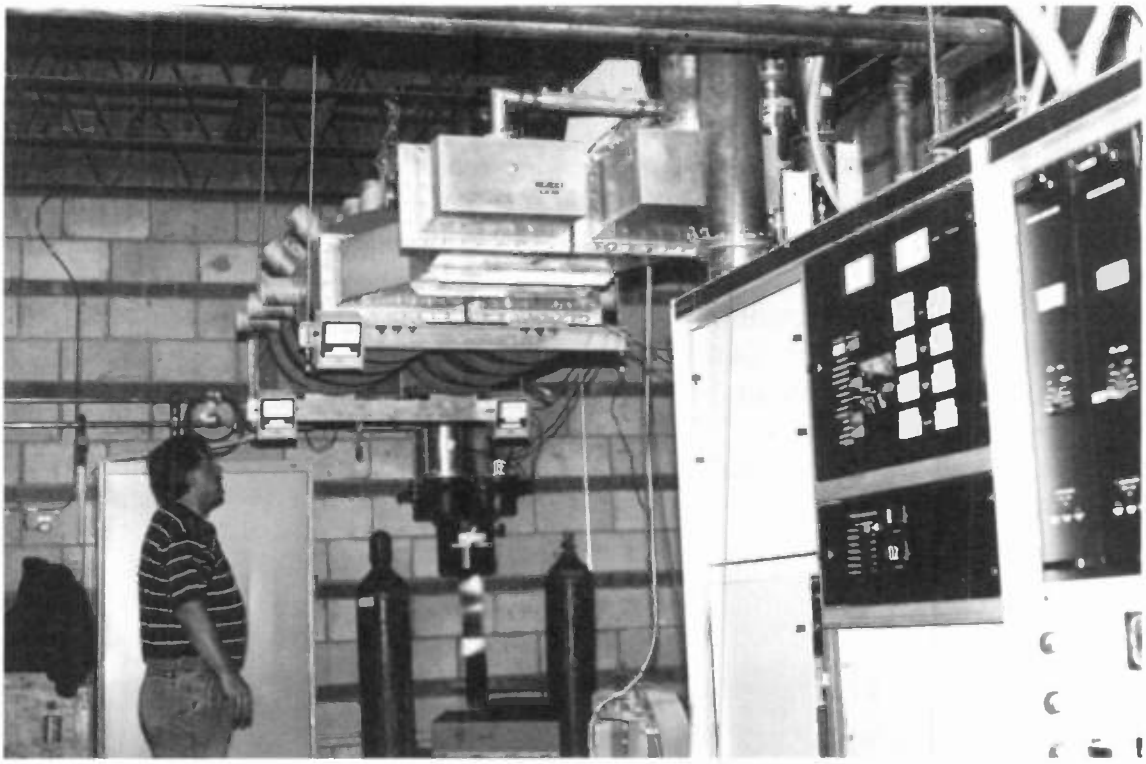

Figure 18. The Au60D 60 KW Transmitter at KASY is to the right of the vertically mounted lowpass filter and the horizontally mounted constant impedance bandpass filter suspended from the overhead. The diacrode cooling water pipes to the heat exchanger are seen near the top of the figure.

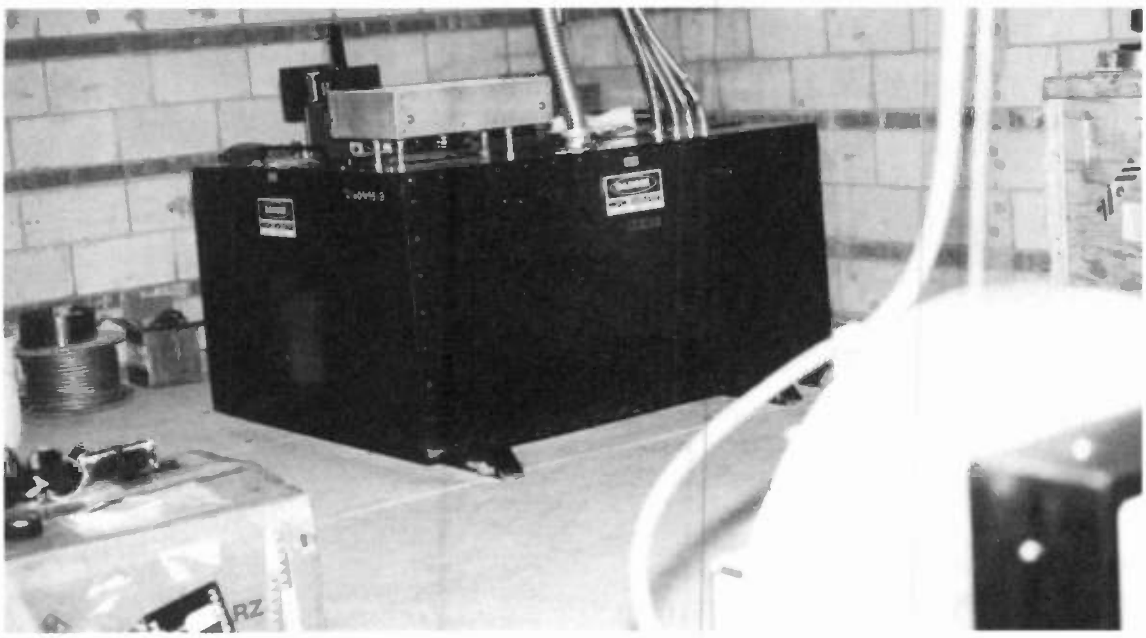

Figure 19. The oil filled power supply, although an outdoor type, is situated in a corner of the transmitter building.

The second Au60D transmitter was installed in December 1995 in Tulsa, Oklahoma at Oral Roberts University station KWMJ and operates on channel 53. The equipment is virtually the same as that at KASY except that Tulsa uses a dry indoor power supply of the same rating, i.e. 8500 volts at 15 amperes. The four solid state driver amplifiers are evident in Figure 20. One significant difference between installations at the two sites is that the layout of the transmitter at one site is the mirror image of the other.

FIGURE 20. KWMJ's Au60D operates on channel 53 in Tulsa, OK.

An uncovered solid state driver amplifier is shown in Figure 21 exposing the four 170 watt PEP capable class AB amplifier stages at KWMJ. These four are driven by the same transistor type, Acrodyne's ACDN301, in class A. These amplifiers are carefully gain and phase matched on the operating channel for maximum power efficiency and minimum power to the combiner reject loads. For 2500 watts of peak of sync drive power, it is not unusual to waste only two or three watts in the dummy loads.

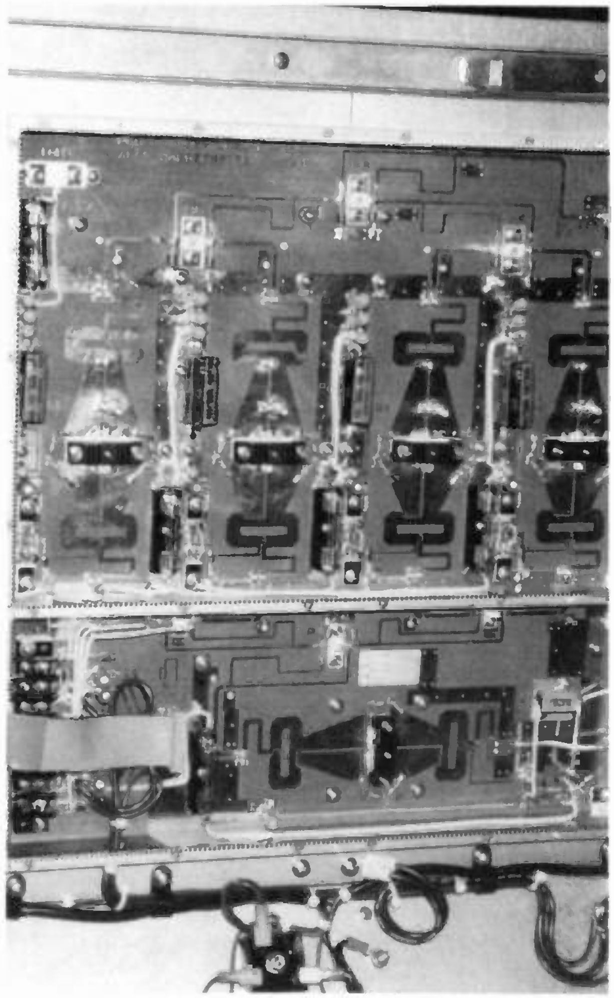

FIGURE 21. One side of a solid state driver amplifier at KWMJ is uncovered showing one Acrodyne ACDN301 transistor in class A driving four in AB.

The heat exchanger installation at KWMJ is shown in Figure 22. Unlike the KASY installation, it is located in the same room as the transmitter. Five panels with a total of twenty fans remove heat from the distilled water. Most heat exchangers are fitted with redundant pumps. Water resistivity is monitored to remain above 200Kohm-cm.

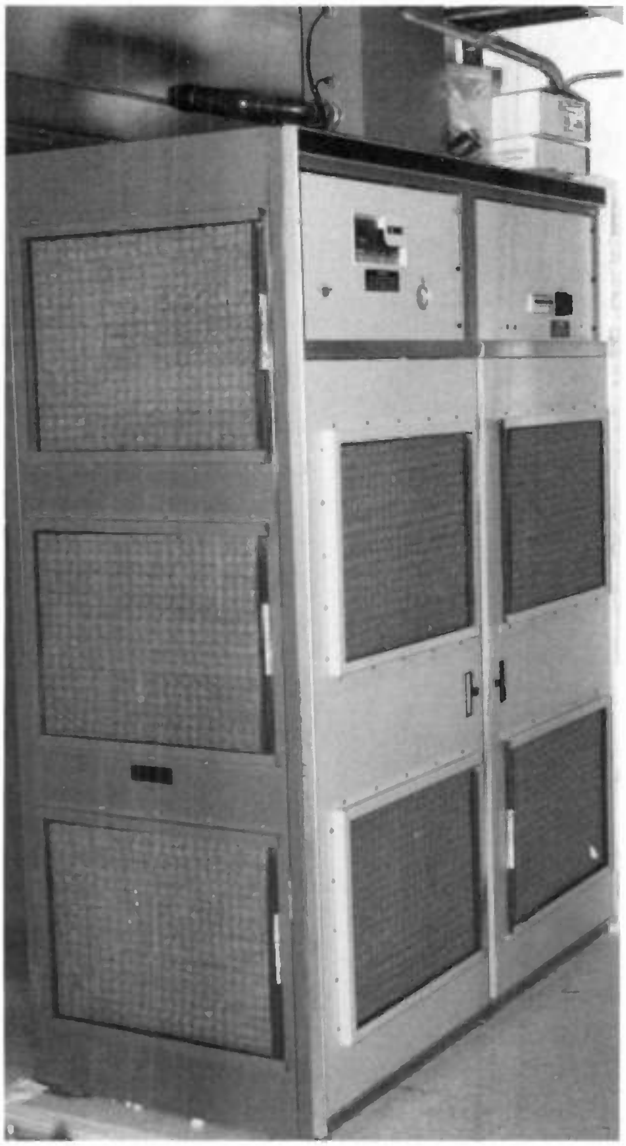

FIGURE 22. The heat exchanger at KWMJ shares the transmitter room

CONCLUSIONS

The diacrode is new, but has not shown any initial life problems. The installations at KASY and KWMJ have been operating for about six weeks as of this writing (January 21, 1996) and a third Au60D is about to go on the air at KQRF in Rockford, Illinois. There have been some startup problems with the first transmitter, but they were not tube related. Upper cavity tuning, at first, seemed to indicate a bad tube while operating it at black picture, but this turned out to be false. Retuning the cavity made everything work just fine.

New introductions are always at risk and no one likes to make claims prematurely, but the diacrode transmitter must be tauted as a well thought out, not before its time innovation that will only prove itself over time. It must be said that heat is the great destroyer of transmitters and that unexplainable hot spots just don't exist in this approach. Every Btu is explainable and accounted for and these numbers point to a very successful product which cannot be dismissed because it's new. The homework has been done and there are no unknowns in the equation of tube life.

The diacrode certainly seems to have a bright future in setting a new power and performance standard. Because it is basically a tetrode, it has all the linearity properties of a tetrode enjoyed by only the VHFers until now. Required correction is no different than the VHF tetrode which is far less demanding than the klystron and all of its derivatives.

If you haven't yet replaced your old klystron transmitter, the decision process just became a little more difficult with the introduction of the Au60D. On the other hand, maybe it just became a lot easier.

FOOTNOTE

® The term "diacrode" is a registered trademark of Thomson Tubes Electroniques.

HIGH POWER MULTIMODE FILTERS FOR ATV, DAB, AND COMMON AMPLIFICATION TV SYSTEMS

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Abstract

Modern TV transmitter systems require more exacting bandpass filter specifications in terms of rejection, insertion loss, and group delay. High Q multimode filters exhibiting finite transmission zeros and/ or delay equalization are discussed for ATV, DAB and common amplification TV systems. The required Q's and the necessary multimode filter structures are discussed for the successful implementation of low loss intermod filters and multi-station (NTSC/HDTV) combiners. Data for intermod, DAB (L - band and 217 - 230 Mhz band) and HDTV filters is presented.

Introduction

The recent rise in the application of common amplification broadcast systems and the future implementation of HDTV systems have redefined the characteristics of high power bandpass filters used in these systems. NTSC common amplification systems require low loss constant impedance intermod filters with significant rejection. HDTV systems using similar amplifiers will require improved rejection (over NTSC systems) near the band edges, further complicating filter structure. Simulcast of HDTV/ NTSC systems as presented in this paper also poses significant filter requirements to maintain signal quality.

Dual-mode filter technology, originally developed for satellite communications systems in a thrust for compact low loss filters requiring high rejections, lends itself nicely to satisfying modern broadcast bandpass filter problems. The use of dual-mode technology allows for the application of couplings between non-adjacent cavities (cross couplings) which produce transmission zeros (notches) in the filter response. Application of the technology allows for the design of preferred elliptic, pseudo-elliptic and delay equalized filters. Without these cross couplings, only monotonic cut-off rates are realized.

Unloaded Cavity Q (Qu)

The size of a UHF filter is determined by the size of each of its resonant cavities, which is basically dictated by three specifications: insertion loss, power handling and rejection. The rejection specification defines the number of resonant cavities required for a given passband vswr while the insertion loss and power handling specifications define the cavity size or volume. Filters with low insertion loss specifications require larger volume, higher Qu cavities than do filters with higher allowable insertion loss.

The two types of resonant cavities under consideration are the rectangular TE10m and the cylindrical TE11m (m = 1, 2, ...) structures shown in figure 1a. The theoretical Qu data [1] for the upper portion of the UHF band is shown in Figure 1b. The subscript 'm' is an indicator of cavity length in units of 1/2 wavelength. The theoretical Qu data plotted is for 1/ 2 wavelength (m=1) rectangular and cylindrical cavities, and full wavelength (m=2) rectangular cavities all of which are manufactured using aluminum. A nearly optimal D/L ratio was used for the cylindrical cavity Qu calculation. Practical Q efficiency of up to 75% can be obtained with proper construction techniques.

Typically, filters employing Chebyshev response parameters are manufactured from 1/2 wavelength rectangular cavities which exhibit relatively low Qu values. Higher Qu is achievable with m > 1, but their size may become a liability. Multi-mode cavity designs discussed in this paper are based on cylindrical TE111 mode cavities which provide a significant increase of unloaded Q for optimum D/L ratios and, therefore, a significant reduction in passband insertion loss.

Other loss reduction techniques are also available. The use of material plays an important role in achievable insertion loss. Manufacturing copper cylindrical cavities will reduce loss by approximately 22% over aluminum cavities. Full wavelength cavities will also reduce loss but they may not be practical due to the required cavity size at UHF. A higher order mode cavity such as TE011 is a potential solution, but again, may not be practical due to the cavity size and the existence of spurious modes if care is not taken in choosing proper D/L ratios.

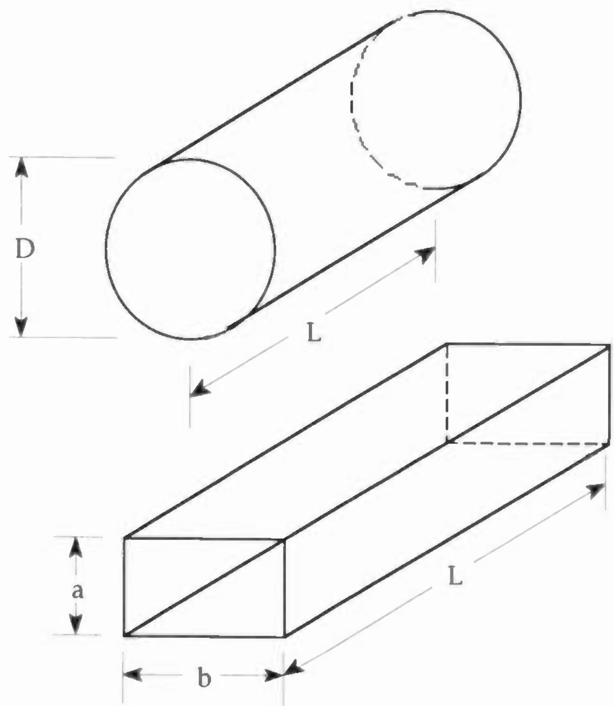

Figure 1a. Resonant cavity structures

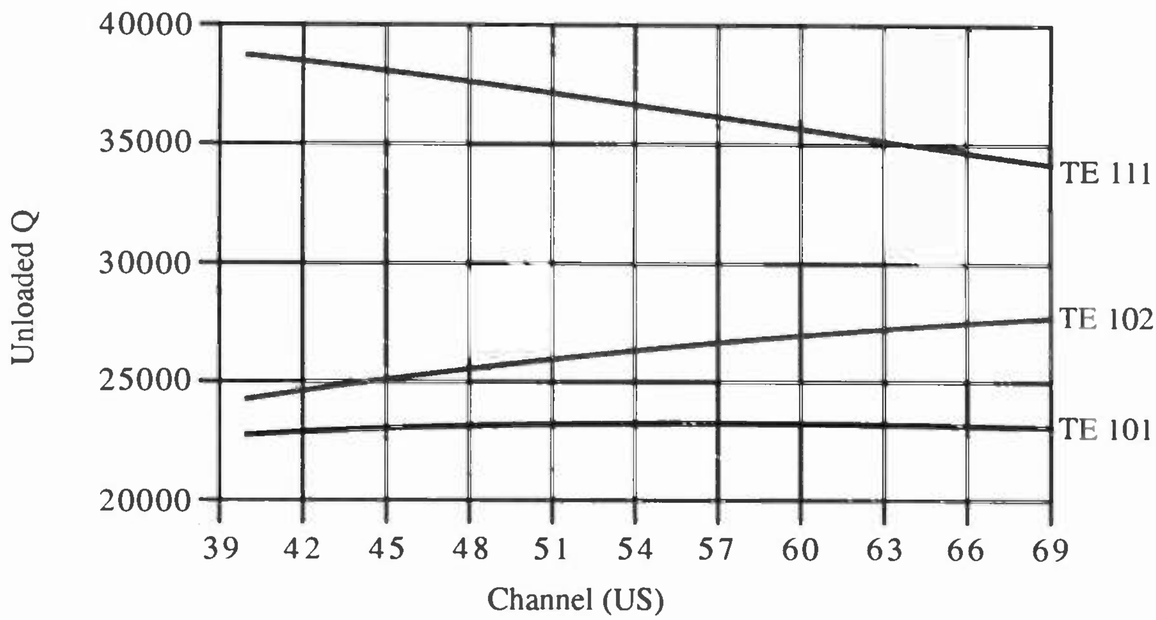

Figure 1b. Qu as a function of frequency

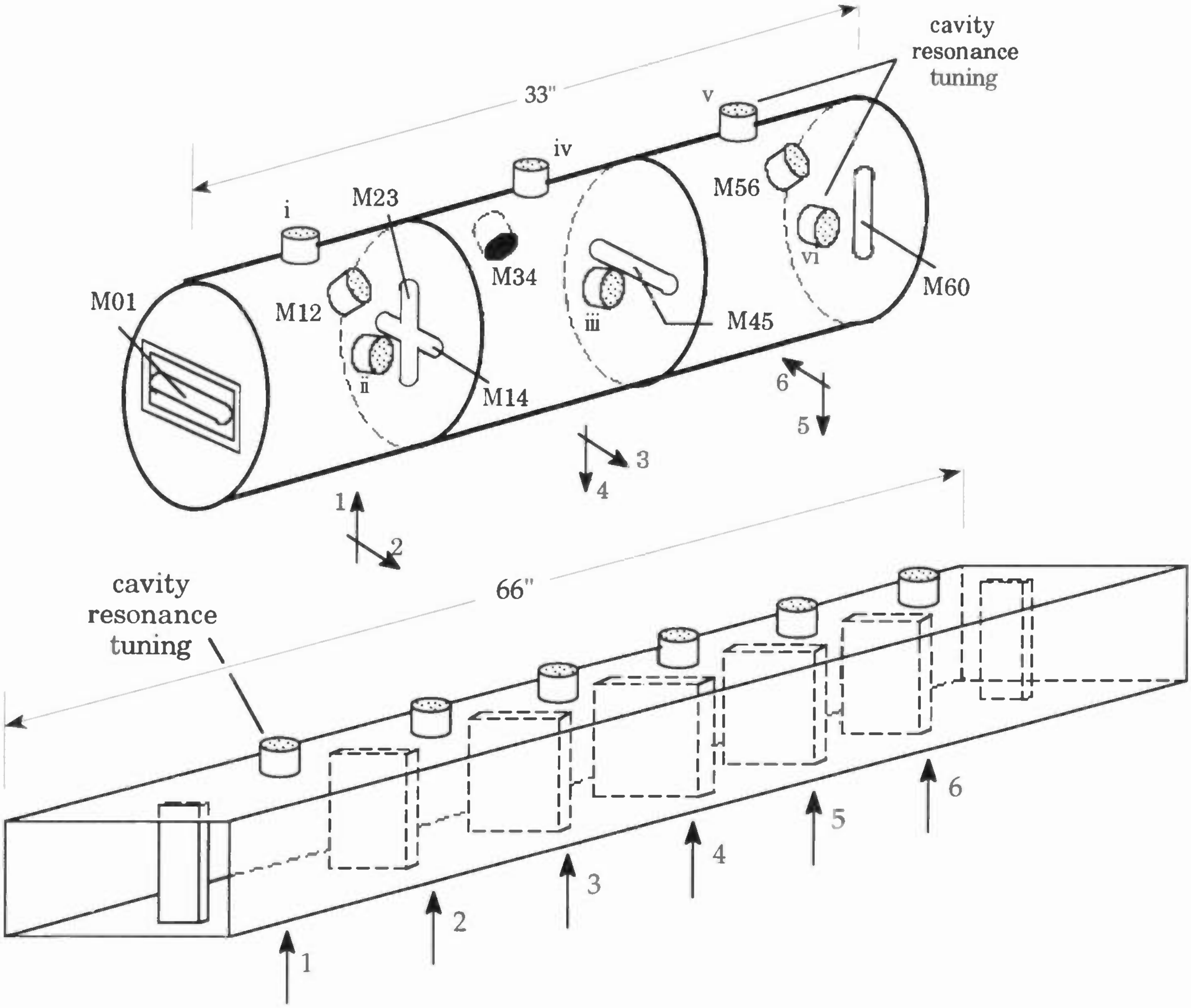

Figure 2. Comparison of single mode cavity filter structure and dual mode cavity structure (top).

Multi-Mode Filter Structures

An example of a single TE101 waveguide mode and dual TE111 mode filter tuned at CH59 is illustrated in figure 2. Each filter contains six resonant cavities (n=6). The single mode cavities are 1/2 wavelength long and schematically depicted by a vertical vector representing the magnitude of electric field polarization. Inductive coupling between adjacent resonant cavities is accomplished using posts and vanes as shown. This technique is well documented and beyond the scope of this paper. Fine tuning of cavity resonance is accomplished using the tuning plungers shown.

The dual mode cavities are also 1/2 wavelength long and contain two orthogonal modes or resonances schematically depicted by orthogonal vectors. The obvious advantage of using dual mode cavities is reduced overall length since one cavity supports two resonances. The two resonances within one cavity are coupled to each other by a tuning plunger located at 45 degrees with respect to the cavity resonant tuning screws. Coupling to adjacent cavities is inductive and accomplished using slotted iris plates. The incentive for using dual-mode cavities over conventional single mode in-line direct coupled cavities arises principally from their ability to couple electrically non-adjacent resonators. Non-adjacent coupling (cross coupling) is achieved using inductive slot indicated in figure 2 as M14. The sign of the main couplings (i.e. M01, M12, M23, ...) is positive throughout the filter. Pseudo-elliptic function responses are readily realized by introducing negative cross couplings, or, couplings opposite to that of the main couplings. This is achieved by placing the 45 degree intercavity coupling screws in such a manner as indicated in figure 2 so that vector 4 is 180 degrees from vector 1. Alternatively, a delay equalized response is realized by maintaining positive cross coupling between resonance 1 and 4.

Common Amplification Dual Mode Filter Response

Another incentive for using dual mode filters is illustrated by comparing the pseudo-elliptic function and Chebyshev response shown in figure 3 for a typical common amplification intermod filter requirement. All filters feature a passband width of 7Mhz with a 1.05 vswr.

As an example, if 25dB rejection is required at fv - 3.58 and +8.08 the six section chebyshev filter will not satisfy the specification while the seven section Chebyshev or six section pseudo-elliptic will. If the filters are tuned at an upper UHF channel, the lower Qu Chebyshev filter constructed from WR1150 waveguide would have approximately 0.09dB more insertion loss over the pseudo-elliptic filter constructed of high Qu cylindrical cavities. This is a significant amount if the balanced intermod filter insertion loss specification is 0.15dB. If 45dB attenuation is required at fv - 4.5Mhz the seven section Chebyshev design now becomes unacceptable. The 7 section Chebyshev filter is usable if it is redesigned to have a narrow passband (i.e. 6.5Mhz) and increased vswr (i.e. 1.08). This may be acceptable since, in a balanced configuration, the actual passband vswr can be minimized by "hiding" the filter behind a hybrid and sacrificing reject load

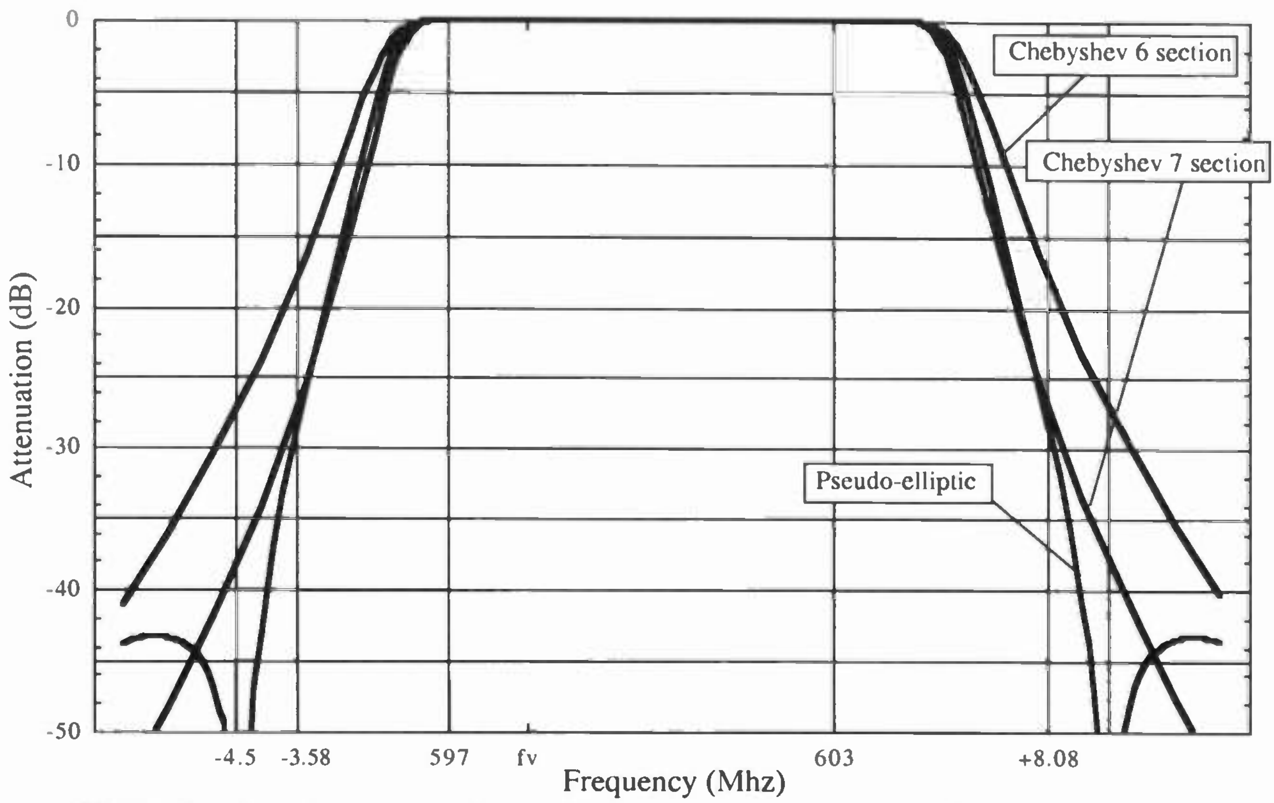

Figure 3. Chebyshev and pseudo-elliptic filter responses for intermod filter requirements.

isolation. Group delay variation is also sacrificed since the passband bandwidth has been reduced. Minimum delay variation is achieved with filters designed for wider passbands than actually used with a minimum number of sections. Therefore, pseudo-elliptic filters can provide increased rejection and a wider passband width over Chebyshev filters. Moreover, pseudo-elliptic filters are readily constructed using dual mode cavity structures which results in a significant reduction in the overall length of the unit..

HDTV Dual Mode Filter Response

ATV out-of-band emission masks have been proposed for systems located N+1/N-1 to NTSC stations Their characteristics depend on mileage between stations. Several transmitter manufacturers have approached Passive Power Products with ATV filter requirements to meet these masks. A typical filter requirement is shown in figure 4. Delay variation over the 6Mhz passband should be less than 100ns.

A Chebyshev design is not practical due to the number of cavities required to meet rejection specifications in addition to excessive group delay variation. A chebyshev filter of lesser degree with a pair of symmetrically placed notches is a possible solution but is more complex and higher insertion loss results due to low cavity Qu. The low cavity Qu also equates to more passband roll-of, or, greater delay variation.

The preferred solution is an eight section pseudo-elliptic filter function using higher Qu dual mode cavities as it provides lower delay variation and lower insertion loss. The filter has two non-adjacent cross couplings to produce two transmission zero pairs, or four notches. One cross coupling controls the +/- 1.25Mhz from band edge notches while the other controls the +/-6.75Mhz band edge notches. Figure 5 illustrates theoretical and measured response of the ATV prototype shown in figure 6. Group delay variation was 98ns over the 6Mhz passband.

A potential problem that may occur with these filters is how the group delay specification relates to the +/-1.25Mhz rejection specification. As more rejection is required, group delay variation must be allowed to increase. There are solutions to flattening the variation over a percentage of the passband, but only at the expense of sacrificing rejection. However, the effects of delay flattening occur mostly at midband, and are less effective at the passband edge.

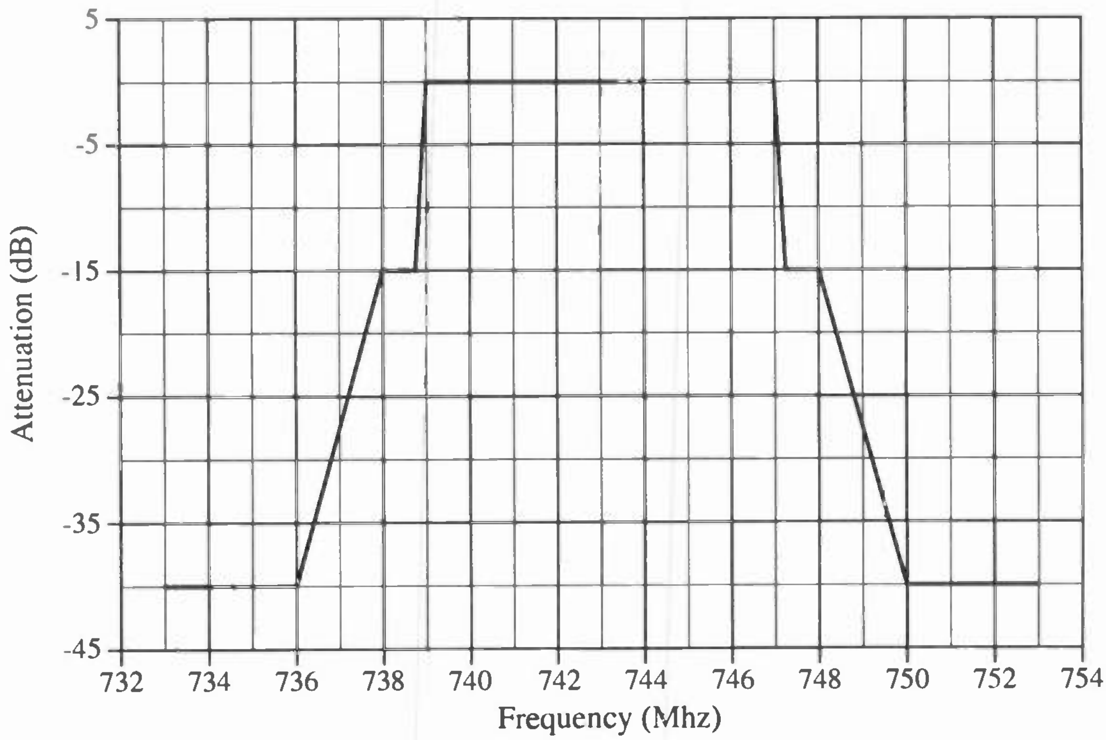

Figure 4. Typical ATV filter requirement

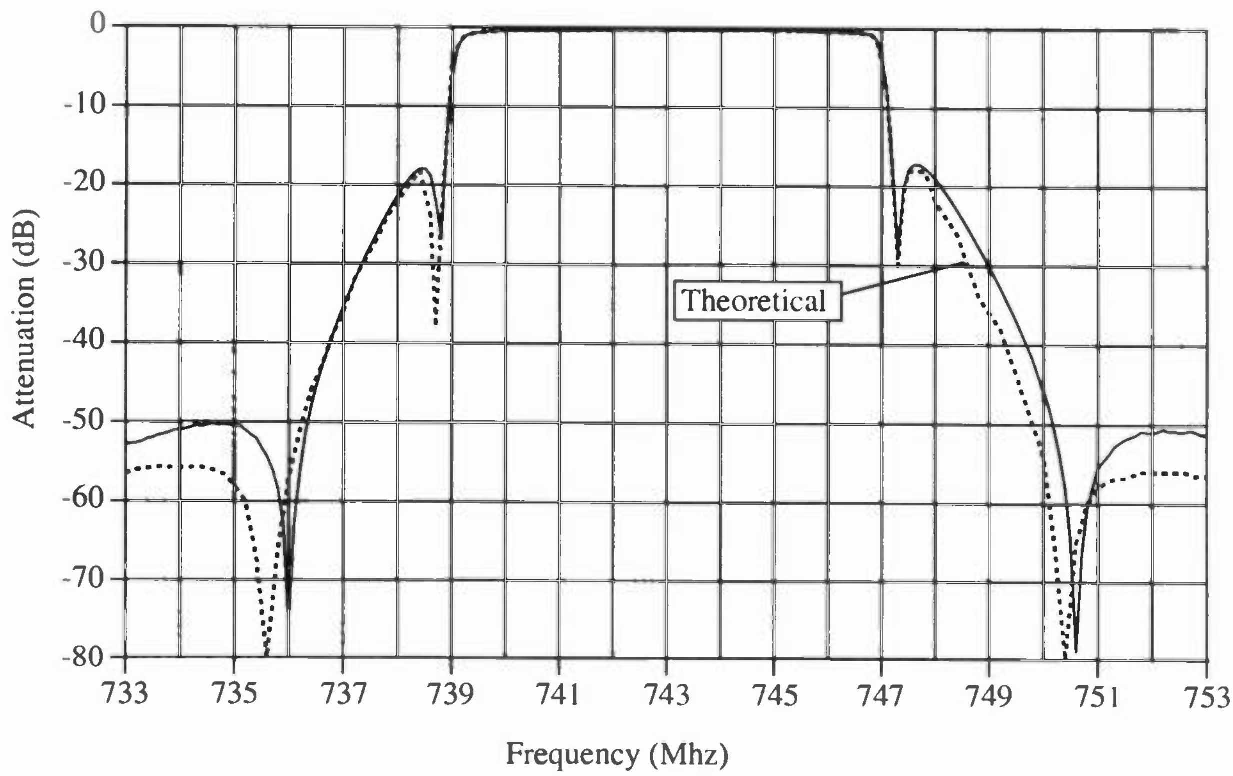

Figure 5. Theoretical and measured response for ATV prototype filter

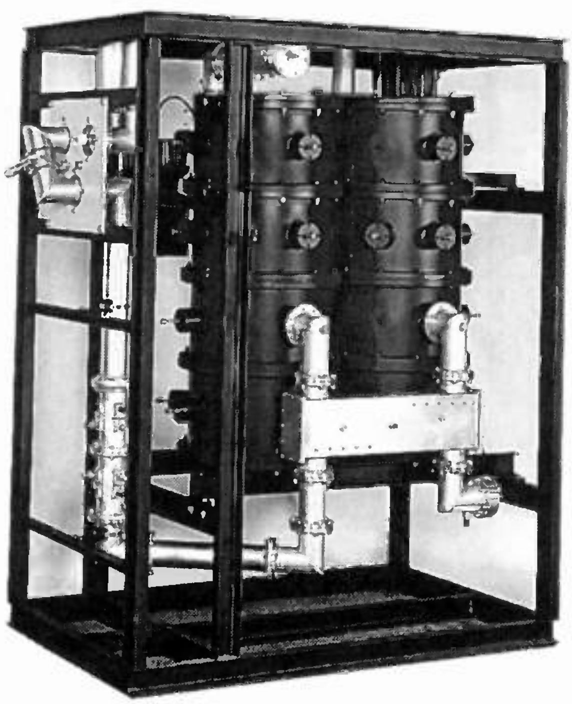

Figure 6. ATV combiner prototype

DAB Filter using Dual Mode Filters

Passive Power Products has manufactured Digital Audio Broadcast filters for L-band and VHF band III frequency ranges. Table 1 reflects typical specifications for these filters. DAB filters, similar to ATV filters, require a significant amount of rejection close to the passband edges. The VHF filter calls for 25 dB rejection at 230 Khz from the band edges. Achieving this level of rejection requires an 8 section cascade quadruplet (CQ) filter circuit [2] which is illustrated in figure 7a along with the measured filter response shown in figure 7b. Each dot represents a resonant cavity and the heavy lines represent the main couplings between adjacent cavities denoted as M12, M23, ...M78. The main couplings in the filter structure are inductive, or, of positive (+) sign. The smaller cross couplings (M14, M58) between non-adjacent cavities are represented by capacitors and are negatively coupled to produce transmission zero pairs in the filter response.

TABLE 1

Specification	VHF Band III Filter	L-Band Filter
Fo:	217 - 230 Mhz	1452 - 1492 Mhz
Passband Bandwidth:	1.54 Mhz	1.7 Mhz
Insertion Loss:	1.5 dB max.	0.85 dB max.
Rejection:		
Fo +/- 1.0 Mhz	-25 dB	xx
Fo +/- 1.5 Mhz	-60 dB	xx
Fo +/- 2.0 Mhz	xx	-15 dB
Fo +/- 3.0 Mhz	xx	-30 dB
VSWR:	1.10	1.10
Power:	10 kW Peak	1 kW Peak

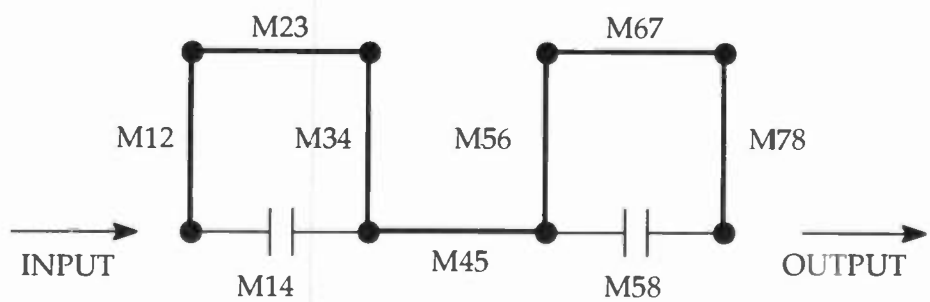

Figure 7a. Cascade quadruplet circuit showing signal flow

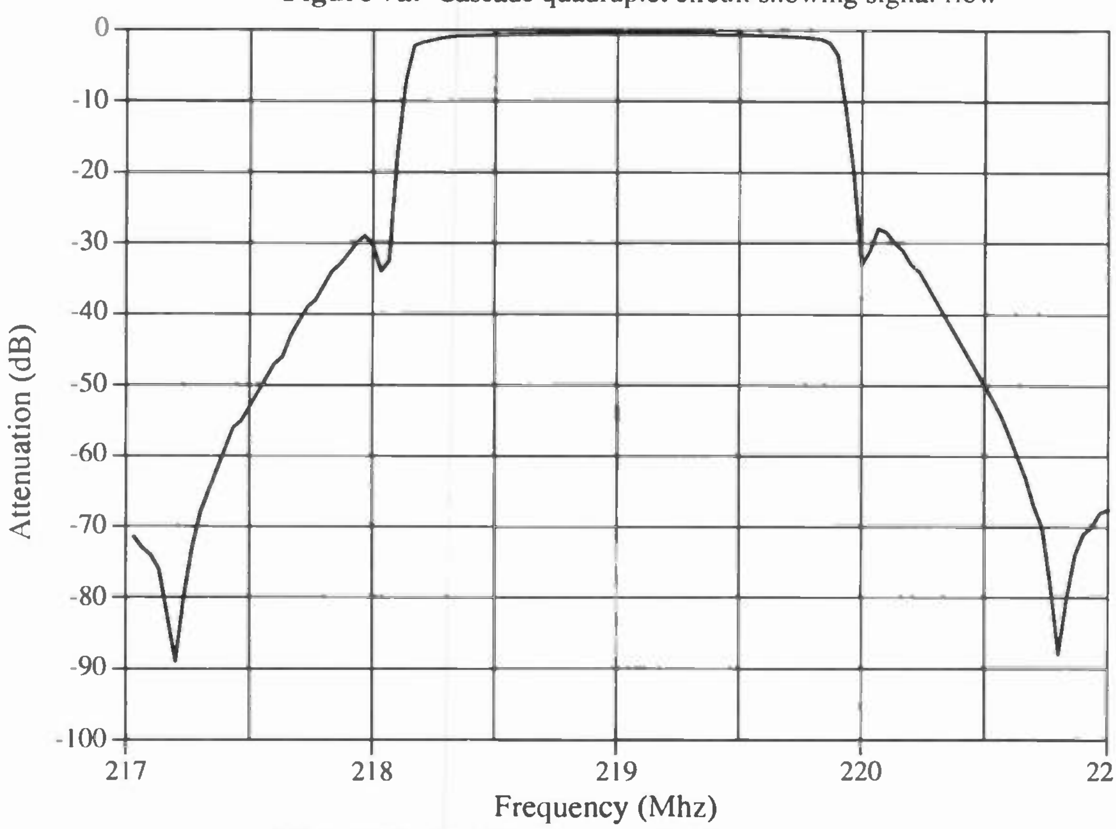

Figure 7b. VHF Band III DAB Filter, measured data.

CQ refers to a cascade of quadruplet coupled circuits. A quadruplet coupled circuit is one where four resonant cavities are adjacently coupled (M12, M23, and M34) and a single non-adjacent coupling exists between cavities 1 and 4 (M14). The primary advantage of a CQ circuit is that the cross couplings, M14 and M58 (figure 7a), can be tuned independently. Stated otherwise, the M14 coupling, which controls a pair of symmetrically placed poles in the transmission response, can be tuned to move the pole pair without affecting the location of the M58 pole pair. This feature is not possible in elliptic function filters; i.e., the tuning of one pole pair effects all other poles, making filter tuning difficult and tedious.

Refer to figure 7b. The M14 cross coupling is tuned to place poles +/- 230 Khz from passband edges to meet rejection specifications. The M58 cross coupling is tuned to provide 60 dB attenuation +/- 1.77 Mhz relative to Fo.

Dual mode cavities were not used in the VHF design due to size limitations and allowed insertion loss. Lower Qu coaxial resonators provided adequate insertion loss and a much more compact design.

The L-band DAB filter specifications are somewhat relaxed compared to the VHF specifications. The L-band specifications dictated a six section elliptic function response using dual mode cavities. Triple mode (TE111,TE111,and TM010) cavities were investigated to realize the filter response, but, the lower Qu TM010 mode hindered the insertion loss. The L-band filter is unique to previously discussed filters because it provides an elliptic function response using dual mode cavities. This is achieved by restructuring the vectors in figure 2 to accommodate two non-adjacent cross couplings. Data is presented in figure 8.

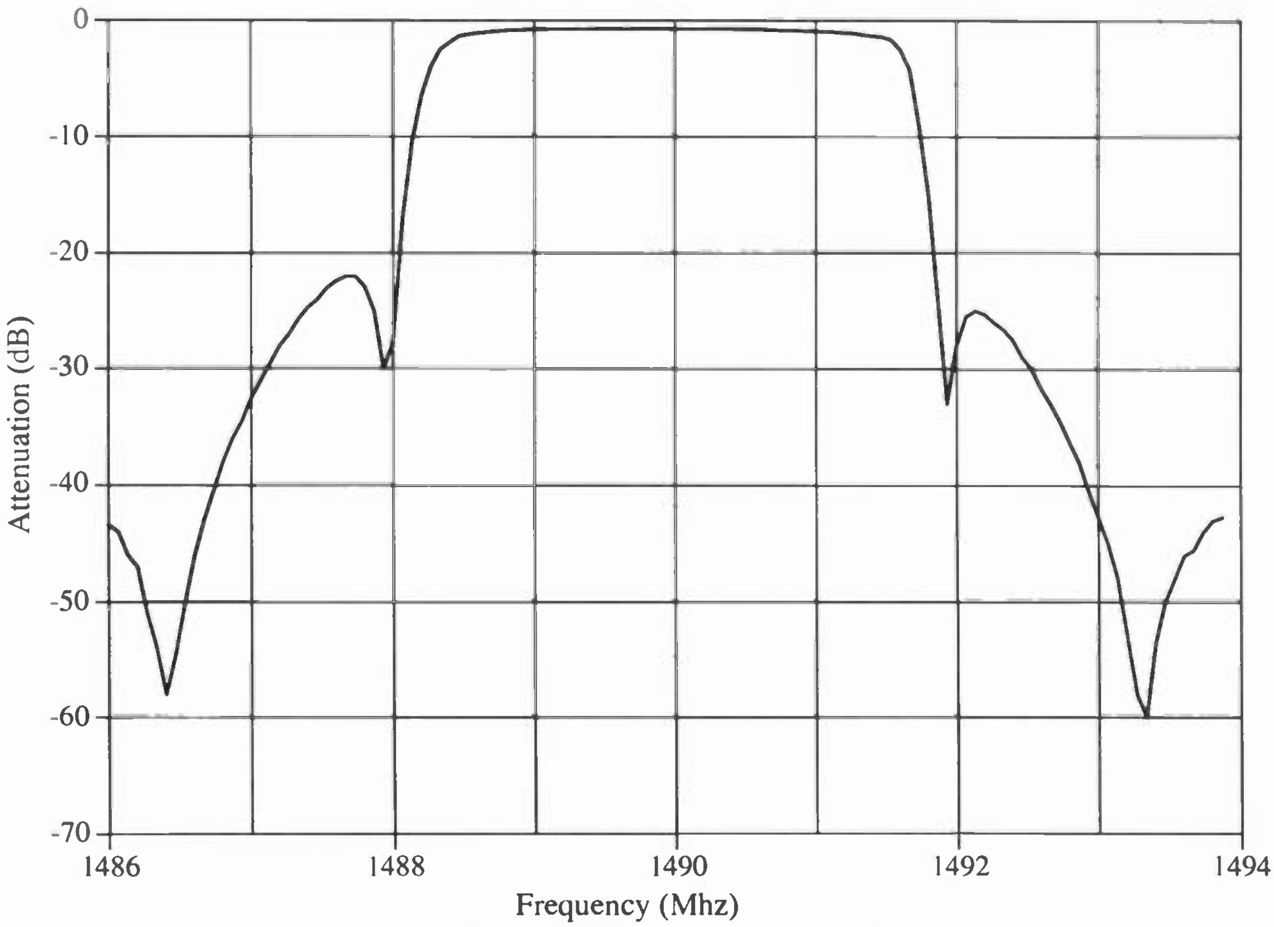

Figure 8. L-Band DAB Filter, measured data.

ATV/NTSC Channel Combining

The most challenging task for rf systems manufacturers may be the successful design of an adjacent channel ATV/NTSC combiner . Adjacent channel combiners will require complex filter designs using high Qu cavities to meet diplexer rejection requirements. A guard band will have to be allowed for between the channels. The guard bandwidth will depend on how much group delay and insertion loss variation is tolerable for each channel. Less guard bandwidth means high degree complex filters.

With the anticipation of future requirements for adjacent channel combining, Passive Power Products is currently investigating the feasibility of using a CONSTANT IMPEDANCE combining system shown in figure 9. The system is similar to a constant impedance intermod filter with the ballast load removed for the ATV signal. The system consists of two hybrids and two bandpass filters tuned to pass the NTSC channel and reject the ATV signal. The NTSC signal is delivered to the input hybrid where it is split and delivered to the filters. The signals are passed through the filters and are recombined at the output hybrid and sent to the antenna. The ATV signal is delivered to the output hybrid where it is split and sent to the filters tuned to reject it. Rejected signals are sent back to the output hybrid where they are recombined and delivered to the antenna with the NTSC signal.

Filter requirements for this type of diplexer will primarily depend on the channel spacing of the diplexed signals. Semi adjacent (or greater) channel combining is easily accomplished and possible with existing common amplification constant impedance intermod filters. This provides current broadcasters that use common amplification intermod filters a cheaper solution for simulcasting ATV/NTSC signals. Retro fitting the intermod filter would consist of removing the ballast load and plumbing the ATV signal to that port. RF tuning may be required, but will be minimized by diplexing an ATV signal that is semi adjacent.
Adjacent channel combining, however, will require NTSC filters significantly more complex. A lower adjacent ATV channel combiner, for example, may require a 750 Khz guard band between it and the NTSC channel. The guard band could occupy the first 750Khz of the NTSC channel. This translates to large group delay variations in the NTSC signal near the visual carrier (possibly up to 400ns). ATV delay variations can be minimized by providing enough rejection at the lower adjacent ATV signal in the filter. Loss variations must also be considered with delay variations of this magnitude.

Conclusion

The results presented in this paper have illustrated that multimode cavity filters ehibiting a pseudo-elliptic response provide a viable and cost effective solution to the demanding bandpass filter specifications for modern TV transmitter systems. The utility of these filters has been established in that they are readily implemented into TV systems as low loss intermod filters or multistation combiners.

The attractive features of multimode cavity filters

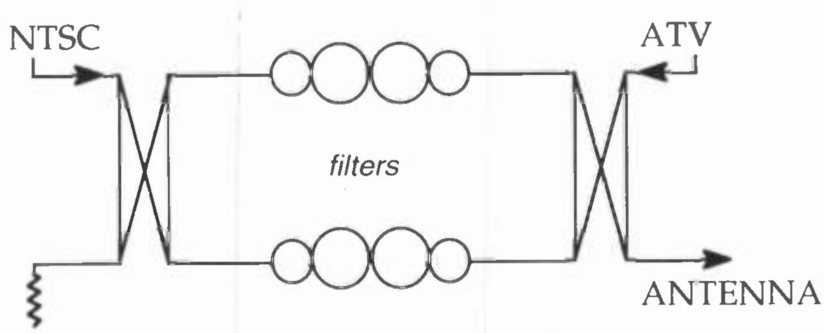

Figure 9. RF flow diagram, NTSC/ATV combiner

over conventional cavity filters include significantly reduced size and weight as well as their ability to couple electrically non-adjacent resonators.

Data from working proptotypes has shown that the performance of these filters can fulfill the requirements for such applications as ATV, DAB and common amplification systems.

Acknowledgements

The author wishes to thank his colleagues at Passive Power Products for their valuable technical contributions and support, without which, much of the work and development described herein would have been exceedingly more difficult.

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STANDARD REFERENCE CURVES FOR ATV FILTER MASK

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ABSTRACT

This paper presents transmission loss and group delay curves for bandpass filters that will serve as a reference to broadcast engineers who are involved with ATV filter specification. The types of filter responses that are considered here include Chebyshev (equal ripple) and Cauer-Darlington (elliptic function) bandpass filter responses of various orders, passband ripple levels and stopband attenuation levels.

BACKGROUND

The possibility of ATV channel assignments adjacent to existing NTSC channels in the same broadcast area has renewed interest in NTSC signal sensitivity to undesired signals [1] [2]. At the heart of this matter lie two fundamental questions: (1) what measure should be applied to quantify the deleterious effects of an undesired/interfering signal and (2) once defined, how can this measured/calculated quantity be correlated to picture quality.

Picture quality is typically graded by a subjective *mean impairment rating*, wherein an impairment rating of 5 signifies 'imperceptible" picture degradation and an impairment rating of 1 signifies a "very annoying" picture. At a mean impairment rating of 5, the undesired signal is said to be at the *threshold of visibility* (TOV). An impairment rating, however, is a subjective quality and is not directly measureable.

A common measure of received signal quality is the ratio of desired to undesired (d/u) signal power in the channel's frequency band. The "desired" signal in NTSC applications has historically been quantified by the peak power amplitude of the NTSC sync pulse. This is a reasonable approach because the NTSC signal spectrum is well defined and consistently predictable with respect to the sync level. Unfortunately, quantifying the undesired signal level is not always such a straightforward matter and an accurate correlation between d/u ratio and picture impairment requires careful consideration of the nature of the undesired signal.

If an undesired/interfering signal is uniformly distributed in frequency over an NTSC channel, then it may properly be quantified by its average power level over the channel's 6 MHz band*. In studies where this has been the case, a d/u (peak sync to average noise) ratio of 50 dB has been determined to produce a grade 5 picture and thus represent the threshold of visibility for uniformly distributed noise. On the other hand, if the interfering

* This average power level would ideally be measured by passing the undesired signal through a perfect bandpass filter into an rf power meter and time averaging the measured power level.

signal is known to vary in power across the channel, as is the case with ATV transmitter spillover, then the 50 dB d/u ratio should no longer be considered to be the threshold of visibility for that interferor. Studies which are summarized in [1] indicate that the NTSC signal sensitivity to interfering signals varies markedly with the spectral distribution of the interference. This spectral variation is modeled in [1] where it is referred to as the *subjective weighting function*. A *weighted average power* is then defined as the integrated product over the channel of the interferor's power spectrum and the subjective weighting function. From the known power level of uniformly distributed noise that is at the threshold of visibility, a *weighted threshold of visibility* is defined as the weighted average power corresponding to that level of uniform noise. The weighted TOV thus provides a robust criterion for the measurement of interference that applies equally to interferors of arbitrary spectral distribution, within the validity of the subjective weighting function.

A further complication in predicting ATV interference to adjacent channels is the simple fact that the relative power levels of the two stations will vary nonuniformly over the broadcast coverage zones. If the two stations are co-located and have identical radiation patterns, then the NTSC station's d/u ratio with respect to the ATV interferor will be constant over it's coverage zone. If, on the other hand, the ATV transmitter is located near the perimeter of the NTSC coverage zone, then it will have a very significant power advantage over the NTSC signal in the neighborhood of the ATV antenna and a corresponding disadvantage at the diametrically opposed region of the NTSC stations's coverage zone. This factor is considered in the analysis presented in [1] where two simplifying assumptions are made: (1) the average in-band power of the transmitted ATV spectrum is 23 dB below the peak sync amplitude of the transmitted NTSC signal and (2) both stations have identical antenna patterns (which are assumed). While these assumptions were necessary to keep the analysis tractable, a full accounting of this spatial power advantage relationship may only be taken on a case by case basis with knowlege of both stations' radiated power levels and antenna patterns.

Consider now the task of determining a suitable bandpass filter mask for ATV operation with adjacent NTSC channels. It seems apparent from the foregoing discussion that a rigorous determination of an optimal filter mask requires knowledge of (1) the radiated powers of each station, (2) the associated antenna patterns and (3) the spectral characteristics of the ATV transmitter spillover. Even with this knowlege, more than one type of filter can be found to satisfy a weighted TOV requirement since spillover emissions may be traded off between various segments of the frequency band. Given sufficient information, though, a filter manufacturer would well be able to recommend quasi-optimal custom filters for specific ATV transmitter applications.

For many broadcasters, a weighted TOV calculation will likely be considered to be excessively complex. These broadcasters will prefer a fixed emission mask to comply with. Neglecting the spatial power advantage previously mentioned, this is a straightforward task (if excessive filtering is not objectionable). Simply require all out of band emissions to lie below the uniformly distributed noise level that corresponds to the TOV in the visual portion of the adjacent channel spectrum and below the uniform noise level corresponding to the threshold of audibility (TOA) † in the aural portion of the adjacent channel spectrum.

† The threshold of audibility (TOA) is defined analogously to the threshold of visibility.

At the time of this writing, the Advisory Committee on Advanced Television Service (ACATS) is working towards the establishment of an ATV emission mask standard and verification protocol. In light of the fact that only an estimated ten percent of ATV channel assignments will be adjacent to existing NTSC channels, perhaps multiple emission mask requirements will be established based upon adjacent channel occupancy. Unfortunately, since the emission/filter mask debate is still underway, specific bandpass filter requirements for ATV cannot at this time be accurately predicted. It shall thus be the purpose of the remainder of this paper to describe the basic filter response options and tradeoffs that will be available to the broadcaster once the requirements are set forth.

FILTER SPECIFICATION

Terminology

There are several characteristics of bandpass filters that shall be discussed throughout the remainder of this paper. In order to avoid any possible confusion, these terms shall now be defined. It shall be assumed that the reader is familiar with the widely agreed upon definitions of two-port device parameters including: scattering parameters (s-parameters), return loss, vswr, insertion loss, isolation and group delay.

The *passband* of a filter refers to that range of frequencies in which signals are expected to pass through the filter with less attenuation than some specified maximum value that is referred to as the *passband insertion loss*. Occasionally, however, the passband definition is broadened to include frequencies at which signals will pass through the filter with less attenuation than some other given amount. In such cases, if the maximum allowable attenuation for a specified passband were, for example, 3 dB, then that passband would be distinguished from the strictly defined lowest loss passband by referring to it as the '3 dB passband".

Just beyond the passband of a filter, on both upper and lower sides, lie the *transition regions*. The filter's attenuation in the transition regions increases from the passband insertion loss at the edges of the passband to a specified minimum level of attenuation referred to as *the minimum stopband attenuation*. This attenuation level is first reached at frequencies that lie above and below the passband and define the end of the transition regions and which are referred to as *the upper and lower stopband frequencies*. The steep slopes of the filter response function in the transition region are frequently referred to as the *skirts* of the filter. The characteristic curveature of the response function as it transitions from a somewhat flat passband into the steep skirts of the transition band is referred to as the *rolloff* of the filter response.

The two ranges of frequencies that lie below the lower stopband frequncy and above the upper stopband frequncy of a bandpass filter are referred to respectively as the lower and upper stopbands. Although it is often assumed that a bandpass filter will continue to attenuate signals by at least the stopband attenuation level for all frequencies ranging from the lower stopband frequency down to zero frequency as well as from the upper stopband frequency up to infinity, this is seldom the case in actuality. A real bandpass filter is a resonant device that will usually exhibit some degree of harmonic response which depends upon the construction of its resonant structures and inter-structure coupling mechanisms. If the existence of additional passbands at harmonics of the fundamental passband are a concern, then this should be addressed explicitly in a filter's specification.

Bandwidth refers to the amount of separation between two given frequencies, the 'lower frequency" and the "upper frequency", and is usually expressed one of three ways. The most straightforward definition of bandwidth is the *absolute bandwidth* which is simply the arithmatic difference between the upper and lower frequencies. This measure of bandwidth is expressed in units of hertz (Hz), kilohertz (KHz), megahertz (MHz)... etc. Sometimes a more significant expression of bandwidth is the *fractional bandwidth* which is defined as the absolute bandwidth divided by the center frequency of the band under consideration. If the fractional bandwidth is multiplied by 100, then it may still be referred to as the fractional bandwidth, or it may possibly be called the *percent bandwidth*.

FILTER RESPONSE FUNCTIONS

The frequency-selective attenuation (frequency response) of any given bandpass filter is derived from a low pass "prototype" filter response that is subsequently transformed into its bandpass "equivalent". These low pass prototype filter responses are modeled by and derive their names from special mathematical functions. The names of these filter response functions are also used as names for the corresponding bandpass equivalent filters. The filtering functions that shall be considered in this paper include the Chebyshev polynomials and the so-called elliptic (or Zolatarev) functions.

Chebyshev filters.

The Chebyshev family of filters are often referred to as *equal ripple* filters because the passband insertion loss exhibits ripples of a given, pre-specified amplitude. A Chebyshev filter may be specified by the following set of five parameters: (1) the center frequency, (2) the percentage bandwidth of the passband, (3) the percentage bandwidth between the upper and lower stopband frequencies, (4) the stopband attenuation level and (5) the passband ripple. The combination of these parameters dictates the minimum required *order* of the filter. For a bandpass filter, the order is equal to the number of resonant structures that are required to realize the filter response in an actual bandpass filter[‡]. An important fact to realize in regards to the elementary [3][§] Chebyshev and elliptic filters that are considered in this paper is that all energy that does not pass through the filter, with the exception of the dissipative portion of the insertion loss, is reflected back towards the source. This means that the insertion loss due to passband ripple is reflected energy. Therefore, the maximum ripple that is allowed to exist in the passband of a filter is limited by the maximum allowable passband vswr. The vswr that results from passband ripple is given in Table 1 for several values of vswr.

VSWR	RL (dB)	Ripple (dB)
1.00	0	0
1.05	32.3	.002
1.10	26.4	.010
1.15	23.1	.021
1.20	20.8	.036
1.25	19.1	.054
1.30	17.7	.075

Table 1. Maximum Allowable Passband Ripple for a Given VSWR or Return Loss Specification

[‡] The "order" of the filter actually refers to the order of the Chebyshev polynomial that describes the low pass response function and is one-half the order of the equivalent bandpass response polynomial.

[§] The term *elementary* is used here to distinguish simple two-port reflective filters from composite "constant impedance" filters that are discussed in a paper by Heymans.

In order to illustrate the transmission characteristics of Chebyshev filters, the following families of filter response curves are presented. Figures 1-4 illustrate the effect that increasing order and passband ripple level has on the steepness of the filter skirts. These curves depict Chebyshev bandpass response functions for various orders, ranging from 5 to 10, and various passband ripple levels that are listed in Table 1 ranging from 0.002 to 0.036. The corresponding group delay curves for these filter responses are depicted in Figures 5-8. The out of band attenuation performance of these filters is readily apparent. However, the interpretation of the group delay curves is rather difficult. Group delay is a rather obtuse indicator of the waveform stretching properties of a filter. It is important to understand that, when comparing group delay curves, the best available indicator as to the relative amount of waveform stretching that will be caused by any given filter is the *variation in the group delay* across the transmitted part of the spectrum. There are, perhaps, better ways to describe the distorting properties of a filter, such as the filter's impulse or step response. Unfortunately, an analysis of pulse distortion due to group delay variation cannot be included within the confines of the present paper.

A few points should now be made concerning the format of the depicted data as well as the choice of parameters that are considered in this paper. The frequency axes that appear in these figures and throughout this report are labelled as *relative frequency*. This shall be understood to mean frequency relative to the lowest frequency of the 6 MHz band that that part of the curve occupies in the NTSC channel spectrum (i.e. modulo 6 MHz). All frequency axes in this report are thus seen to begin in the 'n-1" lower adjacent NTSC channel and either end within the main (ATV) channel, which the subject filter is intended to pass, or end in the 'n+1" upper adjacent NTSC channel. Where appropriate, the visual carrier, chroma subcarrier and aural subcarrier frequencies in the adjacent NTSC channels are denoted respectively by the letters V, C and A appearing at the tops of the graphs as well as by vertical dashed lines that appear below these letters.

Unless otherwise noted, the passbands of all of the filter response curves that appear in this paper are 6 MHz wide. The rational for choosing this passband width is as follows. The intent of this paper is to serve as a guide in the selection of a bandpass filter for ATV transmitters. From previously published reports, it appears that the desired portion of an ATV transmitter spectrum spans 5 MHz of bandwidth that extends from 500 KHz above the lower edge of the channel to 500 KHz below the upper edge of the channel. This would seem to indicate the need for a 5 MHz filter passband. However, thermal drift is a significant factor that must be considered in the selection and tuning of waveguide resonator filters. Strictly controlled experiments have been performed at Micro Communications Inc. that indicate a negative shift in the overall frequency response of a waveguide resonator filter that is approximated by a thermal drift coefficient equal to -8.3 KHz per degree fahrenheit of temperature increase. Furthermore, practical experience has shown that a thermal drift of up to approximately 500 KHz is to be expected for these filters under typical combined average power and ambient temperature broadcasting conditions. Therefore, in order to maintain low loss passband performance over the desired center 5 MHz of the channel under conditions that range from low power, normal room temperature up to high power, warm ambient temperature, the passband of a filter should span 5.5 KHz. Such a filter would be properly tuned under low power, normal room temperature conditions to exhibit a low loss band that extended from 500

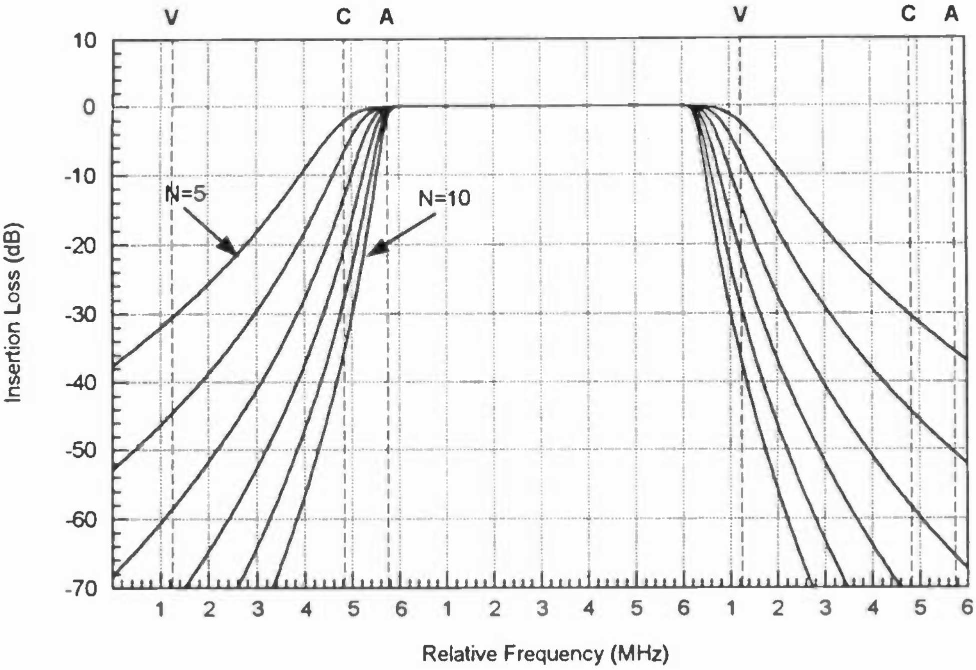

FIGURE 1. Chebyshev filter responses of varying order N ranging from 5 to 10 having 6 MHz passband widths and .002 dB passband ripple.

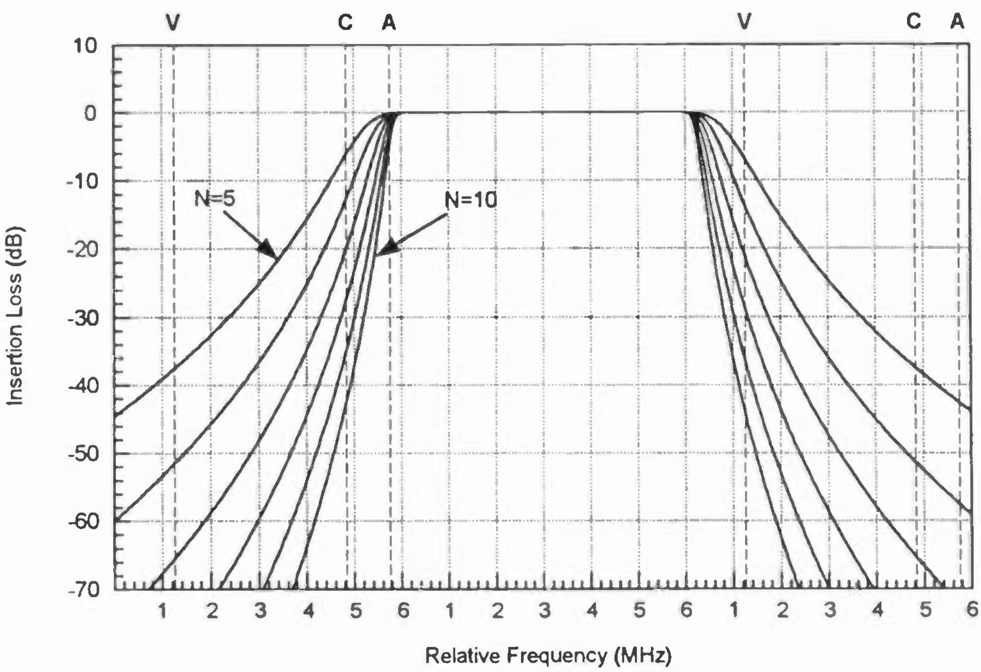

FIGURE 2. Chebyshev filter responses of varying order N ranging from 5 to 10 having 6 MHz passband widths and .010 dB passband ripple.

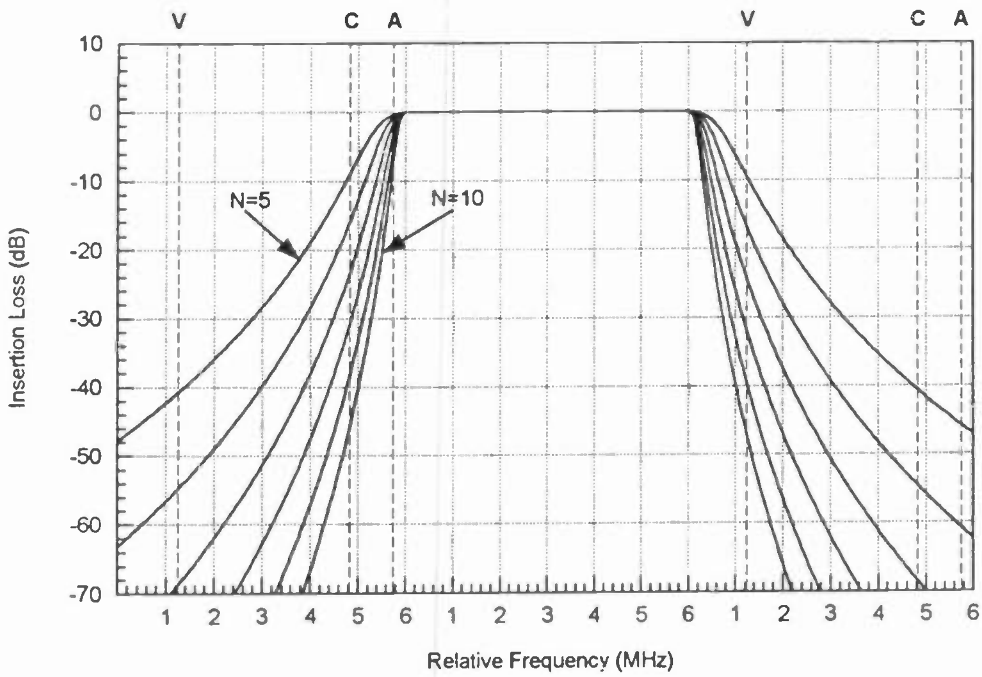

FIGURE 3. Chebyshev filter responses of varying order N ranging from 5 to 10 having 6 MHz passband widths and .021 dB passband ripple.

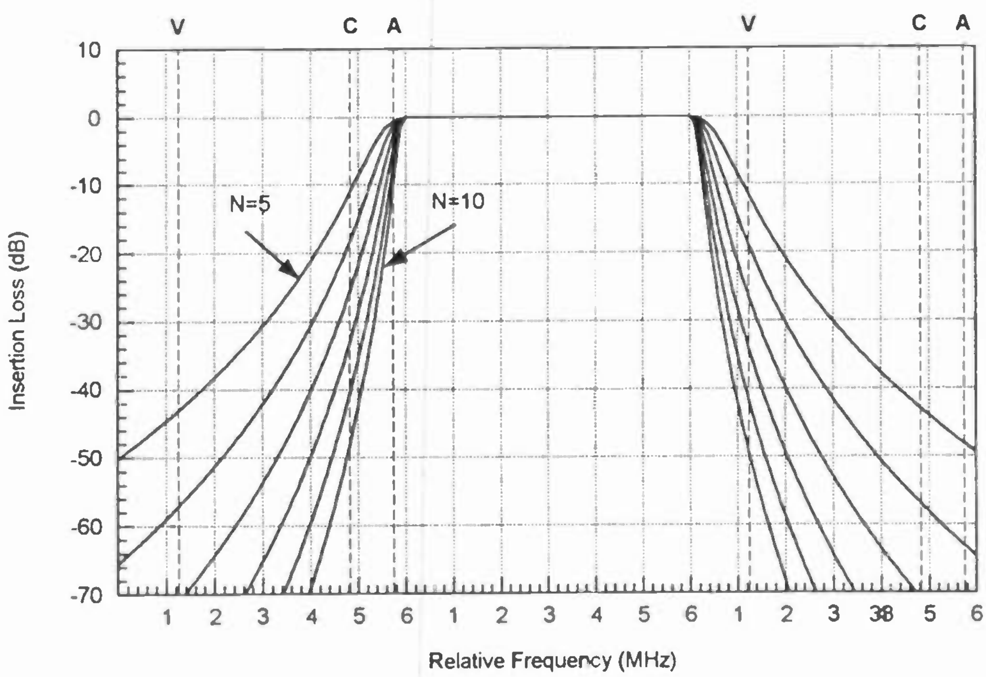

FIGURE 4. Chebyshev filters responses of varying order N ranging from 5 to 10 having 6 MHz passband widths and .036 dB passband ripple.

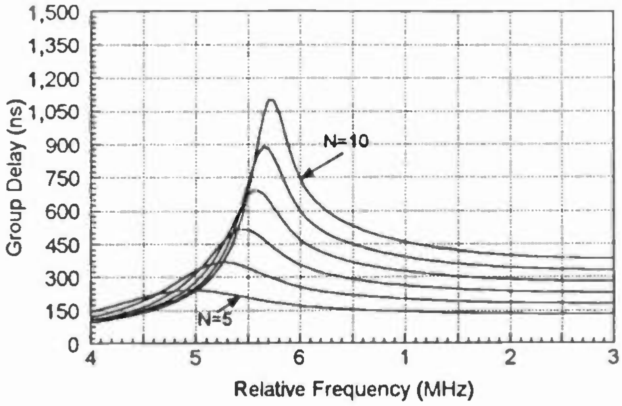

FIGURE 5. Group delay curves of Chebyshev filters having 6 MHz passband widths, .002 dB passband ripple and various order N ranging from 5 to 10. The frequencies depicted represent the upper 2 MHz of the lower adjacent channel and the lower 3 MHz of the main (ATV) channel.

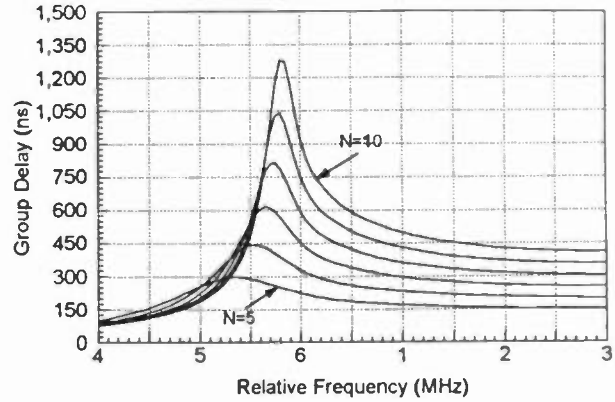

FIGURE 6. Group delay curves of Chebyshev filters having 6 MHz passband widths, .010 dB passband ripple and various order N ranging from 5 to 10. The frequencies depicted represent the upper 2 MHz of the lower adjacent channel and the lower 3 MHz of the main (ATV) channel.

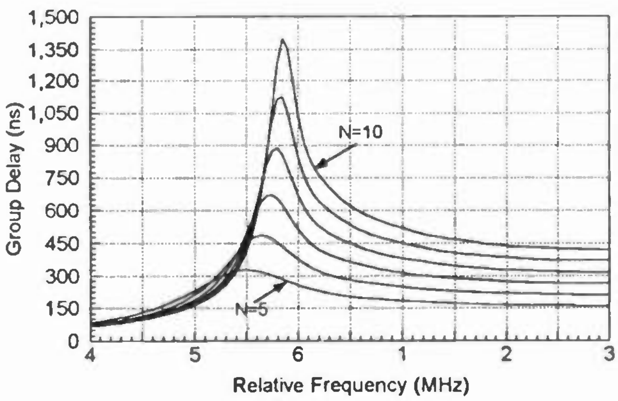

FIGURE 7. Group delay curves of Chebyshev filters having 6 MHz passband widths, .021 dB passband ripple and various order N ranging from 5 to 10. The frequencies depicted represent the upper 2 MHz of the lower adjacent channel and the lower 3 MHz of the main (ATV) channel.

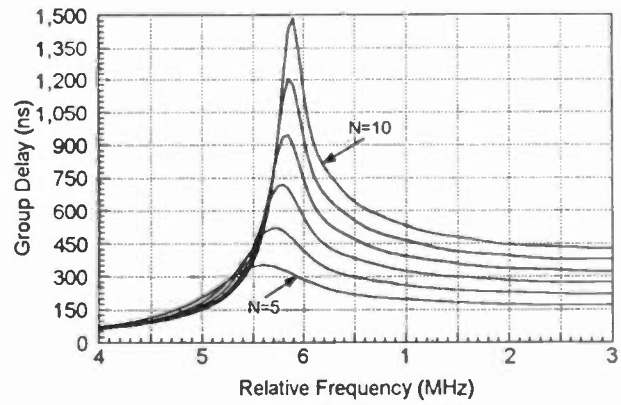

FIGURE 8. Group delay curves of Chebyshev filters having 6 MHz passband widths, .036 dB passband ripple and various order N ranging from 5 to 10. The frequencies depicted represent the upper 2 MHz of the lower adjacent channel and the lower 3 MHz of the main (ATV) channel.

KHz to 6 MHz relative channel frequency and would provide worst case out of band attenuation in the upper adjacent 'n+1" channel under these conditions. Then, under high power, warm temperature conditions, the same filter's passband would extend from 0 to 5.5 KHz relative channel frequency providing worst case out of band attenuation in the lower adjacent 'n-1" channel. The 6 MHz passband bandwidths that are depicted in this paper have thus been chosen to represent an approximation of the worst-case out of band rejection performance of 5.5 MHz passband filters over the total expected range of operating conditions. This presentation of filter responses makes it unnecessary to shift the curves in order to verify compliance with a given filter mask under high power conditions. In order to validate this approximation, filter response curves have been computed for two Chebyshev filters of identical order, passband ripple level and lower passband frequency, but having different passband widths of 6 MHz and 5.5 MHz. The results of these calculations are depicted in Figure 9. The filter having the narrower 5.5 MHz passband exhibits slightly steeper attenuation skirts than the wider passband filter. This means that the 6 MHz filter approximation that is used in this paper errs slightly on the side of caution. Not included in the depiction is the variation in group delay between the two filters. This variation has been calculated and seen to be even less significant than the variation in attenuation response.

One final point to be made with respect to these filter response curves is that no absorptive effects are included in the underlying filter models. In older filter studies, especially of lumped element circuit filters, additional attenuation was observed to affect the out of band attenuation of filters by an appreciable amount. The filters that this paper is concerned with, however, are of such low loss that the effect of the losses on the out of band attenuation response is virtually transparent. The effects of absorptive losses for these filters is significant only within the passband and very close to its edges. This point is illustrated in Figure 10 which depicts the passband response of a series of otherwise identical Chebyshev filters having varying amounts of insertion loss due to absorption in the passband. The passband ripples are seen to be superimposed on somewhat parabolic loss curves that increase monotonically as the edges of the passband are approached from the center. Just beyond the passband edges, the filter response curves merge closely enough that they are indistinguishable, when depicted on a scale such as that used in Figures 1-4.

Elliptic Function Filters.

The so-called elliptic and pseudo-elliptic filter response functions are distinguishable from the Chebyshev family of filter response functions by the presence of additional ripples of attenuation in the filter stopbands that range from a fixed minimum level of attenuation at the tops of the ripples to infinite or near infinite (lossy case) attenuation at certain discrete frequencies within the stopband. These ripples of attenuation are made possible by an intercavity (inter-resonator) coupling scheme that goes beyond the serial connected arrangement of resonators that is used to produce a Chebyshev filter response to a serial/parallel arrangement of coupled resonators.

For the practical filters that are currently available to broadcasters, only even order responses are available. The filter responses for elliptic filters of order 6, 8 and 10; passband ripple levels of .002 and .010 dB and minumum stopband attenuation levels of 40, 45, 50 and 55 dB are depicted in Figures 11-16**. Figure 17 depicts the group delay of elliptic filters of order 6, 8 and 10 with a passband ripple level of .002 and minimum stopband attenuation level of 40 dB. It is seen from this figure that the group delay takes on negative values at frequencies of peak attenuation where the group velocity description of wave propagation is not valid.

The effect on the response function of reducing the passband width from 6.0 MHz to 5.5 MHz has been investigated for the case of the elliptic filter and seen to produce a slight compression of the response in towards the passband center as was seen in the case of the Chebyshev filter response. The effect of introducing absorptive losses into the model was also investigated and found to produce negligible variations in the stopband characteristics as was seen in the case of Chebyshev filters.

** The reader should now be cautioned that while all of these filters *can* be realized (at least approximately), none of the higher order filters (8 and 10) that are depicted have been advertised as being available for broadcast applications.

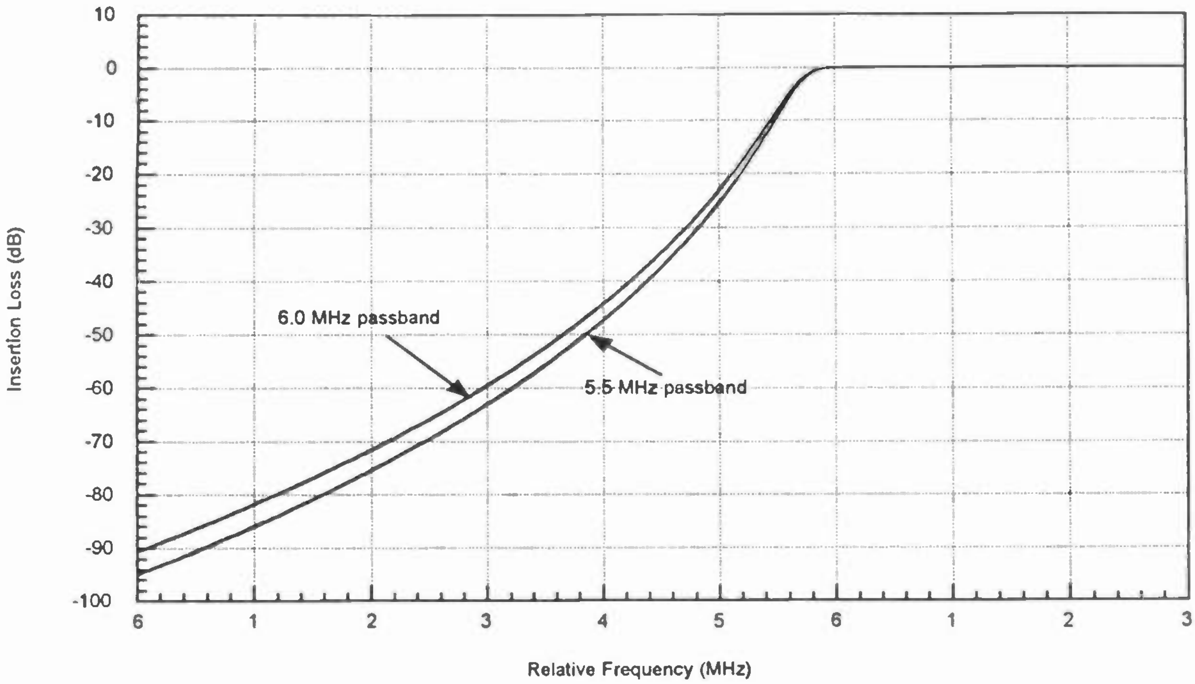

FIGURE 9 : Comparison of two 8th order Chebyshev filter responses having identical passband ripple levels Rp=.010 and lower passband frequencies; but having different passband widths of 5.5 MHz and 6.0 MHz. The frequencies depicted represent the entire lower adjacent channel and one half of the main (passband) channel.

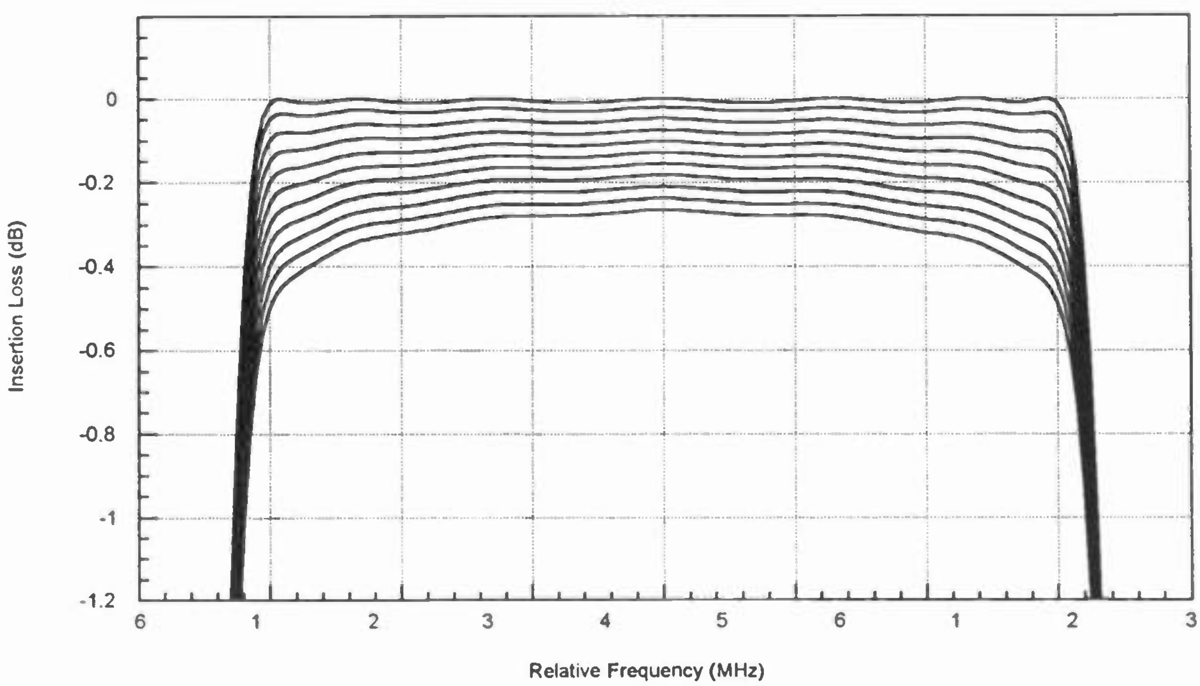

FIGURE 10 : Chebyshev filter passband responses for varying amounts of insertion loss up to 0.5 dB (measured at band edge). Each response curve depicts a 7th order filter with .010 dB passband ripple before adding absorptive losses.

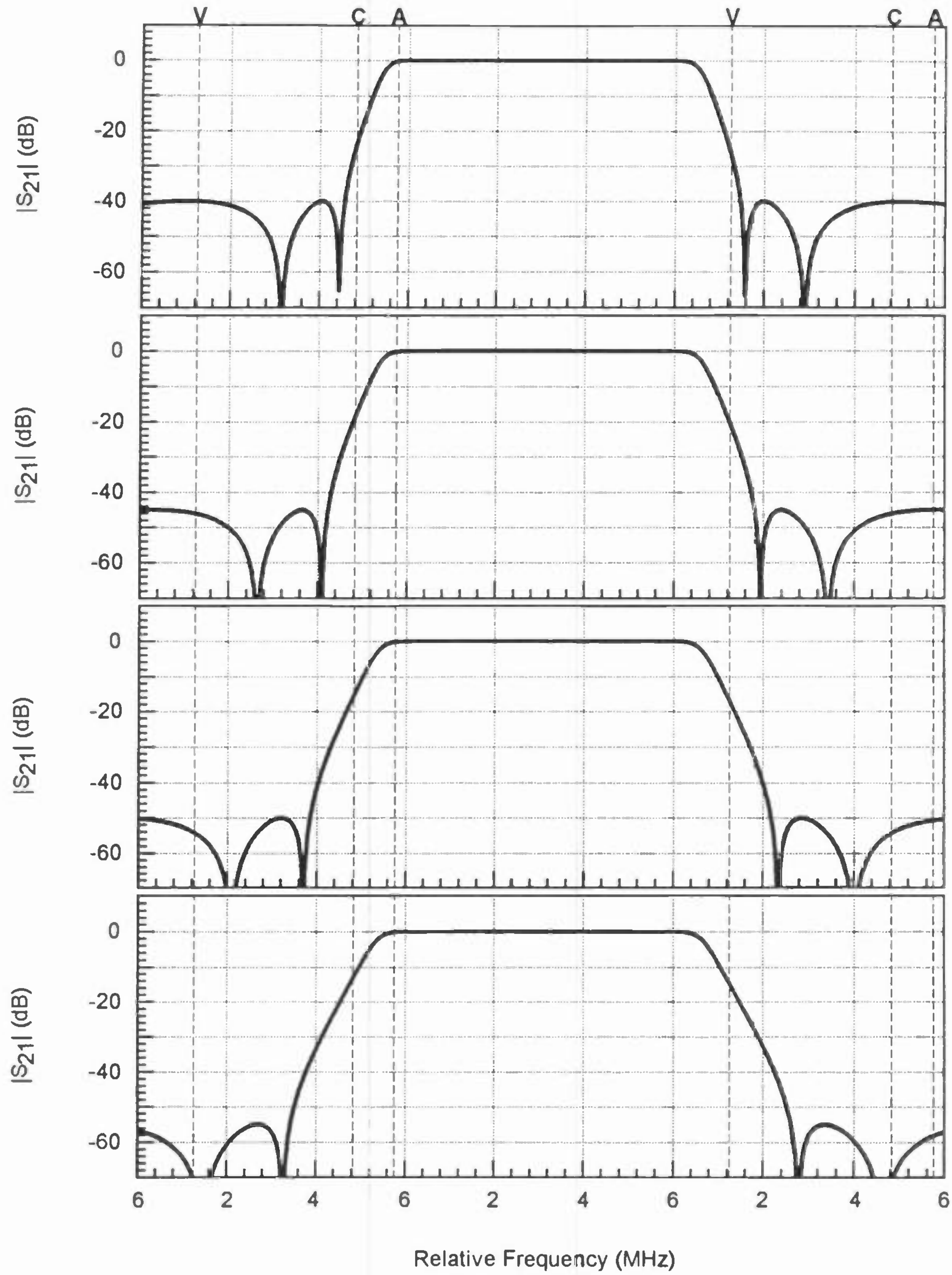

FIGURE 11 Insertion loss vs. relative frequency for elliptic function filters of order 6 with .002 dB passband ripple and minimum stopband attenuation levels ranging from 40 to 55 dB.

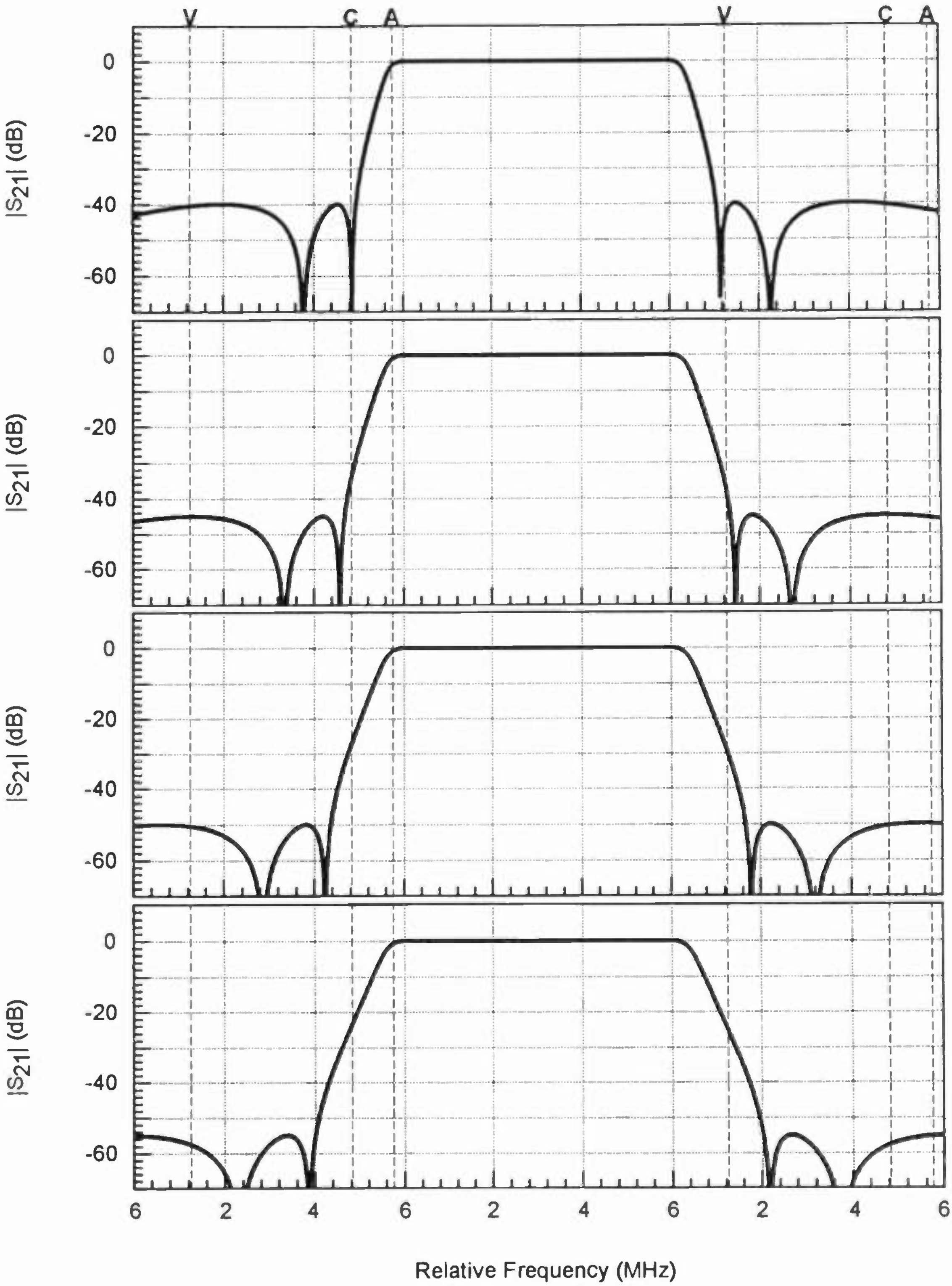

FIGURE 12 Insertion loss vs. relative frequency for elliptic function filters of order 6 with .010 dB passband ripple and minimum stopband attenuation levels ranging from 40 to 55 dB.

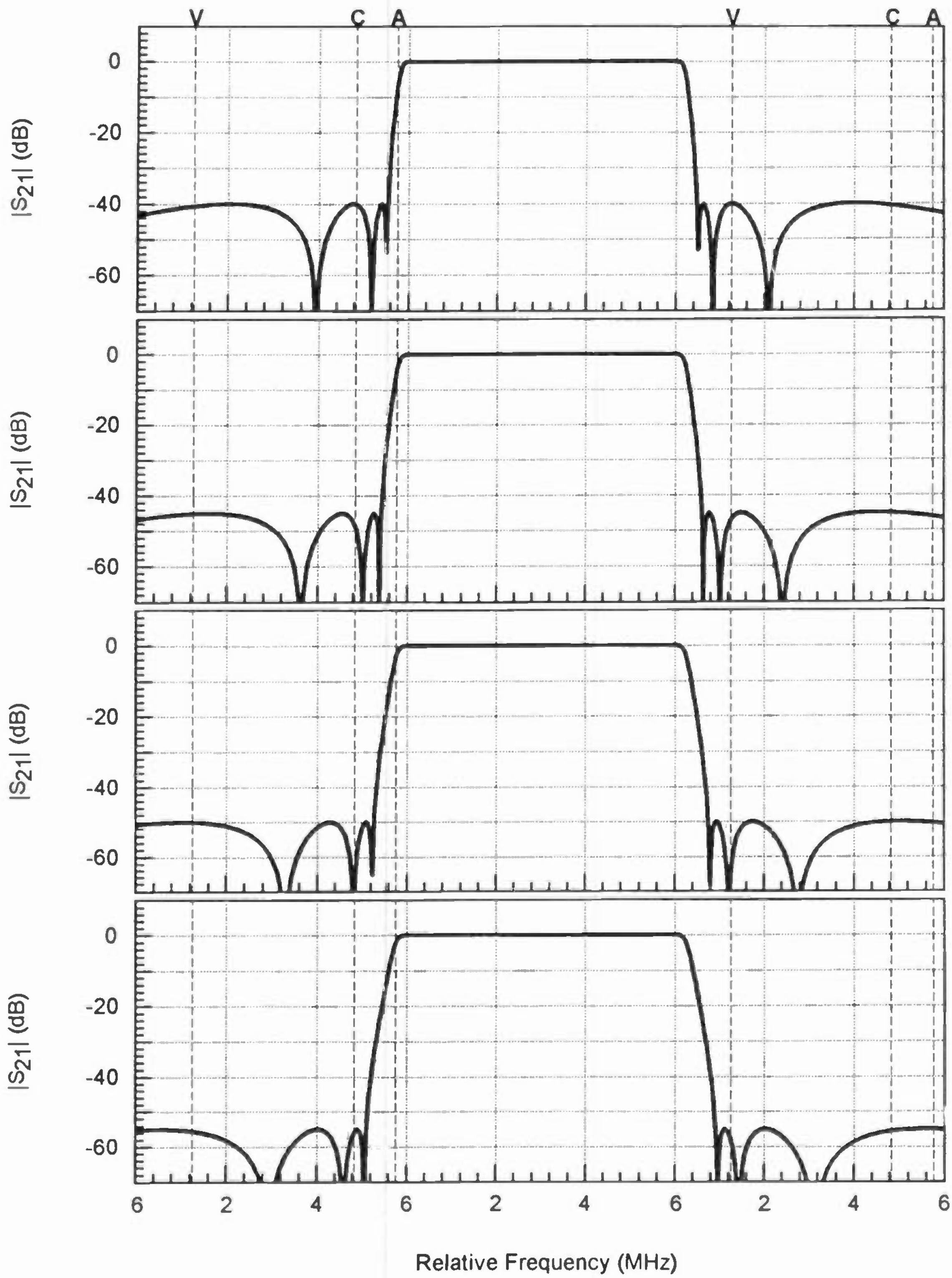

FIGURE 13 Insertion loss vs. relative frequency for elliptic function filters of order 8 with .002 dB passband ripple and minimum stopband attenuation levels ranging from 40 to 55 dB.

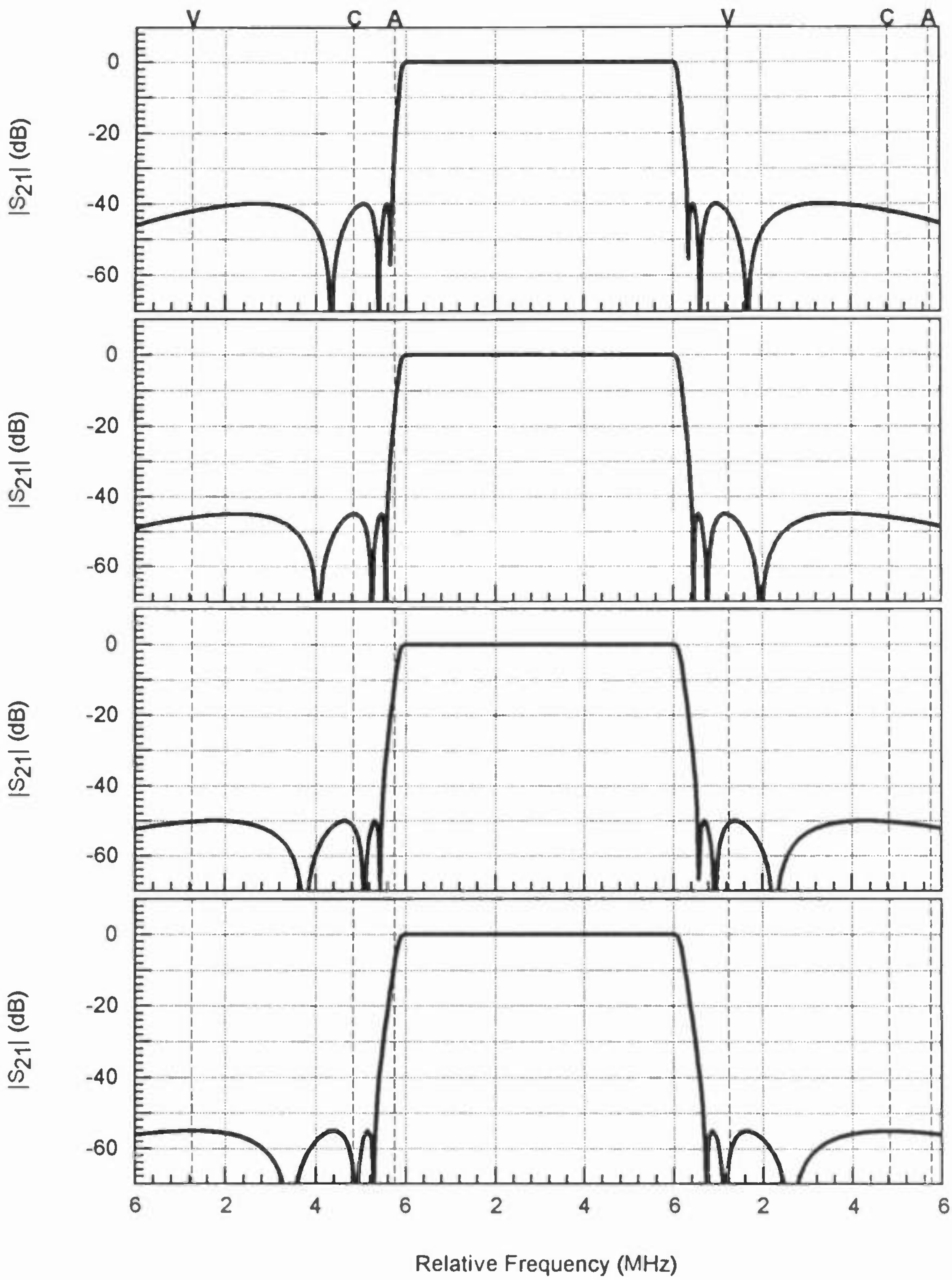

FIGURE 14 Insertion loss vs. relative frequency for elliptic function filters of order 8 with .010 dB passband ripple and minimum stopband attenuation levels ranging from 40 to 55 dB.

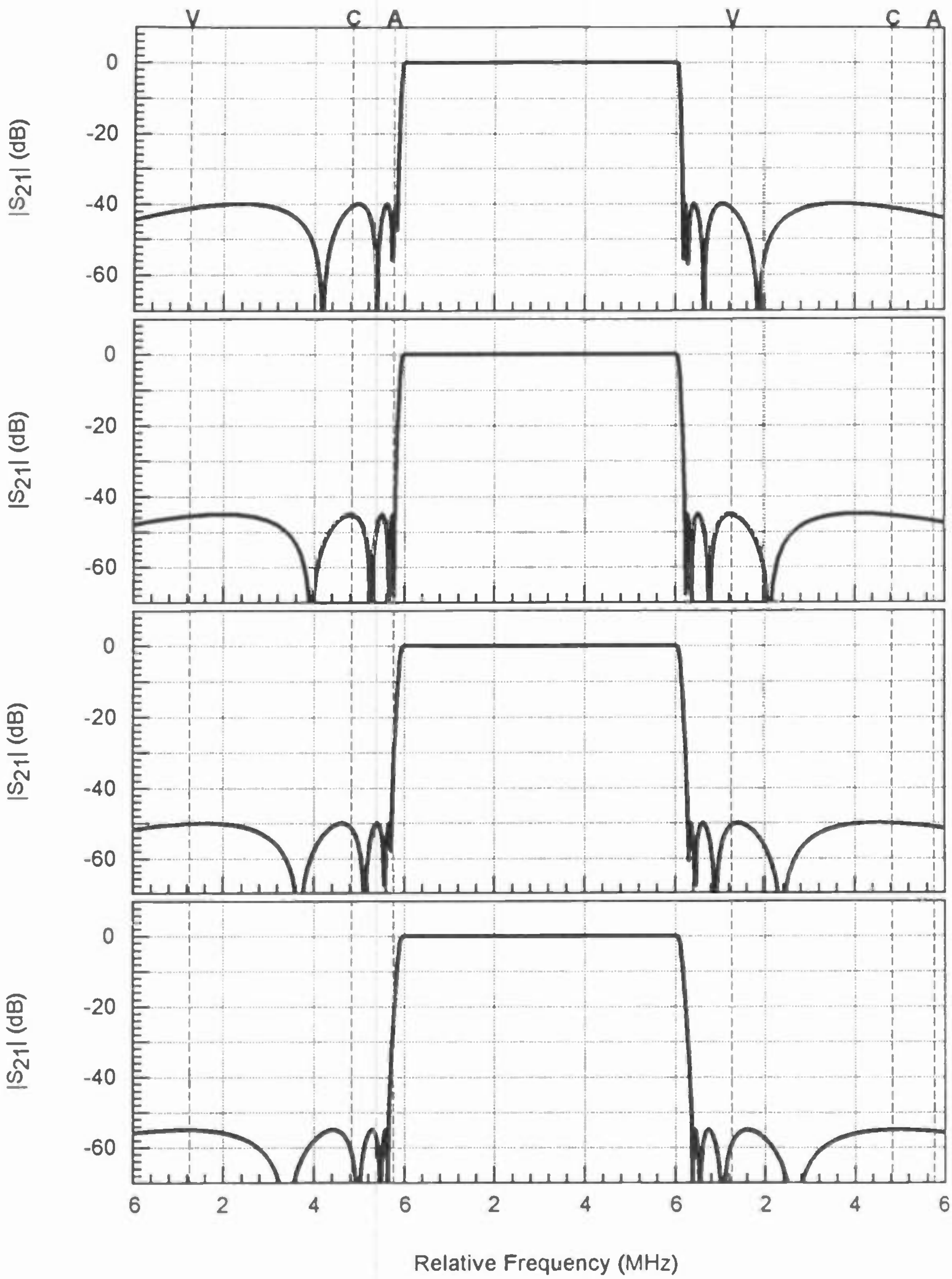

FIGURE 15 Insertion loss vs. relative frequency for elliptic function filters of order 10 with .002 dB passband ripple and minimum stopband attenuation levels ranging from 40 to 55 dB.

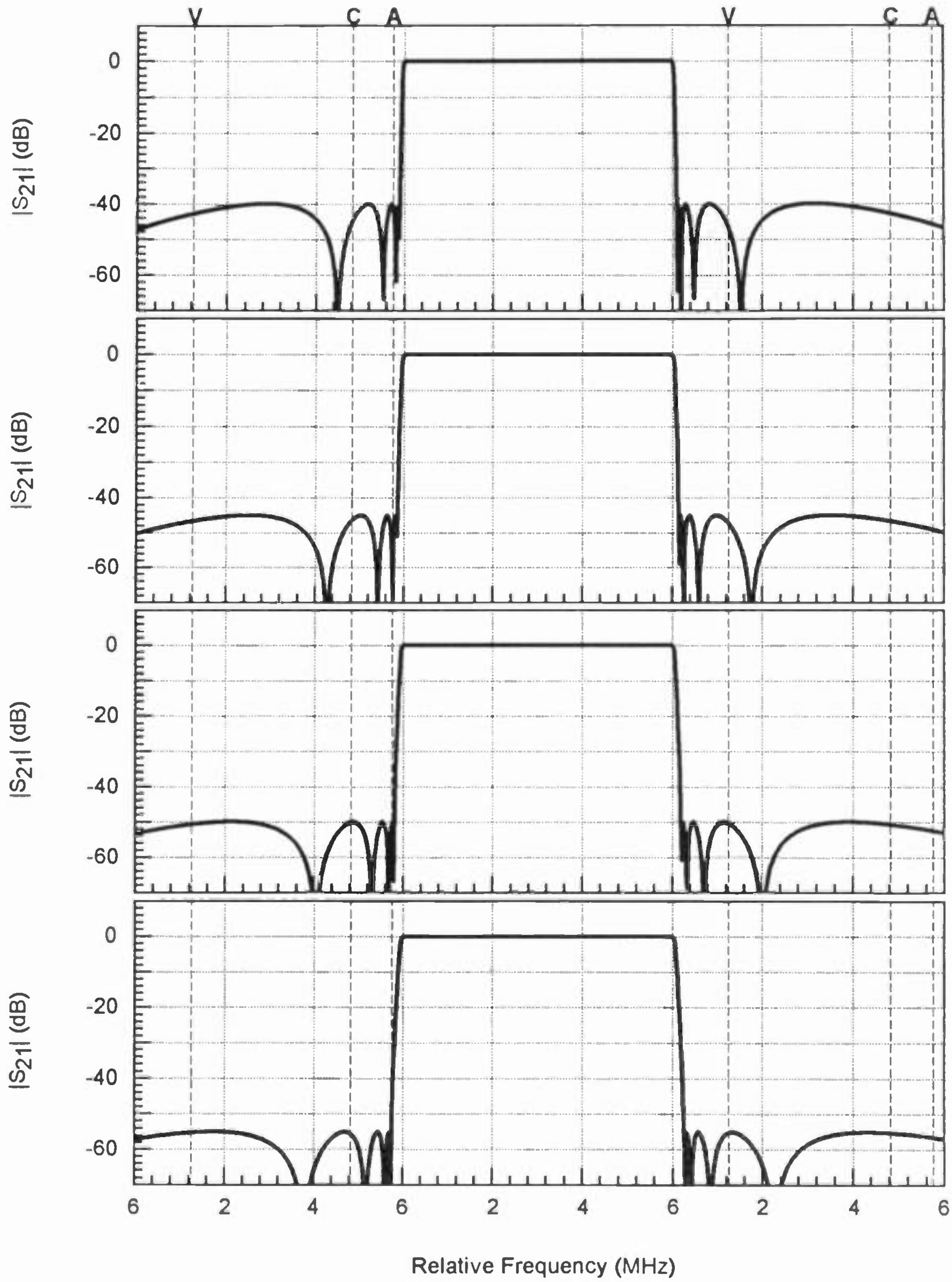

FIGURE 16 Insertion loss vs. relative frequency for elliptic function filters of order 10 with .010 dB passband ripple and minimum stopband attenuation levels ranging from 40 to 55 dB.

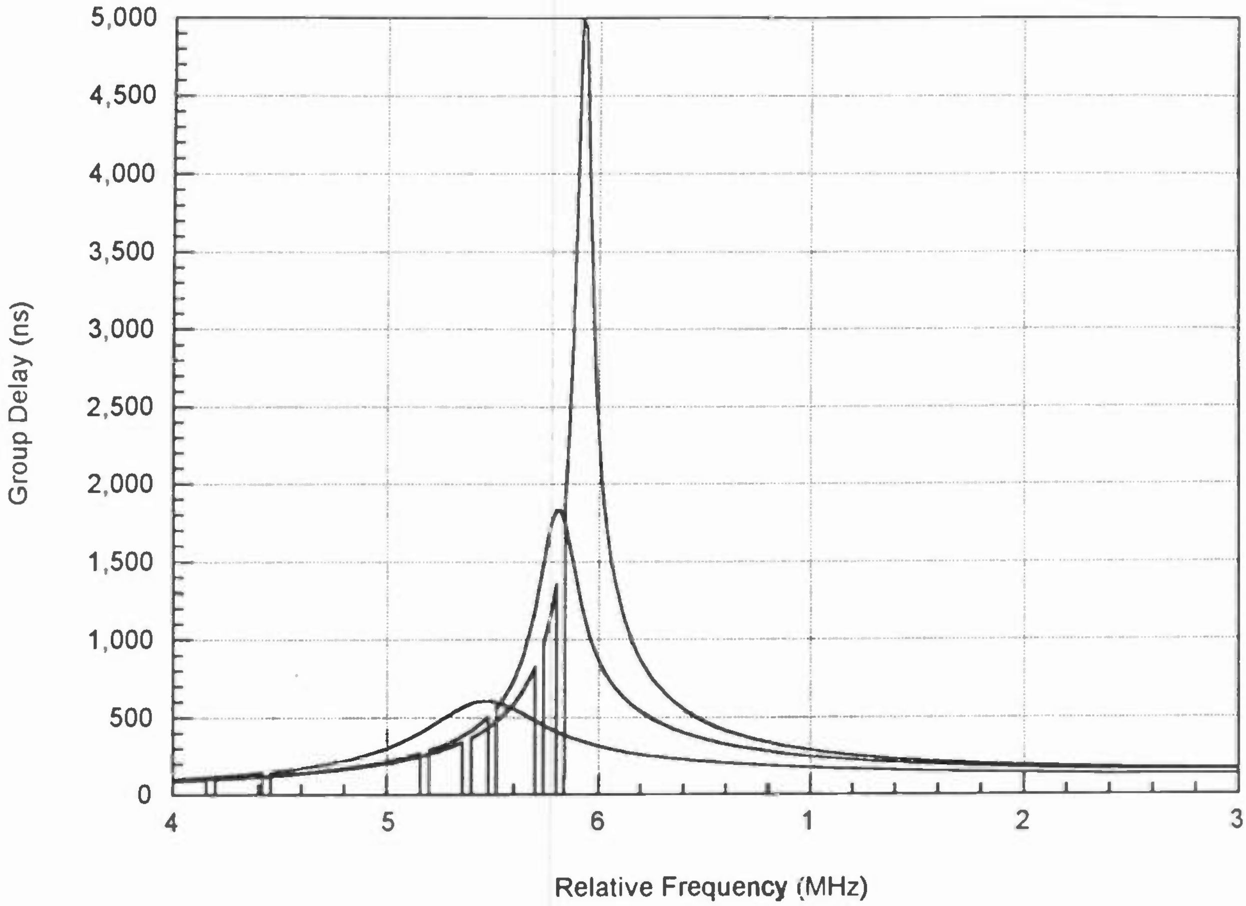

FIGURE 17 Group delay vs. relative frequency for 6 MHz passband elliptic filters of orders N = 6, 8 and 10 having a passband ripple level Rp = .002 dB and minimum stopband attenuation (ripple) level of Rs = 40 dB. The frequencies depicted represent the upper 2 MHz of the lower adjacent channel and the lower 3 MHz of the main (ATV) channel.

Discussion

In a comparison between Chebyshev and elliptic filter responses, it is readily seen that the rolloffs of the elliptic filter response functions far exceed those of equivalent order Chebyshev filters. Increased skirt steepness is certainly appealing when considering out of band rejection alone. Sharp filtering performance, however, comes at a price of complexity in manufacturing. Tables 2 and 3 list critical performance parameters of the Chebyshev and elliptic filter responses that are considered in this paper. If Chebyshev and elliptic responses of equal skirt steepness are compared, the group delay variations are shown to be roughly equivalent. Therefore, the comparison must be extended to include insertion loss, cost and attenuation at frequencies that are well into the stopband, where the Chebyshev response provides increasing attenuation.

		Attenuation (dB) at Critical Frequencies			Group Delay Variations (ns)	
Order	Minimum Stopband Attenuation (dB)	n-1 Chroma	n-1 Aural	n+1 Visual	Peak-Fc	Fl-Fc
.002 dB Passband Ripple						
6	40	23	0	26	473	185
	45	18	0	20	414	167
	50	15	0	16	368	151
	55	13	0	14	334	139
8	40	40	9	40	1669	713
	45	50	6	47	1420	629
	50	64	4	70	1230	561
	55	55	3	57	1083	505
10	40	41	55	41	4816	2185
	45	45	45	45	4064	1867
	50	51	43	50	3407	1620
	55	62	29	59	2895	1425
.010 dB Passband Ripple						
6	40	55	1	47	625	317
	45	34	1	37	542	283
	50	27	1	29	481	256
	55	23	1	25	433	233
8	40	42	28	42	2275	1207
	45	45	16	45	1903	1040
	50	52	13	51	1622	910
	55	68	10	63	1412	808
10	40	44	41	46	7053	3846
	45	46	56	47	5622	3167
	50	50	52	50	4575	2668
	55	56	87	55	3925	2292

Table 2. Critical performance parameters of elliptic-function filters. Group delay variations have been calculated from the center frequency (Fc) of the ATV channel to the frequency at which the group delay is at its peak as well as to the ATV channel's lowest frequency (Fl).

		Attenuation (dB) at Critical Frequencies			Group Delay Variations (ns)	
Ripple (dB)	Order	n-1 Chroma	n-1 Aural	n+1 Visual	Peak-Fc	Fl-Fc
.002	5	2	0	2	112	41
	6	6	0	7	189	78
	7	13	0	14	289	128
	8	20	0	22	410	191
	9	27	1	29	556	270
	10	35	2	37	717	361
.010	5	6	0	7	145	73
	6	13	0	14	242	130
	7	20	1	21	364	201
	8	27	2	29	510	292
	9	34	3	36	683	398
	10	42	5	44	874	522
.021	5	9	0	10	166	97
	6	16	1	17	274	165
	7	23	1	24	408	252
	8	30	3	32	574	360
	9	38	5	40	758	483
	10	45	8	47	982	629
.036	5	1	1	12	187	119
	6	18	1	19	303	201
	7	25	2	27	447	299
	8	33	4	34	624	424
	9	40	7	42	831	566
	10	47	10	50	1060	726

Table 3. Critical performance parameters of Chebyshev-function filters. Group delay variations have been calculated from the center frequency (Fc) of the ATV channel to the frequency at which the group delay is at its peak as well as to the ATV channel's lowest frequency (Fl).

Conclusions.

The choice between Chebyshev and elliptic filter responses for ATV applications may ultimately hinge upon how much pulse distortion can be "pre-corrected" for in the transmitter. It may also be found that notch filters will be a preferred method to protect the lower adjacent channel's chroma and aural subcarriers as well as the upper adjacent channel's visual carrier. This would ease the requirement for ultra-steep skirt bandpass filters. An informed decision as to what the best filter configuration will be for ATV should probably be preceded by a thorough investigation of filter-induced pulse distortion and pre-correction options.

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[1] "Development of a High Definition Television (HDTV) Terrestrial Broadcasting Emission Mask", Carl G. Eilers, IEEE Transactions on Broadcasting, Vol. 41, No. 4, December, 1995.

[2] "Proposal for ATV Out of Band Radiation Mask", Robert J. Plonka, Harris Corporation Broadcast Products, Jan 4, 1995.

[3] "Channel Combining in an ATV/NTSC Environment", Dennis Heymans, 50th Broadcast Engineering Conference, NAB '96.

AN INDUCTIVE OUTPUT TUBE AND CIRCUIT OPTIMIZED FOR DIGITAL ATV

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ABSTRACT

The Inductive Output Tube (IOT) is now used worldwide as the final amplifier in u.h.f. analogue TV transmitters. The transfer characteristics of this class of tube are inherently more linear than those of a klystron and this is of particular value for the coming digital ATV services. A low power 20 kW IOT has now been developed and its performance when amplifying a digital OFDM signal has been measured. This paper describes the tube, summarises its analogue performance, and reports and discusses the results of the digital tests. These demonstrate that the IOT should be an excellent amplifier for any of the currently proposed digital terrestrial television transmission standards.

INTRODUCTION

The first high power u.h.f. TV transmitter incorporating EEV IOTs went into service in 1991. Since then the installed base has become large and worldwide and continues to grow at an impressive rate. Presently IOT transmitters are installed in 18 countries, over a hundred further systems have been ordered and well over 3 million hours of operational life have been achieved. The two key features of an IOT on which this success has been established are its very high Figure of Merit (FOM) and its excellent linearity. These enable very efficient transmitters to be operated in a common amplifier mode – a FOM of about 100% can be obtained when amplifying signals in accordance with the system M (NTSC) standards.

The first IOT to enter service was the IOT7340, rated to give a peak sync. power of up to 45 kW in vision-only service and a peak envelope power (PEP) of up to 56 kW in common amplifier mode. This corresponds to a peak sync. power of 32 kW for a mono sound system having a 10:1 vision to aural power ratio. The IOT7340 was closely followed by the IOT7360, rated at 64 kW peak sync. vision and 75 kW PEP. Some early problems were encountered – these problems and their solutions were fully described and discussed at NAB '95[1]. Experience since then has been excellent – presently there are 40 IOT7360s and IOT7340s in service with lives over 15 000 hours.

Tube Type	Cooling Method	Peak Sync. Output (kW)[A]	Common Amplifier Mode PEP (kW)	Common Amplifier Mode Peak Sync. (kW)[B]
IOT7320R	Air only	25	40	23
IOT7330R	Air only	35	40	23
IOT7330	Air/Liquid	35	40	23
IOT7340R	Air only	45	56	32
IOT7340	Air/Liquid	45	56	32
IOT7360	Air/Liquid	64	75	43

Note A: Vision-only operation
Note B: Peak sync. power corresponding to the PEP, assuming a 10:1 vision to aural power ratio.

Table 1. Range of IOTs

Due to the undoubted success of this technology, both users and transmitter manufacturers have encouraged EEV to extend the power range of IOTs. The full range of tubes is shown in Table 1. The major features of the design of the lower power tube, designated IOT7320R, were described at NAB '95.

The future of television transmission is digital. Analogue television transmission systems will soon be superseded by new digital services offering better quality, wider viewer choice and new multimedia applications. The dawn of the new digital television age has already broken with the launch of digital satellite services, such as DirecTV in the USA. Digital terrestrial television broadcasting is more difficult to achieve since the channel impairments suffered by a digital terrestrial television signal are much greater than those suffered by satellite signals. However, practical implementations of digital terrestrial television are rapidly approaching as the European and American standardisation processes near completion. Digital terrestrial television at consumer prices has been made possible by the rapid advances in semiconductor technology which mean that extremely complex digital signal processing circuit designs can now be produced in high volumes. As the beginning of digital transmission approaches (pilot systems are expected in the UK later this year), it is becoming increasingly important to design television transmitters optimised for the new digital services.

Hence an important reason for the development of the IOT7320R tube was its potential capability for use for amplifying digital ATV signals. This paper reviews its design, outlines its analogue performance and presents and discusses the results of digital tests on the tube. These tests were performed using an experimental OFDM DTV modem designed and manufactured by DigiMedia Vision under contract to the Independent Television Commission[2].

IOT7320R: DESIGN AND ANALOGUE PERFORMANCE

A new IOT system specifically designed for lower power analogue u.h.f. TV transmitters was described in detail at NAB '95. Since then, a number of changes have been made.

Firstly, the inter-cavity coupling adjustment arrangement between the primary and secondary output cavities has been modified. The inter-cavity coupling transition has been shortened so that the width of the whole system has been reduced and it can now be installed in a 19-inch rack.

Secondly, the input cavity has been changed to a slotted cavity type (Figure 1). The tuning is effected by manually adjusting the lower ring which is attached to the tuning door to the required position. The r.f. connection ring, the upper ring, is positioned for optimum match. With this system, the whole frequency range (470 to 860 MHz) can be covered by a single mode

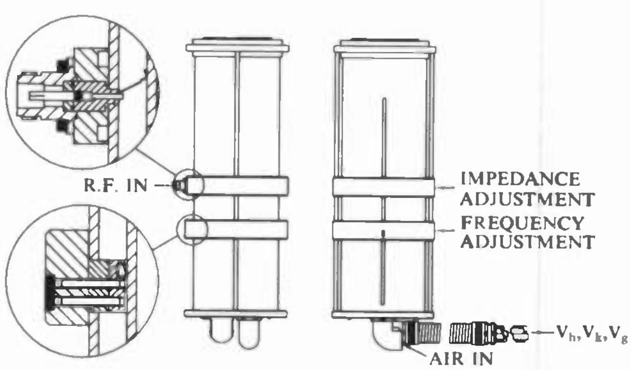

Figure 1. Slotted Input Cavity Design

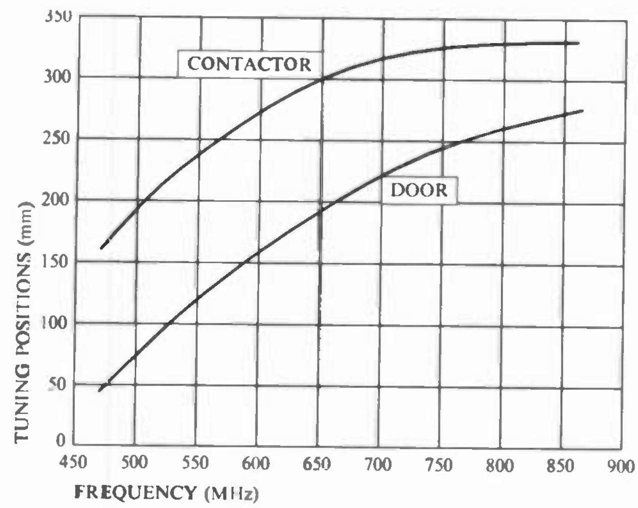

Figure 2. Input Tuning Curves for IOT7320R

(see Figure 2) and there is no requirement to have a double-slug matching section. Test results for IOT7320R serial number 12 when operated in common amplifier mode at European channel 43 are shown in Table 2.

Peak sync. vision output	24.6	18.5	12.8	kW
Aural output power	2.46	1.85	1.28	kW
Peak vision drive power	115	78	70	W
Aural drive	10	6	5	W
Beam voltage	25.9	23.9	21.9	kV
Beam current (mid-grey)	0.84	0.75	0.72	A
Beam current (d.c.)	0.47	0.46	0.49	A
FOM	124.4	113.5	89.3	%
L.F. linearity	11	8	10	%
ICPM	2	2	2	°
Differential phase	1	1	1	°
Differential gain	8	7	9	%

Table 2. Performance of IOT7320R at European channel 43

The gain of the tube is in excess of 22 dB and the FOM at the highest power level is 124%, falling to 89% at the lowest power level. The linearity characteristics of the tube – values of various parameters are given in the table – are good. The FOM of the tube over the u.h.f. band is shown for various power levels in Figure 3, and Figure 4 shows the variation of gain with frequency. The values of l.f. linearity and differential gain vary over the frequency range – the largest values measured were 22% and 15% respectively at European channel 69 (vision carrier frequency 855.25 MHz). Both ICPM and differential phase remained less than 5° throughout.

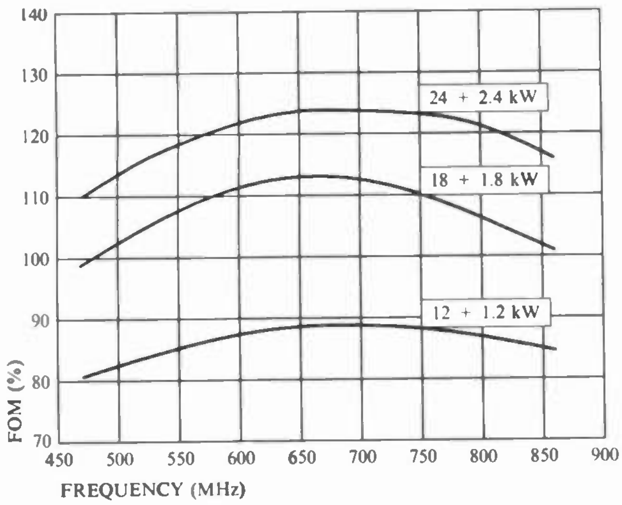

Figure 3. FOM versus Frequency for IOT7320R

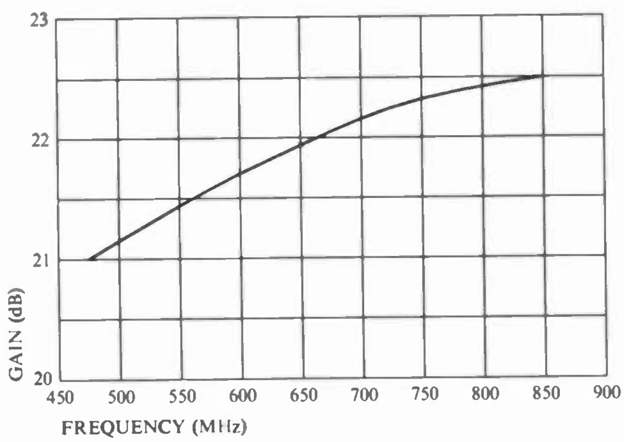

Figure 4. Gain versus Frequency for IOT7320R

A parameter of particular importance to the broadcaster is the intermodulation product (IP) of the system. This topic must be treated with care as measured values vary appreciably depending upon the test method being used. A variety of test signals has been employed. For example, the vision carrier may be amplitude modulated with a constant picture level at mid-grey, or at −10 dB or −4 dB with respect to peak sync. Alternatively, a ramp may be employed. Various colour sub-carrier power levels may also be used. Some test signals are shown in Table 3.

Test Signal	Vision Carrier Picture Level	Colour Sub-carrier Level	Aural Level
1	Mid-grey	−17 dB	−10 dB
2	−10 dB	−20 dB	−10 dB
3	−4 dB	−20 dB	−10 dB
4	Ramp	−26 dB	−10 dB

Table 3. IP Test Signals

Using these various signals, the measured IPs of IOT7320R serial number 12 when operated at European Channel 43 at the 24 kW power level are shown in Table 4. The only correction used was that necessary to correct the sync. pulse height.

	IP Frequency (MHz)		
Test Signal	**f_v − 1.07**	**f_v + 1.07**	**f_v + 6.57**
1	−52	−50	−58
2	−57	−54	−58
3	−52	−51	−60
4	−57	−56	−60

Note: f_v is the vision carrier frequency.

Table 4. Values of Intermodulation Products (dBc)

These results clearly show the variation in measured values which can be obtained using different test signals. The intermodulation product of the tube is a function of the non-linearity of the transfer characteristic. The performance of the tube when amplifying digital signals is also strongly related to its linearity and to its ability to handle the short duration high power transients present in digital terrestrial television signals.

DIGITAL SIGNALS AND THE EFFECTS OF AMPLIFICATION

Digital terrestrial television signals exhibit a much larger dynamic range than that of an analogue television signal. This is because digital TV signals, like noise, have approximately gaussian amplitude distributions which means they have high peak-to-mean power ratios. In order to accommodate the larger dynamic amplitude range of the digital signals without distortion, High Power Amplifiers (HPAs) may need to be 'backed-off' (i.e. run to a lower average power), compared with their analogue TV rating, to ensure that they are operating linearly. A power amplifier also

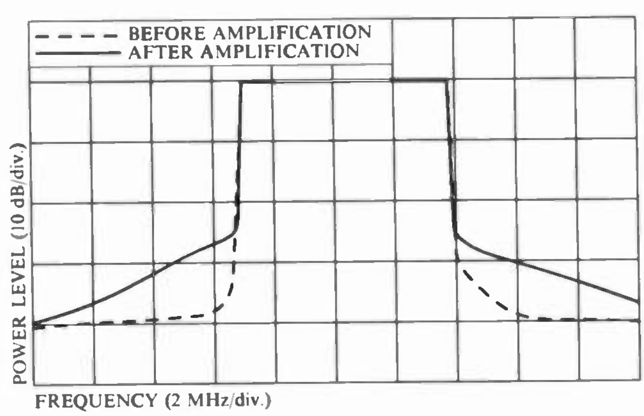

Figure 5. Comparison of Digital TV Signals, showing the effects of Amplification

needs to be operated at an average power level that ensures sufficient headroom for the high power peaks.

When a digital signal is passed through any HPA system it will be degraded due to the non-linear transfer characteristic of the amplifier. As the output power of the HPA is increased, the degradation due to the transfer characteristic becomes more evident. This degradation may be observed in two ways:

Firstly, the sidebands outside the actual digital signal spectrum start to rise out of the noise floor. Figure 5 shows two digital TV signal spectra superimposed. The first is the signal before amplification and the second shows the effect that amplification has on the sidebands of the transmitted signal. Secondly, as the signal degrades, the Bit Error Rate (BER) of the received digital signal increases. Both effects are caused by non-linearity which generates Intermodulation Products (IPs). In-band IPs increase the residual BER, and out-of-band IPs increase the sideband level.

Proposed Digital Systems

Presently, there are two alternative digital systems proposed for digital terrestrial television broadcasting:

1. Orthogonal Frequency Division Multiplexing

The OFDM concept[3] which has been adopted for the transmission of digital terrestrial television in Europe, and in a number of other areas, is based on spreading the data to be transmitted over a large number of orthogonal carriers, each being modulated at a low bit rate. In a conventional frequency division multiplex, the carriers are individually filtered to ensure there is no spectral overlap. There is therefore no inter-symbol interference between carriers but the available spectrum is not used with maximum efficiency. If, however, the carrier spacing is chosen so that the carriers are orthogonal over the symbol period (i.e. the product of any two carriers integrated over the period equals zero), then symbols can be recovered without interference, even with a degree of spectral overlap. For maximum spectral efficiency, the carrier spacing equals the reciprocal of the symbol period. The orthogonal multiplex of carriers may be conveniently generated digitally using the inverse Fast Fourier Transform (FFT) process.

The European DVB digital terrestrial standard defines two modes of operation, one based on 2048 (2k) carriers and one based on 8192 (8k) carriers. The active carriers carry either data or synchronisation information. The active carriers are either QPSK, 16 QAM or 64 QAM modulated. The system allows a trade-off between data rate and transmission ruggedness; data rates range from 4.98 to 31.67 Mbit/s in an 8 MHz channel.

2. 8-Level Vestigial Sideband ATV System

This digital terrestrial transmission system, which is favoured for adoption in the USA, is essentially a single carrier transmission system. A single pilot signal is provided to allow frequency synchronisation. The transmitted data is split into data frames. It incorporates a 4-symbol segment sync. and a blind equalisation technique is used to compensate for channel impairments.

The 8-VSB system proposed for terrestrial broadcasting has one mode of operation defined. The cable transmission system will be 16-VSB. The modulation format is 8-VSB, symbol rate 10.76 Msymbols/s, giving a payload data rate of 19.29 Mbit/s in a 6 MHz bandwidth.

MEASUREMENTS OF AN OFDM SYSTEM APPLIED TO THE IOT7320R

HPA Reference Point

To make comparisons between different transmitter systems it would be advantageous to compare the rated average OFDM power of an amplifying tube or solid-state amplifier unit with a common reference point that could be defined for every amplifier. One such reference point could be the amplifier's 1 dB power compression point. However, in practice this data does not appear to be generally available for transmitter HPAs. Therefore another suitable reference point is required.

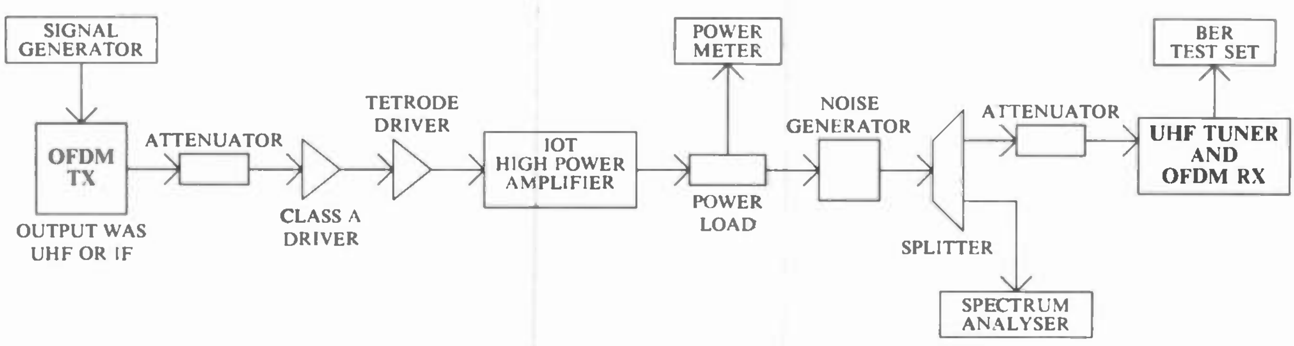

Figure 6. U.H.F. Power Amplifier Measurement System

The peak envelope power (PEP) which is defined for analogue systems appears a more readily available, useful figure. When an amplifier is normally operated in common amplification mode with a Visual:Aural ratio of 10:1, the PEP is a factor of 1.73 (about 2.5 dB) higher than the peak sync. level. In this paper the PEP has been used as a reference point against which the HPA back-off is determined. The IOT7320R amplifier system tested has a peak sync. vision rating of 22 kW when used in a common amplifier mode at a beam voltage of 26 kV. This corresponds to a 38 kW PEP rating. The PEP rating corresponding to a beam voltage of 22 kV is 25 kW.

Measurement Procedure

Figure 6 shows the system configuration that was used to make the u.h.f. power amplifier measurements. The class A solid-state power amplifier used to drive the tetrode pre-amplifier was tested without pre-correction. The attenuator after the OFDM modulator was used to vary the drive level into the HPA and hence vary its output power. The noise generator was placed after a calibrated probe from the power load and allowed the introduction of a defined gaussian signal-to-noise level so that BER v. S/N ratio measurements could be made. The second attenuator before the u.h.f. tuner was used to maintain a constant input level into the tuner as the HPA output power was changed.

The calibration for the S/N power measurements was made using the spectrum analyser. The S/N ratios quoted are thus equivalent to signal power density/noise power density. The BER measurements were made using a BER test set which measured the time interval for a given number of individual bit errors in a Pseudo Random Binary Sequence (PRBS). The PRBS chosen was a $2^{23}-1$ sequence used for telecommunications BER testing; the sequence is internally generated in the modulator. No error coding was applied to the data so uncoded BER curves could be constructed.

The measurements were made without inner error correction coding. If this had been employed the transmitter would have had to be driven well beyond its normal peak power rating in order to see any significant change in the S/N ratio corresponding to a BER of 10^{-4}, after decoding the inner Viterbi error correction code.

Back-to-back u.h.f. measurements were initially made without either the tetrode or the IOT present in the system. The S/N curve obtained was the control reference measurement to compare against the system performance with the HPA.

The procedure for the 'loss of noise margin' sensitivity test is defined below:

1. A reference S/N ratio for a BER of 10^{-4} was obtained at a well backed-off power level corresponding to just the class A solid-state pre-amplifier present in the system.
2. With the HPA in circuit, BER curves were plotted against S/N ratio at different output power levels. The S/N value for a given BER of 10^{-4} was obtained from these curves.

The tetrode driver amplifier was also characterised in isolation from the IOT using the above procedure.

The OFDM System Configuration

The OFDM signal used during the measurements was a 432 carrier multiplex with no spectral holes. Each of the 432 carriers was 16 QAM modulated. The bandwidth of this configuration is 6.75 MHz. The raw data rate is 26.18 Mbit/s.

IOT Power Amplifier Configuration

The IOT transmitter on test comprised the IOT7320R in its IM7320R r.f. circuit, driven by a 1 kW peak sync. vision-only rated tetrode pre-amplifier which was itself driven by a class A solid-state 30 W peak sync. linear amplifier. Initially, the driver stages were tested in isolation from the IOT system so that their effect on the total system performance could be determined. Measurements were made using a u.h.f. modulated source from the OFDM modem, with no pre-correction. The IOT amplifier has two main variable parameters: the beam voltage and grid bias level which controls the IOT 'idle current'. For this investigation only the beam voltage was varied. Two voltages were tested: 22 kV and 26 kV.

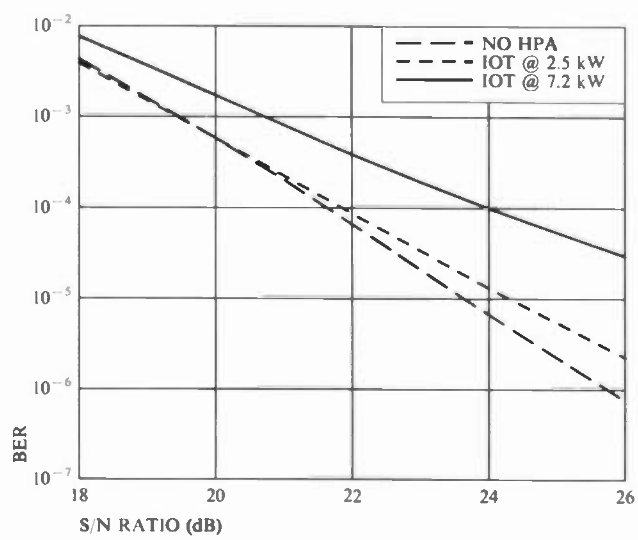

Figure 7. BER versus S/N Ratio at 22 kV Beam Voltage

Results of Digital Tests

22 kV Operation

The total transmitter system was tested with the IOT beam voltage set to 22 kV. The output power from the system was varied from 2.5 kW up to 7.2 kW average OFDM power. BER v S/N curves were taken at three different power levels, two of which are plotted on Figure 7. At 7.2 kW from the IOT, the (1 kW peak sync.) tetrode driver needed to produce only 90 W of average OFDM power. A BER v S/N curve for the drive alone was measured at this power and is also plotted on Figure 7. The tetrode introduces 0.8 dB degradation at a BER of 10^{-4}.

The characteristic 'tail' due to non-linearity is beginning to be noticeable on the IOT curve at low BERs at the 7.2 kW power level. The sideband performance of the OFDM signal was recorded at the different IOT power levels, and showed increasing sideband level as the average signal power increased.

At 7.2 kW the sidebands were 26 dB with respect to the main OFDM carriers. This may not be adequate for high power operation. However, the application of appropriate correction in the system would significantly reduce the sideband level. Previous experience suggests improvements in the order of 10 to 15 dB can be realised.

Peak OFDM power was measured using an HP8991 peak power meter. Table 5 summarises the 22 kV performance.

IOT Output Power	**2.5**	**5.1**	**7.2**	**kW**
Back-off w.r.t PEP	10.0	6.9	5.4	dB
S/N degradation @ 10^{-4}	0.26	0.74	2.28	dB
Sideband	28	27.5	25.8	dBc
Peak power	18.1	25.4	28	kW
Peak-mean ratio	8.6	7	5.9	dB

Table 5 Summary of 22 kV IOT Performance

26 kV Operation

The total transmitter system was then tested with the IOT beam voltage set to 26 kV. The output power from the system was varied from 7.4 kW up to 14.5 kW average OFDM power. BER v S/N curves were taken at four different power levels, two of which are plotted on Figure 8. The sideband performance and peak output power measurements are all summarised in Table 6.

IOT Output Power	**7.4**	**12.1**	**14.5**	**kW**
Back-off w.r.t. PEP	7.1	5	4.2	dB
S/N degradation @ 10^{-4}	0.7	1.9	4.4	dB
Sideband	28	24.9	23.6	dBc
Peak Power	39.75	42.6	47.86	kW
Peak-Mean ratio	7.3	6.1	5.2	dB

Table 6 Summary of 26 kV IOT Performance

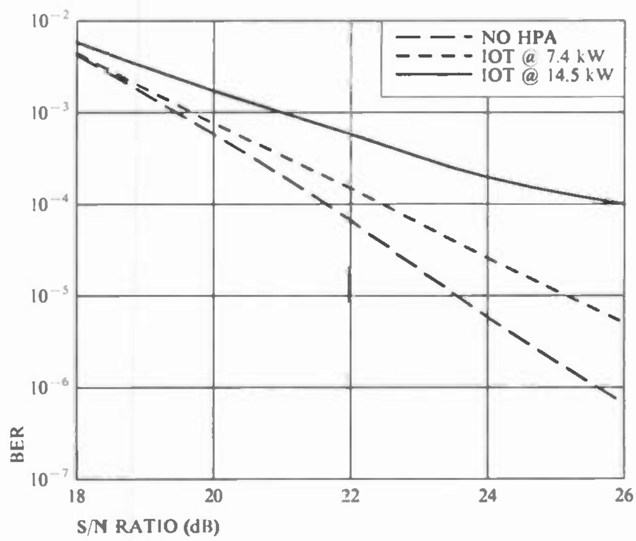

Figure 8. BER versus S/N Ratio at 26 kV Beam Voltage

DISCUSSION

As the residual bit error rate (BER) inherent in a transmitted digital ATV signal increases due to HPA non-linearity, so the capability of a consumer receiver's error correction subsystem to deal with channel impairments, such as multipath reception and noise, decreases. This is because more of the error correcting capability is employed dealing with transmitter non-linearity effects.

Therefore the BER due to the HPA ought to be kept to an acceptably low figure. This is the case with all the power levels measured on the IOT, with a possible exception at the 14.5 kW OFDM average power level. None of the other measured BER figures due to the HPA non-linearity would significantly affect the performance of a consumer's receiver.

The digital ATV signal-to-sideband ratio is important. If digital ATV is to be transmitted at equivalent average powers to the analogue services of today, the worst case power spectral density of the measured sidebands would only be about 26 dB lower than the in-band power spectral density of the digital ATV signal in the adjacent channels. The worst case out-of-band sidebands produced by the digital signal would therefore be equivalent to a S/N ratio of about 28 to 34 dB in the analogue signal.

This may or may not be acceptable. However, it must be remembered that no correction was used in the system that was tested. If a suitable correction system had been employed, a significant improvement in out-of-band sideband noise (probably between 10 and 15 dB) could be anticipated.

It might still be necessary to provide some additional high power filtering. However, more work is required to determine whether or not this is required. The asymmetry of the sidebands from the IOT suggest that the tuning of the IOT cavity required for an analogue system may not be optimum for an OFDM signal.

As expected, with increasing output power, the peak-to-mean ratio decreases as more and more limiting occurs in the IOT. However, with the IOT operating at a mean power of 14.5 kW the measured peak powers produced by the IOT were up to 47.9 kW. This is well above the vision-only peak sync. PAL rating of the IOT7320R tube and its 38 kW maximum peak envelope power rating. It must also be noted that although a back-off ratio of about 6 dB looks to be required for an OFDM signal the back-off ratio compares the **peak** envelope power of an analogue signal with the **mean** power of the digital signal. If the back-off ratio calculation used the ratio between mean analogue power and mean digital power, then the quoted back-offs would only be about 1 or 2 dB.

The results indicate that the 20 kW IOT7320R transmitter system can satisfactorily amplify the OFDM signal. The IOT has an inherently linear transfer characteristic and is capable of producing peak powers much in excess of its average power rating. These characteristics make it ideal for amplifying high power digital signals.

The absence of a suitable power reference point for HPAs makes comparison of HPAs difficult. A new set of standards and comparisons needs to be defined for evaluating digital transmitters. These will probably involve the definition of new digital test signals.

The work reported here has demonstrated the suitability of the IOT for amplifying digital signals. Since the digital tests were done, further modifications have been made to the input cavity resulting in IPs improved by about 5 dB to the values given in Table 4. Since this parameter is a measure of the linearity of the tube, it is expected that the digital performance will be somewhat better than that reported above.

These digital tests were done using an OFDM signal. However, there is no doubt that the tube is also very suitable for amplifying 8-VSB signals – an IOT was used in the successful ATV field trials in Charlotte, NC[4].

Additionally, theoretical work has been reported[5]. A computer model was used to simulate 8-VSB signal generation, an IOT transmitter transfer characteristic and receiver data detection functions. The model estimated the effects of both amplitude and phase non-linearity on symbol error rates and out-of-band noise. It was concluded that satisfactory performance can be obtained provided that low-level curvature of the transfer characteristic is avoided and the mean digital power level is no more than −8 dB below the IOT peak output capability. It was also concluded that with correction the mean digital power level can be increased by 2 to 3 dB whilst maintaining an acceptable symbol error rate performance and a 40 dB ratio of in-band to out-of-band noise density.

CONCLUSION

The experience of the last five years has shown that IOTs are an excellent choice as the final amplifier in analogue u.h.f. TV transmitters. There has been some concern expressed regarding the 'back-off' required when amplifying digital television signals. These measurements strongly suggest that, with the IOT technology, it is possible to obtain much higher average output powers than has been suggested previously. Therefore, in practice, the HPA 'back-off' required for digital signals should not present a significant problem. The results of the work reported here show that the IOT is also an excellent choice for digital transmitters.

ACKNOWLEDGEMENTS

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The authors wish to thank their many colleagues at EEV Limited in England and EEV Inc. in the USA for their valuable technical contributions to the work described.

The views expressed are those of the authors and not necessarily those of the General Electric Company of England.

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IMPROVING AND CONTROLLING SIGNAL QUALITY

Wednesday, April 17, 1996
2:00 - 5:00 pm

Session Chairperson:
Dane Ericksen, Hammett & Edison, San Francisco, CA

PANEL ON RFR EXPOSURE

Panelists: Robert Cleveland, Federal Communications Commission,Washington, DC; Jules Cohen, Consultant, Washington, DC; William Hammett, Hammett & Edison, Inc.,San Francisco, CA; Richard Tell, Richard Tell & Associates.,Las Vegas, NV

DIGITAL NETWORKING TECHNIQUES APPLIED TO SINGLE AND MULTIPLE SITE REMOTE TRANSMITTER OPERATION

John E. Leonard, Jr.
Gentner Communications Corporation
Salt Lake City, UT

MEGAWAVE/NAB JOINT TECHNOLOGY DEVELOPMENT IMPROVED ANTENNAS FOR NTSC OFF-THE-AIR TV RECEPTION

Glenda O. Benham
MegaWave Corporation
Boylston, MA

A DSP-BASED BTSC STEREO DECODER

Mark T. Grant
Belar Electronics Laboratory, Inc.
Devon, PA

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John E. Leonard, Jr.
Gentner Communications Corporation
Salt Lake City, UT

ABSTRACT

This paper addresses the application of networking techniques to the operation of broadcast transmitters and the equipment co-located at those sites.

Current trends in station operation, such as duopolies, LMA's, group owership and broadcast networks now run from a central location, are changing stations' requirements for remote control. Operations can include many transmitters scattered around the city, state or even the country. Beyond being responsive to the FCC regulatory requirements, staying on the air is the most important need to broadcasters.

With the October 2, 1995 action (FCC 95-412) from the Notice of Proposed Rule Making (MM94-130) previous needs for operation of remotely located broadcast transmitters have changed. Unattended operation is now permited of all broadcast (AM, FM and TV) transmitters. Clarification was given in a number of related areas. This paper will only address some of those areas.

After 10 years of remote control with dial circuit operation, it is now thought that digital communications concepts can provide a distinct advantage in transmitter operation. A new approach to site control, utilizing networking, is now being developed.

Networking concepts can be applied to the operation of the transmitter site and the other sites comprising the entire broadcast facilities. With operations now having from two to hundreds of locations, a network is ideally suited to the task. Not only can the transmitter and building-specific functions be successfully applied to a network, but other items such as modulation levels, video measurements, EAS (EBS), etc. could be added to the data occupying this network.

This paper explores the founding of such a network and the possible establishment of an open protocol to this network as applied to broadcast facilities.

OVERVIEW

Transmitter remote control is a somewhat recent addition to broadcast. Remote control or operation of non-directional AM broadcast transmitters up to 10 kW in power was first permitted in 1953. Further changes in broadcast regulations, that affected various segments of the industry, were made in subsequent years. The change in FCC Rules and Regulations for remote operation of transmitters in November of 1984 has had the most dramatic effect on transmitter operation to date.

After the 1984 change in regulations, the first dial-up transmitter remote control system was introduced by Gentner. Their microprocessor based unit, with a digitally synthesized voice, was designed to operate on the switched (or dial) telephone system. This system, along with other dial-up systems that followed the Gentner concept, enabled operators to call the transmitter and both learn of conditions at the site and make corrections using a telephone keypad. This type of remote control fulfilled regulatory requirements when correctly applied.

However, for full unattended transmitter operation to occur, amendment of the Communications Act of 1934 was required. This occurred last October with the FCC Report and Order from MM Docket No. 94-130. This Report and Order established new requirements for remote and unattended operation of all broadcast stations.

Unattended transmitter operation is much more important today than in the past. Significant changes have occured over the past ten years in ownership and operation of broadcast facilities. Multiple station ownership in a metropolitan area is now permiited; Duopolies and LMAs are prevalent; engineering staffs have been dramatically reduced, or entirely eliminated in some cases and replaced with contract engineers. The days of fully staffed transmitter sites are virtually gone. In fact, many studios operate with a staff of just a few people and run their equipment through automated means.

Paired with the change in station operation is the proliferation of communications technology. PCs are abundant and inexpensive. Digital communications lines are being established worldwide. The Internet and its associated communications power has been "discovered" by millions of people. Gentner believes that the changes in station operation, combined with easy access to advanced communication tools, mandates the development of a digital control network.

Regardless of the changes in station operation, one thing has remained constant: the most important thing to any broadcaster is to be on the air. This is as true for a day-time AM broadcaster as it is for a network owned-and-operated television station in any major market.

The current state of the remote control market consists of a 1:1 relationship between equipment at the transmitter and studio or PC equipment. In LMAs and other facilities with multiple transmitters, much of the remote control equipment is redundant, especially the required studio or PC equipment.

As the 1995 Notice of Proposed Rule Making was released, Gentner had began work on a new remote control product which establishes a network for unattended operation of broadcast transmitters and studio equipment, whether remote or not.

The network created by this product is client/server based. Not only does it provide functions for transmitter operation, but it has the capability of carrying data from other equipment such as modulation monitors, video monitors, alarm systems, STL switching, EAS monitors, etc. to their appropriate applications running at the PC.

One side of this network is personal computer (PC) based. The other side uses embedded processors. There are, in fact, two networks that can exist, depending upon equipment location and modes of operation. One network is a local area network (LAN) that exists between embedded processors; the other is the network that exists between the embedded processors and the PC.

Three prime considerations were paramount in the creation of this network concept. These are security, reliability, and usability.

SECURITY

For a system of this type to remain secure, the highest possible degree of authenticity must be maintained on information and the control of access. In a network of this type a number of levels of security must exist.

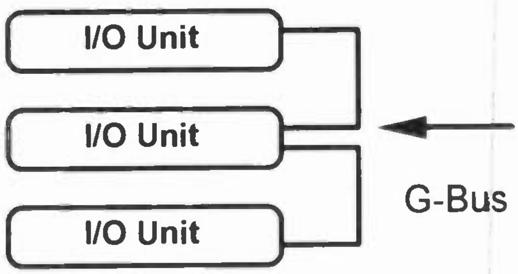
With dial-up remote controls, touch-tone (DTMF) generated passwords are used to control inward access. The dial telephone system is used for outward security. This has worked reliably for the last decade.

Given the security concern, a protocol had to be created for the new system that supports site (or LAN) security, while also providing overall network security.

The new system's network protocol supports the actual data gathering effort with a unit that has been named the I/O Unit

(Input/Output Unit). This embedded micro-controller unit collects all analog metering inputs, all status inputs and issues all command outputs. The I/O unit also contains a real-time clock, backed-up memory for the processor, etc. Connections to the "outside world" (external equipment, or a LAN) begin at the I/O unit.

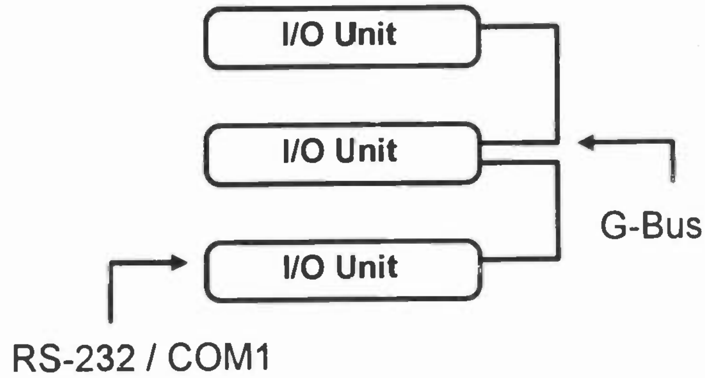
The LAN created includes "daisy chained" I/O units -



The above sketch depicts what Gentner calls a "G-Bus." G-Bus communications occur frequently between I/O units. The G-Bus between units may run up to 600 feet in length (such as in elevator shafts, etc.). These potential "run" distances, coupled with the firmware capability of automatic commands between I/O Units, mandates significant security at this level.

Packet protocol is used with various levels of cross checking. Each I/O Unit has unit-specific identification assigned by the associated Windows software. Each unit also contains a unit-specific electronic serial number of an unspecified length. Unit ID is checked whenever communications are initiated.

The outside world communication with I/O Units has the more apparent need for security. This occurs from what can be viewed as the other side of the I/O Unit. This side is added in the following sketch-



I/O Units all have two standard RS-232 ports. The above sketch only depicts one. This COM port can be used to connect to a PC, a modem or similar communications path. In the sketch above, this location or site would have a total of six (6) COM ports.

The communications server embedded in the I/O Unit provides a number of security aspects. As mentioned, each I/O Unit has its own unique electronic serial number. This serial number is used in the initiation of any communications with this site.

User selected Passwords exist in the PC Server Software that accompany this concept. These Passwords have several levels of operation and functionality associated with them. This is similar to the User, Operator and System Access Codes used in the dial-up remote controls. These, however, are not DTMF, but data.

The Communications Server also contains a site generated random key that is used for encrypting all passwords and other security related items. The Communications Server will not permit the PC Server to transfer any of these items without the use of the encryption random key first being passed and acknowledged. While this provides a strong level of security, we are not prepared to say that it is totally "hacker" proof. However, there are many levels of security that a "hacker" would have to work through in order to gain access to the system.

RELIABILITY

Once a secure network exits, the reliability of that network becomes the next concern. As with security, reliability has two sides: the LAN side and the outside world side.

The I/O Unit reliability is of the utmost importance; fortunately, technology has taken major steps forward in reliability during the past ten years. First, memory types that can be used with the processor have changed drastically. A judicial selection of memory types now assures memory retention without the extensive use of battery back. Flash EPROM and CMOS static RAM have been used. Batteries are now needed only to keep the real-time clock active.

Other areas, such as RF interference protection and a universal AC power supply (operation from 90 VAC to 240 VAC) also have importance in the I/O unit and are being carefully addressed.

USABILITY

If a network has the reliability to exist forever, it has no real value if it doesn't provide a useful service. Key to the network's usability, after consideration of the raw inputs and outputs, is the operating software. That software is both embedded and PC based. PC software runs as a Windows® application, enabling "point and click" operation. Embedded software has been designed for simple upgrades via a diskette or BBS (FTP) download.

The network protocol controls what can occur in the system. From the inception of this project, private data packets have had a presence. To thoroughly use a network, data from other sources must be able to exist. A number of immediate uses are apparent for these private data packets. Some manufacturers, such as Belar and Tektronix, offer products that communicate via modems to PCs using their software. Gentner anticipates being able to move the data from the RS-232 output of their equipment to their software operating on the PC employing the Gentner PC Server software. EAS also has an RS-232 requirement, and will present potential applications.

This network can only do its job in a Client/Server environment. Client/Server segments exist throughout this network.

In the I/O Unit the software embedded kernels must exist for one selected unit to provide the outside world communications, while also supporting G-Bus communications to other I/O Units.

With networking, size is no longer a constraint in configuring a control system for broadcast sites and transmitters. Each site in the system may support up to 16 I/O units, and the network will support up to 256 sites. In its largest configuration, a site control network could have 4096 I/O units (65,536 channels). Nor does this system ignore the needs of the small broadcaster; in its simplest configuration, it will support one site with eight channels each of metering, status and control (control channels are configured with two outputs each). LMAs and other facilties requiring multiple control from one location may control all transmitters, plus the studio site if needed, with one PC and with I/O units located at the studio and each transmitter site.

Specific information on the configuration and operation of this networked site control system is available from Gentner Communications.

RULES AND REGULATIONS

The rules that came from MM Docket No. 94-130 have provided clarification in the area addressed by this paper. While not wanting to address the area of the Rules specifically, it is felt that several deserve being repeated here. These include, from the Discussion section, the following paragraphs from the Report and Order, and are intended only to provide background. Emphasis and underlining have been added by the author of this paper.

Unattended operation.

7. The Commission, based upon its experience in enforcing broadcast rules, concurs with the majority opinion that waiver of Section 318 of the Act to permit unattended operation is not likely to result in an increase in operation outside the tolerances specified in the Rules or the station authorization and will not adversely affect the public interest.

Significant technical malfunctions should quickly become obvious and we do not anticipate their continuance for significant periods of time. The waiver appears further justified for reasons of efficiency, in order that our broadcast licensees can best decide how to allocate resources to ensure compliance. **Therefore, the rules relating to station operators will be eliminated or revised as proposed in the Notice so as to permit unattended broadcast station operation.**

Universal application.

10. The comments indicate that there are no technical obstacles to the automation of any type of broadcast station. It is clear, for instance, that the maintenance of the antenna sampling system is much more important than whether the particular system has been formally approved. The Commission concurs with the comment that stations without approved sampling systems, because of licensees' adherence to appropriate maintenance procedures, may be just as stable as stations with approved sampling systems and would be more stable than a station with an approved sampling system that is inadequately maintained.

Imposing new regulatory restraints upon licensees of such stations does not appear to serve any useful purpose. In any event, the requirement remains that the antenna system must operate properly, regardless of the type of sampling system. **Therefore, all types of standard broadcast (AM, FM and TV) stations, as well as international broadcast stations will be permitted to operate unattended**. The same permission applies to low power TV ("LPTV") stations with locally-originated programming.

Monitoring, measurement and calibration requirements.

22. Accordingly, the rules will be adopted as proposed so as to indicate the operating parameters requiring periodic licensee attention. However, our proposal to specifically require log entries of corrective actions will not be included. The intent of that proposal was simply to require that licensees maintain some minimal record of their transmission system measurement, adjustment and maintenance procedures. The Commission always reserves the right to question licensees found violating the technical rules in order to determine whether or not a good-faith effort has been made to comply with the rules. Whether a licensee's response to such an inquiry takes the form of the proposed log entry or as a reply to an interrogatory letter at the time of a perceived problem would not appear to be of such consequence as to require specification in the rules.

23. The intended effect of proposed Section 73.1350(c)(2) seems to have been misunderstood by the commenters. **It was intended merely as a clarification of long-standing policy and not as a change in measurement procedure or as an effective change in operating tolerances**. An example should suffice to illustrate the objective. Section 73.1560 requires that a station's power be maintained as near as practicable to the authorized value and may not be less than 90% nor more than 105% of the authorized value for AM and FM stations.

Thus, for an AM station authorized 1,000 watts, the maximum permissible output power is 1,050 watts, which is the absolute limit. The proposed rule merely makes explicit the implicit requirement that the licensee be aware of the error inherent in the measurement instruments employed. Section 73.1215 requires that indicating instruments used at broadcast stations have an accuracy of 2%. The rule merely acknowledges reality in recognizing that the economical manufacture of perfectly accurate meters and other measurement devices is impossible. However, producing measurement devices with an error rate not exceeding 2% is economical. But even though such devices may not be perfect, they can be compared or calibrated against higher standard instruments so that their inherent inaccuracy may be known and taken into consideration.

24. For the example cited, the power would typically be calculated by taking the common point antenna impedance (for the purpose of the example, 50 ohms) and multiplying it by the square of the radio frequency ("RF") current. Thus, the correct RF current (assuming a perfect indication) would be 4.47 amperes for 1,000 watts and 4.58 amperes for 1,050 watts. All the proposed rule requires is that the licensee be familiar with how high or low relative to the true value a meter reads, so that the actual indication can be used to adjust the transmitter to the authorized limit. Thus, if a licensee had an RF ammeter with a full-scale reading of 5 amperes which indicated 1% below the true current value, the indicated error would be -0.05 ampere. That would need to be subtracted from the ideal values in order to determine the true power. Thus, the indication for the authorized power would be 4.42 amperes (instead of the expected 4.47 amperes) and the licensee should be

concerned about overpower operation if the indicated value goes above 4.53 amperes (which, in reality, reflects a true value of 4.58 amperes).

25. **The example given above should demonstrate clearly that the proposed rule amendment does nothing to alter the practical application of specified parameter tolerances. However, it does serve to correct the erroneous idea that the Commission is only concerned with indicated, rather than actual, parameter values.** Therefore, the proposed rule will be retained.

Permissible methods for remote transmitter control

40. The Commission agrees with those in favor of relying on the PSTN for transmitter control. There is no doubt that the reliability of the PSTN is very high, and evidence that dedicated leased lines receive higher priority from the local telephone companies has not been provided. Moreover, the Commission is not persuaded by the arguments that dedicated switched lines should be used for purposes other than transmitter control, even if such use is expected to be small. It is impossible to predict when ATS/AMC equipment may need to contact a responsible person, or to know when designated supervisory personnel may want to call the transmitter site to ascertain the status of the equipment. **Therefore, the rules will be amended to permit the use of a dedicated, switched telephone line (or number) for transmitter control purposes, in lieu of a dedicated, continual use leased line**.

SUMMARY

In summary, the ability and needs for operation of broadcast transmitters have come great distances, particularly in the last decade. With a Client/Server Network established, it is hoped that the coming decade will see even more advances.

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MEGAWAVE/NAB JOINT TECHNOLOGY DEVELOPMENT IMPROVED ANTENNAS FOR NTSC OFF-THE-AIR TV RECEPTION

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ABSTRACT

This paper describes the design and analysis results, and performance achieved for the NTSC off-the-air TV antenna developed by MegaWave Corporation, as part of a Joint Technology Development program with the National Association of Broadcasters. The model used to evaluate the performance of the antenna in terms of user perception i.e. received picture quality, is discussed. Measured performance parameters, including VSWR and radiation pattern, are also presented. Finally, statistical results of received picture quality relative to other available TV set-top antennas are shown to demonstrate the performance improvements that have been achieved.

BACKGROUND

In June 1994, the National Association of Broadcasters (NAB) issued a competitive RFP to industry requesting proposals to develop improved TV set-top antennas for NTSC off-the-air reception. Most current antenna designs for off-the-air reception are based on "rabbit-ears" and loop antennas, or variations thereof. There have been no major technological advances in this area for more than forty or fifty years. Based on its proprietary antenna technology, MegaWave Corporation was selected by NAB in October 1994, to participate in a joint technology development to fabricate and test prototype antennas to meet the defined program objectives.

LIMITATIONS OF EXISTING OFF-THE-AIR ANTENNAS	PERFORMANCE GOALS FOR THE JOINT TECHNOLOGY DEVELOPMENT
• Large physical size • Appearance • Separate antennas for VHF and UHF • The need to adjust the length and orientation for different channels • Coupling to body when adjusting the antenna for best reception	• To develop a compact design • No length adjustment needed • Reduced coupling to nearby people and objects • Aesthetic packaging • Single antenna covering VHF and UHF • Near constant performance at VHF and UHF • Comparable performance to "rabbit-ears"/loop • Low cost design

Table 1. Performance Goals

The principal goal of the program was to overcome the limitations of existing antenna designs as summarized in Table 1. The performance goals for the MegaWave antenna, based on these limitations, are also listed in Table 1.

"Rabbit-ears" and loops, like many antennas, are narrowband devices i.e. their performance, VSWR and radiation pattern, are constant only over a very narrow frequency range. Outside of this frequency range, the patterns undergo large fluctuations and become multi-lobed as illustrated in Figures 1 and 2 for the "rabbit-ears" and loop respectively. Their VSWR also shows extreme variations with frequency, such that the antenna

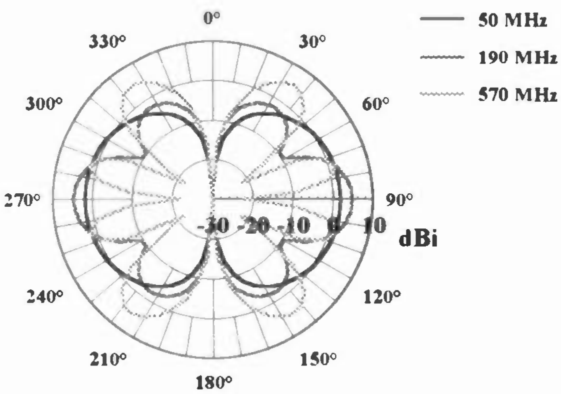


Figure 1 Computed radiation patterns of 2 meter dipole

is not well matched at most frequencies. The measured VSWR of a typical "rabbit-ears"/loop antenna is shown in Figure 3. The antenna designed during this Joint Technology Development program aims to overcome these fundamental performance limitations.

DESIGN ISSUES AND PERFORMANCE MODELING

Within the goals for the program, one of the major design challenges is to achieve the same performance in the low VHF band, with a compact antenna, as a 2 meter long dipole, the latter being 0.36 wavelengths long at Channel 2 (54 MHz).

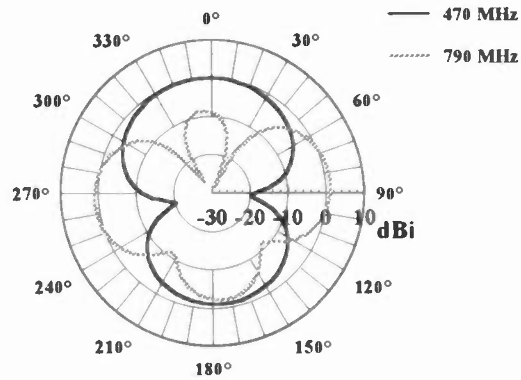


Figure 2 Computed patterns of 8 inch diameter loop

A number of different antenna shapes were considered during the design process, the design of each being optimized to obtain the best performance characteristics. A two-fold approach was used during the design process. Firstly, the Method of Moments Code NEC-4.1[1] was used to compute input impedance and radiation patterns of each antenna configuration considered. However, while this gives information about the antenna parameters, it provides no information relative to the end-user's perception of the antenna performance. A second model was therefore developed which uses the antenna parameters computed by NEC-4.1, and incorporates a statistical propagation analysis and noise models for different areas (e.g. rural, urban) to compute the received signal-to-noise ratio.

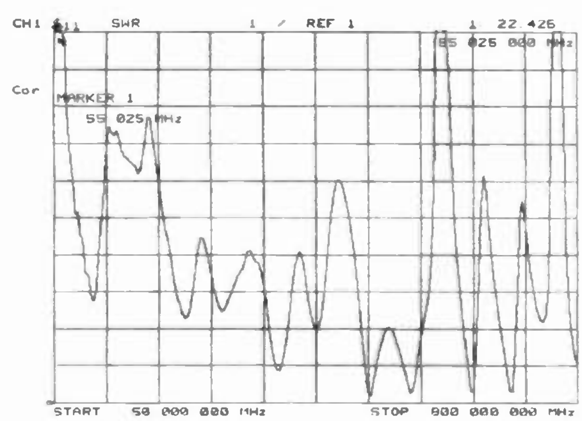


Figure 3 Measured VSWR of Typical "Rabbit-Ears" and Loop Antenna

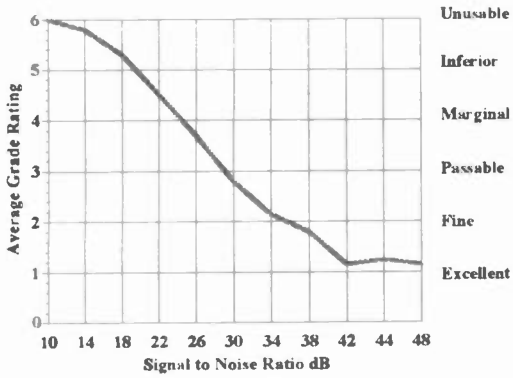


Figure 4 Average Observer Ratings vs S/N Ratio - NTSC Color Signals

The picture quality is then derived from the received signal-to-noise ratio using the standard psophometric weightings as shown in Figure 4. The other benefit of this model is that it allowed the MegaWave antenna to be benchmarked against other antennas before any prototypes were built and therefore serves as a virtual test bed. The antenna performance benchmark is designed to simulate the picture quality seen by end-users in metropolitan, urban and suburban markets. It assumes that the transmitter sources are clustered as shown in Figure 5.

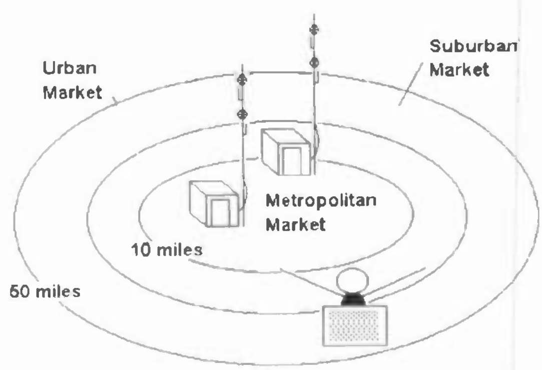


Figure 5 Antenna Performance Benchmark Scenario

The model can be used with any set of input parameters for transmitter height and power, propagation path hill height, receive antenna height, and receiver noise figure. The specific parameters chosen for the results presented are listed in Table 2.

PARAMETER	VALUE
TX antenna height	300 m
TX power	100kW (low VHF) 316kW (high VHF) 5MW (UHF)
Path roughness	90 m
RX antenna height	10 m
RX noise figure	6dB (low VHF) 9dB (high VHF) 13dB (UHF)

Table 2. Input parameters assumed for picture quality model

The virtual test bed model was applied to the different antenna candidates considered and used to evaluate and rank the candidates prior to building the prototypes. The design which came closest to meeting the performance goals has a maximum envelope 20 inches long by 8 inches wide by 4 inches high. A computer visualization of one possible implementation of the antenna is shown in Figure 6.

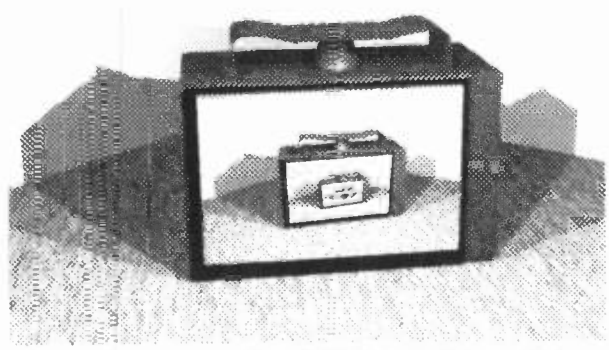


Figure 6 A Possible Implementation of the MegaWave Antenna

The results for three cases, a 2 meter long dipole, an 8 inch diameter loop, and the design finally selected for MegaWave antenna, are shown in Figure 7(a)-(c) where picture quality is presented as a function of channel number on the vertical axis and distance from the transmitter on the horizontal axis. It is assumed in all cases that the antenna is pointing in a fixed direction. With the 2 meter dipole, it is seen that due to the resonant behavior of the dipole, very poor performance is observed around Channel 28. With the 8 inch diameter loop, the performance in the low VHF channels is very poor, as would be expected since the antenna is electrically small. Under the assumed conditions, excellent picture quality is observed out to approximately 22 miles with the loop. Figure 7(c) however shows that with the MegaWave antenna, the excellent picture quality contour is nearly uniform across all channels at an average range of ~ 31 miles. The path roughness of 90 meters corresponds to hilly terrain. With lower values for the roughness, the excellent picture contours would lie further away from the transmitter, however, the relative results between the antennas would be the same.

If a low noise figure amplifier is added to the MegaWave antenna, then the excellent picture contours for the MegaWave antenna would extend by 8-9 miles at UHF and 6-7 miles at high VHF with little improvement at low VHF. With lower noise figure TV receivers than assumed here, the improvements are less significant.

MEASURED PERFORMANCE

The VSWR and radiation patterns of the final antenna were measured. The VSWR, shown in Figure 8 is seen to be less than 3:1 over entire VHF and UHF bands and less than 2:1 over approximately two-thirds of this range. This is a marked improvement over the VSWR of the "rabbit-ears"/loop of Figure 3, where the VSWR lies anywhere between 1:1 and greater than 11:1.

The azimuthal radiation patterns were measured in the low and high VHF bands using signals-of-opportunity from local TV stations. Due to multipath problems at UHF, patterns were measured using a local signal source. The measured radiation patterns are shown in Figures 9(a)-(c). In Figure 9(c), the equivalent UHF channel numbers are given for the selected frequencies used during testing. Figure 9 demonstrates the uniformity of the radiation pattern at all frequencies from Channel 2 to Channel 69. The patterns are figure-of-eight in all channels with a slight broadening of the pattern at the high end of the UHF band. When compared to the computed patterns of the "rabbit-ears"/loop in Figures 1 and 2, the improvement is significant.

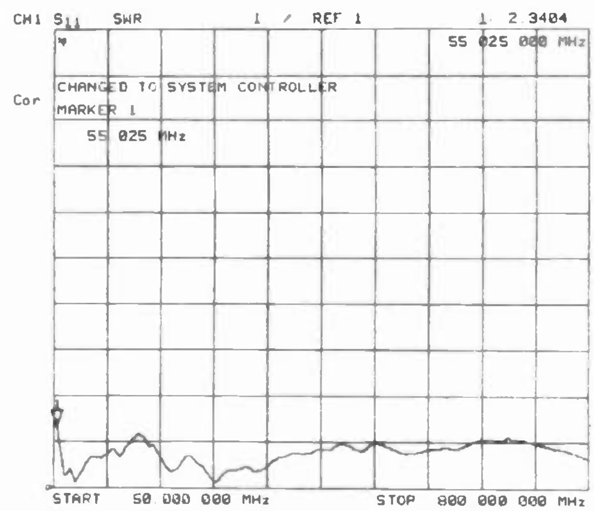
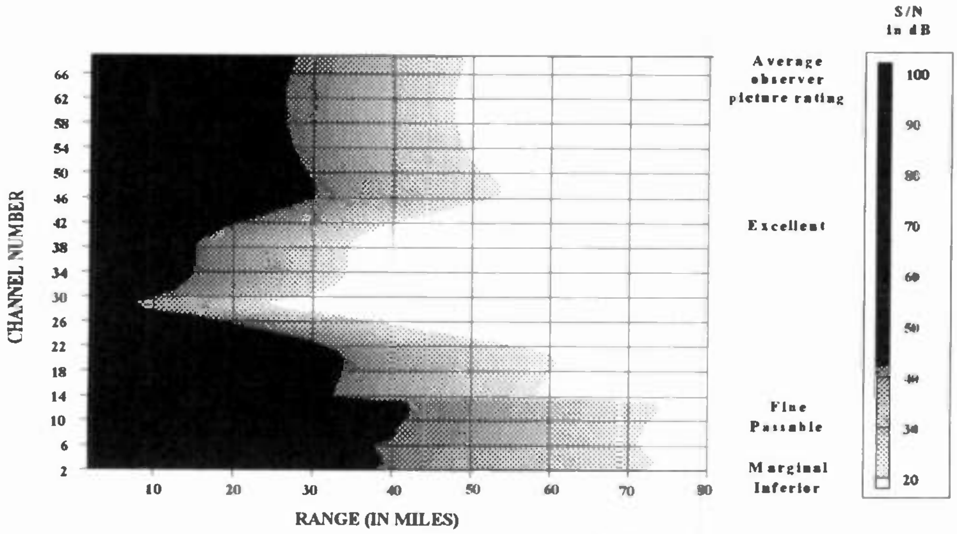


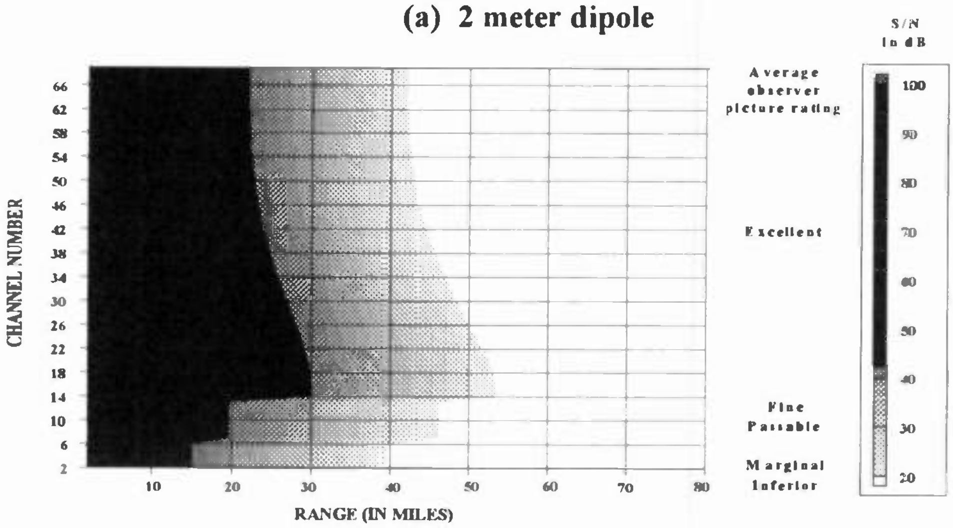
Figure 8 Measured VSWR of MegaWave Antenna

MegaWave has also conducted extensive subjective picture quality testing against other available TV set-top antennas in several locations. Testing of this nature needs to be carefully conducted. Due to localised variations in field strength, testing using side-by-side TV sets was performed with the antennas located on both sets.

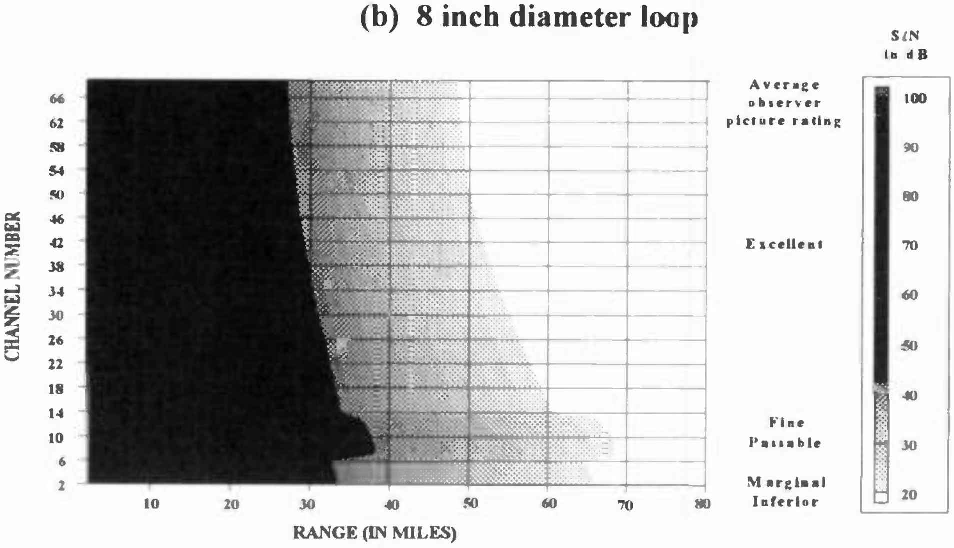
Results collected on this basis, should be viewed statistically. Figure 10 shows the subjective picture quality testing results as a function of TV channel number. Ratings were scored as:



(a) 2 meter dipole

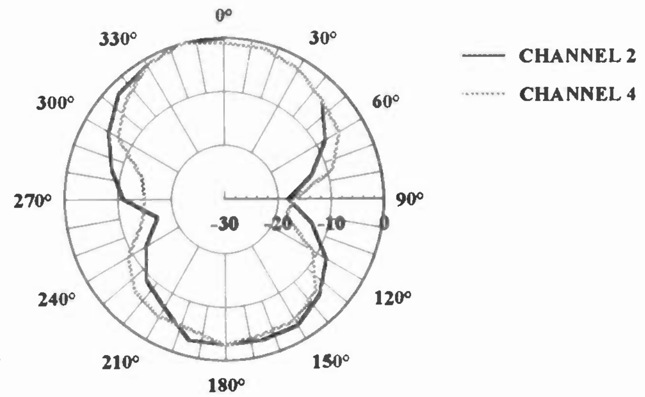


(b) 8 inch diameter loop

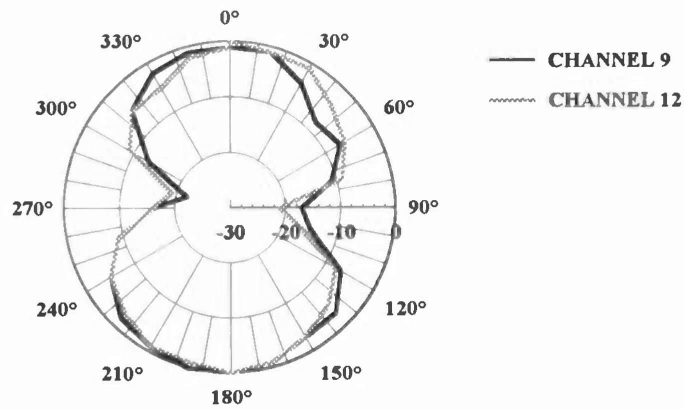


(c) MegaWave antenna

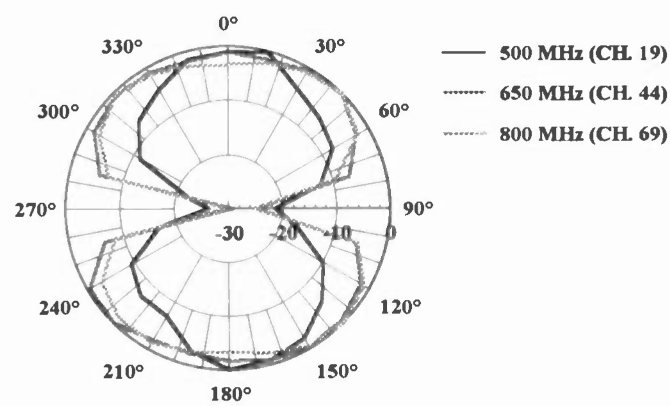
Figure 7 Picture Quality Contours vs. Channel Number and Range



(a)



(b)



(c)

Figure 9 Measured Radiation Patterns of the MegaWave Antenna

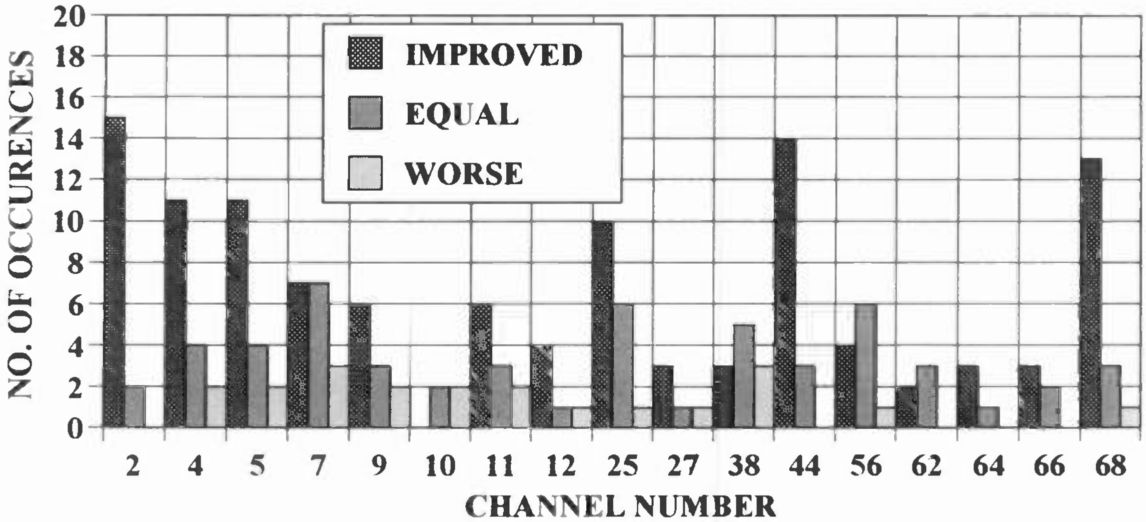


Figure 10 Subjective Picture Quality Test Results (Cumulative Basis)

MegaWave antenna better than the other antenna, equal to the other antenna and worse than the other antenna. The scores on a cumulative percentage basis are shown in Figure 11. The results show conclusively that the MegaWave antenna outperforms typical available antennas. In any location, a typical user would expect a better or equal picture to that received by other antennas for 90% of channels received. The results also show the virtual test bed results of Figure 7 are optimistic for the 2 meter "rabbit-ears" in the low VHF band, since testing shows the MegaWave antenna to be superior at channels 2, 4 and 5. A possible explanation for the difference between the virtual test bed and the test results at low frequencies, may be due to non-ideal construction techniques for the "rabbit-ears". Another reason is that the 2 meter dipole and loop were modeled in isolation, whereas in practice, since they are collocated, there will be significant coupling between them. This is one of the reasons why they are difficult to adjust. Also there are an infinite number of positional combinations for the "rabbit-ears" and loop such that it would be impossible to model all configurations. At UHF, the results are consistent with the virtual test-bed results and are found to be consistently better than most "rabbit-ears"/loops.

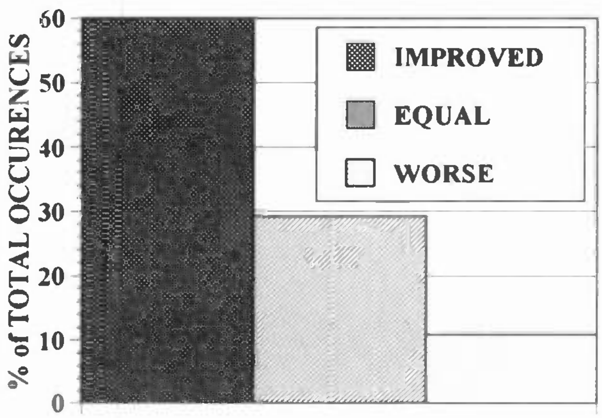


Figure 11 Picture quality test results on a percentage basis

CONCLUSIONS

The goals of the Joint Technology Development program with the National Association of Broadcasters have been achieved. The antenna design resulting from this program is not only compact, but requires no length adjustment. It overcomes the performance limitations of other available set-top antennas by providing uniform VSWR and radiation patterns without adjustment or switches. Because of its size and envelope, it can be placed on top of the TV set or on an adjacent surface.

Consistent with picture quality computer modeling, subjective testing supports and in some instances exceeds the analysis, that the MegaWave antenna has superior reception to a two-meter long dipole and an eight inch diameter loop. Results of subjective testing show that a typical user in any location would expect an equivalent or better picture with the MegaWave antenna for the majority of channels received without adjustment.

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A DSP-BASED BTSC STEREO DECODER

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ABSTRACT

This paper describes a BTSC stereo audio decoder for television that digitizes the aural baseband signal and decodes the stereo multiplex portion using digital signal processing techniques. A block diagram of the stereo decoder is presented. For each block the performance advantages over an analog system are explored. Tables illustrate the allowable errors in a stereo decoder to obtain a given performance. Finally, the measured performance of a fully implemented stereo decoder is discussed.

INTRODUCTION

The decoding of a stereo composite signal into its components Left, Right, L+R, L−R, pilot, and stereo subcarrier is a well defined process. In a traditional analog stereo decoder one would find several high order filters, a switching demodulator with some form of product cancellation and cross-coupling, along with numerous precision components and trimmers. With the advent of high-speed digital signal processors, the prospect of implementing a stereo decoder in software has become feasible. (The FMSA-1, Belar's first DSP-based stereo decoder for FM broadcast, was demonstrated in 1993.) The enormous amount of computation necessary for a BTSC stereo decoder requires the speed of today's faster processors. The processors in the stereo decoder described here execute 100 million instructions per second to perform their function.

This paper describes a BTSC stereo audio decoder for

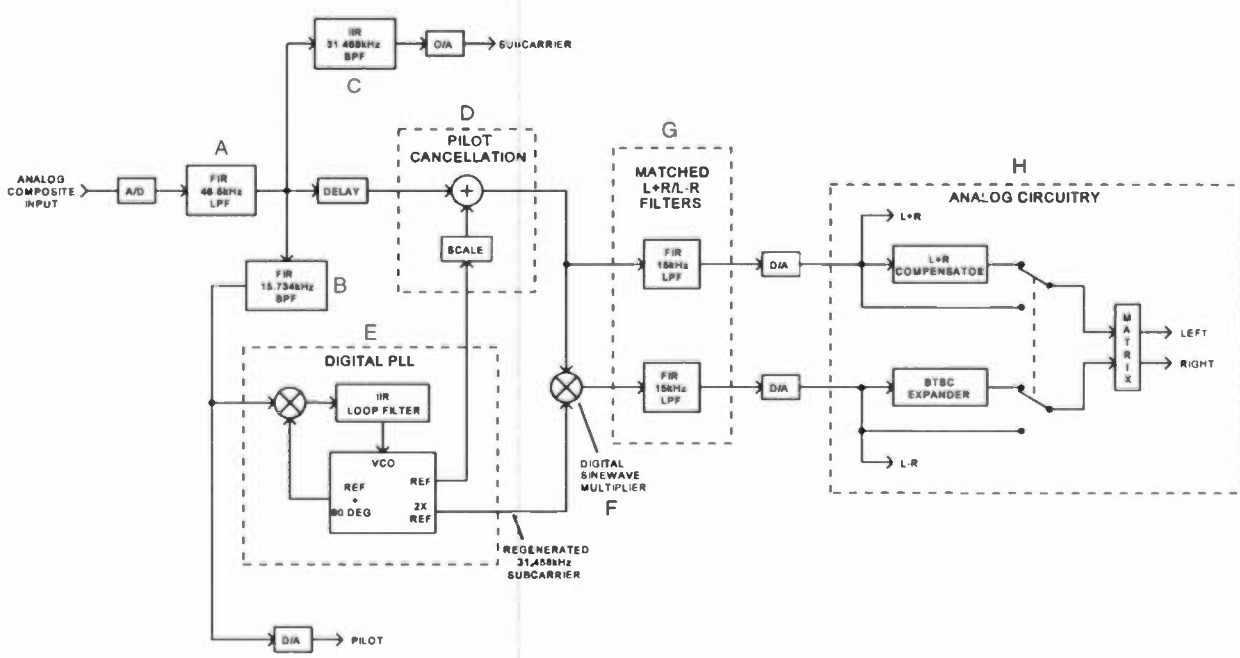


Figure 1 - BTSC Stereo Decoder Block Diagram

television that digitizes the aural baseband signal and decodes the stereo multiplex portion using digital signal processing techniques. The decoding functions implemented in software include an FIR pilot bandpass filter, a digital phase-locked loop, a pilot cancellation system, an IIR subcarrier suppression bandpass filter, a sinewave multiplication demodulator, and matched FIR L+R/L−R lowpass filters.

A block diagram of the digital BTSC decoder described in this paper is shown in *Figure 1*. In the following paragraphs, the performance advantages over an analog system for each block are explored. Measured performance of a fully implemented BTSC stereo decoder is presented at the end of this paper.

INPUT

The analog aural BTSC baseband signal, possibly containing Second Audio Program (SAP) and Professional Channel (PRO) carriers, is sampled and converted to a digital representation in a precision analog-to-digital converter. A linear-phase FIR low-pass filter with 46.5 kHz cutoff (labeled "A" in *Figure 1*) is implemented in the digital signal processor array. This filter removes the SAP and PRO signals without affecting the character of the stereo signal. The digital, low-pass filtered stereo signal passes to the pilot filter ("B"), the subcarrier filter ("C"), the pilot cancellation system ("D"), and the demodulator system ("F").

PILOT FILTER

The pilot bandpass filter (BPF) in a BTSC decoder must be capable of separating the pilot from the stereo composite while preserving both its amplitude and phase. The pilot amplitude is important because it is used to determine the pilot injection level. The FCC specifies that the stereophonic pilot must modulate the aural carrier to 5 kHz ±0.5 kHz peak deviation. In order to measure the injection, the amplitude stability of the pilot BPF should be at least 10 times better then the FCC specification. This requires a BPF with a ±1% amplitude stability over time and temperature. The pilot phase is critical because it determines the phase relationship between the L+R and L−R sidebands. Recovering this phase information accurately is necessary for maintaining separation. *Table 1* illustrates the relationship between separation and pilot phase error.

From the table we see that if a separation of 80 dB is to be maintained, the pilot BPF must have less than 1.0 degree of phase error. In the DSP-based stereo decoder, the pilot BPF is implemented as a linear-phase FIR filter ("B" of *Figure 1*). This results in excellent amplitude and phase characteristics. The linear-phase constraint of this filter means that a constant delay, which can be an integer multiple of the sampling rate, is applied to the pilot. This constant delay is easily compensated for by delaying the composite an equal amount, thus preserving the phase relationship between the filtered pilot and the L+R and L−R sidebands. As a comparison, an analog pilot filter would have to maintain both a 1% amplitude error and less then a 1 degree phase error to achieve similar performance. The design of such an analog filter is very difficult when filter tuning, component tolerances, and temperature stability are considered.

Separation (dB)	Pilot Phase Error (degrees)
−20	17.55
−30	10.08
−40	5.71
−50	3.22
−60	1.81
−70	1.02
−80	0.57
−90	0.32
−100	0.19

Table 1 - Allowable Regenerated Pilot Phase Error for Given Separation Ratios

PHASE-LOCKED LOOP

The PLL circuit in a stereo decoder locks the regenerated stereo subcarrier to the incoming pilot. (See "E" of *Figure 1*.) An analog PLL has three components—the phase detector, loop filter, and VCO. The DSP-based PLL simulates these elements in software.

For the phase detector, the analog balanced mixer is replaced by a digital multiplier. After sampling, the pilot is filtered from the stereo composite and multiplied by a computed sine wave reference. The output of the multiplier is equivalent to the analog phase detector output.

Next, the analog loop filter is replaced by a digital IIR filter. The digital IIR filter filters the phase error signal output from the multiplier. This operation is analogous to the analog loop filter function.

Finally, in place of the analog VCO is a digital integrator and sinewave lookup table. The output of the digital loop filter is integrated to determine the phase of the next reference sample. Using a sinewave lookup table, a new reference sample is then computed for multiplication by the next pilot input sample. The digital VCO also generates a reference signal for pilot cancellation, and regenerates the stereo subcarrier at twice the pilot frequency. This VCO block acts like an analog VCO with an external divide-by-2 in the feedback loop to the phase detector.

Since the software PLL has no analog components, it is not subject to some of the problems which can complicate an analog design. These advantages include:

- No initial or periodic adjustments.
- No dc offset errors.
- High quality, *i.e.* 24 bit/144 dB dynamic range, sinewave VCO reference, pilot cancellation, and regenerated subcarrier signals.
- Analog crystal-VCO performance with frequency range that is easily set and controlled.

PILOT CANCELLATION

Pilot cancellation in a BTSC stereo decoder is a necessity. The 15.734 kHz pilot is so close to the L+R audio that removal of the pilot with a 15-kHz low-pass filter is not practical. A conventional analog stereo decoder may cancel the pilot by inverting and summing the bandpass-filtered pilot with the stereo composite. This approach assumes the analog pilot filter is stable enough to maintain an adequate null. As previously discussed, such a filter is difficult to build.

Another disadvantage of canceling the pilot by inverting and summing is that the inverse filter affects the stereo composite. When the output of the pilot filter is subtracted from the composite, the amplitude and phase response of the pilot filter appears as a notch in the stereo composite. This can cause unwanted phase shift and frequency response problems in the L+R and the lower L–R sideband at high audio frequencies.

Since the sinewave lookup-table VCO implemented in the DSP-based decoder can also generate a pure sinewave in phase with the incoming pilot, the aforementioned inverse filter effects can be avoided. This is accomplished by scaling the regenerated pilot to the correct amplitude and subtracting it from the stereo composite samples. (See "D" in *Figure 1*.) In order to track changes in pilot injection, this scaling is performed in a software control loop. The control loop automatically adjusts the scaling to maintain the pilot null.

Finally, because the regenerated pilot is a low-noise, single-frequency tone, using it to cancel the incoming pilot adds no spurious components to the composite stereo signal.

SUBCARRIER SUPPRESSION FILTER

A stereo subcarrier suppression filter is a bandpass filter centered at 31.468 kHz used to measure any residual subcarrier which may have been introduced by the stereo generator. To test a stereo generator, a 5 to 14 kHz modulating signal is applied to either the left or right channel while the output of the subcarrier bandpass filter is observed.

Since this is strictly a magnitude measurement, the phase response of the subcarrier filter is not important. Therefore, a digital IIR filter was used to perform this operation. (See "C" in *Figure 1*.) There is no substantial performance improvement in using a digital filter over an analog design. However, the digital filter is much easier to design and does not require the use of close tolerance analog components.

As an alternative to using a bandpass filter to recover the subcarrier, it is possible to make use of the fact that the DC component of the demodulated L–R is proportional to the subcarrier amplitude. While this is true if the subcarrier and the residual are in phase, any other phase relationship results in erroneous measurements. The only way to escape the phase related errors is to multiply the residue by two subcarriers: one in phase and one in quadrature. For the DSP-based decoder presented here it was more efficient to use a IIR bandpass filter than to decode by the quadrature method.

DEMODULATOR

The typical analog stereo decoder recovers the left and right channels by multiplying the stereo composite by a square wave switching function. The switching function weights the L+R and L–R amplitudes differently and requires a cross-coupling network to correct for this imbalance. The amount of cross-coupling between the left and right channels varies depending on the complexity of the switching function, but cannot be eliminated in practice. While the exact cross-coupling necessary can be calculated, because of component tolerances and other circuit elements, final adjustment is typically done during setup with the aid of a stereo generator. A stereo generator, however, is not perfect and adjustment based on its performance can result in setup errors in the stereo decoder. These errors can be significant, as in the following example.

Suppose a stereo generator has an amplitude error in the L–R sidebands which is constant with audio frequency. An amplitude error of only 0.05 dB in the L–R sidebands would result in –51 dB separation. This amplitude error would be difficult to detect using a oscilloscope and could be canceled entirely in the stereo decoder by incorrectly adjusting the cross-coupling.

The FCC recognized this problem and included a test for it in the pre-1983 Type Approval rules for FM stereo monitors. The test the FCC performed was to apply a stereo test signal known to have −35 dB separation. The stereo monitor was required to read this separation correctly to within ±3 dB. Since the −35 dB error could be verified using a specially calibrated oscilloscope, this technique provided a way to ensure a prospective monitor met a minimum performance standard without requiring a high-accuracy source.

Improved stereo transmission systems now require the capability of measuring separations greater than first required by the FCC. *Table 2* shows the stereo decoder performance required to measure known stereo signals to within given tolerances. From *Table 2* we see that to guarantee that a stereo decoder is able to provide separation performance to

Actual Separ-ation (dB)	Maximum Measurement Error (dB)				
	±6	±3	±2	**±1**	±0.5
−35	−41	−46	−49	−54	−60
−40	−46	−51	−54	−59	−65
−45	−51	−56	−59	−64	−70
−50	−56	−61	−64	−69	−75
−55	−61	−66	−69	−74	−80
−60	−66	−71	−74	**−79**	−85
−65	−71	−76	−79	−84	−90
−70	−76	−81	−84	−89	−95
−75	−81	−86	−89	−94	−100

Table 2 - Minimum Required Monitor Separation (dB)

within ±1 dB at the −60 dB level, it must have nearly −80 dB residual separation. This extreme requirement is a direct result of possible cancellation effects between a stereo generator and the decoder.

Unfortunately, the old test does not provide us with a means to check separations in the −70 dB or greater range. The amplitudes and phases of the L+R and L−R sidebands cannot be verified to the great precision that is required. This highlights one of the significant advantages of implementing the stereo decoder in a digital signal processor.

The DSP stereo decoder described in this paper uses a pure sinewave demodulator. (See "F" of *Figure 1*.) This means that the L−R sidebands are multiplied by a sinewave, which is phase-locked to the pilot and twice its frequency. This demodulation technique does not result in any amplitude imbalance between the L+R and L−R sidebands, as is the case with an analog switching demodulator. The lack of amplitude imbalance eliminates the need for cross-coupling and makes the decoder adjustment free. The mathematical representation of this demodulation process is shown below.

Stereo composite input after pilot cancellation:

$$Composite = \left(\frac{L+R}{2}\right) + (L-R)\cos(\omega_s t)$$

where $\omega_s = 2\pi f_s$ and f_s = frequency of subcarrier.

The regenerated subcarrier is:

$$Subcarrier = \cos(\omega_s t) = \cos(2\omega_{pilot} t)$$

Multiply the composite by the regenerated subcarrier:

$$(Composite)*(Subcarrier) =$$

$$\left(\frac{L-R}{2}\right) + \left(\frac{L+R}{2}\right)\cos(\omega_s t) + \left(\frac{L-R}{2}\right)\cos(2\omega_s t)$$

Filter the composite and (Composite*Subcarrier) with identical 15 kHz LPF's:

$$(L-R)' = (Composite*Subcarrier)' = \left(\frac{L-R}{2}\right)$$

$$(L+R)' = Composite' = \left(\frac{L+R}{2}\right)$$

Matrix to get L and R:

$$L' = (L+R)' + (L-R)' = \left(\frac{L+R}{2}\right) + \left(\frac{L-R}{2}\right) = L$$

$$R' = (L+R)' - (L-R)' = \left(\frac{L+R}{2}\right) - \left(\frac{L-R}{2}\right) = R$$

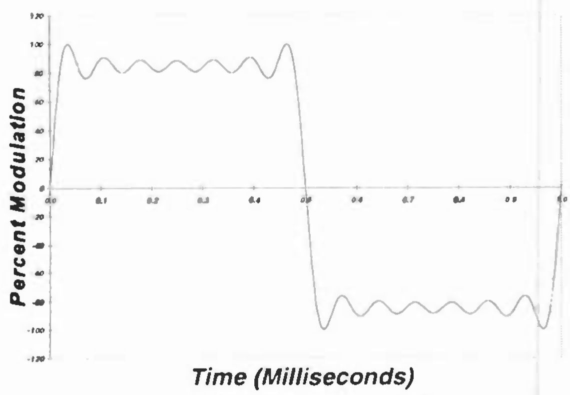
The matrixing functions are implemented in block H of *Figure 1*.

MATCHED L+R/L−R FILTERS

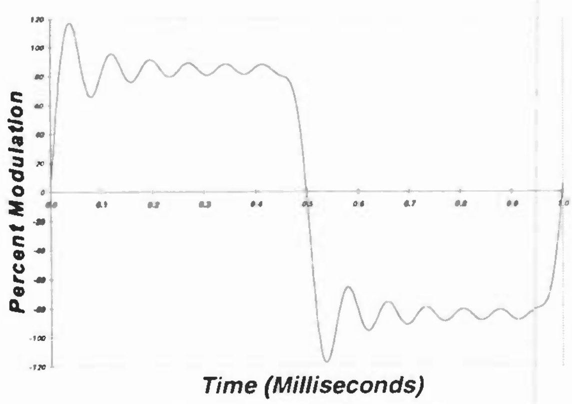
The stereo decoding of the BTSC signal requires that the demodulated L−R audio be expanded using the BTSC algorithm. The demodulated L–R signal cannot be fed directly to the BTSC expander without first removing the products above 15 kHz generated during demodulation. These products are present whether the demodulation is done using a pure sinewave or a switching function. Unfortunately, it is not possible to filter the L–R without introducing some amplitude and phase errors. These errors will directly effect the separation of the left and right channels. However, if equal amplitude and phase errors are applied to the L+R signal, no loss of separation results. This suggests that the L+R and L–R signals should be filtered by identical, or matched, 15-kHz lowpass filters. In practice, this is what is typically done.

Preserving separation through the matched L+R/L–R filters necessitates some very tight filter tolerances. An amplitude and phase mismatch of just 0.05 dB and 1.1 degrees results in –40 dB separation. The high-order analog lowpass filters required to suppress the unwanted products above 15 kHz are very difficult to build and tune. Together, the high order and imperative matching of these filters make this section of the analog stereo decoder crucial if stereo separation performance is to be maintained.

The implementation of matched filters in the DSP domain of the stereo decoder described here is much easier than in the



a) Signal before analog filter



b) Signal after analog filter

Figure 2 - Effects of Analog Audio Low-Pass Filters

analog filter case. The FIR filters, marked "G" in *Figure 1*, have identical filter coefficients and produce matched filter characteristics that are only limited by the precision of the multiplications and additions performed. These numerical errors are very small with today's 24 bit processors and have negligible effects.

Another advantage of implementing the L+R/L–R filters with FIR filters is their linear-phase response. A linear-phase filter delays all frequencies it passes by the same amount. While this characteristic has no bearing on separation or crosstalk performance, it does affect peak modulation readings. As an example of how dramatic these peak modulation errors can be, consider the following scenario.

An L+R signal composed of the first seven harmonics of a 1 kHz square wave is passed through a 9th order elliptic lowpass filter. The 15 kHz elliptic filter has 0.1 dB passband ripple, 1.20 transition region, and a –74 dB stopband attenuation. The waveforms plotted in *Figure 2* show the signal before and after the filter.

As the graphs show, the nonlinear phase characteristic of this high-order lowpass filter result in a 18 percent error in the L+R peak modulation reading. This error will vary according to program material but its effects make the Left, Right, L+R , and L–R peak modulation readings difficult to interpret accurately.

PERFORMANCE

A prototype stereo decoder with the block diagram presented has been constructed. The performance of the prototype was measured using a stereo generator and a FFT analysis program. Separation, crosstalk, and distortion measurements are limited to the wideband SNR unless a spectrum analyzer or tracking filter is used. However, by using an FFT analysis of the analog outputs, signal components as low as –100dB can be reliably measured. The FFT data is generated by digitizing the analog left and right outputs of the decoder. Both the stereo generator and FFT analysis program were developed in-house for the purpose of making these measurements. The stereo parameters measured include separation, crosstalk, distortion and signal-to-noise ratio.

Using an FFT analyzer, the separation may be measured by applying a Left or Right stereo signal and observing the residual reading in the unwanted channel. The separations are measured without BTSC compression of the L–R so that errors in the BTSC expander will not affect the readings. Separations of –80 dB or better were measured using this technique. If BTSC compression/expansion is used, the separations are significantly reduced. This loss of separation results from the amplitude and phase errors of a typical expander circuit.

Crosstalk is measured by applying an L+R or L−R signal to the decoder and determining the residual reading in the unwanted channel. The ability of this decoder to measure crosstalk is directly proportional to the stopband attenuation of the L+R/L−R filters. The FIR L+R/L−R filters in the decoder presented have a stopband attenuation greater than −100 dB. Because the attenuation of the filters is so great, it is difficult to separate the residual crosstalk of the monitor from the noise floor.

The distortion of the decoder is measured by applying a Left or Right signal and calculating the ratio of the RMS value of the fundamental to its harmonics in the output of the respective channel. The distortions are measured without the BTSC L−R compression so that distortion in the BTSC expander will not influence the readings. Using the FFT analyzer, the distortion components of a 1-kHz modulating frequency were measured. The RMS value of the distortion was then calculated to be 0.005% THD.

The signal-to-noise ratio is the amplitude ratio of a 100% modulated Left or Right signal to an unmodulated signal with only the pilot present. This measurement is typically made with de-emphasis. Using a wide-band analog RMS detector the SNR was −77 dB. Without de-emphasis the SNR was −72 dB.

CONCLUSIONS

The implementation of a BTSC stereo decoder using DSP techniques offers many advantages over its analog counterpart. These advantages include reduction of tight tolerance components and trimmers, elimination of temperature and dc stability problems, true sinewave demodulation without cross-coupling, and FIR linear-phase filters which eliminate phase distortion. Together these advances produce a stereo decoder with excellent performance and virtually no adjustments.

ACKNOWLEDGMENTS

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MOBILE TELEVISION: ENG, DSNG, REMOTE VEHICLE

Wednesday, April 17, 1996
2:00 - 5:00 pm

Session Chairperson:
Jerry Butler, WETA-TV, Arlington, VA

SOUND PRODUCTION OF SIMULTANEOUS MUSICAL PERFORMANCE IN FIVE COUNTRIES OF THE WORLD, BY USING THE INTERACTIVE SATELLITE LINK
Kazuyuki Jimma
NHK
Tokyo, Japan

***DIGITAL TELEVISION SATELLITE TRANSMISSION**
Stanley Soonachan
CBS
New York, NY

***DISK-BASED VIDEO FIELD ACQUISITION**
Jim McKain
Avid Technology, Inc
Tewksbury, MA

ISSUES AND ENGINEERING FOR FRACTIONAL TRANSPONDERS EMPLOYING MULTIPLE DIGITAL SIGNALS
Kenneth J. Burns
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LIVE COMPUTER GRAPHICS
Tim Heidmann
Shoreline Studios
Mountain View, CA

THE LONG WAY TO CHHOMOLUNGMA (MT. EVEREST) - PROCESS OF OVERCOMING DIFFICULTIES OF PRODUCING IN HDTV
Fumihiko Saito
NHK
Tokyo, Japan

*Paper not available at the time of publication.

SOUND PRODUCTION OF SIMULTANEOUS MUSICAL PERFORMANCE IN FIVE COUNTRIES OF THE WORLD, BY USING THE INTERACTIVE SATELLITE LINK

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ABSTRACT

A TV Program "Symphony for The Earth" had been broadcasted live, in which Tokyo was linked up with nine places of the world via interactive satellite relay. On the discussions by some panelists of the world, six languages were used and therefore the number of simultaneous interpreters reached eighteen. The communication via the satellite link was going smoothly by the circuits of the complicated interactive connecting line.

A high point of this program was the simultaneous musical performance by a symphony orchestra in Tokyo and other musicians in five places of the world. Although there has been very few such precedent and many difficulties were estimated, the live performance was a great success.

This paper describes the sound production of the simultaneous musical performance and interactive satellite broadcasting.

INTRODUCTION

A TV program "Symphony for The Earth", which had a theme of "Sound and Voice on the Earth", was broadcasted live for three hours on 1st of January 1995. The actual life of people living on the Earth was reported in this program.

Seiji Ozawa , a famous conductor , took charge of a main caster on this program. He conducted the Saitho-Kinen symphony orchestra and also talked with some guests in the world by satellite relay.

This program was constructed with following contents.

1. Seiji Ozawa talks with some panelists of the world about a theme of this program.
2. Musical performance by the Saitho-Kinen Symphony Orchestra.
3. The simultaneous musical performance by a symphony orchestra in Tokyo and other musicians in five places of the world.
4. Messages from some big names and musical performances by famous musicians on VTR.

AN OUTLINE OF SOUND SYSTEM

This program was produced by using two TV studios at NHK Broadcasting Center in Tokyo. Main set which was made up with orchestra set and talk set was built in a TV studio called CT-101.

The other studio called CT-104 was the key station of this program. Tokyo was linked with nine places of the world via interactive satellite relay.

Schematic drawing of sound system is shown in Figure 1. Two mixing consoles were used in studio CT-104. An orchestra was recorded in studio CT-101 and stereo mix was fed to CT-104.

Every language was interpreted simultaneously at interpreting booths which were built in CT-104.

Mix-1, fold back for casters and interpreters, were mixed by using a Mix-1 matrix console which is installed in CT-104. Every direct out from each channel of mixing desk are fed to the

matrix console. While the matrix console provides 16 outputs, these all outputs were used completely for this program.
The Mix-1 matrix console enabled the simultaneous interpretation of six languages and simultaneous musical performance, by means of its ten scene memories. In this program over 20 times changes of the scene memory were necessary.

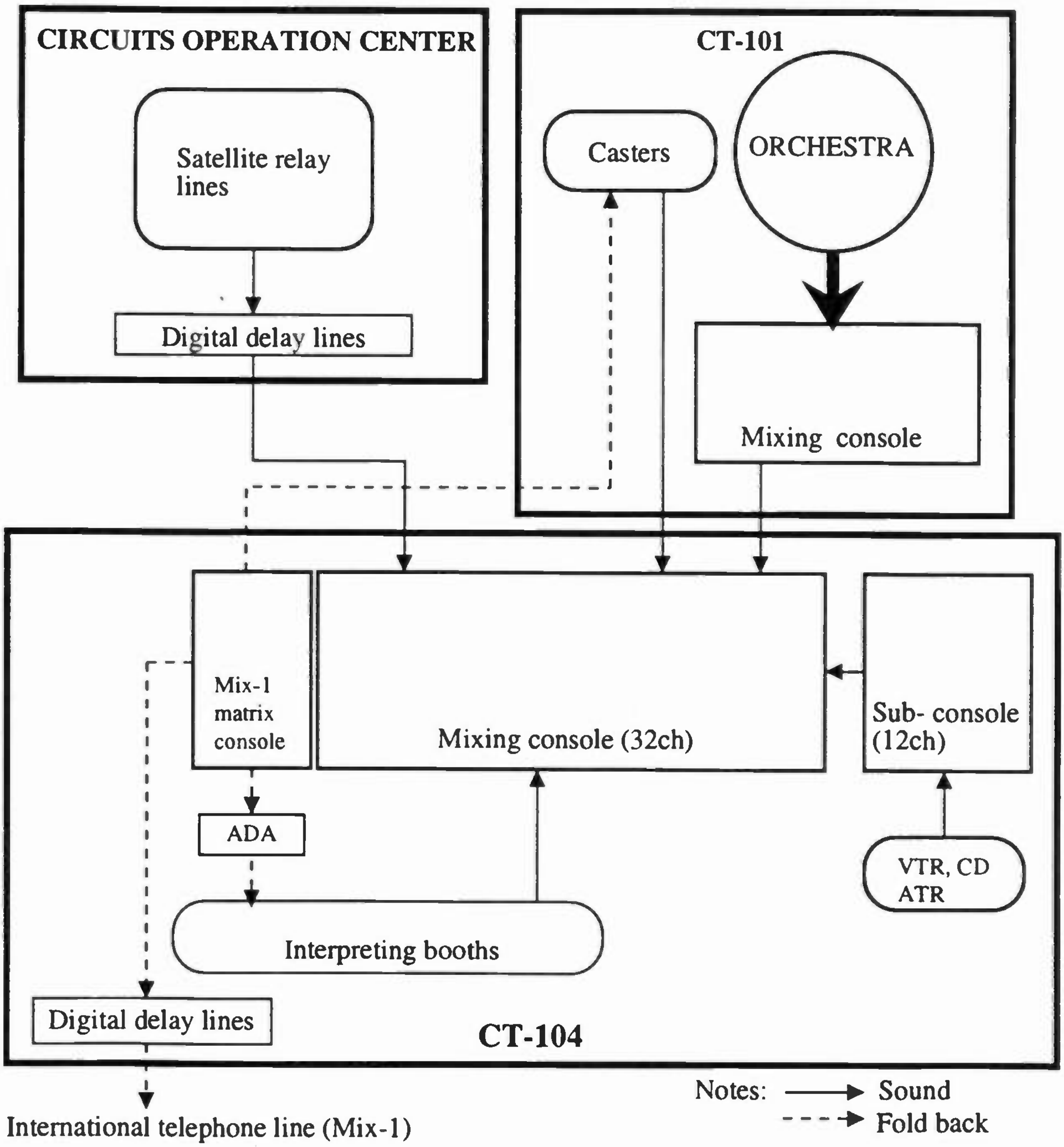

Figure1. Schematic drawing of sound system

Interactive connecting line via the satellite relay

The worldwide interactive connecting line via the satellite relaies had been realized, by which panelists of the world could grasp the details of this program and could talk with each other.
Satellite relay lines were used for sending video and audio from abroad to Tokyo. Otherwise Mix-1 sound for communication was sent by international telephone line. In addition to such relay lines, it was necessary that information line of video had to be sent to panelists in foreign countries for conversation and the simultaneous musical performance on this program.
For this purpose, only one information line of video was sent by satellites and received at each place. Schematic drawing of the satellites relay is shown in Figure 2.
Since Mix-1 signal would reach each place of the world early compared with video signal, for synchronizing with information line of video and Mix-1 sound, digital delay lines had been inserted into each Mix-1 lines.
Six languages, such as Japanese, English, German, French, Russian and Chinese, were used on the program and therefore the number of simultaneous interpreters reached eighteen.
A details of the program, which had been interpreted into English, was always sent to every panelists of the world on Mix-1 line. In case that a panelist who speaks English, their conversation was sent on Mix-1 lines, because main caster on this program could speak English. In the case of any other language excepted English, interpreted voice was sent back by each own Mix-1 line, and English interpretation of their conversation was sent to another panelists.
Two kinds of fold back for casters in the studio were prepared. One of them was for casters who could speak English, when they talked with a panelist by English, voice of panelist was sent directly on fold back line. And when they talked with a panelist by other languages except English, Japanese interpretation was sent on it. The other was for casters who could not speak English, Japanese interpretation was always sent on another fold back line.
The concerns messages of some big names by VTR, had been substituted for Japanese voice-over for broadcasting, also they had been sent just as original voices to overseas panelists.

The Simultaneous Musical Performance

A piece of "Satellite Celebration" composed by John Williams had been simultaneously played by the Saitho-Kinen Symphony Orchestra in Tokyo and various famous musicians in other five places of the world as a high point of this program. To realize of the theme of this program, this music was played in corporation with many musicians over the world.
The musicians who joined this performance were Yo-Yo Ma played the cello in Boston, Isaac Stern on the violin in Connecticut, Dick Lee and Ai Jing sang in Hong Kong, The choir of Assisi in Italy and an African percussions group led Dou Dou N'diaye Rose in Senegal.
Musical performance had started in Tokyo and passed to Yo-Yo Ma in Boston. Then it had relayed in Connecticut, and played between Boston and Connecticut. Next, performance had returned to Tokyo and joined chorus in Hong Kong and Assisi. After that percussion group had joined, finally all musician of the world played this piece together.
Schematic drawing of fundamental system of simultaneous musical performance is shown in Figure 3. As a general rule, musicians play with orchestra in accordance with a conductor. In case that musicians play apart from orchestra, the subject is how to synchronize his performance with orchestra. Therefore it was required that a picture of conductor and sound of orchestra which would make musicians easy to follow the orchestra, were sent to all musicians. A picture of conductor was sent by information line of video and sound of orchestra was by the Mix-1 line.
However, to carry out this musical performance, some difficulties with regard to the time lag by

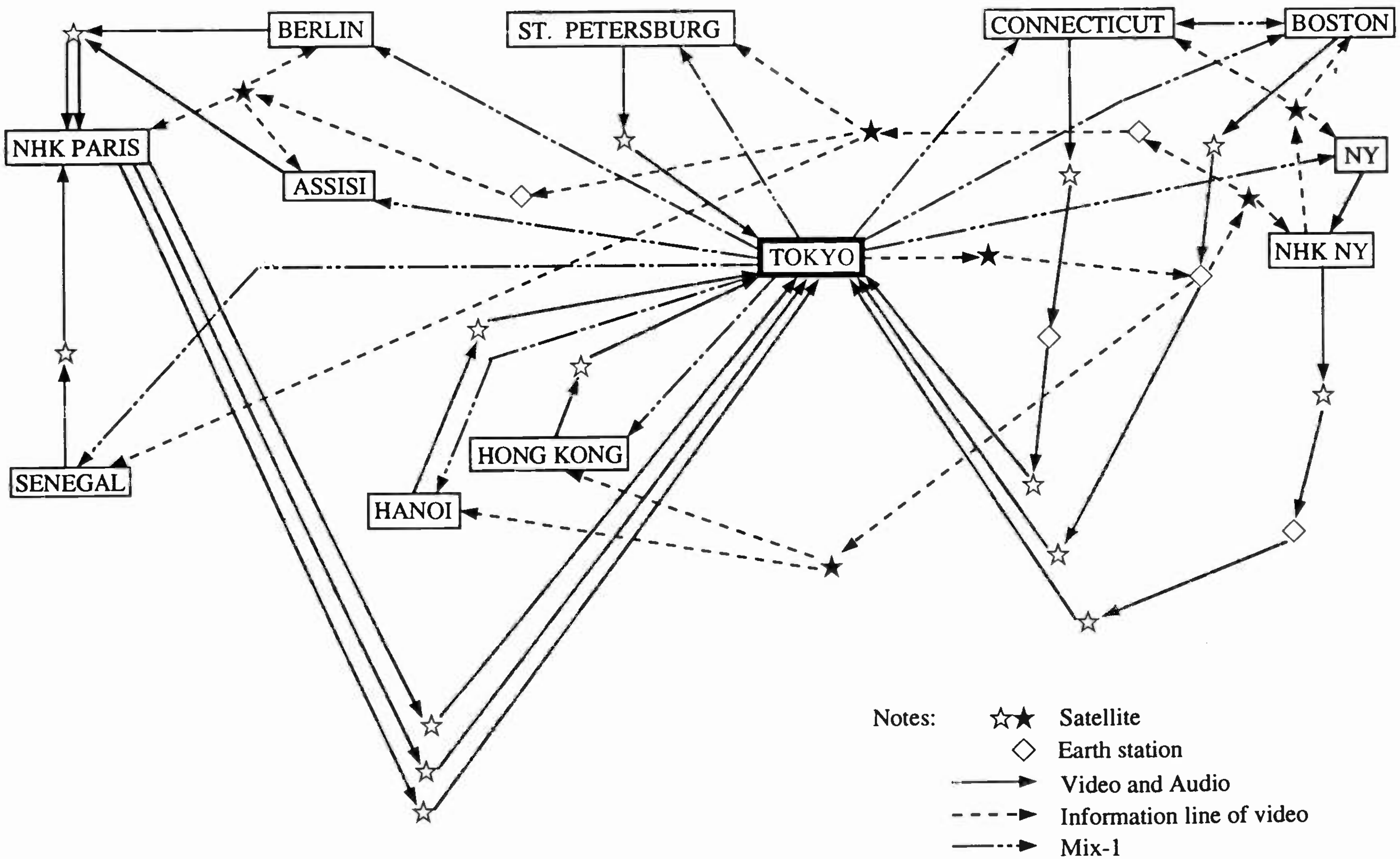

Figure 2. Schematic drawing of satellite relay

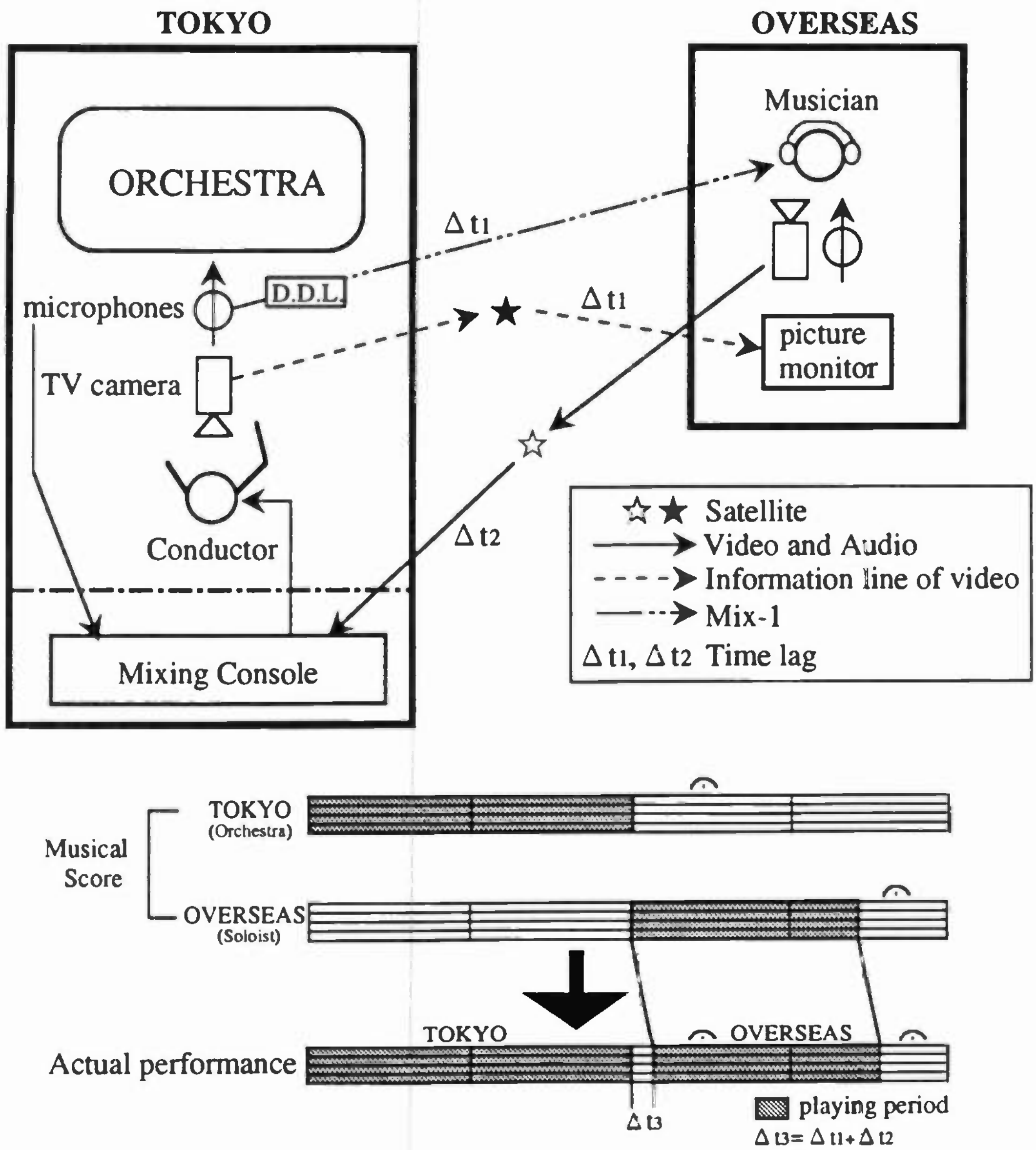

Figure 3. Schematic drawing of fundamental system of simultaneous musical performance

the satellite relay were expected. In the case of this program, there were two kind of time lag. One of them was on the information line of video($=\Delta t_1$). The other was on the video and audio line($=\Delta t_2$) which sent performances of musician from abroad to Tokyo.

The time lag, i.e. Δt_1 and Δt_2, between Tokyo and each place of the world was different with each route.

Video and audio signal had been sent via one or two satellites from each place. On the other hand, it need up to four satellites for sending the information of video. In order to synchronize Mix-1 to information line of video, D.D.L., Digital Delay Line, were inserted into every lines. Delay times of D.D.L. for Mix-1 lines were as follows;

Tokyo to Boston: approximately 1000msec
to Connecticut: approximately 1000msec
to Hong Kong: approximately 750msec
to Assisi: approximately 1400msec
to Senegal: approximately 850msec

Finally each sound of overseas performance reached to Tokyo with different time lag that is $\Delta t_3=\Delta t_1+\Delta t_2$.

In this musical performance, the time difference, Δt_3, had been expected from 1000msec to 2000msec.

As above, for mixing this musical performance, it was not possible to remove the total time lag Δt_3 between orchestra in Tokyo and overseas performance. Of course, if the orchestra mix could be inserted with appropriate delay, orchestra and overseas performance would be synchronized. However, Δt_3 was different to each countries and it was not practical to delay the orchestra mix due to synchronization with video of orchestra performance.

From the beginning, the problem of time lag on mixing had been surmised. So that We had requested a composer to clear this problem by inventing something in the musical score. Example of musical score is shown in Figure 4.

For example, when performance was passed from Tokyo to Boston, orchestra were continuing their playing as 𝄐 , "fermata". Therefore TV viewers did not feel a sense of incongruity even if musician in Boston started his playing a little bit later.

During musical performance, only sound of orchestra in Tokyo had been sent on Mix-1 line for not to confuse musicians by a time lag. Additionally it had planed that telephone line had been connected between Boston and Connecticut, because the musical piece "Satellite Celebration" requested these two musicians in Boston and in Connecticut play simultaneously in tempo. The musicians in those places could not play simultaneously without hearing with each other. In consequence, performance between Boston and Connecticut had went very well.

Although rehearsal of this performance with all musicians together had not been done, because of difference in time and their schedules, the live

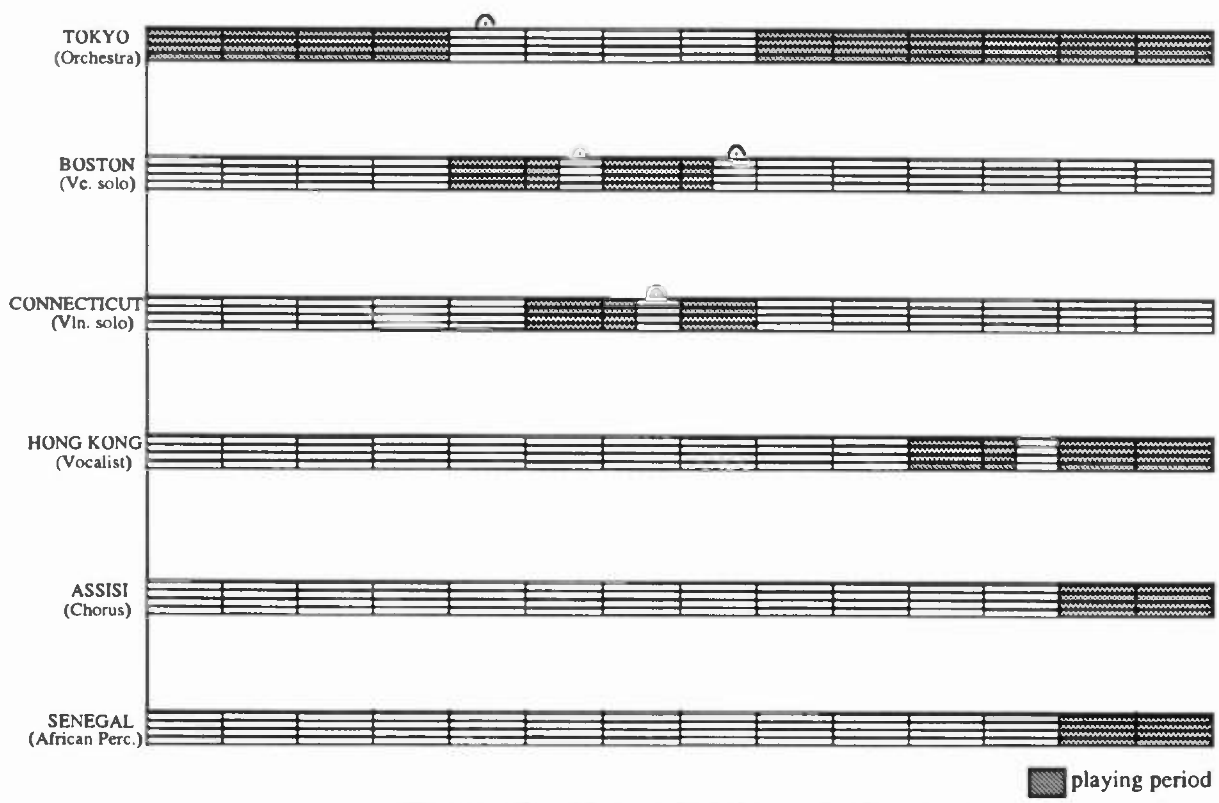

Figure 4. Example of musical score

performance made a great success and so that it had realized the purpose of this program, the cooperation over the world with the music.

Conclusion

This program had been broadcasted in stereo, but audio of satellite relay had been monaural and its quality had not been satisfied as a music program. It would be necessary that transmission by digital signal such as ISDN is examined on next time.

And considering with the simultaneous musical performance, a number of satellite should be minimized as possible.

In the future, if television signal will be able to be transmitted by a fiber cable after being established a technology on B-ISDN, the time lag will be short and the simultaneous musical performance will be more easier.

Acknowledgments

A success of the simultaneous musical performance and interactive satellite broadcasting was due to all staffs that had been investigated and prepared for four months. The author would like to thank all staffs of this program.

ISSUES AND ENGINEERING FOR FRACTIONAL TRANSPONDERS EMPLOYING MULTIPLE DIGITAL SIGNALS

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Abstract

Due to the increasing popularity of compressed digital video, fractional transponder operations will be more common than in the past and will see more part time and occasional feed traffic.

This paper will discuss the engineering issues and tradeoffs that must be considered for successful fractional operations. These issues include earth station equipment (especially receive antenna size) available transponder power, bandwidth, and linearity, and coordination with both co-satellite and adjacent satellite operations especially FM television. In particular the paper will show how the these issues affect the number of digital carriers that can be accommodated, as well as carrier loading within the transponder, carrier operating point, and expected performance.

Introduction

SCPC (Single Channel Per Carrier) compressed digital video systems use QPSK modulation with total bit rates between 3 and 14 Mb/s. The 3 and 6 Mb/s rates are becoming popular among educational and business television users; while other users whose applications are less tolerant of occasional picture artifacts tend to favor the higher rates. In cases where multiple compressed video signals originate from a common point it may be more advantageous to use MCPC (Multiple Channel Per Carrier) systems. In these systems a number of 3 to 6 Mb/s compressed video bit streams are time division multiplexed to create a larger stream on the order of 20 to 30 Mb/s for transmission over a common carrier.

Some of the information presented in this paper is the direct result of measurements and some, particularly the intermodulation noise levels and system design examples, are calculated results. These calculations are based on years of real world experience gained by members of the AT&T-Bell Labs Satellite Laboratory of which the author is a member.

The author begins with a brief overview of QPSK and some of the differences between it and FM including the effects of passage through a non-linear amplifier. Following this overview is a more detailed discussion of the following link design considerations and the tradeoffs between them:

- Transponder Operating Point and Intermodulation Noise.
- Carrier Spacing Methods with Respect to Transponder Bandwidth and Intermodulation Noise.
- The Effect of Transponder Operating Point on Customer Power Level Setting.
- Thermal Noise Performance with Respect to Receive Antenna Size and Uplink Power Levels.

- Offsetting of Digital Carrier Frequency From the Center Frequency of FM Television to Minimize Interference.

A Brief Overview of QPSK

A QPSK signal is composed of two BPSK signals in quadrature. A BPSK signal is a Double Sideband - Suppressed Carrier (DSB-SC) signal in which the modulating signal is a series of pulses. The equivalence of BPSK to a DSB-SC signal is shown by Lucky et al [1]. To make analysis simple, many texts consider the pulses to be square resulting in a modulated signal with no amplitude variations. However in practice the pulses are shaped to optimize bandwidth and reduce intersymbol interference. This results in a modulated signal with amplitude variations. In practice QPSK is a special case of AM, or more precisely QAM, and not a true "angle modulated" signal like FM since the phase of the carrier is not continuously varied during a pulse period. A QPSK signal's spectrum can be changed by passage through a non linear amplifier such as TWTA in saturation. Figures 1 and 2 shows two spectral plots of a QPSK signal at the output of a hardware satellite payload simulator. Figure 1 shows operation through a TWTA operating with a 6 dB output backoff which puts it in fairly linear operation. Figure 2 shows operation in saturation. Note the extended sidebands of the signal in Figure 2. The extra sideband lobes that come up during non-linear operation are sometimes called "Spectral Regrowth" and can be a cause of interference. This is especially true if the saturated TWTA is an uplink HPA (High Power Amplifier). Also please keep in mind that within an earth station signal chain any non-linear device, any amplifier in saturation will cause this effect. Uplink operators who are making the switch to digital would be well advised to check their systems for linearity before transmitting.

Another difference between QPSK and FM is that the QPSK signal spectrum from video encoder modulators tends to be constant regardless of picture content or even if there is no video at all. This is mainly because the modulators scramble the digital signal to prevent long strings of logical zeroes or ones from occurring. Also note that unlike FM the QPSK signal has a simple almost rectangular shape. For most purposes, the signal can be modeled as having a constant average power density over a bandwidth equal to the symbol rate or one half the bit rate. The occupied bandwidth is generally considered to be 1.2 times this value.

QPSK Signal 6 dB OBO

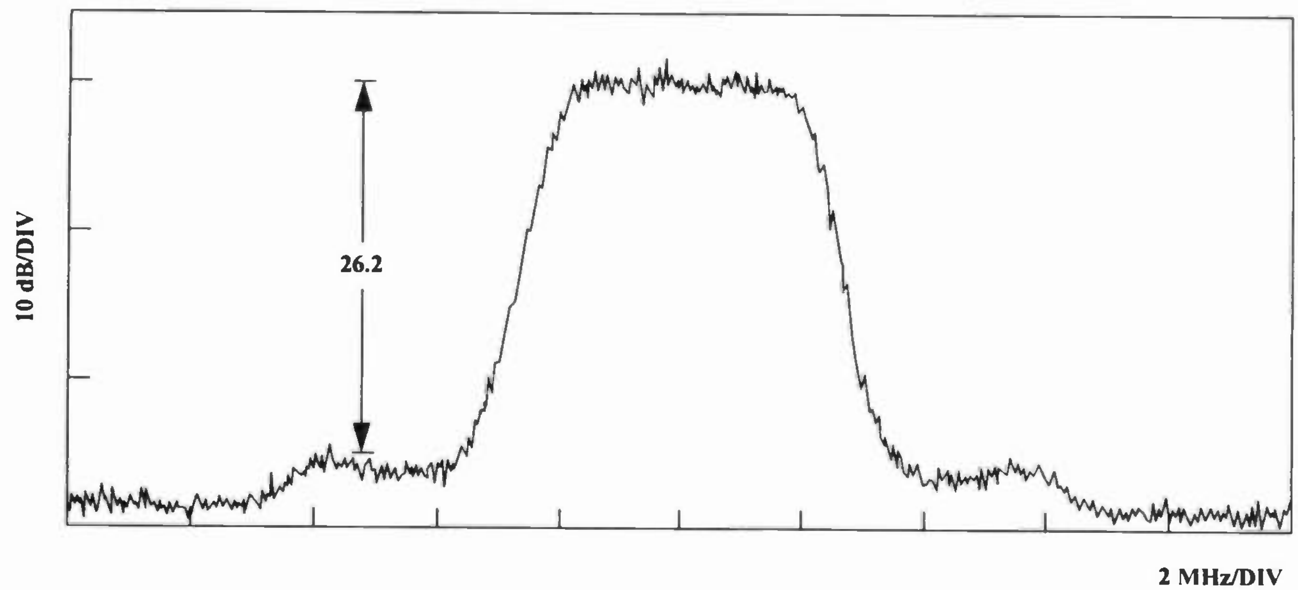


Figure 1

QPSK Signal Saturation

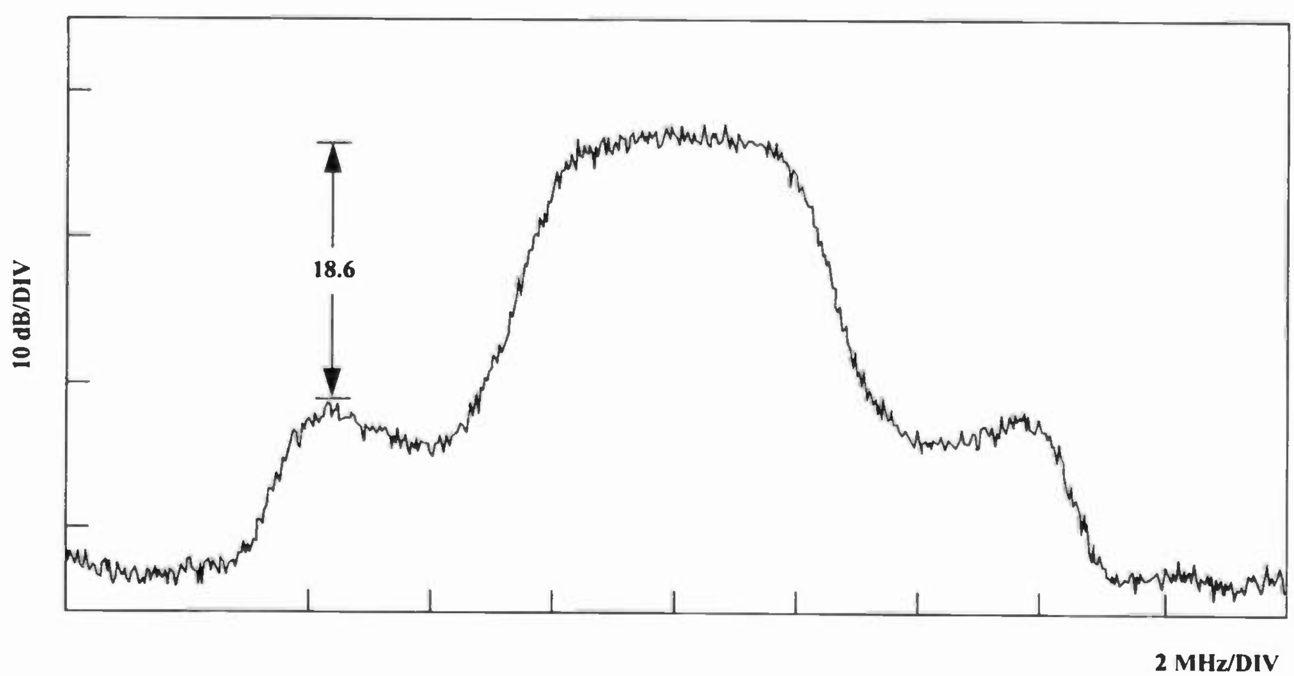


Figure 2

Power, Bandwidth and Linearity Constraints

Cost, ease of installation and aesthetic considerations drive the size of receive antennas towards the smallest size possible resulting in the need for high satellite downlink EIRP levels. However if the transponder is carrying a multi-channel load, it requires a total output backoff of between 3 and 6 dB. This means that from 3 to 6 dB must be subtracted from the saturation EIRP before the remaining power can be divided up among the carriers. This is one reason why a single MCPC carrier may be more desirable than multiple SCPC signals. In the MCPC case the transponder can be operated at or near saturation. A small (0.5 dB) C/N performance penalty may be incurred due to AM-PM conversion effects in TWTAs. Virtually no penalty should be incurred for SSPA equipped transponders. The sideband spectral regrowth energy extending beyond the transponder bandwidth will most likely be well attenuated by the transponder's output multiplex filter.

Three dB total output backoff (OBO) multicarrier operation is possible with certain transponder channelization plans in a carefully controlled uplink situation, such as a single uplink site or a small number of well understood earth stations with experienced operators. In the more general case especially those involving a large number of uplink stations which are not well known to the satellite operator, a total output backoff in the area of 5 to 6 dB is recommended.

As a transponder is driven harder (lower output backoff) the level of intermodulation noise rises higher. This noise is generally not flat across the transponder but will have peaks and valleys depending on the signal bandwidth and the spacing between the signals. It has been shown by Babcock [1], and by Fang & Sandrin [2], that it is possible to create a frequency plan for a transponder (amplifier) such that most of the intermodulation noise power falls in between the signals resulting in a best C/Im (Carrier-to-Intermodulation noise). The necessary condition for these plans to work is to have much more bandwidth in the transponder than is taken up by all of the carriers combined. For example: If 6 carriers are to be assigned to a transponder with minimum intermodulation noise, there must be enough bandwidth available for 18 carriers. Using the methods described by the cited authors, 6 out of the 18 possible slots are chosen for occupancy. A design example follows: Assume that six 3.3 Mb/s QPSK carriers are to be loaded on a 54 MHz wide Ku transponder. The occupied bandwidth of each carrier is 2 MHz and the desired center frequency separation is 1.5 times the occupied bandwidth resulting in 3 MHz wide channels. Eighteen channels at 3 MHz each will exactly fill the 54 MHz transponder bandwidth. The individual channels are numbered 1 to 18 starting at the low end of the transponder as shown in Figure 3 below:

Figure 3

12003	12006	12009	12012	Downlink Freq. in MHz	12051	12054
1	2	3	4	- - - - - - - - - - - - - - - - - -	17	18

54 MHz Bandwidth Divided Into Eighteen 3 MHz Channels

The possible channel assignments for minimum intermodulation noise are shown in Table 1. Babcock has only one assignment plan while Fang & Sandrin have three.

Table 1

	Babcock	Fang & Sandrin
Channel Numbers	1, 2, 5, 11, 13, 18	1, 2, 5, 11, 13, 18
Channel Numbers		1, 2, 5, 11,16, 18
Channel Numbers		1, 2, 9, 12, 14, 18

The following tables give channel assignments and C/Im (Carrier-to-Intermodulation) ratios at 3 and 6 dB total output backoff for three types of channel spacing: evenly spaced carriers, randomly spaced carriers, and carriers assigned by the Babcock - Fang & Sandrin plan. The values of C/Im are computed using the Tel/Com Sciences, Inc SIAP2 program.

Table 2 Evenly Spaced Carriers

Channel Number	Center Freq (MHz)	C/Im 6 dB OBO	C/Im 3 dB OBO
2	12006	22.5 dB	19.2 dB
5	12015	20.9 dB	17.6 dB
8	12024	20.2 dB	17.0 dB
11	12033	20.2 dB	17.0 dB
14	12042	20.9 dB	17.6 dB
17	12051	22.5 dB	19.2 dB

Table 3 Randomly Spaced Carriers

Channel Number	Center Freq. (MHz)	C/Im 6 dB OBO	C/IM 3 dB OBO
1	12003	37.0 dB	33.7 dB
5	12015	33.4 dB	30.3 dB
7	12021	28.4 dB	25.1 dB
10	12030	25.6 dB	22.3 dB
15	12045	27.7 dB	24.5 dB
18	12054	28.6 dB	25.3 dB

Table 4 Babcock - Fang & Sandrin Spacing

Channel Number	Center Freq (MHz)	C/Im 6 dB OBO	C/Im 3 dB OBO
1	12003	39.9 dB	36.7 dB
2	12006	40.0 dB	36.8 dB
5	12015	36.7 dB	33.4 dB
11	12033	36.4 dB	33.1 dB
13	12039	36.4 dB	33.1 dB
18	12054	37.5 dB	34.3 dB

As these tables show, evenly spaced channels produce the worst C/Im situation. This is because the intermodulation products fall right on top of the carriers. As predicted the Babcock - Fang & Sandrin spacing plan results in the best C/Im performance, but also note that random spacing can produce C/Im results that are quite acceptable. This is encouraging because the need to avoid certain parts of the transponder due to interference considerations could result in a compromise channel plan that looks more like random channel spacing.

In cases where there are signals of various bandwidths the picture becomes more complicated. For example suppose it became necessary to load four 3.3 Mb/s signals and one 6.6 Mb/s signal onto the same transponder. One could still use Babcock's plan by assigning the 6.6 Mb/s signal to the band occupied by channels 1 and 2. The 3.3 Mb/s signals can then be assigned to the remaining channels (5, 11, 13, & 18).

Even though careful channel spacing can allow a 3 dB OBO from a C/Im point of view, careful attention should be paid to the effect that changes in transponder load or a change in a single customer uplink drive will have on the downlink levels of the other signals. In a five channel QPSK 3 dB output backoff case recently investigated by the author, the output level of any one channel increased by 0.5 dB for every channel removed from the total load. This means that the downlink EIRP of any one signal can change by up to 2 dB with no change in its uplink drive level depending on the other traffic on the transponder. This also means that the satellite control center will have to take the existing transponder load into consideration when setting the level of an additional user. In the five channel QPSK case the satellite control center will have to set the level of the first on board customer's signal 2 dB higher than normal to compensate for the fact that his level will be pulled down toward the normal full load operating power, 0.5 dB at a time, by each additional customer who comes on.

One possible way to simplify the level setting procedure in these low output backoff cases is to maintain a low level CW pilot tone on the transponder whose EIRP is 10 or more dB below that of a single channel. The pilot level will move with the changing load conditions and will serve as a benchmark indicating what EIRP density the latest customer should be set to. Since the EIRP of the pilot is very low compared to any one of the signals on the transponder it will not materially effect the transponder operating point. If the transponder is being used as part of a network that has a signal serving as a "Home Channel" which is active continuously during business hours, the "Home Channel" will perform the same function as the pilot. However if either of these approaches are used, the pilot or home channel transmitter should be equipped with uplink power control especially when operating on the Ku band.

In contrast, operations at total output backoffs of 5 to 6 dB can be considered virtually linear. In these cases the downlink level of any one signal does not noticeably change as a function of total transponder load making downlink EIRP setting and monitoring straightforward. This more linear type of operation also provides more protection against single uplink station drive fluctuations.

Thermal Noise Performance Issues

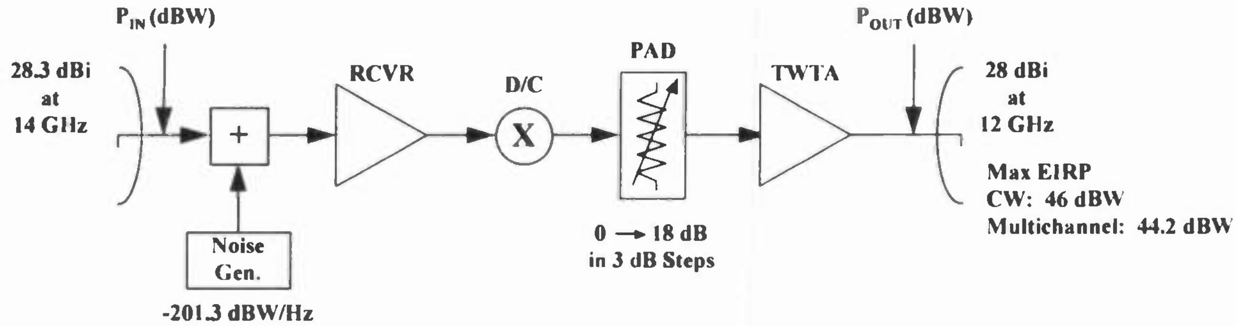
When small receive antennas (1.8 meter at Ku) are used the downlink C/N (Carrier-thermal Noise) ratio tends to be the limiting factor in a well designed system, which is in essence a two hop microwave radio circuit with a linear translator repeater in the middle.

The receiving antenna on the satellite looks down at the earth and "sees" the uplink signal plus some noise radiated by the earth itself since it is a massive body at a temperature of about 290 K In addition the receiver and other elements of the transponder generate some noise of their own. In order to keep analysis simple, the earth, antenna, and transponder electronics are modeled in Figure 4 as noiseless devices and all of the noise that they would otherwise make is shown as coming from a fictional noise generator applied to one input of a summer. The other input is the uplink signal. The combination of signal and thermal noise, known as the uplink C/Nth ratio passes through the frequency translator and then to the power amplifier (TWT or SSPA as required). The first stage of the power amplifier contains the flux control attenuator or "Pad". The pad adjusts the overall gain of the transponder and by doing so determines how strong of an uplink signal is required to saturate the transponder.

In the downlink, the earth station receiving antenna looks out into space and for elevation angles greater than about 25 degrees it sees a noise temperature of about 36K, mostly from earth noise coming in the sidelobes of the antenna pattern. The antenna is followed by a feed mounted LNA or LNB (Low Noise Amplifier or Low Noise Block converter) with noise temperatures in the 70 to 100 K range for the Ku band. The LNA(B) drives a feedline that brings the signal to the receiver where demodulation and decoding takes place to get video and audio.

Taking all this together it can be seen that the overall C/Nth at the receiver is a combination of the uplink and downlink C/Nth ratios. Therefore if the downlink C/Nth is the limiting factor in the link, the uplink C/N should be made as high as possible so as to have minimal effect on the overall C/Nth. In FM TV operations the normal procedure was to saturate the transponder in order to get maximum downlink C/Nth. Improvements to the uplink C/Nth were made by raising the pad on the satellite so that the transponder would saturate at a higher uplink EIRP. With a multicarrier system load the transponder must be operated linear or semi-linear. This requires that the uplink EIRP level and the pad setting must be coordinated such that the proper transponder operating point is maintained as well as the necessary uplink C/Nth.

The best way to illustrate the effect of C/Nth ratios is to use a system design example. The system is a six 3.3 Mb/s QPSK carrier design assuming the model Ku satellite shown in Figure 4. The transponder TWTA is operating with a total output backoff of 3 dB and assume that the C/IM could be made no better than 20 dB. The receiving antenna is a 1.8 meter with



$P_{OUT} = -.0009\ X^3 - .0552\ X^2 - .136\ X + 16.08$

$X = (P_{IN} - PAD + 112)$

P_{IN} is Full Transponder Load Input Power

Figure 4

a gain of 44.8 dBi and a total receive system G/T of 23.2 dB/K. The required system Eb/No is 7 dB with the No consisting of both thermal noise and intermodulation plus interference noises. The 7 dB Eb/No corresponds to a C/N of 10 dB in a 1.65 MHz bandwidth.

Table 5 gives calculated uplink EIRP/carrier, uplink C/Nth, satellite pad setting, downlink C/Nth, overall C/Nth and the margin against system failure for both clear sky and downlink rain fade conditions (Crane Model Region D2, 99.95% availability).

Table 5 C/Nth and Margins for 1.8 Meter Receive Antenna

Uplink EIRP (dBw)	Pad dB	Uplink C/Nth (dB)	Downlink C/Nth (dB)	Total C/Nth (dB)	Margin Clear(dB)	Margin Rain (dB)
55.7	3	15.3	18.8	13.7	1.1	-1.3
58.7	6	18.3	18.8	15.6	2.6	-0.6
61.7	9	21.3	18.8	16.9	3.7	-0.1
64.7	12	24.3	18.8	17.8	4.3	0.1
67.7	15	27.3	18.8	18.3	4.6	0.3
70.7	18	30.3	18.8	18.5	4.8	0.3

As can be seen in the above table, raising the EIRP results in a dB for dB improvement in the uplink C/Nth and a somewhat less than that improvement rate in the total C/Nth until 61.7 dBw is reached, thereafter the improvement in total C/Nth is slight. This is because at lower uplink powers the uplink noise dominates. As the uplink power is raised the uplink C/Nth becomes great enough such that the downlink C/Nth and C/Im takeover and begin to dominate. That is to say that with very high uplink power the uplink looks noiseless compared to the downlink. Also notice that the satellite pad setting was increased with the uplink power so that the transponder operating point (total output backoff) remained at the same value. The step size in uplink power that can be made is then based on the step size of the satellite pad.

Also keep in mind that for a fixed antenna size higher EIRP means higher HPA power. The HPA must be sized so that it can be operated with sufficient OBO such that intermodulation noise generated within it is not a significant contributor to the total noise. If the HPA is a TWTA one must consider the levels of intermodulation products that extend beyond the transponder frequency band since TWTA's are very broadband devices. These products can be properly attenuated either by linear operation or the addition of an external channel filter. Even in the single channel per HPA case non-linear operation will result in unwanted sideband regrowth in the QPSK signal's spectrum which could be a source of interference. If the uplink station in this example were an SNG truck transmitting a single channel with a 2.4 meter antenna and 2 dB of feed loss, the HPA output power would have to be 17.7 dBw when working with a 12 dB satellite pad. If the HPA output backoff is 6 dB then the HPA must be capable of 23.7 dBw (234 watts). This means that the typical 300 watt TWT HPA would be limited to operations with a pad of 12 dB or less.

If the receive antenna size in the design example is increased to 3 meters, then the calculated C/Nth, margins and other parameters are shown in Table 6.

Table 6 C/Nth and Margins for a 3 Meter Receive Antenna

Uplink EIRP(dBw)	Pad dB	Uplink C/Nth (dB)	Downlink C/Nth (dB)	Total C/Nth (dB)	Margin Clear (dB)	Margin Rain (dB)
55.7	3	15.3	23.3	14.7	1.9	0.6
58.7	6	18.3	23.3	17.1	3.7	2.0
61.7	9	21.3	23.3	19.2	5.1	2.8
64.7	12	24.3	23.3	20.8	5.9	3.3
67.7	15	27.3	23.3	21.8	6.5	3.6
70.7	19	30.3	23.3	22.5	6.7	3.7

With the larger antenna it is now possible to achieve a 3.7 dB clear sky margin with 3 dB less uplink power than in the previous case. Also we now have margin during the rain fade whereas before the margin had just extinguished during the rain fade. Note that the Downlink C/Nth is 23.3 dB and the C/Im is still 20 dB. This means that the intermodulation noise is starting to control the downlink margin. This will become more evident with even larger antennas. Also if the C/IM dominates the downlink then the difference between clear sky and rain margins will become smaller. This is because during a rain fade both the signal and intermodulation noise are attenuated together. In a downlink thermal noise dominated situation the rain attenuates the incoming signal but the LNA noise remains constant.

Interference

FM television is probably the most troublesome interference source due to the nature of its spectrum which contains high peaks and is very much picture content dependent. The region 2.5 MHz either side of center frequency there are intermittent spectral peaks that within a 100 KHz bandwidth can contain powers equal to the unmodulated carrier power. Of course only one peak can occur at a time. Narrowband SCPC signals generally should not be placed same satellite crosspol or 2 degree adjacent satellite co-pol within this 5 MHz wide central region.

A series of experiments were performed by A.R. Rehwinkel [4] in which FM TV interference was added to an SCPC signal until a certain BER (Bit Error Rate) was achieved. The SCPC signal was then offset from the center of the FM signal and the interference level was raised until the same BER occurred. This was repeated for different values of frequency offset. The amount of extra interference power that had to be added to counteract the beneficial effect of moving away from the center of the FM interferer is referred to as the C/I advantage. In other words the extra interference power that has to be added is equal to the additional protection that the QPSK signal receives by being offset from the FM center frequency. Figure 5 is a plot of C/I advantage versus frequency offset from an FM interferer center frequency for a 6.6 Mb/s QPSK signal. Figure 6 shows the results achieved for an 11.3 Mb/s QPSK signal. Given that the 3 dB or "noise" bandwidths of the receivers are 3.3 and 5.7 MHz respectively, it can be seen that once the QPSK passband skirt moves far enough away from the FM center frequency such that the passband is beyond the +/- 2.5 MHz region of peaks, the C/I improves.

C/I Advantage for Center Frequency Offset

6.6 Mbps, 3/4 Rate FEC, Video Interferer

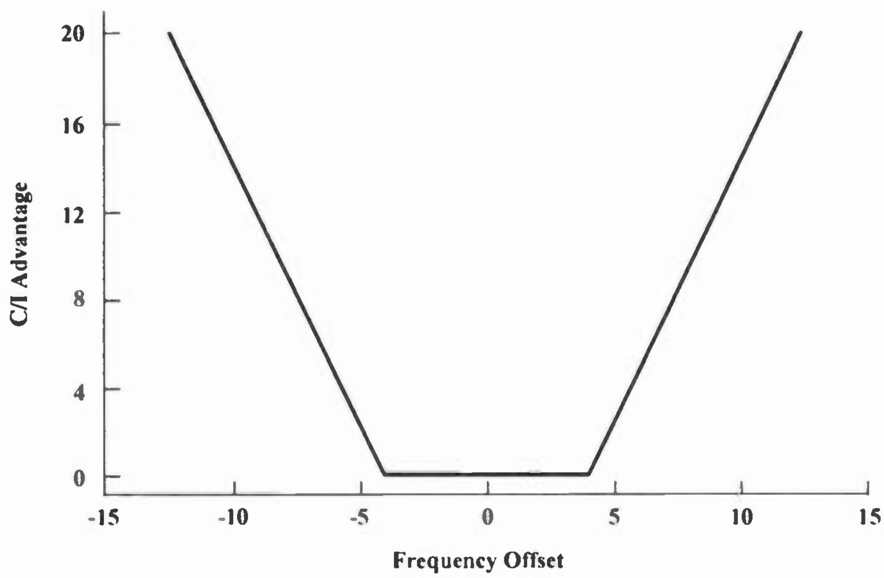


Figure 5

C/I Advantage for Center Frequency Offset

11.3 Mbps, 3/4 Rate FEC, Video Interferer

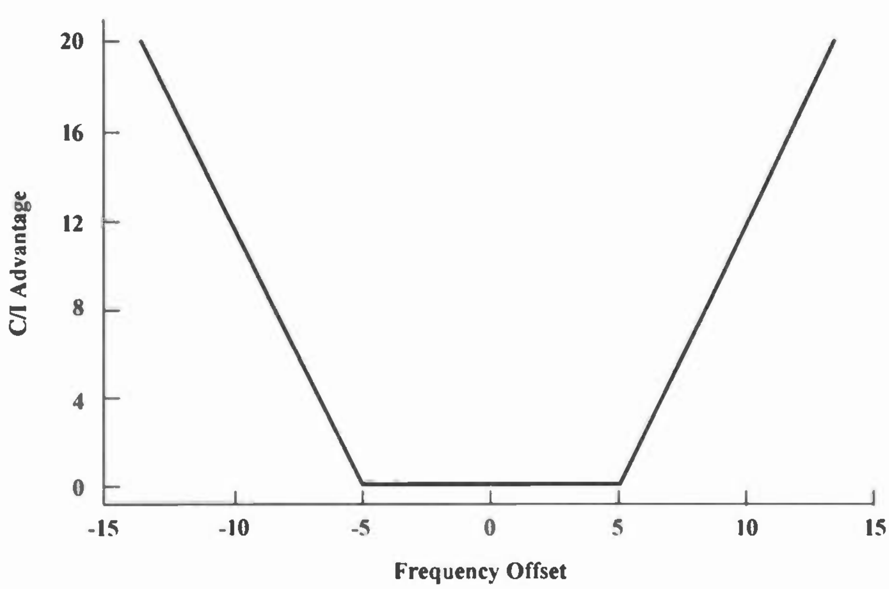


Figure 6

Acknowledgment

The author wishes to acknowledge the contributions of the members of the AT&T Bell Labs Satellite Laboratory and the AT&T Skynet Engineering and Operations Groups.

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LIVE COMPUTER GRAPHICS

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Abstract

General purpose graphics workstations are a valuable tool to the television animator, providing a powerful and responsive environment in which to create models, set up animation, and preview effects. The creation of these visuals generally does not happen in real time, but off-line, taking up to several minutes per frame.

The graphics performance and features of these general purpose workstations has continued to improve, to the point now where the graphics which can be generated in real-time is of a quality suitable for use on-air. The process for creating software and imagery to be presented in real time is completely different from traditional computer animation and effects, both in how the software is organized and how the graphics should be used on air. This paper looks at two example applications and the technology and design issues surrounding their use.

The Yacht3D System

In 1991, Trimble Navigation, manufacturer of electronic global navigation equipment and a sponsor of the America's Cup, approached Silicon Graphics Inc. (SGI), manufacturer of graphics workstations, for their help in providing some graphical elements related to the boats' relative positions and distance from the racing marks for use on air. SGI devised a system which would provide 3D images of boats, viewable from any direction by virtual cameras, including the trails indicating the path the boats had taken. A limited, though very successful system was built for the 1992 broadcast, and in 1995 a new system was created, based on the experience gained and ideas generated during the previous event.

GPS satellite receivers aboard the yachts calculated their position twice a second with sub-meter accuracy. Electronic compasses and interfaces to the yacht's onboard computers monitored the boats' heading and performance parameters. This information was continually transmitted to radio receivers on shore. A single workstation on shore listened to data from all the yachts and several officiating boats, then retransmitted the data over an Ethernet network to several computers creating graphics or analyzing the data. Two Silicon Graphics Reality Engine deskside computers, a primary and a spare, generated the animation. These computers were controlled by a single SGI Indy computer located in the graphics truck. In addition, an SGI Indigo2 computer was located in the commentators' studio for analysis of data and creation of scenarios. A separate Indy collected all race data and acted as a server for database requests when replays were called up. Although all these functions could have been done on a single computer, distributing the task allowed for tremendous reliability and flexibility, where several people could be working in parallel on different aspects of the graphics coverage.

The range of possibilities of this new technology is hinted at by the features included in the Yacht3D system:

- *Realistic 3D virtual-reality views of the race*, including yachts, their paths through the ocean, racing marks. All hulls and sails, ocean surface, and sky are fully texture-mapped with real photographic images for realism and visual interest.
- *Imaginary lines made visible*. The race course appears as a featureless ocean to the camera, so the graphics can illustrate important boundaries, such as the start-finish line, the laylines, the ahead-behind line.
- *Automatic camera animation*, which tracks yachts as they move, does smooth moves to any viewing angle, smooth transitions to any framing of yachts and marks, does compensated moves from sea level to miles above the ocean for a "God's-eye view", all programmed as simple button-press presets.
- *Ancillary stats and graphics*, boat speed, wind direction & speed, lead, projected time to marks and laylines are represented as animated text or graphical elements. A range of screen layouts is available to the director at the push of a button.
- *Replay*. The paths of the yachts are stored on disk in

the computers, so the computers can replay a previous segment of the current race, perhaps at high speed, or a segment of a previous day's racing.

- *Simulations*. A specially-designed sailboat simulator lets a commentator create imaginary tactical sailing situations quickly, then display them using the live graphics system to illustrate rules or maneuvers. Scenarios can be generated in advance and called up quickly, or they can even be created within a couple minutes for use immediately.
- *Transitions into race and between races*. A satellite photograph of San Diego was digitized and mapped into the scene to provide an effective transition from a map of Southern California into the race course just off-shore. Since several races often occurred simultaneously, using the map to transition from one racecourse to another made the coverage particularly clear and interesting.
- *Air traffic control*. The backup graphics computer was set up to provide continuous overhead views of each of the races in progress. Though never broadcast, this view was a tremendous aid to the producers in planning, coordinating, and timing the live coverage of the several races happening in parallel.

Using General Purpose Graphics Workstations in Live Broadcast

Graphics workstations combine the ability to create 2D & 3D animation, manage changing textures, do image processing and effects, generate sound, perform external device sensing and control, plus do powerful communication and calculation. It is possible to write an application which will run on anything from a sub-$10,000 computer to a $1,000,000+ multi-processor, multi-graphics pipeline supercomputer, to a distributed network of heterogeneous machines.

The use of these machines in Live Broadcast has been limited by the quality of graphics which could be generated in real time, preferably at 60fps, by the cost of high-end workstations, and by the non-deterministic behavior of these machines: A great strength of Unix-based workstations is their ability to run several tasks simultaneously, even transparently. The downside of this feature is that each task has no guarantee that it will be able to respond to an external event in a given amount of time, or that it will be guaranteed a certain amount of processor time over a given interval. This situation can be controlled with good programming techniques, and by the use of new operating system features being introduced continually.

Though general purpose machines offer tremendous flexibility, to achieve the machine's maximum performance, it is important to do several things correctly. For example,

- *Separate Control from Animation*. Running more than one graphics activity on a single workstation introduces overhead in the graphics subsystem to switch between them. User-interface elements such as buttons, sliders, and menus should run on a separate machine and communicate with the graphics process via the network.
- *Manage Complexity and Culling of Models*. In general, the geometric detail necessary to make a model look good when viewed up close is to too high to use when several objects are on the screen at once. This problem is well-known to people developing flight simulators and is referred to as managing level-of-detail. Several versions of a model at different complexities must be maintained and the appropriate one drawn in each situation. Also, objects which are obscured or behind the camera must not be drawn to avoid bogging down to graphics hardware.
- *Separate the Animation process from the Drawing process*. To feed data into the graphics hardware at the highest possible rate, it is important to dedicate a process to doing just that, separate from the process to create the animation and manage the geometry. Software libraries such as SGI's "Performer" can help manage the job and provide low-level tools for efficient drawing.
- *Accurate Synchronization & Time Management*. Often several parts of a real-time system run at different rates. Sometimes it is necessary to introduce delays into parts of a system to ensure that the graphics are accurate or in sync with external video. Often graphics must be generated field-accurately with respect to external video equipment. All these situations require absolutely accurate management of synchronization and timekeeping, often between several machines on a network.
- *Absolute control of the video pipeline*. Some applications involve taking video into a workstation and using it as a texture map. All applications require sending 30 or 60fps graphics out video interfaces. Unless the buffering, timing, and filtering of every field is under absolute control of the application, huge artifacts can appear.

What Makes for Successful Live Graphics

In creating the Yacht3D system, and in evaluating several other potential applications, we have identified several useful criteria for a successful live graphics system:

- *If you can do it with videotape, use videotape*. You have infinitely more control of the visual if you can use traditional animation packages and techniques.

Live graphics works best where the data really is live, such as vehicle or object tracking, visualization of changing data, or in cases where there must be an extremely short production turnaround, such as in live news coverage.

- *Absolutely first-rate graphics*. Broadcast does not allow for technical flaws, such as aliased edges, dithered or banded colors, dropped or mis-ordered fields. Nor does broadcast allow for less than first-rate graphic design - in composition, object appearance, motion, or important nuances such as shadows, reflections, highlights, and light flares.
- *Complement the other visual elements*. In Yacht3D, it was tempting to create "gee-whiz" imagery which duplicated the helicopter shots and moves in virtual reality, but the whole goal was to complement the live pictures, by offering overhead map-style views, easy-to-read schematic views of race strategy, and otherwise impossible shots.
- *Fit into the flow of production*. Yacht3D operation was designed in such a way that the director could cut to it, have it go through its precise moves in a few seconds, then finish cleanly allowing the director to cut back to the live picture. Such scripted improvisation required extensive use of automatic animation and button-driven presets to control the motion. In addition, the graphics must support the larger style of the show. For example, it is easy to generate and analyze replays on the computer, but the style of television coverage emphasized the action happening live, so the graphics capabilities had to be developed accordingly.
- *Use the network*. Just about any live graphics application can benefit from making data and control accessible over the network. All our projects make extensive use of high-speed high-interaction communication between independent machines.
- *Rapid prototyping, rapid modification*. The way these graphics are used changes daily, even during a broadcast. A tight relationship between development and production, and tools for rapidly creating new presentations and modes of operation is crucial to exploiting the full flexibility of these systems.
- *Cheat to the full extent of the hardware*. In a real-time environment, we do not have the luxury of all the features and effects available to a traditional non-real-time animator. By keeping a bag full of clever graphics tricks, and understanding the strengths and capabilities of the graphics hardware, we can still do pretty well.

Other Applications

In early 1995, some of the engineers from Silicon Graphics who created Yacht3D formed Shoreline Studios in Mountain View, California to work full time on creating live computer graphics applications. The first project completed by Shoreline was the software for Fox Sports' FoxTrax system. Fox designed the FoxTrax system to use sophisticated tracking techniques to locate players and objects in sporting events and generate a graphical highlight over the video picture to help the viewer follow the game. Its first implementation appeared January 20, 1996 in the broadcast of the NHL Hockey All-star game. Electronics embedded in the puck allowed it to be detected by sensors around the rink, and computers calculated its position and displayed a faint highlight over the action. When a shot was made on goal, the highlight turned into a red comet tail and the speed of the shot appeared as a separate graphic on the screen.

The FoxTrax system for Hockey consists of several components:

- Specially made pucks are ringed with 20 infrared emitters which strobe for 1/10,000 sec 30 times per second.
- 20 IR photodetectors mounted around the rink detect the strobing and keep a master Puck Sync signal in phase with the puck.
- 8 IR-sensitive cameras mounted in the rafters and around the arena are shuttered in sync with the puck and capture the image of the rink in which the puck appears as a bright spot. The image is captured by custom pixel-processing boards mounted in a PC.
- The broadcast cameras for which a highlight is to be generated are mounted on instrumented heads and their pan and tilt position, as well as zoom and focus positions, are captured 30 times per second by additional PC's.
- A single SGI Indigo2 Impact reads the data from all IR camera PC's and broadcast camera tracking PC's, rejects false targets, calculates a 3D fix for the puck 30 times per second, then interpolates to create 60 field per second animation of a foreground element and independent linear key signal. The keying currently happens downstream of the video switcher, so the Impact also monitors the production switcher tally to decide, on a frame-by-frame basis, the camera for which it is generating the highlight.
- For instant replay, the VTR's record clean video, then in replay, the Impact monitors VITC timecode from the active deck and generates the graphic highlight for that frame from puck and camera data stored on disk. The look of the highlight can be changed as desired.
- The Impact detects shots, deflections, moves between zones, etc., and can synthesize sound effects to emphasize these events.

- The communication, calculation, display, and video output process introduces a delay of about 5 video frames, so the program video and audio are delayed to ensure the graphics and sound effects are in perfect sync with the live video and sound.

FoxTrax lies on the other end of spectrum from Yacht3D in a number of areas: The graphics are relatively simple - a fuzzy disk or ring, changing to a comet trail, accompanied by some text. The communication and calculation requirements are much more demanding - locating and generating a 3D fix on a small fast-moving object 30 times per second, with sub-millisecond timing accuracy, then displaying a constantly changing highlight plus external key 60 fields per second, all with only a few frames of delay. The fact that data could be collected from 8 separate sensors and 6 broadcast camera trackers, complex fix calculation and false target rejection done, and 60fps graphics drawn all on one desktop SGI Indigo2 Impact workstation provides a glimpse of the exciting possibilities with this technology.

Future Directions

As new applications of live graphics to different kinds of sports and news broadcast are explored, we expect to see more demanding uses, more compelling visuals and effects, and more complex animation.

The ability of live graphics software to run completely automatically introduces the possibility of broadcasting additional channels to accompany live coverage of an event, channels which are completely synthetic and custom targeted at one particular aspect of that event.

The ability to make exciting and informative pictures from data has many applications with Internet browsers, offering the possibility that every participant on the net has the ability to play director, calling up whatever graphics the viewer desires to accompany a sporting or news event.

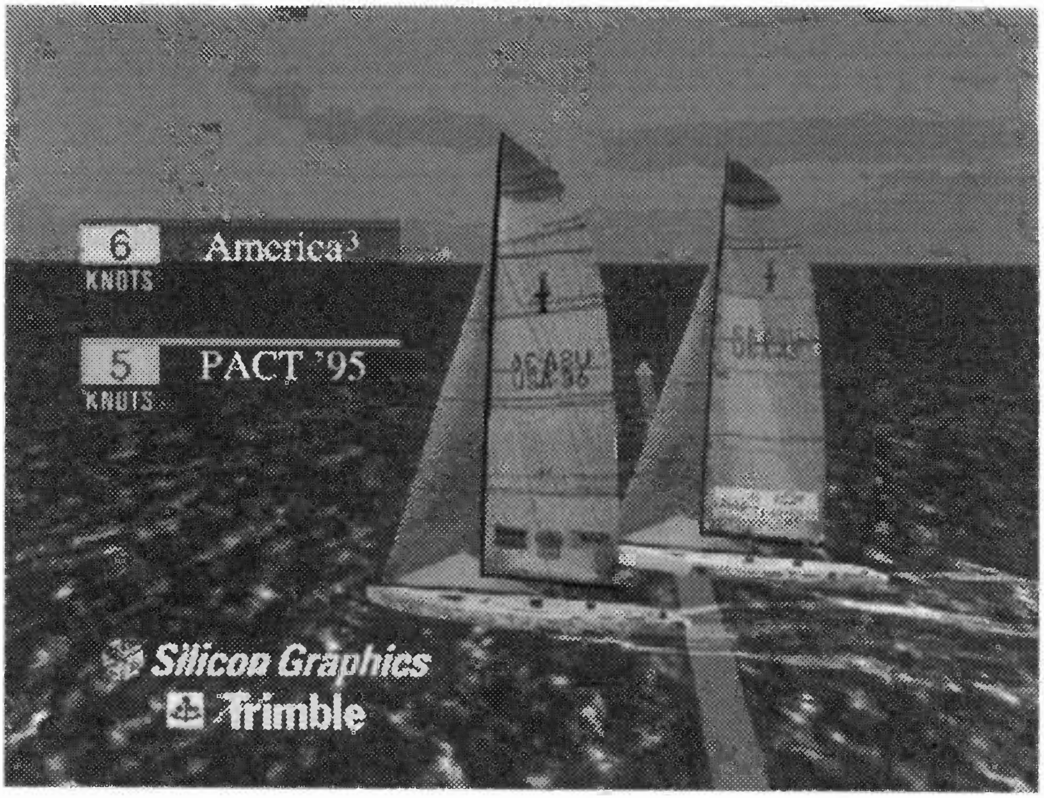


Figure 1 - 3D Texture-mapped Racing Yachts Cross the Starting Line

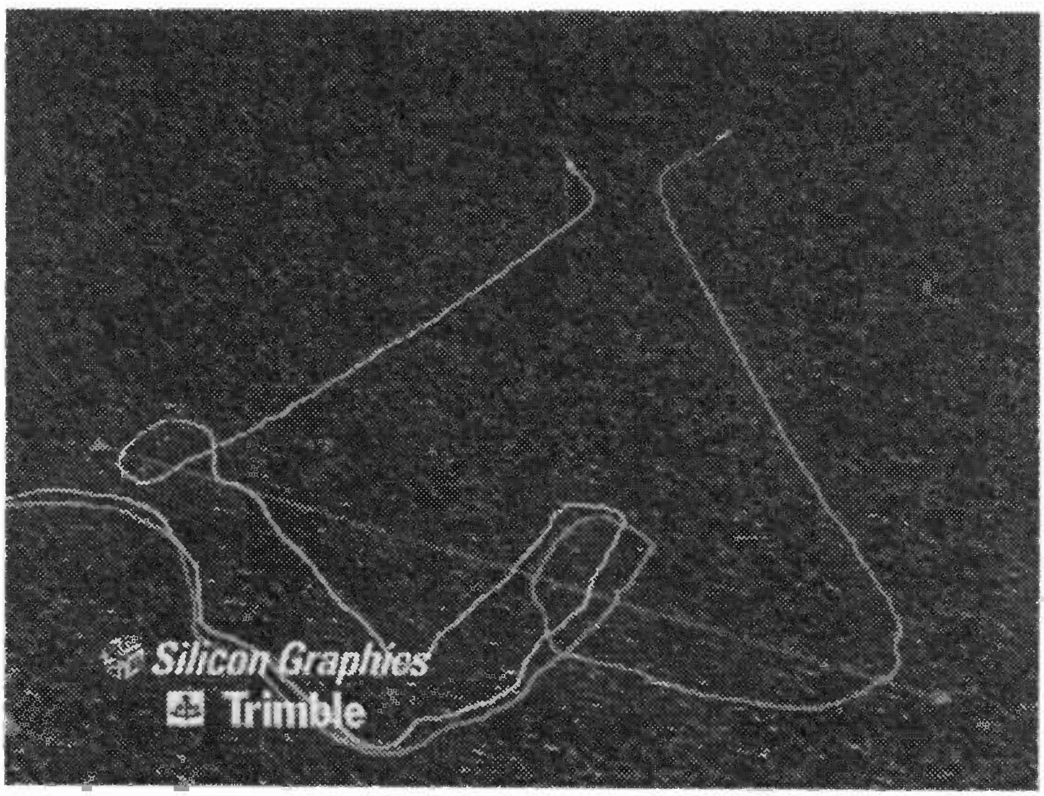


Figure 2 - "God's-eye" View of Starting Line Showing Boat Trails

Figure 3 - Shot on Goal Becomes a Comet in FoxTrax

THE LONG WAY TO CHHOMOLUNGMA (MT. EVEREST) - PROCESS OF OVERCOMING DIFFICULTIES OF PRODUCTION IN HDTV

Fumihiko Saito and Shigeru Kurata
NHK
Tokyo, Japan

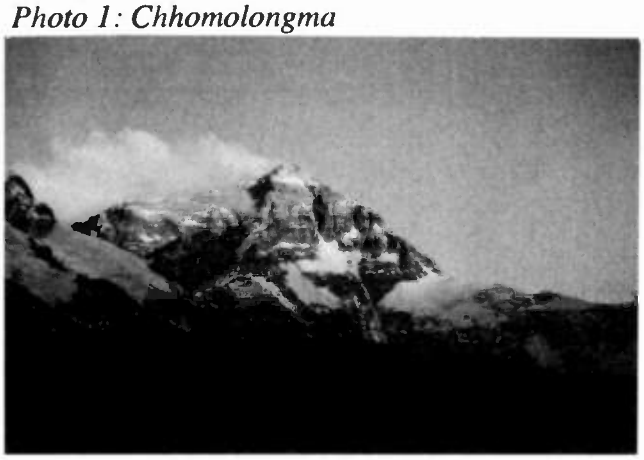
ABSTRACT

Chhomolongma, or Mt. Everest , at 8848 meters above sea level, is the tallest mountain in the world. Chhomolongma was first scaled by an English party in 1953. But the harshest, northeastern ridge remained unscaled despite many top climber's attempt through the years. The ridge was finally scaled in May, 1995, by a Japanese party from Nihon University in Tokyo.

The Nihon University party was accompanied by an HDTV crew from NHK, and the footage was broadcast as the documentary,"The Long Way to Chhomolongma." The period of time necessary to shoot the documentary was four long months of enduring altitudes of more than 5000 meters and its consequences, namely the extremely low temperatures and thin air.

This report will explain the experience of filming in HDTV and the process of overcoming difficulties encountered, using footage from the shoot.

Photo 1: Chhomolongma



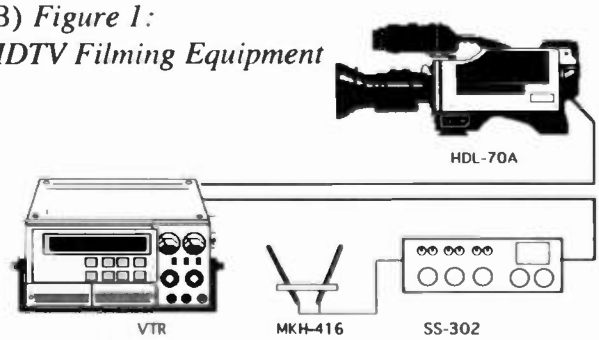
1. Equipment and Crew

Special care was taken to ease the process of acclimatization to the high altitudes, and ample supplies, such as food and fuel, were taken on the shoot. Countermeasures against the extreme cold were taken for the equipment, and as little equipment was used as possible, to lighten the load.

(A) *Table 1:Equipment (Total weight:200Kg)*

Camera,Lens	VTR
HDL-70A AH12×8.5ERD 2/3inch/4CCD DVW-700 (Aspect Ratio 9:6) TR-2000(Hi8)	AU-HD1400 (UNI-HI)
Power & Battery	**Audio**
Engine Generator (1.2KW 550W) Solar system Battery TBA-40 MAGNUM14,DP-55	MKH-416 SS-302 DAT TCD-D7

(B) *Figure 1:*
HDTV Filming Equipment



(C) The film crew consisted of 3 cameraman, 2 video engineers, 1 audio engineer, and 5 Sherpas.

2. Problems encountered and Countermeasures taken

(A) Preparation of the equipment
The camera and VTR were tested in the special laboratory, where temperatures as low as -30℃ could be attained. Table 2 lists some of the problems that occurred at this low temperature.

(B)Problem
Heat was added to both sides of the camera and cooling fans were turned off to prevent the problems from occurring during the shoot.
The low temperature caused defocus of lens and mis-trucking of tapes, however, taking the countermeasures and the heating mentioned below, the shoot was successful.

(C) Power Supplies
(1) Selected Batteries
Temperature characteristics for common batteries are shown in Figure 2. According to the figure, discharge capacity at -20℃ is about half of the capacity at room temperature.
In order to reduce the amount of equipment necessary, it is desirable to use a single type of battery for all of the equipment. However, this increases the risk factor due to difference of charge efficiency and battery-related equipment failure. To account for the increased risk, three types of batteries, namely lead sealed, nickel-cadmium, and lithium, were used.

The batteries were kept in a box and used with covers to avoid cooling down, consequently, they worked well.

Figure 2:Battery of Temperature Characteristic

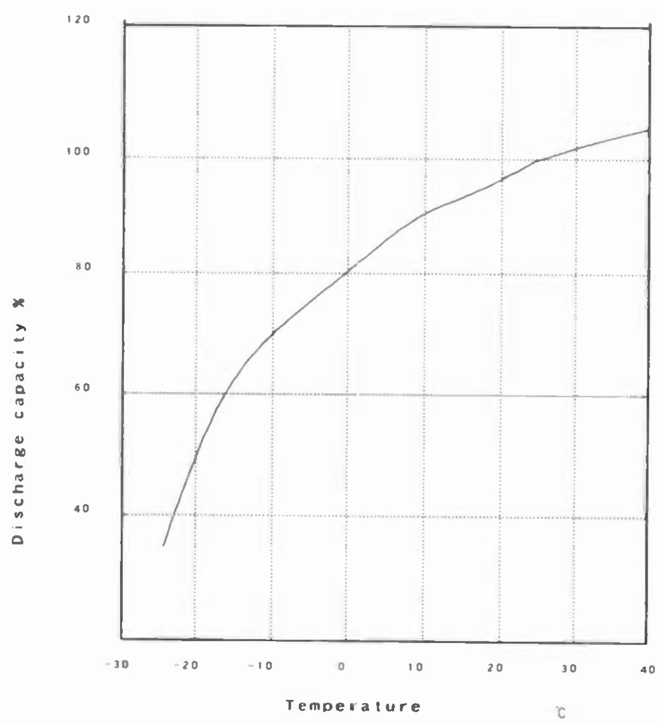
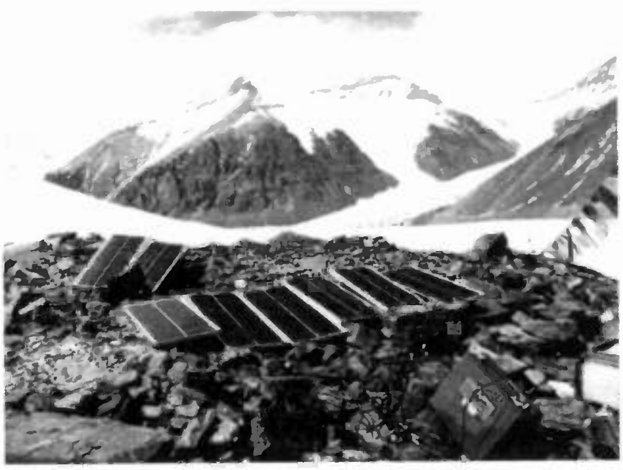


Table 2 : Problem and Countermeasure

Problem	Countermeasure
CCD Streaking	Heat DTL board
Decrease in S/N Ratio	Heat and exchange parts
Shading	Select"Quick Auto Setup" and preheat
View Finder mis-clamp	Change parameters of Sync circuit
Hardening of Lens Gears	Apply lubricants
Hardening of Cables	Cover with silicon tubes and heat
Unable to load cassette	Stop internal fan and heat

(2) Solar Panels and Electric Generating Unit
Generators and solar panels were used at the base camp to power equipment and to charge the batteries. Measures were taken so that the generators would function in the thin air of the camp. It is said that generator output power decreases by 10% for every 1,000 meters of rose above sea level. Two generators were taken on the trip to account for this loss of output. The generators were preheated with burners and sunlight before ignition. Batteries were charged mainly by solar power. Sixteen high-capacity panels were used to achieve approximately 200 w/hour output. In sunny weather, the panels provided more than enough power to charge the batteries. The panels could also be used in cloudy weather, but any snow must be removed from the panels.

Photo 2: Solar Panels



3. Safety Measures

The air at 5300 meters is only about half as thick as at sea level. The crew received ample training to ease the process of acclimatization to the thin air. Precautions for acuting mountain illness were taken to provide enough rest, water, and oxygen mask to the crew.

4. Conclusion

The Nihon University party succeeded in scaling the most difficult mountain in the world. The NHK HDTV crew was also successful, through the aforementioned preparations.
The HDTV camera, set at the 7050 meters point, shot footage at the peak. To 8020 meters above sea level, footage was shot by the HDTV crew using DVW-7000. However, it was too dangerous for the HDTV crew to scale and was not possible to take the HDTV equipment rest of the way to the peak. Consequently, footage at the peak was shot by a member of the Nihon University party using a portable 8mm video camera. In the near future, however, we are confident that the entire program will be able to be shot in HDTV, due to recent advances in technology.
Finally, we would like to appraise the courage shown by the Nihon University party, and thank the authorities in China and Nepal for their cooperation.

RADIO ON-LINE: INTERNET APPLICATIONS

Thursday, April 18, 1996

9:00 am - 12:00 pm

Session Chairperson:
Bill Ruck, KNBR-AM/KFOG-FM, San Francisco, CA

***CYBERCASTING: HOW TO MAKE YOUR WEB SITE A HIT?**
Rob Glaser
Progressive Networks
Seattle, WA

EXPANDING HORIZONS: BROADCASTING ON THE INTERNET
Diane Williamson
Bunyip Information Systems
Willowdale, Ontario, Canada

***THE INTERNET AND RADIO BROADCASTING**
Steve Church
Telos Systems
Cleveland, OH

RADIO ON THE INTERNET: A GIMMICK OR GOOD BUSINESS?
Gary L. Hawke
The University of Kansas
Lawrence, KS

*Paper not available at the time of publication.

EXPANDING HORIZONS: BROADCASTING ON THE INTERNET

Diane Williamson
Bunyip Information Systems
Montreal, Canada

ABSTRACT

1993 saw the first Broadcaster experiment by putting sound files of programs on the Internet. Less than three years later enormous advances have been made, with the introduction of the World Wide Web, Xing Stream technology and Real Audio. Where will the next two years take Radio Broadcasters as they continue to grow into a new world of broadcasting.

How to differentiate yourself when your market is the world. Where is the Internet headed? Broadcast standards? Competition or Cooperation? This paper will provide some insights, some forecasts and do a little future gazing.

RADIO ON-LINE. WHERE ARE WE? WHERE ARE WE GOING?

There has been a major push by media to get on the web in the past year. Some have done it better than others. Some squeezed budgets to find some money without having their primary business suffer. Others have spent considerable resources. There are so many broadcasters on-line now, they are too numerous to mention. The beneficiaries are all of the, ever increasing, number of users accessing that content on the Web. However for broadcasters, they now must find ways of supporting these new activities through either advertising, sponsorship, subscription or some combination of all three.

Recent research by Forrester Research(Cambridge,Mass.) shows that the amount of advertising revenue of on-line content providers will increase significantly.

On-line Ads 1995-2000

total U.S. ads 1995	37 million
total U.S. ads 2000	2.577 billion
ads per user 1997	$10
ads per user 2000	$100

In fact, broadcasters may lose advertising revenue from their "main" service, as it migrates to their Internet sites. This is where integrated marketing becomes important, but that is another paper! Forresters research found that ad/sponsorship dollars will allow content providers to produce superior audience draws at what may turn out to be lower costs-per-user than other media. Combine this with one of the key findings of a Yahoo/Jupiter Web User Survey which founf that 61% of respondents said they are watching less television, and spending more time on-line.

The Internet is a new medium, which should have content produced for it, not simply re-formatted. This of course takes time, and it will evolve. Not unlike when television was introduced, and radio producers just did radio on-camera! There is no "mass market" culture group on the Internet. There are virtual communities however, and this equals virtual markets. Traditional radio listening is a passive activity, where the "sit forward" computer activity is more active, and participatory. The user can select their own music or information, when they want it, to suit their schedules.

IMPROVED DELIVERY SYSTEMS, SOFTWARE AND CLIENTS

A number of features are contributing to the ability to offer a more diverse on-line product. The Internet infrastructure continues to be improved to facilitate the rapid transport of multi-media products and services. Improved band width will mean faster and more affordable delivery to both home and office. Similarly new software is emerging, and will continue to emerge to take advantage of the increased speed and capacity of these networks. Java from Sun Microsystems will permit www browsers to execute programs on the users machines. New client and server software will provide users with expanded control, intelligent agents, improved resource discovery and smoother integration of graphics, sound and video.

MEDIA CROSSING BOUNDARIES

The recent trend of magazines which become successful television and/ or radio programs and vice versa will now continue to include the Internet. We may see a service start on Internet as a "webzine" and then add the traditional on the rack magazine, and/or radio show. Rather than trying to be all things to all people globally, these communities of interest, or niche products are more likely to succeed.

DIFFERENTIATE YOUR SERVICE

Within existing broadcast markets you know who your competitors are. When you are on the Internet anyone can be a broadcaster. And the market is global. There are a number of ways you can differentiate your service however to not just become a part of the "background noise". Cooperative infrastructures are one way to approach building your service, whether it is shared resources, databases, or programs. Add on-line columnists, exchange programs with other countries, offer more than one language on your site, add interactive games and chat sessions. Users can chat with program hosts, but with other users too, if you create the interactive environment.

While it has been said that "content is king", but what is content without order , user control, or the ability for a user to set preferences?

If information is food for the mind, when it comes to the Internet, we need to balance out diet!

Having logical ways to archive, index, and use directories will encourage new users to your site and keep loyal users coming back. Put control in the hands of the users, by allowing them to "personalize" to their taste and schedule. Ensure that you are running your service with the capacity required, as indexing and profiling also uses capacity. By including user profiles your service will achieve two important goals. It will allow a user to tailor programs to their interests as well as provide valuable feedback on which parts of the service are most popular.

Beyond local indexing, global indexing systems form the basis for much of the Internet information infrastructure. These are evolving into more powerful and interoperable information systems.

ALLIANCES & PARTNERSHIPS

The Internet has brought about many partnerships and alliances between companies that have a common issue to overcome for differing goals. Banks and software companies, broadcasters and computer hardware, telcos and software developers. The World Radio Network has created a very unique satellite service which brings together content from a wide variety of broadcasters. While each of these broadcasters exists for reasons other than the World Radio Network, a unique and valued service has been created by bringing these parts together to form a different "whole".

The very infrastructure which is the Internet is based on shared resources, established standards and a cooperative approach. This model could translate

very effectively to a kind of United Nations of broadcasters on-line.
This would reduce the individual burden of broadcasts. Create an environment that recognizes the 24 hour global demands, and meets users needs for content that changes frequently, offers variety that they can not find elsewhere. By incorporating your content with that of other broadcasters, it will be introduced to an audience that you may never have reached, flying solo.
The Internet is not likely to put traditional forms of media out of business, just as newspapers didn't die with the advent of radio, nor radio when TV arrived. So media producers must recognized that the terrain has shifted and that the Internet is moving into the mainstream quickly and ahead of what analysts predicted.

NON-TEXTUAL SEARCHES

With the advent of greater amounts of multi-media available on the Internet, and the previously stated need for effective indexing for specific searches, it is expected that some effort will be put into indexing of non-textual objects, such as images and sound files. Thematic key words may suffice for rough content searches, but what will be stored in order to retrieve musical composition by fragment sample? These are some of the challenges facing developers, and they will overcome them.
Software currently exists that allows users to interact in their native language and character set.

CONCLUSION

Net years move much more rapidly than calendar years. The adoption rate of new technologies continues increase, and the Internet has in just two years already had a significant impact on broadcasters. The next two years will continue to advance capabilities, and the sands will continue to shift.

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Interactive Video News.Aug. 7/95

RADIO ON THE INTERNET: A GIMMICK OR GOOD BUSINESS?

Gary L. Hawke
The University of Kansas
Lawrence, KS

Everyone's Doing It

Your stations may be one of the hundreds that have gone On-Line in the last year. It was *the* thing to do in 1995. For many stations, creating a Web site is still a far reaching concept to management. NAB's Conference sessions for the past 18 months have been loaded with "how to" topics on creating your Web site. Too often, I fear, this visual radio has been a fad. It's as if one station in the market buys a big remote van, followed by every other station feeling that's the way to stay hip and on the cutting edge, so vans spring up all over the market. I'm here to say that being O-line is great but if all you're using it for is to profile your jocks, sell promotional material or list your program schedule, you may be missing a lot. I think the 'net should be used for many more technical applications and these can provide genuine station rewards, cost very little to implement and actually save you money!

The Internet offers the opportunity to Video Conference, Interchange stereo programming and commercial material, provide Network capabilities, Direct interaction with clients and even provide real time Video programming and commercial production interchange. The Internet offers a great chance to interact with your listeners and viewers, but let's explore the opportunities for intra-corporation exchange and group owned interact.

As LMAs, leasing and group ownerships grow, you need a better way to communicate with your stations. The Internet has it for you at very low costs.

Video Conference

At the University of Kansas through the student radio station, KJHK, we've pioneered real time, continuous live radio broadcasts over the Internet. Since December of 1994 we've been broadcasting a live continuous signal of our on-air programming over the 'net. This can be heard using 14.4 modems through the software of *Netphone*. On January 2nd of this year we also provided a live, real time, *video* program to the 'net using *CU-See Me*. Operating this readily available software and low end computer hardware you can be producing all of the programming types I've mentioned above.

Long distance teleconferencing is what *CU-See Me* was created to do for your company. Since it was developed through a government grant, you can down load this software free. Originally, it was only available for use on Mac systems with T-1 connections. Recently, however, it is available for PC systems.

You may obtain *CU-See Me* specs by visiting their Web site:

ftp://gated.cornell.edu/pub/video/

Obviously, the higher speed transmissions capabilities are the recommended methods. *CU-See Me* offers your management the ability to hold staff meetings, sales seminars, training and what have you using computers over non-toll phone lines. All you have to do is hook up your system to a large TV monitor for classroom style communications...and this is interactive communications. You can connect 20 or more sites, simultaneously, for instant worldwide communications. You can do this with cameras that can cost as little as $100.00, a 486 computer or MAC 6100 or better and your Internet connection. This feature, alone, is enough to get your stations on-line and one that even the most frugal management should see the advantages.

Interchange of Stereo Programming and Commercial Material

Using current technology broadcasters can transmit program material, commercial and promotional spots, copy and billing information and the like, without the need for long distance telephone chargers or satellite uplink fees. Again, stations should invest in ISDN local connections. This will allow them wider streams of data transmission and reception.

There are software creators currently offering the framework to get this done. One such is Xing Technology in Arroyo Grande, California. They offer broadcasters a complete listing of their services and costs just by visiting their Web site at *http://www.xingtech.com*
Users can choose their best transmission rate and produce real time audio and/or video. They may also compress certain signals for easier shipment.

Xing Technology can provide you "on demand" streams for previously recorded material or "live" capabilities for sharing between company locations. Product descriptions and prices are available from the company via their Web site. Typical users merely need an Internet Web site with Sparc, HP, SGI or Linux Web server. Linux is a free program system giving you Unix type use. You will

have to make sure you have the proper network connection for your use. This may be ISDN level or T-1 capacity. Obviously, the speed at which you can send and the number of

Now playing on a computer near you.

those who can connect to you at one time will be effected by your network connection. ISDN lines can be as little as $200 to install and about $50 per month to operate. A T-1 connection can cost around $2000 to install and as much as $1500 a month to operate.

You will need a file server with a least a Pentium 90 speed and 32meg RAM. This should cost about $3000-4000. *Xing's* server software is in the $3500 range. You will need an encoder for each audio stream at about $2500 or video/audio at about $6500.

This will now allow for the transfer of mono or stereo commercial announcements, live or "taped" programming, company-wide training or information presentations and meetings. This distance communications can be done without the need for long distance telephone access or through costly satellite uplink charges.

Users will need a minimum of a 486 PC, or an Apple 68030 computer with 28.8 modem. Software for audio transmissions can be down loaded from several sources. KJHK has worked with the folks at *Netphone* for some time. Their software will allow even 14.4 modems to carry real time audio. You can check out their products at *http://www.emagic.com.* *Netphone* provides very reasonably priced software for your delivery and free software for your clients to receive your feed. With the ISDN line and higher speed computers the process will function even faster.

This interfacing can also be arranged with large agency clients or rep firms. The broadcaster may want to invest in their own equipment for these connections to insure a place on the desk of your clients or they can certainly use equipment already in place at major agencies. You may receive copy material, make presentations and submit avails all live and interactive if you wish. Through CU-See Me and audio/video software, you have a direct connection to the desk of your clients and representatives.

So, if you think you're exhausting your technical envelope with a Web site presence, you're missing some very real money saving advantages and management flexibility. You should be commended for venturing onto the Web, but there is good business in increasing your interactive capabilities rather than just going with the gang in the gimmicky world of Home Pages.

IMPLEMENTING VIDEO DATACASTING

Thursday, April 18, 1996
9:00 - 10:40 am

Session Chairperson:
Andy Butler, PBS, Inc., Alexandria, VA

***THE ADVANTAGES OF DIGITAL SERVICES ON TERRESTRIAL TRANSMITTERS**
Udo Scalla
Deutsche Telekom
Bonn, Germany

Bernd Furstos
Deutsche Telekom
Freiburg im Breisgau, Germany

IMPLEMENTATION CONSIDERATIONS FOR DATA BROADCASTING BY NTSC TELEVISION STATIONS
Kenneth D. Springer
Datacast Partners
Reston, VA

CLOSED CAPTIONING WITH RANDOM-ACCESS VIDEO
Bruce Anderson
IMMAD Broadcast Services
Markham, Ontario, Canada

***INTERCAST PRESENTATION**
John Kirby
Intel Corporation
Hillsboro, OR

A LOW-COST/FLEXIBLE DE-INTERLACING SOLUTION
Jordan Du Val
Genesis Microchip Inc.
Mountain View, CA

IMPLEMENTATION CONSIDERATIONS FOR DATA BROADCASTING BY NTSC TELEVISION STATIONS

Kenneth D. Springer
Datacast Partners
Reston, VA

ABSTRACT

Station level implementation considerations are presented for a new data broadcasting service directly to personal computers. Requirements for a data combiner, a data modulator and a multimedia data workstation are discussed.

INTRODUCTION

Until now, television data broadcasting has consisted primarily of stations leasing capacity to service providers which transmit VBI-based textually oriented information to business users. Such data broadcasting operations require negligible effort for stations to maintain and operate and offer commensurately minimal revenues.

Advanced digital television will make possible a completely new paradigm. By definition, all television broadcasters will be in the data broadcasting business with a 20 Mbps bitstream. Meanwhile, the oft-discussed merger of computers and televisions will equip consumers with intelligent, random-access, video-savvy, data receivers. Broadcasters will then be able to more cost effectively deliver a broader range of viewer demographics to their advertisers through broadcasts of non-linear, threaded, multidimensional programs which combine elements of both broadcasting and narrowcasting.

In the interim, new high-speed (> 500 kbps) NTSC data broadcast systems offer the ability to broadcast a wide range of ad-supported multimedia content directly to personal computers. Such multimedia programs may include not only text and graphics, but short video clips and digitized audio segments. This presents broadcasters with new possibilities to establish national multimedia broadcasting networks for personal computers, providing a mix of national and local programming of news, sports, information, and entertainment.

This paper will address the components required to implement such a data broadcasting system, specifically in the context of the Digideck D-channel system. This paper will discuss generally the transmission chain requirements including data encoding, the studio-to-transmitter link, and insertion into the vestigial sideband. Additionally, this paper will discuss the requirements for a station-level multimedia data workstation for local ad and program insertion within a packetized data broadcast signal. The paper will also address receiver hardware and software requirements.

A NATIONAL MULTIMEDIA BROADCASTING NETWORK

A useful model for a new, ad-supported, multimedia broadcasting service is the existing television broadcast system. Following this model, a multimedia broadcast system requires a network center, a national distribution network, local broadcast station affiliates, and receivers. **Figure 1** illustrates the major components of such a national multimedia broadcasting network.

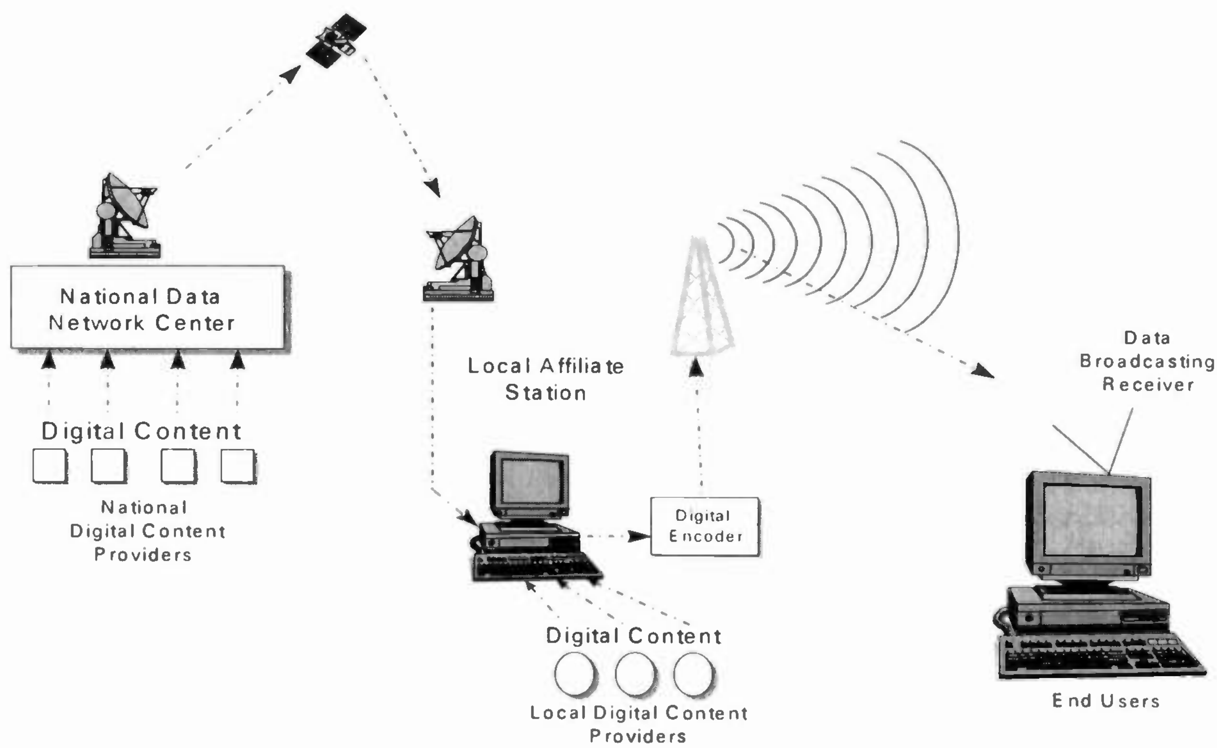

FIGURE 1

Programming is distributed nationally by landline and/or satellite from a national data network center to local affiliate broadcast stations. The data network center may transmit a number of different national, regional, or local data network feeds to local stations.

At the local station, the network data feed(s) is received, reformatted, and broadcast over the D-channel in the NTSC vestigial sideband. Stations not initially desiring to distribute local ad or program content may have a simple hardwired passthrough to a data encoder and modulator installed in their existing transmitter. As the FCC will probably require that the broadcaster be able to inhibit undesired data transmissions, the data combiner must provide an inhibit control signal to the station.

Stations wanting to insert local programming and ads into the data broadcast signal require a data workstation. The data workstation formats locally originated programming and seamlessly integrates it into the network feed to generate a local broadcast signal. The workstation also handles traffic management including scheduling, logging, and billing for any subscription broadcasts.

The data broadcast is received by the audience with a PC-based data receiver. The receiver includes a special chip set which may be in an OEM computer, or purchased separately as an add-in card or external unit. The data receiver operates in conjunction with supplied software to reconstruct data programs and format them for presentation to the user.

THE DATA RECEIVER AND OPERATING ENVIRONMENT

To better understand data broadcast equipment requirements at the station level, an understanding of the data broadcast signal and the operation of the data broadcasting service itself is helpful. The system which will be discussed is based upon the use of a vestigial sideband based digital signal such as the Digideck D-channel, currently under evaluation by the National Data Broadcasting Committee (NDBC). The vestigial sideband transmission system can be considered to comprise the physical and data link layers of the broadcast signal in an OSI model of the data broadcast communication link.

Figure 2 illustrates the spectrum of an NTSC transmission modified to include a digital signal in the vestigial sideband.

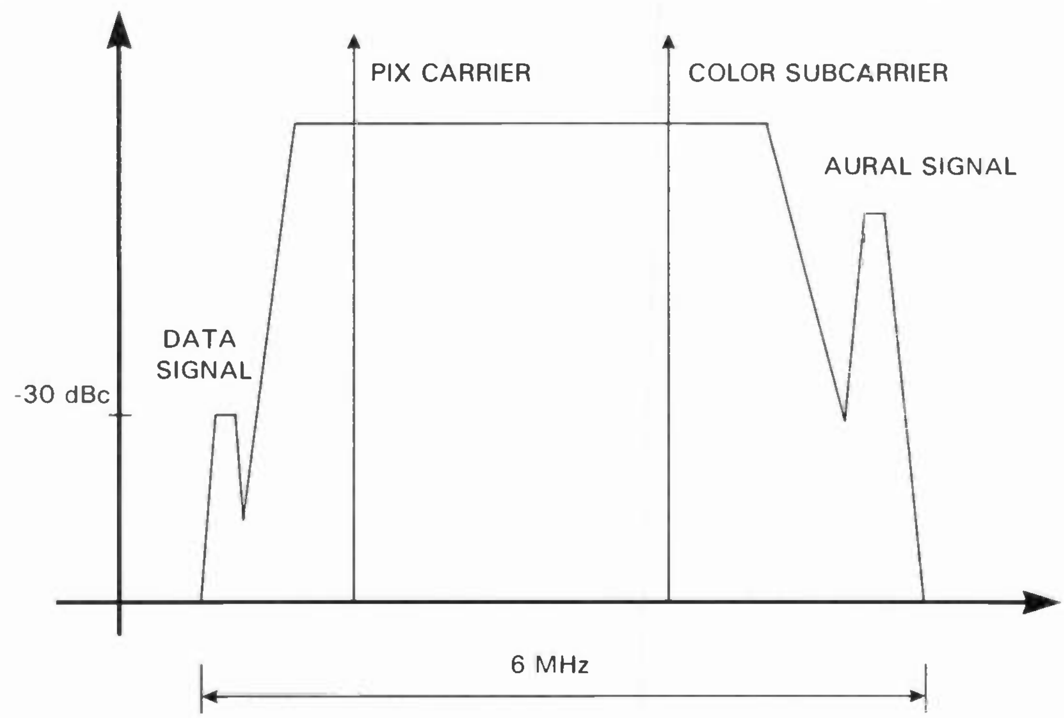

FIGURE 2

The digital signal is inserted approximately 1 MHz below the picture carrier at a level of about -30 dB below peak of video sync. To accommodate the digital signal, the NTSC spectrum is modified to more rapidly attenuate the vestigial sideband components. Optimally, sideband video components are filtered to be 20 dB below the data carrier. Laboratory tests with expert viewers from the NDBC have demonstrated that insertion of the data signal creates virtually no perceptible effect upon the television picture. More detailed information on the D-channel signal can be found in CONNER, A.B., "*D-Channel - A Transition into Datacasting*" published in these proceedings.

Figure 3 is a block diagram of an exemplary data receiver. The tuner, covering the VHF and UHF broadcast bands, has a channel selector which operates under software control of the user's PC. The data signal is filtered, demodulated, and error corrected to produce a 525 kbps data bitstream. The receiver analyzes received data packets and reassembles them into their respective data programs.

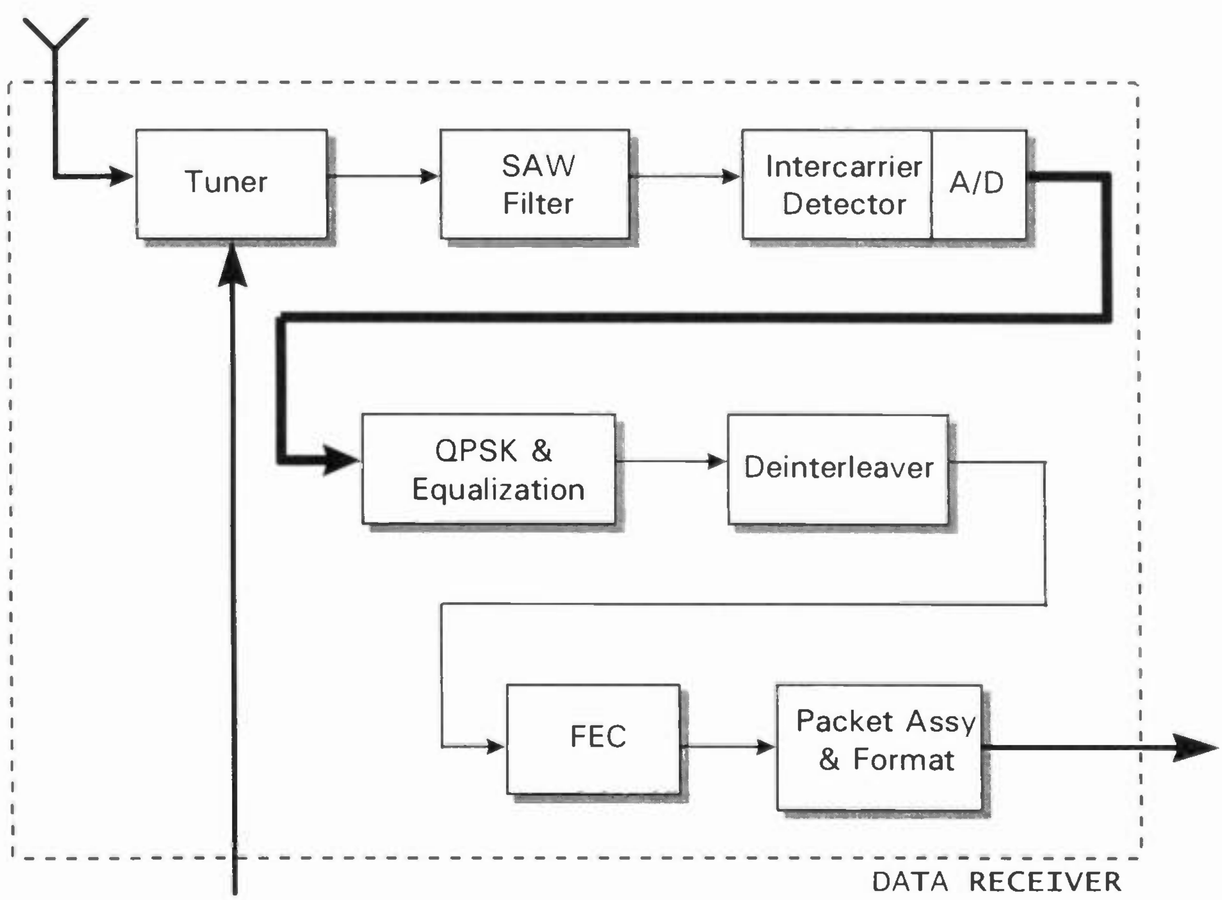

FIGURE 3

The data receiver works in conjunction with supplied firmware and computer software elements to deliver and present data applications and programs to the end user. Receiver software elements include Communication Connectivity software, Operating Environment software, and Application software. These software elements work together with the receiver to transform the received bitstream into flexible, PC-based multimedia application programs with graphics, pictures, text and short video and sound clips.

Communication Connectivity software manages reception, storage, and presentation of data packets received over the channel. Each data packet is identified in its header by a unique program ID and data packet number. The initial packet(s) of each data program contain a program length field which specifies the number of packets for the program. For a particular program, data packets may be transmitted sequentially or interspersed with packets for other programs on a data channel. Parts of a program may even be transmitted over several different channels. The Communication Connectivity software is responsible for reassembling data packets into complete programs.

Although the system incorporates powerful error correction techniques, on rare occasions a packet will be received in error. Because the data broadcasting channel has no return path, the receiver is unable to request that the packet be retransmitted. Communication Connectivity software creates a Dummy Packet Table for each program which identifies the packet numbers of any packets received in error. When the program is retransmitted at some later time, the software will search out and capture only those packets listed in the Dummy Packet Table which are needed to complete the program. The Communication Connectivity software presents the received data packets to the PC across a standard PC-based communication interface.

The Operating Environment software is the heart of the system and the engine which drives Application software modules which support each data application or program available over the broadcast channel. The Operating Environment software contains a Data Management Software Module and a User Interface Software Module (UISM). The Data Management Software Module (DMSM) handles data delivery, including decryption and address-matching, data storage, and data retrieval. When a non-real time data application is received, the DMSM stores the program on the user's hard disk drive and notifies the UISM of the program ID and description. For real time data, the DMSM notifies UISM that real time data is being received.

The User Interface Software Module UISM is the core software module which creates a consistent user interface and hosts multimedia-based Application Software Modules (ASMs). The UISM provides the user with a familiar Windows-based user interface with pull-down menus.

The UISM includes a User Profile component which customizes system operation for each particular user. The UISM also includes a Program Log component which maintains an index of currently available and upcoming multimedia data programs. Finally, the UISM includes an Application Support component.

The User Profile component includes a Receiver Set-up algorithm and a Viewer Filter algorithm. The Receiver Set-up algorithm allows the user to scan all of the television channels in the market to create a directory of available data signals. The Set-up algorithm also allows the user to adjust the receiver antenna for maximum reception by providing an icon with real-time feedback on signal quality. The Viewer Filter algorithm allows the user to specify areas of interest such that the Data Management Software Module will only capture and store data programs matching the user's areas of interest. The Viewer Filter works in conjunction with a Station Program Guide which is periodically broadcast to identify the schedule and other pertinent information concerning upcoming programs.

The Application Support component is the heart of the Operating Environment software. The Application Support component uses Object Linking and Embedding (OLE) technology and contains drivers supporting objects comprising graphic images, sound bites, and full motion video clips. The Application Support component allows common objects to be shared across applications and programs. It also facilitates links from one ad or program to another, across applications.

For example, an automobile advertisement icon appearing in a sports ticker may contain links to a separate application which presents a multimedia program about the vehicle. By pointing a mouse and clicking on the icon, a user will leave the sports ticker application and open the multimedia vehicle brochure. Moreover, the same icon may appear in a variety of different data programs and may be placed in different locations within each program.

Updates and revisions to the Operating Environment software, such as a more powerful Viewer Filter algorithm, for example, may be broadcast to the data receivers and automatically installed.

Application software operates in conjunction with received data and the User Interface Software Module to present multimedia data programs to users. Examples of application programs might include multimedia yellow pages, classified ads, electronic multimedia newspapers, and sports tickers. Each of these applications operates with a core program which periodically receives digitized multimedia content via the data broadcast signal. As new data broadcast applications are developed, they may also be delivered to users automatically over the broadcast channel.

TV STATION DATA BROADCASTING TRANSMISSION EQUIPMENT

In a minimal configuration, the TV station data broadcast equipment consists of a network data receiver, a data encoder, and a data combiner as shown in **Figure 4**.

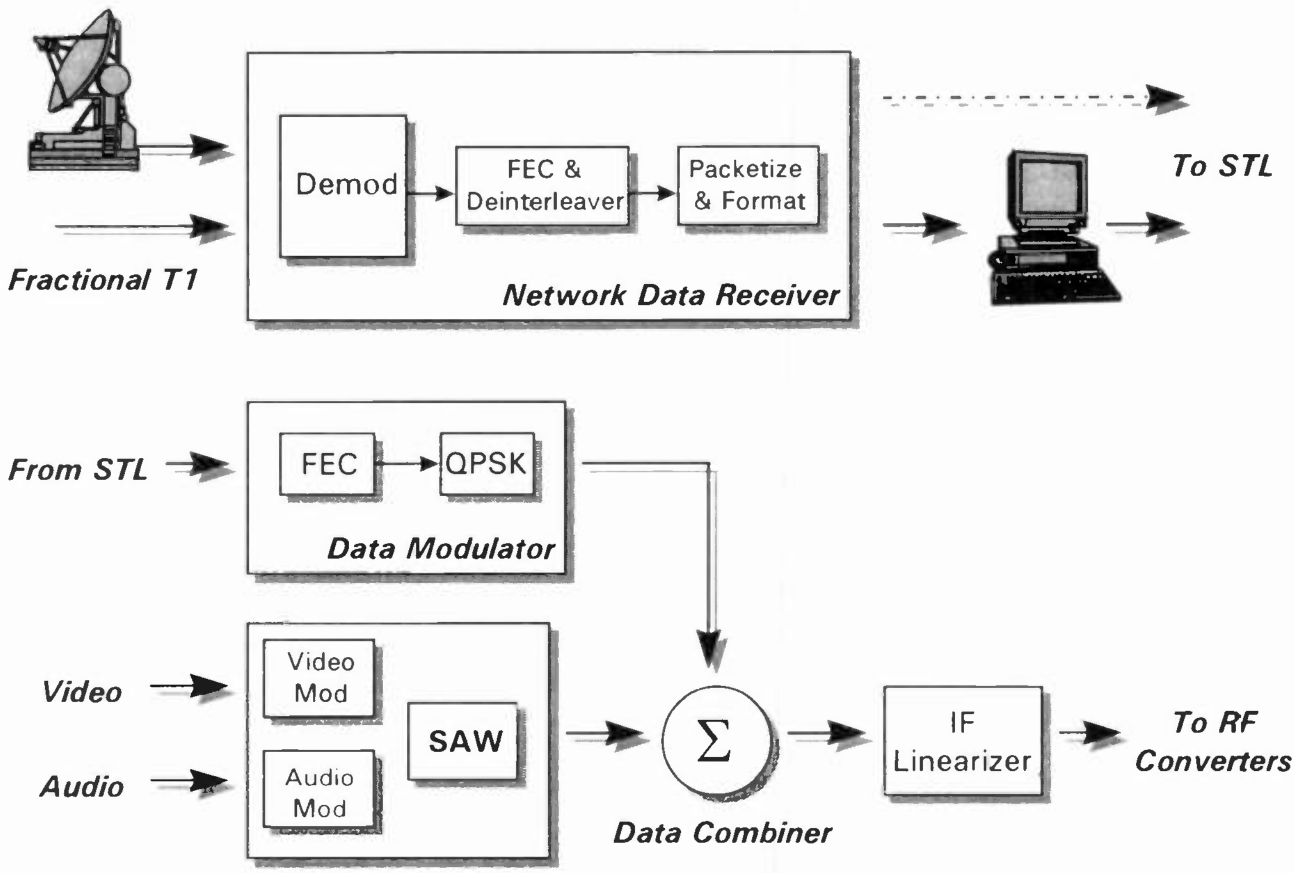

FIGURE 4

The network data receiver may be connected to a VSAT terminal or to a fractional T1 line. The network data receiver demodulates the network data feed and applies interleaving and error correction. For a simple passthrough installation, the network data receiver produces a packetized data broadcast signal and passes the packets directly to the data encoder located at the transmitter site over a studio-to-transmitter link (STL). For stations providing local ads and content, the network data receiver output is connected to a data workstation, as discussed below.

The packetized data to be broadcast is provided to a data modulator as shown in **Figure 4**. In some stations, the data modulator may be installed as an add-in card to an existing modular exciter. In other cases, the data modulator and data combiner cards may be mounted in a separate, standalone, data broadcast box which is located at the station's main transmitter site.

The data modulator receives the packetized data stream at approximately 525 kbps and applies error correction and interleaving. The resultant 700 kbps bitstream is applied to a differential

QPSK modulator. The data modulator is also connected to an internal station reference signal to lock it to the transmitter IF and carrier signals.

The modulated data signal is provided to a data combiner where it is added to the modulated visual and aural signals to be broadcast. The data combiner structure will differ from station to station depending upon the transmitter and exciter configuration in use.

Installation of the data combiner also involves replacing the existing vestigial sideband (VSB) SAW filter with a new filter with steeper roll-off characteristics. The QPSK-modulated data signal is inserted in the VSB of the modulated video signal, approximately 1MHz away from the picture carrier.

The output of the data combiner is provided to the broadcaster's existing transmitter output stages and broadcast over the existing transmit antenna.

DATA WORKSTATION REQUIREMENTS

If a station desires to provide locally originating data transmissions, it will require a data workstation to integrate local data with the network feed. The data workstation automates a number of tasks which allow the local television station to create a unique local data broadcast signal which integrates local and national programs and ads. The data workstation receives network data programs via the network data receiver, integrates local content into the signal, and provides a packetized data signal, via STL, to the data modulator at the transmitter site.

The data workstation receives a network program schedule from the network feed. The network program schedule provides ID numbers, program descriptions, program lengths and scheduled times for upcoming programs. At the data workstation, the network program schedule is integrated with a locally generated schedule of local programs to create a station program schedule of network and local data programs. The station program schedule is used by the data workstation to queue programs for local broadcast. The station program schedule is also broadcast as a Station Program Guide over the data channel to data receivers to be used by the User Interface Software Module to deliver programs to the user.

The data workstation receives network programs as packetized data transmissions. It decodes information in the header fields of each received packet to identify the program with which it is associated. The workstation also identifies "live" network data signals for direct passthrough to the data encoder for real-time broadcast. Examples might include breaking news and sports information or programming which may be tied to the station's current television program.

The data workstation includes a mass storage unit where network programs may be stored for later broadcast or rebroadcast. Local data programs and ads may also reside on the mass storage unit. For programs which are to be stored locally, the data workstation assembles data packets into complete programs and adds the program ID, description and length to its program index.

For each network program, information may be decoded by the workstation which allows the workstation to insert local ads of varying sizes and formats, where appropriate. For example, a network sports ticker may have packet slots identified for insertion of a local ad billboard. During one score update, a local broadcast station may insert an ad for a local car dealer. During a subsequent score update, the station may run a station promo in the same billboard.

The data workstation can automatically mix and match ads with programs based on a variety of parameters entered by a local station operator. Among other things, this allows the broadcaster to control ad frequency and to ensure compatibility of an ad message and sponsor with its associated program content. Local ads may comprise a mix of graphic and text based "billboards" as well as icons, short video clips, and sound bites. Each of these different ad objects requires a different number of packets to be broadcast. For example, a short video clip requires many more packets than a static text ad. The data workstation is responsible for matching available ad slots within programs with appropriately sized ad content.

Generally, local programming and ads are produced on disks or CD-ROM as input media for the data workstation. The data workstation may also include authoring tools for producing local data programs and ads.

The data workstation also maintains a broadcast log of ads and programs which have been broadcast. If desired, the workstation may generate a billing report detailing the times when each sponsor's ads were broadcast.

SUMMARY

Station level requirements for a network affiliate multimedia data broadcasting service have been discussed. The model for this service is the existing, ad-supported, television program distribution and broadcast business. The components of this system are currently under development and expected to be available to interested television stations within the next year.

CLOSED CAPTIONING WITH RANDOM-ACCESS VIDEO

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ABSTRACT

The passage of the **Americans With Disabilities Act** has brought new attention to the field of closed captioning. This has led and will continue to lead to a dramatic increase in the total volume of captioning that must be done. Since captioning is a very labor intensive activity, two key business issues will arise. The obvious one is an effort to contain costs; the less obvious is a shortage of skilled captioners.

The dramatic advances that have occurred in the last few years in disk storage costs, computer speed, hardware support for video compression and decompression, and the proliferation of video editing capability have made possible a new generation of tools to support closed captioning; tools that will help contain or reduce costs as well as open up new market possibilities.

This paper describes one such tool in detail; identifies the characteristics associated with technology advance and points to likely future directions.

Product Background

Before we can sensibly discuss economic issues, some background is in order - both to put IMMAD's project in perspective, and to provide a reference point on what the building blocks are and how they fit together.

In 1993, YTV commissioned IMMAD Broadcast Services to develop a state-of-the-art captioning authoring system. The original concept was to establish a network server which would capture and playback all of the required video, and integrate it with the library management system to provide maximum productivity for the caption editors.

The RFP called for using state-of-the art workstation video technology, state-of-the-art video networking, and creating a new standard for state-of-the-art in user interface.

The project was envisioned as something that could be brought to production status by Spring of 94. What actually happened is rather different. We learned quite a few lessons, changed many of the original "must-have's" and we were not in actually in production until early 96!

Project Lessons

1) Pushing the envelope in several directions at once leads to remarkable complexity, not to mention cost.

2) Windows NT was nowhere near ready to support such a system in 1993.

3) Video Server technology is heavily biased towards asymmetrical codecs - since that is what video training & set-top boxes need. Caption authoring requires symmetrical codecs - and the economics are not there even today.

4) There is both a lot of complexity and a lot of technology risk associated with attempting to insert captions at the very last point in the chain. We set this requirement aside, to wait until the technology was more mature.

5) A "simple" thing like word-wrap, so deeply a part of conventional word processors that we rarely think of it, turns out to be conceptually very difficult when dealing with captions.

6) Microsoft's AVI technology, in 1993, was oriented to postage stamp sized video at 15 frames per second. The notion that anyone might want to track SMPTE timecode and associate it with specific captured frames was simply not part of the design at that time. Sound synchronization worked well for the first couple of hundred seconds - that was all that most people were doing. After 30 minutes of video, the sound could be as much as several seconds out of sync.

Microsoft's new "Quartz" video technology may solve a lot of these problems. But -- as this paper goes to press, Quartz is not even here yet.

Intel's new Indeo codec, which uses wavelet compression represents a big step in the right direction; when combined with Quartz and NT, it may well resolve all of the issues from the original AVI.

Needless to say, discovering all this required exploring a lot of dead ends. We finally settled on licensing the Montage Video Engine - based on the original mature DVI video technology. The DVI technology isn't good enough to go to air with, but the captioning process does not require that quality level.

The Building Blocks

FutureCap incorporates several new concepts in captioning and the associated workflow, so before we can sensibly discuss what could be, we must build a common frame of reference.

A FutureCap installation consists of two types of components: Authoring Stations and Encoding Stations. The Authoring Station works much like a very expensive Dictaphone. Feed in video and audio; add labor; output an encoding file. The encoding file is similar to an EDL - it is a list, frame by frame of which codes to insert. The Encoding Station takes in the encoding file and, by monitoring timecodes from the master tape, controls the timing of commands to a caption encoder (e.g. EDS 400, EEG-270). The Encoding Station does not require the expensive video compression hardware, so it is normally set up as a separate machine. The functions can be performed by the Authoring Station in a minimal installation.

The screen in the Authoring Station is laid out in three main parts:

a. The Video window. This window displays the current position of the video file, and displays the simulated caption. The caption decoder simulator is able to simulate both TC2 and FCC decoders. The is the What- You- See- Is- What- You- Get part of the system.

b. The editor window. This is simply a text editor with a few special features to handle captioning special situations (e.g. passing on pre-existing captions).

c. The Timeline. This is the window which looks most different about FutureCap. There is a lot of information packed into the five bands of data on this window. On the X axis, each pixel represents from 1/4 frame to 4 frames, user selectable. From the top down, the bands contain:

i. the secondary caption channel. It contains information only when captioning two channels in the same field.

ii. the conflict bar. This shows which frames are used to transmit data, and by color difference, shows red whenever an attempt is made to transmit two different sets of codes on the same frame.

iii. the primary caption channel. This line shows the transmit time, the InPoint and the OutPoint of each caption (when **EOC** and **EDM** are executed).

iv. the summarized audio display. Except during continuous loud music or background noise, this display makes it fairly easy to pick out individual words. The more spread out the display, the easier it is to pick out words.

v. the repeat loop. We provide the ability to continuously repeat a section of audio that may be difficult to transcribe. This bar shows the audio segment that is being repeated.

Wheat From Chaff

What are the differences that make the difference?

1. The timeline provides important visual feedback of how the InPoint and OutPoint of each caption relates to audio, video and other captions.

2. We provide two mechanisms to achieve rough caption timing: a) by tracking keystroke timing during transcription, the workstation can approximate the caption positions in time; b) the editor can complete transcription and then replay the video, placing each caption InPoint on the timeline as it goes by. Our early conclusion is that the method of choice is highly dependent on the target quality level. For all but the highest quality, the second method seems to be faster. The reason is that this technique places a high percentage of the captions "close enough", while it is quite time-consuming to carefully listen to each sound phrase and adjust the caption timing. For the highest quality levels, where this will be done for all captions anyway, the first method is faster, as it eliminates a pass through the source video.

3) Having multiple decoder simulators and an easy way to switch from one to the other has proven very important when working with TC2/FCC.

4) A surprisingly important feature is the ability to perform edits during the Line21 encoding pass (e.g. pull out a freeze frame into a commercial break, etc.). This step is bottleneck in many operations, and anything that will free up operator or machine time eases the pressure.

5) Overall, the skill level required for "average quality" captioning jobs is significantly reduced.

Work Flow

The FutureCap workflow proceeds as follows:

1) **Capture**. During this phase, FutureCap is set up much like a VCR with a timecode display. The source VCR is controlled from the screen. Video and audio are monitored during capture. Once the video and audio are on disk, an audio summary file is created. This file is used to provide a visual reference on the timeline during captioning. With removable drives, this step can be accomplished on a separate Authoring Station.

2) the Caption Editor then works the station like a dictation system. There are normally at least three passes of the material required:

a) transcription - get the words typed.

b) timing - place the words WHEN the Editor wants them.

c) embellishment - this covers placement on the screen, italics, music notes, etc.

In general, the higher the quality level, the more passes are required.

3) when the captions are finished to the editor's satisfaction, the Encoding file is created, and passed on to the Encoding Station.

4) the Encoding Station controls the Line 21 Encoder and causes it to insert the appropriate codes into the appropriate frames. This is done on a frame by frame basis and allows for editing during the captioning pass and to make use of previously captioned segments of the master. As long as the master used to capture to disk has timecodes identical to the master used for the final dub, the system will tolerate any amount of forward and back, inserts, etc.

Business Cost Issues

When a provider sends a tape off to a captioning service and gets back a captioned master, it is very easy to compute the cost of captioning. As soon as some of the work moves in-house, it becomes much harder to calculate. To make matters more complicated, the new generation of captioning tools provide the caption editors with the possibility of doing extremely high quality work - work that will dramatically escalate the cost of captioning - work that in many cases will not be cost effective. Experienced Caption Editors are hard to come by, and expensive.

FutureCap (and competitors) do two things to the cost equations for captioning: 1) the what-you-see-is-what-you-get technology lowers the skill level required for captioning (and hence labor costs); 2) allow caption editors to perform a given quality level of captioning more rapidly that older technology; 3) provide a platform whereby service providers can select the captioning quality level from a very broad range - from quick roll-ups to frame accurate and carefully evaluated pop-ons.

Item 1 (less experienced labor force) will mean that captioning activities will have to be more carefully managed; Item 2 will be discussed below; Item 3 requires very careful management attention lest it become a Pandora's Box.

Basic captioning - essentially transcription with minimal attention to embellishments, timing or placement on screen - probably will not get much below two and a half three hours for a 30 minute show - a 5 to 1 ratio. This allows only for one pass to dub for captioning, three passes to transcribe, and one pass to encode. A fastidious and creative captioner can easily add another 10 to even 20 passes - for example, very careful positioning and reworking dialog to shorten or simplify. This is a five to one ratio of potential cost - a valuable option, but one which must be carefully managed.

The value added from this generation of captioning engines is that they can significantly reduce the effort required for high quality work (Item 2). Random access to the video, visualization of the timing relationships of the captions, the ability to easily generate a channel 2 script from a channel 1 script - all these factors work to improve productivity.

A cautionary note for anyone evaluating captioning costs - the workflows vary quite a bit from one system to another, and even within a given system, so side by side comparisons of vendor prices seldom gives the whole picture. It is important to examine each step of the process you will use with each approach and evaluate three associated costs: 1) the cost of the equipment tied up for that step; 2) the cost of the people tied up for that step; 3) the cost of the step in terms of interfering with other processes (a chronic problem in small shops). Even with FutureCap, there is a wide range of workflow possibilities but the most efficient only make sense in a large shop.

Early experience with FutureCap suggests that high quality captioning can be performed by editors without significant captioning experience. None of the products are yet mature enough to have undergone the field-tuning that it will take to determine how much faster captioning can be done this way.

New Markets

Line 21 decoders currently provide for two co-existing channels. To date, there has been minimal use of the second channel. There are two root problems: the data rate is so low that some situations won't fit; and it is very difficult (without computer help) to insert captions into both channels and get the timing right.

Once the primary, verbatim transcription is completed and the captions timed, placed and embellished, most of the work is done. Adding a second (or third or fourth) transcription is relatively simple - consider a Spanish translation, or simplified English.

The operator interface required to handle two channels is a simple extension of the existing FutureCap interface. Load two scripts at once; display both on lines in the timeline, and show the conflicts. Then all the editor has to do is adjust wording and timing so the red conflict indicators disappear. This may require content compromises in rapid dialogues, but the training required for the caption editor is minimal.

We firmly believe that the ability to add translations or alternate texts will rapidly open up new markets to providers - markets that did not show interest before now.

Next Steps

All this technology in one box makes for a high end system, both in complexity and cost.

2) Video compression / decompression. This technology had evolved rapidly, and the underlying system encapsulated the interface to the video so that upgrading the system to the latest technology could be done in a cost effective manner.

3) Frame accurate. This required monitoring SMPTE timecode during capture, and was the basis for rejecting a lot of otherwise attractive solutions early on. Caveat: "frame accurate" can be misleading. All of the top quality Line21 handling equipment uses a regeneration technique which causes a 1 frame delay for each regeneration. This is discussed in the EIA 608 standard. The captioner can place any event accurately on any frame in a master tape - but if there are a variable number of regenerations between the master tape and the viewer's set (e.g. direct broadcast vs cable head), the captions will still be close but won't be precisely frame accurate. This can be especially noticable when captions are tied to scene changes.

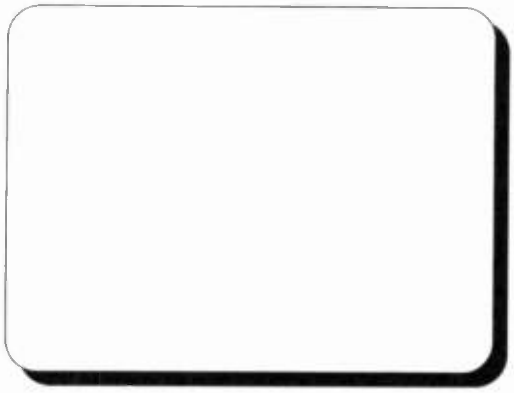
4) Speech recognition. This can be used in two ways: a) as an alternate aproach to transcription - an appealing technology, but not here yet. Actually, it can be done with sufficient accuracy in Chinese but there are other issues to deal with before we can properly penetrate that market. Also, for anything beyond the lowest quality captioning, transcription is not the driving time component. b) as a mechanism for issuing commands to the computer - this technology is here and is viable. At least one competitor is using it today.

5) Speech alignment. This means taking the digitized speech and a transcript and matching the two. This can be done today, but has not been cost-effective for any environments that we have evaluated to date.

6) Timeline display. FutureCap implements this, and our beta test experience suggests that it is very important in reducing the training required before an editor can produce quality captions for air.

7) Multiple channels. This is on the very high priority list, as we firmly believe that there will be major market demand for multiple channels as soon as the capability is in place.

E.g. English and Spanish or French, Verbatim and Simplified Text, etc.



A LOW-COST/FLEXIBLE DE-INTERLACING SOLUTION

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Abstract -- Video de-interlacing is generating new interest in the video and computer industries. The merging of interlaced video and computer display formats into high-end "multimedia" monitors and other display devices is becoming more common; the products are beginning to appear in consumer markets. This paper reviews interlaced and progressive displays, examines various de-interlacing techniques and introduces a low-cost, flexible de-interlacing integrated circuit (IC) from Genesis Microchip Inc.

I. INTRODUCTION

Interlaced video is ubiquitous . . . it is viewed on television screens around the world. Advances in algorithms and integrated circuits have made it possible to improve the viewing quality of interlaced video. One technology known as *de-interlacing, line doubling or proscan conversion* converts interlaced video into a non-interlaced format for display on progressively scanned systems. Image flicker and apparent resolution are improved when video is viewed on a progressively scanned monitor.

Several applications make extensive use of de-interlacing technology. They include:

- Large screen projection systems
- Video-to-computer scan converters
- Up-converters: line doublers and line quadruplers
- Video-to-film converters
- Converting conventional TV to progressive scan ATV
- Pre-processing for MPEG bit-rate compression
- Digital video editing workstations

This paper introduces the concepts of interlaced and progressive (i.e., non-interlaced) scanning. Various methods of de-interlacing are discussed including vertical replication, vertical interpolation, field merging, vertical/temporal filtering and motion adaptive interpolation. It concludes with the presentation of a new de-interlacing integrated circuit from Genesis Microchip Inc.

II. INTERLACED AND PROGRESSIVE DISPLAYS

Interlaced scanning and display originated with the introduction of the NTSC television system in 1941. It was developed as a means to limit video transmission bandwidths and maximize the vertical resolution of the displayed images. Interlaced scanning is a system in which a television scene is represented as two images known as fields. Each field is scanned at different times and consists of one half the number of lines in the original image frame (Fig. 1). In order to provide faithful motion rendition, the NTSC standard set the field scanning rate at 60 fields/sec. or 30 frames/sec. Due to a limited bandwidth, interlaced displays suffer from a number of deficiencies. The interlaced fields have increased apparent vertical resolution over non-interlaced frames of the same size. Flicker can be a problem for fields with high vertical frequency content; "stair stepping" or "jaggies" may also result.

Progressive scanning and displays evolved from the computer industry and were not constrained by video transmission bandwidths. Progressive displays represent the image by a complete set of lines (i.e., a frame) each time the image is scanned (Fig. 1). When displaying the same number of frames as fields in interlaced video, progressive displays have less line flicker and fewer jaggies. Apparent vertical resolution is improved while superior visual results can be obtained if interlaced video is converted and displayed in a progressive scan format.

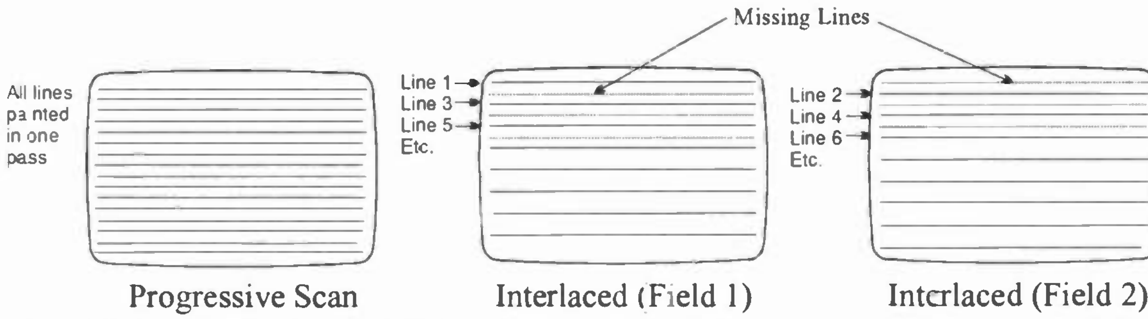


Figure 1: Interlaced and progressive scan displays

III. DE-INTERLACING METHODS

De-interlacing is a method of converting interlaced fields into progressive scanned frames. The "missing" lines in an interlaced field are calculated from one or several temporally displaced interlaced fields. Several de-interlacing solutions have been developed, but all trade vertical and temporal resolution. In areas of no motion, vertical resolution is maximized when adjacent fields are merged. The maximum vertical resolution is equal to the number of lines in a frame. This is equivalent to heavily favoring temporal processing; however, if temporal processing is favored in areas of high motion, scene content changes from many fields are combined and multiple images result. This phenomenon is known as *jutter* (see Fig. 2). In order to avoid jutter in areas of motion, it is desirable to minimize temporal processing and calculate the "missing" lines from the current interlaced field. In this case, intra-field processing is employed limiting the maximum vertical resolution to the number of field lines.

Depending on the image quality and hardware cost/complexity tradeoffs, de-interlacing techniques attempt to balance the conflicting goals of maximizing vertical resolution and minimizing jutter. Popular de-interlacing techniques include:

1. Vertical Replication
2. Vertical Interpolation
3. Field Merging
4. Vertical/Temporal Filtering
5. Motion Adaptive Interpolation

Successful evaluation of these de-interlacing techniques requires the analysis of critical static and motion scene content.

i. Vertical Replication

Also called field or line replication - this is the simplest and least expensive de-interlacing technique and it yields the lowest quality results. This method repeats lines from the same field to create the missing lines. The scheme does not use the signal's temporal information; it achieves, at best, only vertical field resolution. Jutter is

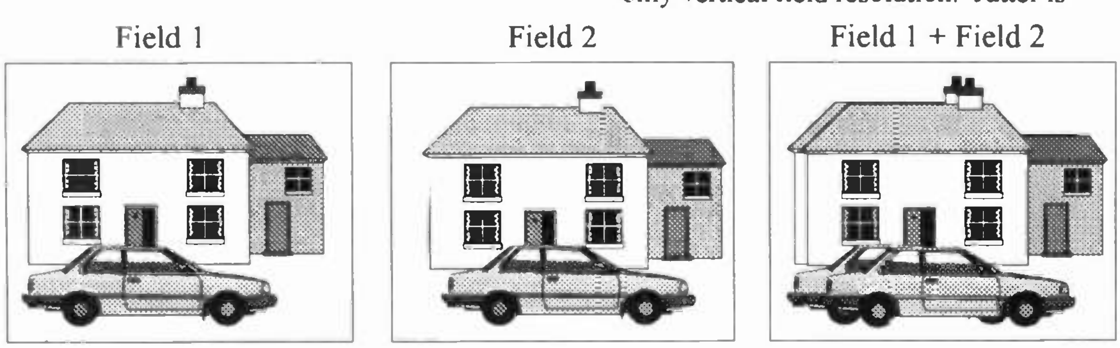


Figure 2: Combining Field 1 and Field 2 when there is motion (e.g., camera panning) results in jutter.

minimized because no temporal information is used. Line replicating does not employ any filtering techniques and can produce a distortion phenomenon called *imaging*, which is revealed as unwanted jaggies visible on picture detail. Imaging is a phenomenon occurring when a signal is upsampled and a filter is not employed to remove the undesired signal repeat harmonics. Fig. 3 illustrates line replication.

ii. Vertical Interpolation

Another simple de-interlacing method is vertical interpolation (Fig. 4). Vertical interpolation uses neighboring lines in the same field to create the missing lines. Intra-field vertical interpolation also limits the maximum possible vertical resolution to *field resolution*. Jutter is also minimized because no temporal information is used. Vertical interpolation, however, often suffers from blurring. The resolution appears less (i.e., the images appear softer compared to those created by vertical replication) because of compromises introduced by the vertical interpolator. A high-quality interpolator - one with many *taps* (numerous lines to interpolate) - will yield a superior response in contrast to a minimum or linear interpolator.

Field merging (Fig. 5) is a de-interlacing technique where two temporally adjacent fields are simply merged together. Maximum resolution (i.e., frame resolution) is achieved. This works well for static images but in areas of motion, jutter is excessive.

iv. Vertical/Temporal Filtering

This method makes use of both vertical and temporal information to create the missing lines. In most cases, a good filter design can provide an acceptable tradeoff between vertical and temporal resolution for static and

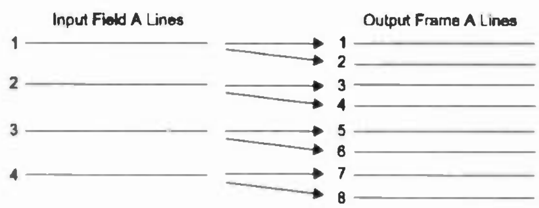
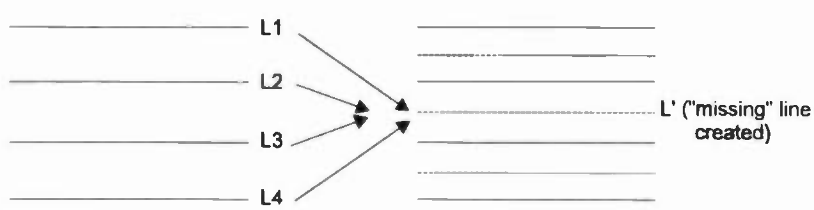


Figure 3: Line Replication, field lines are replicated into frame lines.



L' = aL1 + bL2 + cL3 + dL4, where a,b,c,d are filter coefficients

Figure 4: Vertical Interpolation

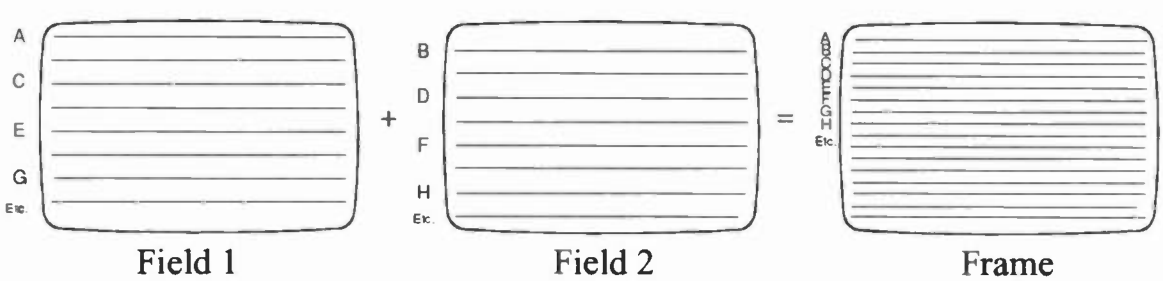


Figure 5: Field Merging

motion scenes. Due to the cost of line/field stores, a typical filter aperture for a vertical/temporal filter can be 2-3 fields and 4-5 lines. However, a vertical/temporal filter approach can have trouble when displaying moving diagonal edges in a scene. Fig. 6 illustrates the construction of a "missing" line from the current, past and future fields.

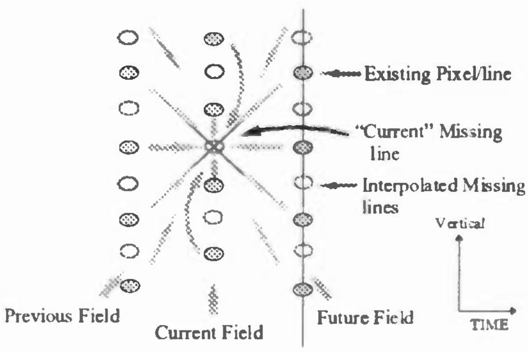


Figure 6: Vertical/Temporal Filter

v. Motion Adaptive Interpolation

In this approach, different filters are applied depending on the amount of motion in the scene. Processing is included to detect static and motion content in the scene. For static scene areas, the vertical image content is maximized by emphasizing the contribution of scene detail from other fields. Field merging is typically applied to static areas of the picture. For areas of motion, the filter changes to minimize the contribution from adjacent fields and the missing lines are constructed from lines within the field. Vertical interpolation is usually applied to motion areas.

The disadvantages of motion adaptive interpolation include cost, complexity, motion detection failure and possible discontinuities or visible changes in the image's resolution. However, in areas of general motion, the reduction in resolution is generally not detectable or appears natural when viewed on a monitor. As a de-interlacing solution, motion adaptive interpolation bears a high implementation cost.

IV. A LOW-COST, FLEXIBLE DE-INTERLACING SOLUTION FROM GENESIS MICROCHIP INC.

Genesis Microchip will introduce at N.A.B. a single-chip video de-interlacer. By using a specific implementation of a vertical temporal filter, this chip's interpolator provides excellent performance for motion and static video scenes. The chip's de-interlacing component comprises an arrangement of delays, adders, subtractors and multipliers. But the greatest feature (other than great overall video de-interlacing performance) is the convenient and simple way it is designed into a variety of systems.

The Genesis de-interlacing (or Video Line Doubling) chip family consists of two parts:

- gmVLD8 - an 8-bit Video Line Doubler
- gmVLD10 - a 10-bit Video Line Doubler

Both chips support the following three modes:

<u>Three De-interlacing Modes</u>

- Mode 1: 3-field processing with temporal and vertical filters (two external field stores required)
- Mode 2: 2-field processing with temporal and vertical filters (one external field store required)
- Mode 3: Field merging with filter bypass (one external field store required)

<u>Glueless Interfacing</u>

- To Philips-style video decoders, SMPTE 125/CCIR 656 supported
- To popular field stores: NEC, Hitachi, Okidata, T.I.
- To Genesis scaling engines: gm833x2, gm833x3, gm833x3F & gm865x1

<u>Other Features</u>

- Onboard color space converter
- Onboard CRT controller
- Direct connection to DAC

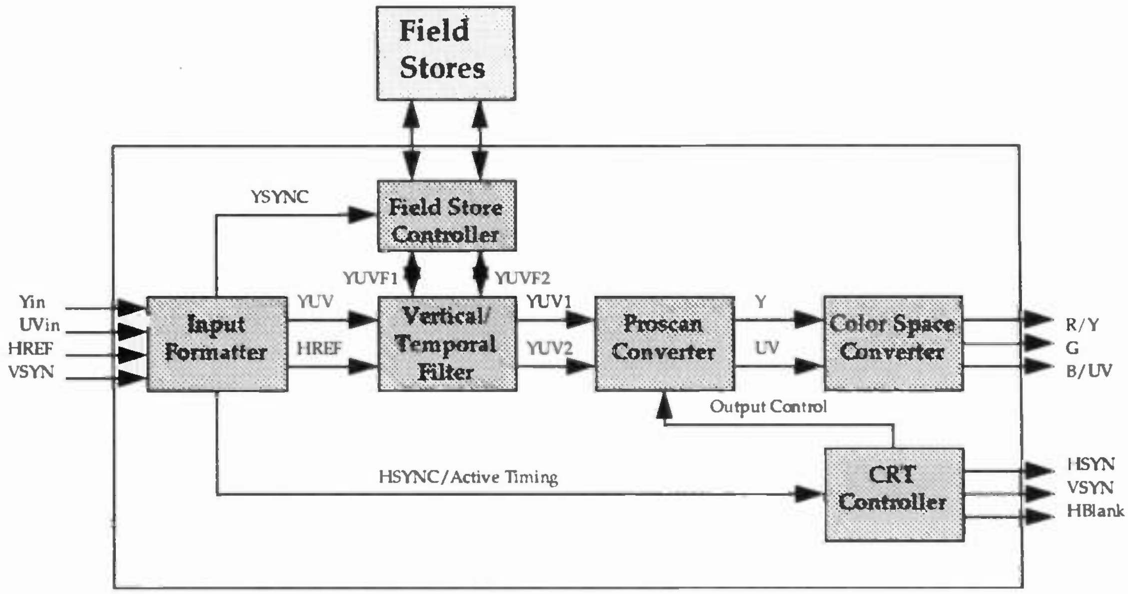


Figure 7: gmVLD8 and gmVLD10 block diagram

V. CONCLUSION

Choosing a de-interlacing technique depends on cost, complexity and desired picture quality. The Genesis Microchip's gmVLD8 and gmVLD10 represent a breakthrough in de-interlacing techniques with the ability to properly de-interlace video streams in a cost-effective, easy-to-implement manner.

TELEVISION ON-LINE: INTERNET APPLICATIONS

Thursday, April 18, 1996
10:45 am - 12:00 pm

Session Chairperson:
Andy Butler, PBS, Inc., Alexandria, VA

***BROADCAST AND INTERNET: LEVERAGING ASSETS**
Sheau Ng
Toshiba America Consumer Products, Inc.
Princeton, NJ

***NEW DEVELOPMENTS IN DIGITAL VIDEO BROADCASTING COST-EFFECTIVE SATELLITE DELIVERY TO PCS FOR CORPORATE ENTERPRISE APPLICATIONS**
James Long
Starlight Networks
Mountain View, CA

***PROFESSIONAL AUDIO AND VIDEO IN AN INTERACTIVE WORLD**
Rob Glaser
Progressive Networks
Seattle, WA

*Paper not available at the time of publication.

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